Protocol design and performance analysis for wireless networks

Juki Wirawan Tantra

2007

Juki, W. T. (2007). Protocol design and performance analysis for wireless networks.
Doctoral thesis, Nanyang Technological University, Singapore.

<https://hdl.handle.net/10356/2485>

<https://doi.org/10.32657/10356/2485>

Nanyang Technological University

Downloaded on 26 Apr 2023 18:21:30 SGT

**NANYANG
TECHNOLOGICAL
UNIVERSITY**

**PROTOCOL DESIGN AND PERFORMANCE
ANALYSIS FOR WIRELESS NETWORKS**

**JUKI WIRAWAN TANTRA
SCHOOL OF COMPUTER ENGINEERING
2007**

Protocol Design and Performance Analysis for Wireless Networks

Juki Wirawan Tantra

School of Computer Engineering

A thesis submitted to the Nanyang Technological University
in fulfilment of the requirement for the degree of
Doctor of Philosophy

2007

Acknowledgment

I am deeply grateful to my advisor Dr. Foh Chuan Heng for his advice and guidance during my candidature period. Without his guidance, my work would not have been fruitful.

I would like to thank Dr. Lee Bu Sung for his kind advice and help for my work. I would also like to thank Dr. Lim Teck Meng for many hours of brainstorming sessions that we had. My gratitude to Dr. Adel Ben Mnaouer for his guidance in the collaboration work that we had. I am indebted to Dr. Yeo Chai Kiat, without whom I would not have started my graduate study. I am thankful for the advices, recommendations, and helps from Dr. Giuseppe Bianchi and Dr. Ilenia Tinnirello. Their contributions have been so valuable to my researches.

I am indebted to the examiners who have spent their precious time to give suggestions to improve the thesis.

Lastly but not the least important, I would like to express my gratitude to Priscilla Laura Sumianto, my lovely wife, for her support and kindness during my candidature period and more. Thank you for being so understanding.

Contents

Acknowledgment	iii
Contents	v
List of Figures	ix
List of Tables	xi
Abstract	xiii
1 Introduction	1
1.1 Thesis Overview	4
1.2 Contributions	7
1.3 Publication List	9
2 Literature Review	13
2.1 Researches on Wireless LANs	14
2.1.1 On History of Wireless LANs MAC Protocols	14
2.1.2 On Improvements of Wireless LANs Protocols	15
2.1.3 On Modeling of the IEEE 802.11 MAC Protocols	18
2.1.4 Quality of Service Support in Wireless LANs	19
2.2 IEEE 802.11 MAC Protocol	21
2.2.1 Distributed Coordination Function	21
2.2.2 Point Coordination Function	23
2.3 IEEE 802.11e Enhanced Distributed Channel Access	24
2.4 Bianchi's Saturation Analysis	26
2.5 Markovian Framework for DCF Analysis	31
3 Refinement of DCF Saturation Model	35
3.1 Saturation Throughput Model	37
3.2 Numerical Results	42
3.3 Summary	44

4	Saturation Analysis of EDCA	45
4.1	Analytical Models	45
4.1.1	Saturation Model	45
4.1.2	Saturation Throughput	53
4.1.3	Frame Dropping Probability	54
4.1.4	Saturation Delay	54
4.2	Results and Discussions	58
4.2.1	Saturation Throughput	58
4.2.2	Frame Dropping Probability	60
4.2.3	Delay Distribution	61
4.2.4	Mean Frame Delay	61
4.3	Summary	64
5	Out-of-Band Signaling Scheme	65
5.1	Description of Out-of-Band Signaling Scheme	65
5.2	Saturation Analysis of Out-of-Band Signaling Scheme	68
5.2.1	Markovian Framework for Saturation Analysis	68
5.2.2	Performance Comparison	73
5.2.3	Signaling Channel Optimization	77
5.3	Analysis of the Out-of-Band Signaling Scheme under Statistical Traffic	79
5.3.1	Markovian Framework for Out-of-Band Signaling Scheme under Statistical Traffic	79
5.3.2	Performance Comparison	83
5.4	Summary	85
6	Out-of-Band Polling Scheme	87
6.1	Scheme Overview	88
6.1.1	Signaling Channel	88
6.1.2	Data Channel	89
6.1.3	Quality-of-Service in OBPS	90
6.2	MAC Delay Analysis	93
6.2.1	OBPS Worst-case Delay for Priority Packets	94
6.2.2	IEEE 802.11 PCF Worst-case Delay	96
6.2.3	IEEE 802.11 DCF Average Delay	97
6.2.4	Upstream Transmission Delay Comparison	98
6.2.5	OBPS Worst-case Delay for Priority Upstream Traffic with Downstream Traffic	100
6.2.6	OBPS Worst-case Delay for Downstream Traffic with Upstream Traffic	101
6.2.7	Simulated MAC Queuing Delay	102

6.3	Throughput Analysis	103
6.3.1	Saturation Throughput without Priority Stations	103
6.3.2	Saturation Throughput for OBPS with Bidirectional Traffic and Priority Stations	106
6.3.3	Simulated Rate Differentiation	108
6.4	Summary	108
7	Contention-Tone Protocol	109
7.1	Protocol Detail	110
7.2	Success Rate of CTP	112
7.3	Throughput Performance of CTP	115
7.4	Delay Performance of CTP	118
7.5	Summary	118
8	Conclusion	121
8.1	Future Considerations	125
	Acronyms and Abbreviations	127
	Bibliography	129

List of Figures

2.1	IEEE 802.11 DCF: (a) Basic access method. (b) Four-way hand-shaking method.	22
2.2	IEEE 802.11 PCF.	24
2.3	Timing of the EDCA mechanism.	25
2.4	Bianchi's Markov chain model of a station backoff process in DCF.	28
2.5	State transition diagram of SD-M/M/1/k SSQ.	32
3.1	IEEE 802.11 basic access method.	38
3.2	The Markov chain representation of the new model.	38
3.3	Probabilities of an idle, a successful transmission, and a collision of various models for $W = 16$ and $m = 6$	43
3.4	Saturation throughput of various models for $W = 16$ and $m = 6$	44
4.1	Markov chain model of the EDCA backoff process.	48
4.2	Probability that a slot contains a successful transmission by stations of each AC.	52
4.3	Probability that a slot is an idle slot and probability that a slot contains a colliding transmission.	53
4.4	Illustration of the backoff process for delay computation.	57
4.5	Saturation throughput for EDCA basic access methods.	59
4.6	Saturation throughput for EDCA RTS/CTS methods.	60
4.7	Probability of dropping a frame.	60
4.8	Delay distribution of EDCA basic access method with five stations of each AC.	62
4.9	Delay distribution of EDCA RTS/CTS method with five stations of each AC.	62
4.10	Saturation delay for basic access methods.	63
4.11	Saturation delay for RTS/CTS methods.	63
5.1	The OBS scheme illustrated.	66
5.2	Saturation throughput with various data channel bit rate.	75

5.3	Saturation throughput with 108 Mbps channel bit rate.	76
5.4	Mean saturation transmission delay with 108 Mbps channel bit rate.	77
5.5	Minimum signaling channel bit rate required to achieve full utilization on the data channel.	79
5.6	Mean transmission delay under statistical traffic.	83
5.7	Throughput versus mean transmission delay under statistical traffic.	84
6.1	Operations of out-of-band polling.	88
6.2	Downstream and upstream transmissions for station A.	90
6.3	Station A downstream packet transmission preempted by station B.	90
6.4	An illustration for the value τ and ϕ	96
6.5	Worst-case delay for upstream transmissions with fixed packet size of 1500 bytes.	99
6.6	Worst-case delay for bidirectional transmissions with fixed packet size of 1500 bytes.	100
6.7	Average delay with fixed packet size of 1500 bytes.	102
6.8	Saturation throughput with fixed packet size of 1500 bytes (the solid lines represent numerical results).	104
6.9	Saturation throughput with fixed packet size of 500 bytes.	105
6.10	Saturation throughput of OBPS with various number of stations.	107
6.11	Throughput attained by each station with fixed packet size of 1500 bytes.	107
7.1	Illustration of contention-tone protocol.	110
7.2	Saturation throughput performance of the proposed CTP compared with the DCF standard.	117
7.3	Saturation delay performance of CTP compared with the DCF standard.	119

List of Tables

2.1	Typical parameters for various ACs.	25
3.1	Parameters used for numerical analysis and simulations.	42
4.1	Parameters used for numerical analysis and simulations.	59
5.1	Parameters used for numerical analysis and simulations.	74
6.1	OBPS worst-case delay for downstream traffic on a 54 Mbps data channel.	101
8.1	Comparison of OBS, OBPS and CTP.	125

Abstract

In recent years, *wireless local area networks* (WLANs) have been very popular for the *last mile* connections. This popularity has fueled the demands for even faster wireless networking technology and also for *quality of service* (QoS) support in wireless networks. This thesis deals with protocol designs and analysis for future high speed wireless networks with some considerations for QoS provisioning on these wireless networks.

In this thesis, first we introduce a refinement to the throughput performance model of the current IEEE 802.11 standard for WLANs. This refinement introduces a correction to the current models by Bianchi and also by Ziouva and Antonakopoulos so that it becomes more standard conforming, increasing the usefulness of the model for the analysis of the performances of the standard. Furthermore, we develop a model to analyze the performance of IEEE 802.11e *enhanced distributed channel access* (EDCA) under saturation condition. With this model, we derive the throughput, the mean delay, and the delay distribution of stations executing EDCA. The contribution of this model is in its applicability for admission control development and protocol optimization.

For future high speed wireless networks, we propose and analyze three protocols which share a common theme: the use of out-of-band signaling. These protocols vary in the complexities, and thus vary in the applications. The first of these protocol is *out-of-band signaling scheme* (OBS), which uses contention based protocol similar with the IEEE 802.11 standard protocol to reserve the channel. The time-wasting contention is done on a separate low speed channel, thus freeing the high speed channel for data transmission. Our analysis shows that off-

loading the contention to a separate channel improves the performance of wireless networks compared with the current in-band signaling scheme.

The second protocol, *out-of-band polling scheme* (OBPS), uses a polling scheme on the signaling channel to identify the stations that need to transmit frames. Inside this thesis, we show that OBPS can provide both delay differentiation between flows and rate differentiation between stations. These characteristics make the scheme attractive for deployment in WLANs that require QoS provisioning.

We call the third protocol *contention-tone protocol* (CTP); CTP implements a novel tone contention scheme on a separate signaling channel to reserve a transmission on the data channel. This scheme can achieve near to the maximum theoretical *medium access control* (MAC) throughput in wireless networks. The results presented in this thesis show that out-of-band signaling scheme is more efficient than the current in-band scheme. Out-of-band scheme is also more scalable to higher speed wireless networks. The three protocols presented here can achieve higher throughput than the current IEEE 802.11 standards. They have various complexities; thus, they can cater for various designs depending on the required device complexity.

Chapter 1

Introduction

In recent years, we have witnessed the popularity of *wireless local area networks* (WLANs) for the *last mile* connections. This wide appeal of wireless networks is hardly surprising considering their ease of use; owing to the introduction of WLANs, we have an almost pervasive Internet connectivity: in the libraries, cafes, and many other crowded places. Owing to this popularity and also the potential for future applications, many works have been focused on wireless networks researches.

The popularity of WLANs was partly promoted by the introduction of the IEEE 802.11 standard [1]. First introduced, the original standard supports physical bit rates of up to 2 Mbps. This standard has been updated with some revisions that provide higher physical bit rates: IEEE 802.11a [2], which provides physical bit rates of up to 54 Mbps in the 5 GHz spectrum band, IEEE 802.11b [3], which provides bit rates of up to 11 Mbps in the 2.4 GHz spectrum band, and recently IEEE 802.11g [4], which support 54 Mbps bit rates in the 2.4 GHz spectrum band and is backward compatible with IEEE 802.11b. Even though the IEEE 802.11a

was introduced about the same time with 802.11b and can provide higher bit rates, IEEE 802.11b proves to be more popular because 11b devices were marketed long before 11a devices and, at the beginning, 11b devices have much longer range than 11a devices [5]. As the popularity of IEEE 802.11a dwindling, IEEE 802.11g is introduced to provide 54 Mbps bit rates in the 2.4 GHz spectrum band.

IEEE 802.11 specifies two access modes: the first is contention based *distributed coordination function* (DCF), and the second is an optional polling based protocol named *point coordination function* (PCF). DCF basically implements Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA). To prevent too many collisions, DCF implements exponential backoff for the random backoff period selections.

Demands for high speed WLANs and *quality of service* (QoS) support have spurred many works and investments in the last few years. With the standard for QoS support in WLANs (designated IEEE 802.11e [6]) has just recently been standardized, there are many works to be done in implementation related researches such as scheduler, admission control, and QoS management and reservation, which the standard lacks. IEEE 802.11e introduces *hybrid coordination function* (HCF), which consists of the contention based *enhanced distributed channel access* (EDCA) and the contention-free *HCF controlled channel access* (HCCA) to support QoS applications on WLANs.

With the ongoing development of IEEE 802.11n, high speed WLANs have generated many researches in both the physical and data link layer. In physical layer, many researches are focused on more efficient modulation technique to support high speed transmission. Researches in data link layer mainly try to improve DCF, which many researches have found to be inefficient with high bit rates

physical layer [7] or a high number of stations [8].

Another recent popular topic in wireless networks *medium access control* (MAC) research is on performance modeling. Performance modeling is important for many applications, for examples: protocol improvements, performance benchmark, and admission control. In 1998, Bianchi introduces his famous model for DCF [9], which deals with saturation scenario, i.e. where all the involved stations always have data frames to transmit. The popularity of this model is due to its tractability and accuracy in predicting the throughput performance of IEEE 802.11 DCF protocols. Since its introduction, many researchers have worked on both improving the model and applying the model for various applications¹.

When I started this thesis, QoS issues and high speed wireless networks are two of the popular topics with the development of IEEE standards for these issues (802.11e² for QoS and 802.11n for high speed WLANs). It can be seen that this consideration has a large influence on the contents of this thesis. The first chapter proposes a refinement to the Bianchi's saturation model, so that it becomes more standard conforming. The next chapter discusses a saturation model that we developed to analyze the throughput and delay performances of EDCA. The end half of this thesis deals with three protocols for high speed wireless networks; they share a common theme, which is the use of an out-of-band signaling channel to improve the performances of high speed wireless networks.

Previously, out-of-band signaling in wireless networks is mostly used to combat hidden-terminal and exposed terminal problems [10, 11]. In-band protocols with their simplicity are considered advantageous until the recent push for high

¹We will describe this further in the next chapter

²As mentioned earlier, IEEE 802.11e has been standardized.

speed wireless networks. In high speed wireless networks, out-of-band signaling is obviously advantageous as it can offload the transmission overheads to another channel. Furthermore, our out-of-band protocols allow easy implementation of QoS support.

We called the first of our out-of-band protocols *out-of-band signaling* (OBS); OBS uses contention based scheme on the signaling channel to reserve a transmission chance on the data channel. The second protocol is *out-of-band polling scheme* (OBPS), which uses a polling scheme on the signaling channel to identify the stations that need to transmit frames. The last protocol, which we named *contention-tone protocol* (CTP), implements a novel tone contention scheme on a separate signaling channel to reserve a transmission on the data channel. The reason of having three protocol designs instead of one is that they have various complexities that suit specific applications depending on the requirements.

1.1 Thesis Overview

This thesis summarizes the works that I have done during my candidature period for the degree of Doctor of Philosophy. What I mean by ‘summarize’ here is that I have only included the significant works in the thesis. It is common in research that we have failed experiments, non-working proposals, and also the experiences of working on good proposals only to find that other people have completed the work. In this thesis, I left out all of the ‘non-successful’ works, and thus I detailed only the significant works that we have done in the last three years.

Most of the content chapters of this thesis are derived from research papers, which have other authors. I only use parts of these research papers that I have

worked on; therefore, some sections from those papers are not presented here.

Chapter 2 of this thesis provides a survey of the wireless networks researches. In this chapter, we provide an overview of past proposals of protocols for wireless networks and the recent advancements in the area of wireless networks. This chapter also briefly summarizes the IEEE 802.11 MAC protocols, Bianchi's saturation model [9, 8], and Markovian analytical framework by Foh and Zukerman [12].

Chapter 3 describes our refinement of Bianchi's saturation model. The refinement deals with the backoff freezing mechanism in the IEEE 802.11 standard, which is missing in Bianchi's model. The backoff freezing mechanism has previously been modeled by Ziouva and Antonakopoulos [13]. However, the model was developed without considering the underlying channel status probabilities, which causes the model to be inaccurate. We first describe the derivation of our Markov Chain model, and then we show the accuracy of the refinement by comparing the analytical results with the simulation results. The benefit of this new model is more accurate computation of the underlying probabilities: probability of idle slots, successful transmissions and collisions. These probabilities in turn determine the accuracy of performance measurements such as frame dropping probability and delay distribution. This chapter is derived from [14].

Chapter 4 introduces a model for EDCA saturation analysis. In this chapter, we extend Bianchi's saturation model to include contention window and inter-frame spacing differentiation, which are defined in the recent IEEE 802.11e standard [6]. We use this model to derive the throughput performance and the frame dropping probability. Furthermore, we derive the delay performance using z-transform analysis. With this analytical method, we can derive the delay dis-

tribution of EDCA; hence, we can capture the delay characteristics of EDCA. The accuracy of our analytical model is verified using simulations. This saturation model is useful for admission control design and protocol optimization. This chapter is derived from [15] and [16].

Chapter 5 introduces the OBS protocol. Firstly, we describe the details of OBS, which implements contention based signaling to reserve for frame transmissions on the data channel. The basic idea of separating the signaling from the frame transmission is that in DCF, idle periods and transmission collisions are the main overheads of frame transmissions. Therefore, by separating the signaling process, the data channel will be better utilized and thus achieve better performance compared with the performance achieved by combining the data and signaling channel. We develop analytical models for the operation of OBS under saturation condition and under statistical traffic. These models are inspired by the Markovian Framework proposed by Foh and Zukerman for the analysis of DCF [12]. We verify these models with simulations, and then we compare the performance of OBS with that of the standard IEEE 802.11 DCF. The results affirm the advantages of OBS compared with the current standard protocols. This chapter is derived from [17, 18, 19].

Chapter 6 discusses the OBPS protocol. We first describe OBPS protocol details and its capability to support bounded latency. This capability is one of the features of polling based signaling implemented by OBPS. OBPS first polls the stations for requests for transmissions before the actual polls for frame transmissions. We analyze the delay performance of OBPS; we compare this delay performance with those of IEEE 802.11 DCF and PCF. After delay analysis, we also compare the throughput performance of OBPS with those of IEEE 802.11

DCF and PCF. From these comparisons, we can conclude that OBPS can achieve better performance than the current standard protocols. This chapter is derived from [20].

Chapter 7 describes CTP. This chapter is opened with the description of the contention tone and the operation of the protocol. Instead of frame contentions implemented in DCF and OBS, CTP implements tone contention on the signaling channel. Briefly described, stations transmit a sequence of tones, and the winner is determined on the end of the sequence; this winner reserves the right to transmit a frame on the data channel. We derive the success probability of the contention tone; this success probability is then used to derive the saturation throughput and delay performances of CTP. We then compare the throughput and delay performances of CTP with those of DCF. We show that CTP can achieve higher throughput and lower delay compared with DCF. This chapter is derived from [21].

Chapter 8 concludes the thesis, and provides possible future applications or extensions. In this chapter, we summarize this thesis and reiterate the major conclusions and contributions of the thesis.

1.2 Contributions

The main contribution of this thesis is threefold. Firstly, we have refined Bianchi's saturation model of DCF to accurately model the backoff freezing mechanism specified in the IEEE 802.11 standard. This refinement corrects the previous backoff freezing model by Ziouva and Antonakopoulos [13], which is inaccurate because the model is developed without considering the underlying channel status

probabilities. Our refinement increases the usefulness of these previous models as the new model is now more standard conforming. Furthermore, our presentation of this model in [14] has also cleared some doubts on the backoff freezing mechanism of DCF. This model has been used in [22, 23, 24].

Secondly, we have introduced an analytical model for saturation analysis of EDCA. Using this model, we can analyze the throughput and delay performances of EDCA. Furthermore, we derive EDCA delay distribution with the z-transform analysis presented in this thesis. This saturation model is useful for admission control design and protocol optimization. This saturation model is presented in [15] and [16]. Our analytical model accurately represents the operation of the EDCA protocol compared with the previous models. With this model, underlying probabilities such as idle, success and collision probabilities can be accurately computed. The z-transform approach for the delay computation also significantly reduces the complexity of the derivations and computations of the numerical results.

Thirdly, we present three protocol designs in this thesis; these protocols share a common theme of the use of out-of-band signaling method. These protocols have various complexities; thus, they can cater for many applications depending on the complexity requirements of the protocol. These out-of-band protocols significantly improve the throughput and delay performance of wireless networks compared with the existing in-band techniques. The details of these protocols are as follows:

- OBS is an out-of-band signaling scheme that uses a contention based protocol for signaling. This scheme improves the throughput and the delay performances of wireless networks with a limited support from the access

point. The complexity of this protocol is shared between the stations and the access point; the access point is slightly more complex as it needs to poll the stations. OBS enables the use of queue management algorithms to better support the QoS applications; hence, it exhibits the benefits of both random access and polling protocols. OBS is discussed in [17, 18, 19].

- OBPS is an out-of-band signaling scheme that uses a time division protocol for signaling. OBPS can provide a certain quality of service, which is required by real-time applications. OBPS requires heavy coordination by the access point; hence, most of the complexity of OBPS is located on the access point whereas the station has low complexity. OBPS can support applications that require specific timings while at the same time reduce the overheads associated with polling protocols. OBPS was reported in [20].
- CTP is an out-of-band signaling scheme that uses contention tone to claim the channel. CTP can achieve high efficiency with or without coordinating access point. CTP requires complex client implementation to support fast tone detection and transmission. With contention tone, CTP can achieve a throughput that is near to the maximum theoretical throughput; this is a significant improvement compared to the throughputs achieved by existing schemes and standards. CTP is presented in [21].

1.3 Publication List

This thesis is based on seven research papers, and a work in progress to be submitted for publication. The list is as follows:

- Journal papers:

1. T. M. Lim, J. W. Tantra, C. H. Foh, and B. S. Lee, "Out-of-band polling scheme for QoS support in wireless LANs," *International Journal of Communication Systems Special Issue on QoS Support and Service Differentiation in Wireless Networks*, vol 17, no. 6, pp. 643-661, Aug. 2004.
2. C. H. Foh and J. W. Tantra, "Comments on IEEE 802.11 saturation throughput analysis with freezing of backoff counters", *IEEE Communications Letters*, Volume 9, Issue 2, Pages 130-132, Feb. 2005.
3. J. W. Tantra and C. H. Foh, "Achieving near maximum throughput in IEEE 802.11 WLANs with contention tone", *IEEE Communications Letters*, Volume 10, Issue 9, Pages 658-660, Sep. 2006.
4. J. W. Tantra, C. H. Foh, I. Tinnirello, and G. Bianchi, "Out-of-band signaling scheme for high speed wireless LANs," to be published in *IEEE Transactions on Wireless Communications*.

- Conference papers:

5. J. W. Tantra, C. H. Foh, and B. S. Lee, "An efficient scheduling scheme for high speed IEEE 802.11 WLANs," in *Proc. IEEE Vehicular Technology Conference*, Orlando, Florida, USA, Oct. 2003.
6. J. W. Tantra, C. H. Foh, G. Bianchi, and I. Tinnirello, "Performance analysis of the out-of-band signaling scheme for high speed wireless LANs," in *Proc. IEEE Globecom*, Dallas, Texas, USA, Nov.-Dec. 2004.

7. J. W. Tantra, C. H. Foh, and A. B. Mnaouer, “Throughput and delay analysis of the IEEE 802.11e EDCA Saturation,” in *Proc. IEEE International Conference on Communications*, Seoul, South Korea, May 2005.

- In preparation:

8. J. W. Tantra, C. H. Foh, and A. B. Mnaouer, “Delay analysis of the IEEE 802.11e EDCA³,” in preparation for journal submission.

³Tentative title.

Chapter 2

Literature Review

This chapter covers recent researches in WLANs, the IEEE standards for WLANs, and also detailed descriptions of some important analytical works which will feature prominently in the subsequent chapters. This chapter serves both as a review of the recent advances in WLANs, and as the basis of the contents of this thesis. As a review, we include many interesting researches in recent years; we admit that, as a review, this chapter is incomplete, and interesting is a subjective evaluation criteria. However, this chapter is complete enough as a support for the contents of this thesis, a basis for the ideas and analysis presented in this thesis. Hopefully, it will be an interesting reading material.

This chapter is organized as follows. This chapter starts with descriptions of many recent WLANs literature. Then, we describe the current IEEE standards for WLANs. Following that, we discuss the analytical framework introduced by Bianchi for saturation analysis of the IEEE 802.11 standard protocols. Lastly, we describe a method by Foh and Zukerman for non-saturation analysis of these standard protocols.

2.1 Researches on Wireless LANs

2.1.1 On History of Wireless LANs MAC Protocols

The first wireless MAC protocol, ALOHA, is introduced by Abramson in 1970 [25]. The basic operation is simple. Stations transmit the frame using a radio to another station. If collision occurs, i.e. more than one station transmits at the same time, the frame transmission is rescheduled. The frame will be retransmitted after a random delay. In 1972, Roberts introduces slotted ALOHA [26] which doubles the throughput of ALOHA at a cost of slot time synchronization.

Later on, *Carrier Sense Multiple Access* (CSMA) protocols are developed to further improve the throughput performance of ALOHA. In carrier sense protocols, a station first attempts to detect whether there is any transmission in progress on the channel; the station starts its transmission only if the channel is free. One notable CSMA protocol is *CSMA with collision detection* (CSMA/CD), which is used in the popular Ethernet local area networks.

In 1990, Karn introduces *Multiple Access with Collision Avoidance* (MACA) protocol for WLANs [27]. In the paper, Karn proposes the *request-to-send / clear-to-send* (RTS/CTS) mechanism to prevent transmissions from neighboring stations. In 1994, Bharghavan *et al.* [28] introduce *MACA for Wireless* (MACAW), which is an improved MACA protocol. MACAW introduces direct acknowledgment of transmitted frames and carrier sensing to the original MACA protocol. These two protocols evolve to become *CSMA with collision avoidance* (CSMA/CA), which is the basis of the IEEE 802.11 DCF standard [1]. The details of the standard is given in section 2.2.

Tobagi and Kleinrock [10] introduce *busy tone multiple access* (BTMA) to

eliminate the hidden-terminal problem. BTMA basically uses a separate narrow signaling channel to transmit the busy tone whenever a transmission takes place. In [11], Haas and Deng propose *dual busy tone multiple access* (DBTMA) that uses two busy tone channels to solve both the hidden-terminal and exposed-terminal problems. Tobagi and Kleinrock [29] also introduce an out-of-band request protocol named *split-channel reservation multiple access* (SRMA). SRMA operates with two signaling channels, which serve as the transmission request channel and the answer channel.

There are many papers that analyze the performances of these protocols. The throughput and delay performances of ALOHA, slotted ALOHA, and CSMA under infinite stations assumption is provided by Tobagi and Kleinrock in [30]. Throughput evaluation of ALOHA under finite stations is in [31] by Kleinrock and Lam. Performance analysis of CSMA/CD is given in [32, 33, 34, 35]. Performance analysis of CSMA/CA is detailed in section 2.1.3.

2.1.2 On Improvements of Wireless LANs Protocols

Many researches have proposed improvements to the IEEE 802.11 DCF. Bianchi et al. [36] propose Adaptive Contention Window mechanism to improve the performance of CSMA/CA. The performance of CSMA/CA is affected by the chosen contention window size. Adaptive Contention Window modifies the contention window dynamically according to network loads; hence, it can improve the performance of CSMA/CA by reducing collision probability.

Cali *et al.* [37, 38] propose and analyze an adaptive backoff mechanism to improve the capacity of the DCF protocol. Using the proposed mechanism, the

backoff process is optimized according to the channel status, hence improving the performance. Adaptive Backoff Mechanism in [37, 38] is similar to the Adaptive Contention Window in [36] in the sense that they modify the backoff mechanism of the IEEE 802.11 MAC protocol according to network loads.

Ni *et al.* propose and analyze slow contention window decrease scheme to improve the performance of DCF. Wang *et al.* [39] also propose a similar scheme, in which the contention window is slowly reduced after a successful transmission following failed transmissions. Compared with the existing scheme where the contention window is reset to the initial value, this mechanism reduces the number of collisions as the stations become less aggressive in trying to access the channel.

Kwon *et al.* [40, 41] introduce fast collision resolution scheme, in which the backoff timers are reduced exponentially when consecutive idle slots are detected. Choi *et al.* [42] introduce a reservation system, in which each station broadcasts its future backoff information on every successful transmission. Stations that receive the information can avoid collisions by choosing different backoff counters. These attempts try to optimize the original protocols with some adjustment on the backoff operations. In [43], Kong *et al.* propose to dynamically adjust the RTS/CTS threshold to make the selection between basic access method and RTS/CTS method more effective.

Many papers [44, 45, 46, 47, 48] propose algorithms for dynamic link adaptation. Dynamic link adaptation adjusts the transmission bit rate according to the channel condition; hence, it reduces transmission error rate. Kim *et al.* [49] propose dynamic fragmentation to further improve the performance of link adaptation.

There are also many proposals to enhance the performance of PCF. Sharon

and Altman [50] propose a polling scheme that makes use of capture effect. This polling scheme has higher throughput and lower access delay compared with the standard polling protocol. Ganz and Phonphoem [51] propose the use of superpoll instead of a single poll to better utilize the channel. With superpoll, the access point polls multiple stations at once with a superpoll frame. After receiving the superpoll, the stations will transmit the data frames according to the sequence broadcasted by the access point in the superpoll. In [52], Lo *et al.* introduce CP-Multipoll where the access point broadcasts the backoff counters of the polled stations; hence, the access point manages the transmissions sequences of the stations through the broadcasted backoff counters in the multipoll. In [53], we propose that stations first register for frame transmissions to the access point before the access point polls the stations for the frames. Similar in the principle, Kim *et al.* [54] propose two-steps multipolling where the access point first polls for the queue status of the stations before it polls for the actual frames from the ready stations. The previous four schemes attempt to reduce the overheads of polling while retaining the advantages of a polling scheme.

For high speed WLANs (IEEE 802.11n), Xiao [55, 56, 57] proposes frame aggregation schemes to improve the performance. The basic idea of aggregation is that because IEEE 802.11 has large overheads for every transmitted frame, we can reduce those overheads by transmitting large frames instead of small frames, thus achieving better overall throughput.

2.1.3 On Modeling of the IEEE 802.11 MAC Protocols

Early analysis of CSMA/CA mechanism is in [58], which assumes simpler retransmission algorithm. Chhaya and Gupta [59] provide an approximate model of the IEEE 802.11 MAC protocol that takes into account the presence of hidden stations. Cali *et al.* [60, 37] provide capacity analysis of CSMA/CA under saturation condition with simpler retransmission scheme. Bianchi [9, 8] provides a complete model for DCF saturation condition. This model captures the details of frame retransmission scheme specified in the IEEE 802.11 standard. These two papers by Bianchi have now become classical works on the analysis of DCF.

Ziouva and Antonakopoulos [13] refine the model by Bianchi and claim that the new model is more standard conforming¹. In this paper, they also calculate the saturation delay of DCF. Wu *et al.* [62] extended the Bianchi's model to include the transmission retry limit that is specified in the standard. Using a similar model, Chatzimisios *et al.* [63] derived the delay of frame transmissions with retransmission limit implemented. In [64], Zanella and Pellegrini introduce a z-transform analysis method to derive the delay distribution of DCF in saturation. German and Heindl [65] use a different approach of analysis with stochastic petri nets to analyze the performance of DCF.

Vishnevsky and Lyakhov [66] analyze the performance of DCF under noisy condition. Hadzi-Velkov and Spasenovski [67] extend Bianchi's model to provide saturation throughput and delay analysis of DCF with channel errors. Carvalho and Garcia-Luna-Aceves [68] propose a bottom up approach to compute the average service time and jitter in the IEEE 802.11 network. In [69], Panda *et*

¹It was later found that the refinement is inaccurate (see [61], [14], and Chapter 3 of this thesis).

al. propose a model for interferences between neighboring cells of IEEE 802.11 WLANs.

Foh and Zukerman [12] provide a model that extends Bianchi's model to statistical traffic conditions. This model uses a state dependent single server queue; the arrival of this queue is the arrival process from the stations, and the service makes use of the results from Bianchi's model. It approximates the queue's service using phase type (PH) distribution (e.g. Erlang). We provide a detailed description of this framework in Section 2.5. Other attempts to model DCF under statistical traffic appear in [70, 71], which model each station as a $G/G/1$ queue.

There are many applications of these models in the literatures. In [72], Bianchi and Tinnirello develop a Kalman filter to estimate the number of competing stations using the results from Bianchi's model. In [73], Patil and Apte uses Foh and Zukerman's Markovian Framework model to find the maximum number of users that can be supported by an access point.

2.1.4 Quality of Service Support in Wireless LANs

Pattara-atikom et al. [74] provide a survey of recent advancements in WLANs QoS; they provide details on many differentiation methods such as inter-frame spacing differentiation [75], contention window differentiation [76], distributed fair scheduling [77], and distributed deficit round robin [78]. There are also methods that utilize both inter-frame spacing and contention window differentiations [6, 79]. In [80], Suzuki and Tasaka evaluate the performance of PCF, which is a centralized protocol, in supporting real-time traffic.

Lindgren et al. [81] provide comparison on popular schemes to achieve service

differentiation. They compare PCF [1], EDCA [6], Distributed Fair Scheduling (DFS) [77] and the Blackburst [82] using simulation studies. They observe that EDCA has better performance compared with PCF. They also observe that Blackburst has the best performance. However, the requirement of constant access interval limits the usefulness of this protocol. Gu and Zhang [83] evaluate the performance of IEEE 802.11e through simulations. Mangold et al. [84] provide simulation studies of the IEEE 802.11e and compare the performance with that of IEEE 802.11.

Ranasinghe et al. [85] propose the adaptation of deficit round robin (DRR) [86] to provide fairness in the IEEE 802.11 WLANs. Ganz et al. [51] introduce a superpoll scheme for the IEEE 802.11 PCF to support multimedia applications. Ergen *et al.* [87] propose a token ring protocol, which is built on top of DCF, that can guarantee bounded latency and reserved bandwidth for real-time applications.

Xiao [88,89] provides and analyzes a simple priority scheme that differentiates the initial window size, the factor for increasing the window size, and the maximum backoff stage. The contribution of this paper is the simple analytical model to derive the saturation throughput and delay of the system.

Some models have also been proposed to analyze the EDCA mechanism. Xiao [61, 90] developed a model to analyze the contention window size differentiation in the EDCA mechanism. However, this model lacks the *Arbitration Inter Frame Space* (AIFS) differentiation and *virtual collision* mechanism specified in the IEEE 802.11e standard. Robinson and Randhawa [91] extend the Bianchi's model to analyze the saturation throughput performance of the EDCA mechanism. Kong *et al.* [92] also extend the Bianchi's model for EDCA analysis; in this paper, they develop recursive solution to analyze the delay performance. Bianchi

and Tinnirello [93] provide a probabilistic approach in an analytical model for schemes that differentiate the interframe space. There are many other attempts to model EDCA [94, 95, 96, 97]. These models can accurately predict the saturation throughput performance of EDCA. However, most of these models lack the derivation of the delay performance.

2.2 IEEE 802.11 MAC Protocol

The IEEE 802.11 [1] specifies two mechanisms for transmission: *distributed coordination function* (DCF) and *point coordination function* (PCF). DCF uses of CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) with binary exponential backoff for contention purpose. PCF is an optional contention-free protocol which uses polling by the centralized *point coordinator* or *access point* (AP).

2.2.1 Distributed Coordination Function

DCF employs two methods for frame transmissions. The default is a two-way handshaking method called *basic access method*. The other technique employs a four-way handshaking method; this technique is known as *Request-To-Send / Clear-To-Send* (RTS/CTS).

In basic access mechanism (Figure 2.1a), when a station has a new frame to transmit, it monitors the channel activity. If the channel is detected idle for a period of time called *distributed inter frame space* (DIFS), the station transmits immediately. If the channel is busy, the station defers the frame transmission and selects a random backoff interval from a uniform distribution. The backoff counter

(a)

(b)

Figure 2.1: IEEE 802.11 DCF: (a) Basic access method. (b) Four-way handshaking method.

is decremented when the channel is sensed idle, frozen when channel activity is detected, and reactivated when the channel is sensed idle for more than a DIFS again. The station transmits its frame when the backoff counter reaches zero.

DCF uses a slotted binary exponential backoff technique. The period following an idle DIFS is slotted and the backoff time is measured in terms of slot time. The slot time is the time needed for a station to detect a transmission from other stations. It accounts for the propagation delay, the time needed to switch from receiving to transmitting state and the time to notify the MAC layer about the state of the channel.

The backoff counter is uniformly chosen in the range of $(0, CW-1)$, where CW is the current contention window. At the first transmission attempt, CW is set equal to the minimum contention window (CW_{min}). After each unsuccessful transmission, CW is doubled until it reaches the maximum contention window (CW_{max}).

When the destination station successfully receives a frame, it transmits an acknowledgment (ACK) frame after a *short inter frame space* (SIFS). If the sender does not receive the ACK frame within a specified ACK timeout period, or it detects another transmission on the channel, it reschedules the frame transmission according to the previous backoff rules.

The transmission and backoff mechanism of RTS/CTS method (Figure 2.1b) is similar with the basic access method. In RTS/CTS method, when a station is ready to transmit a data frame, it first transmits a special RTS frame to reserve the channel. When the receiver receives the RTS frame, it responds with a CTS frame after a SIFS period. The sender transmits the data frame only if it correctly receives the CTS frame.

The RTS and CTS frames carry the information of the length of the frame to be transmitted, which is used to update a *network allocation vector* (NAV) by other stations. The NAV contains the information about the period of time in which the channel will remain busy; hence, a station that detects either RTS or CTS frame will delay its transmission to avoid collision. The RTS/CTS method improves the system performance with large frames, as it reduces the length of the frames involved in the collisions.

2.2.2 Point Coordination Function

The IEEE 802.11 standard also specifies the optional PCF (Figure 2.2) which is implemented on top of DCF. The PCF operation is contention free as it uses polling by the AP to schedule transmissions. In PCF, the AP use *point coordination inter frame space* (PIFS), instead of DIFS, to issue polls. The PIFS is longer

Figure 2.2: IEEE 802.11 PCF.

than SIFS but shorter than DIFS; hence, the AP can take control of the channel and stop all the asynchronous traffic while it issues polls and receives responses. In PCF, the ACK frame can be combined with a data or a poll frame, thus it has less transmission overhead compared to DCF.

2.3 IEEE 802.11e Enhanced Distributed Channel Access

EDCA is specified to provide QoS support on IEEE 802.11 WLANs. It supports up to four access categories (AC), from the lowest priority service class, AC_0 , to the highest priority service class, AC_3 . EDCA differentiates service classes through three mechanisms: contention window size, *arbitrary inter frame space* (AIFS), and transmission opportunity (TXOP) limit differentiations.

Stations that use smaller CW_{min} and CW_{max} receive higher QoS than the other stations as their channel access delays are generally shorter. In EDCA, high priority service class uses smaller CW_{min} and CW_{max} to improve the QoS received by the higher priority classes.

In EDCA, AIFS is used instead of DIFS, where $AIFS = AIFS_n \times \text{slot time} + SIFS$. Each service class can use different AIFS value to differentiate the QoS received by the service class. Stations that use lower AIFS encounter fewer col-

Table 2.1: Typical parameters for various ACs.

AC	CW_{min}	CW_{max}	$AIFS_n$
Background (BK)	32	1024	7
Best Effort (BE)	32	1024	3
Video (VI)	16	32	2
Voice (VO)	8	16	2

Figure 2.3: Timing of the EDCA mechanism.

lisions and count down the backoff counter faster than the other stations; hence, they receive better QoS.

EDCA also allows stations to transmit multiple frames without contending again, known as contention free bursting (CFB). CFB is limited by the TXOP limit specified for each service class. Longer limit means that the service class can transmit more frames; hence, it receives better QoS.

In EDCA, each station implements a queue for each AC. Each queue has its own QoS parameters and backoff counter. A collision within a station is handled virtually, i.e. the frame from the highest priority queue involved in the collision is chosen and transmitted to the access medium. This mechanism is known as *virtual collision*.

Table 2.1 shows the suggested parameters for the various ACs in EDCA. These parameters are based on the IEEE 802.11b [3] parameters. We will use these

parameters in the subsequent analysis. Figure 2.3 shows the timing relations of AC_{VO} and AC_{BE} stations.

2.4 Bianchi's Saturation Analysis

In this section, we describe Bianchi's saturation analysis for WLANs throughput performance. He first publishes the analysis in a letter [9], which he later extends with more results in [8]. His method is popular because of the simplicity of the computation and the accuracy of the analysis. Even though the saturation assumption limits the scope of his method, it is still very useful for a wide range of purposes, for examples: network performance predictions, baseline for hardware performance evaluation, protocol optimizations, and admission control algorithms. As mentioned before, in his model Bianchi assumes no backoff freezing; hence, the backoff counter is decremented on every slot instead of on every idle slot as specified in the standard. He also assumes no retransmission limit, thus a frame will be retransmitted until it is successfully received.

Bianchi defines two stochastic processes to model the backoff counter progressions of a station: $s(t)$ model the progression of backoff stages, i.e. the number of failed transmission attempts of a frame, and $b(t)$ model the progression of backoff counters on each backoff stage, i.e. the counting down of backoff counters. For shorter notations, define W as CW_{min} . Let m denotes the *maximum backoff stage* where $CW_{max} = 2^m W$. The first stochastic process, $s(t)$, captures the value of the backoff stage at time t , hence $s(t) = \{0, 1, \dots, m\}$, whereas $b(t)$ has a range of $\{0, 1, \dots, 2^{s(t)} W\}$.

The key approximation of Bianchi's model is that a frame collides with con-

stant and independent probability p . With this assumption, the stochastic processes $\{s(t), b(t)\}$ of each station become independent of the other stations' backoff process; therefore, it is possible to model the backoff progression of each individual station. In Figure 2.4, we reproduce Bianchi's Markov chain model of the stochastic processes $\{s(t), b(t)\}$. In this section, we follow the notations in [8]; let $P\{i_1, j_1 | i_0, j_0\} = Pr\{s(t) = i_1, b(t) = j_1 | s(t) = i_0, b(t) = j_0\}$. The non-null transition probabilities of this Markov chain are

$$\begin{aligned} Pr\{i, j | i, j + 1\} &= 1 & j &= \{0, 1, \dots, W_i - 2\}, & i &= \{0, 1, \dots, m\} \\ Pr\{0, j | i, 0\} &= \frac{1-p}{W_0} & j &= \{0, 1, \dots, W_0 - 1\}, & i &= \{0, 1, \dots, m\} \\ Pr\{i, j | i - 1, 0\} &= \frac{p}{W_i} & j &= \{0, 1, \dots, W_i - 1\}, & i &= \{1, 2, \dots, m\} \\ Pr\{m, j | m, 0\} &= \frac{p}{W_m} & j &= \{0, 1, \dots, W_m - 1\}. \end{aligned} \quad (2.1)$$

Let $b_{i,j}$ be the stationary distribution of the Markov chain, where $i = \{0, 1, \dots, m\}$ and $j = \{0, 1, \dots, W_i - 1\}$. Solving for $b_{i,j}$, we have the following relations:

$$b_{i,0} = pb_{i-1,0}, \quad i = \{1, 2, \dots, m\} \quad (2.2)$$

$$b_{i,j} = \frac{W_i - j}{W_i} b_{i,0}, \quad i = \{0, 1, \dots, m\}, \quad j = \{0, 1, \dots, W_i - 1\}. \quad (2.3)$$

The value of $b_{0,0}$ is easily computed using the normalization condition

$$1 = \sum_{i=0}^m \sum_{j=0}^{W_i-1} b_{i,j}. \quad (2.4)$$

Figure 2.4: Bianchi's Markov chain model of a station backoff process in DCF.

Solving $b_{0,0}$ gives

$$b_{0,0} = \frac{2(1-2p)(1-p)}{(1-2p)(W+1) + pW(1-(2p)^m)}. \quad (2.5)$$

Define τ as the probability that a station transmits in a randomly chosen slot.

Having the stationary distribution of the chain, we can formulate τ as

$$\tau = \frac{2(1-2p)}{(1-2p)(W+1) + pW(1-(2p)^m)}. \quad (2.6)$$

Now, we are only missing the conditional collision probability p . A transmitted frame collides if at least one of the other stations also transmits a frame, which gives

$$p = 1 - (1 - \tau)^{n-1}, \quad (2.7)$$

where n is the total number of stations. The probabilities τ and p form a set of non-linear equations that can be computed numerically.

With τ and p defined, we can compute the probability P_{tr} that a slot contains at least one transmission by

$$P_{tr} = 1 - (1 - \tau)^n. \quad (2.8)$$

The probability P_s that a non-empty slot contains a successful transmission is then computed by

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{1 - (1-\tau)^n}. \quad (2.9)$$

Nanyang Technological University

Finally, the saturation throughput S is the ratio

$$S = \frac{E[\text{frame payload transmission time within a slot}]}{E[\text{length of a slot}]}. \quad (2.10)$$

Let $E[P]$ be the frame payload transmission time; the saturation throughput can then be computed by

$$S = \frac{P_s P_{tr} E[P]}{(1 - P_{tr})\sigma + P_s P_{tr} T_s + (1 - P_s) P_{tr} T_c}, \quad (2.11)$$

where σ , T_s and T_c are, respectively, the length of a slot time, the average length of a successful transmission, and the average length of a collision. The length of a slot time, σ , is specified in the standard. Assuming that *extended inter frame space* (EIFS) is not implemented, the values of T_s and T_c for basic access method and RTS/CTS method are given by

$$\begin{cases} T_s^{bas} = H + E[P] + SIFS + \delta + ACK + DIFS + \delta, \\ T_c^{bas} = H + E[P^*] + DIFS + \delta, \end{cases} \quad (2.12)$$

$$\begin{cases} T_s^{rts} = RTS + SIFS + \delta + CTS + SIFS + \delta + H \\ \quad + E[P] + SIFS + \delta + ACK + DIFS + \delta, \\ T_c^{rts} = RTS + DIFS + \delta, \end{cases} \quad (2.13)$$

where H denotes the physical and MAC headers, and δ denotes the propagation delay. The value $E[P^*]$ is the average length of the longest frame payload transmission time.

2.5 Markovian Framework for DCF Analysis

Foh and Zukerman introduce a framework to analyze the performance of DCF under statistical traffic in [12]. In this framework, they model the interactions among frame transmitting stations into a *single server queue* (SSQ). An arrival to this SSQ models a frame generated to the MAC layer of one of the stations, whereas a departure from the SSQ models a successful frame transmission by one of the stations. They further use the common assumption that the MAC layer holds only one frame; thus, the queue size of the SSQ represents the number of *active stations*, i.e. the stations that have frames to transmit.

The arrival process of the SSQ is, obviously, the aggregate of the arrival processes of the stations. The arrival rate of the SSQ depends on the number of *idle stations*, i.e. the stations that currently have no frames to transmit, which can be determined from the queue size of the SSQ. On the other hand, the service process depends on the DCF protocol operation. Noting that a network with n active stations is actually a network in a temporary state with n saturated stations, they use the saturation analysis by Bianchi to determine the service rate parameters of the SSQ. Their experiments further show that the service process of DCF can be fitted with a phase-type distribution such as the Erlang distribution. Therefore, assuming Poisson arrivals and a total of k stations, the SSQ is of type state-dependent M/PH/1/k (SD-M/PH/1/k).

Figure 2.5 shows the state transition diagram of a state-dependent M/M/1/k SSQ. The state-dependent arrival rate, $\lambda(n)$, is given by

$$\lambda(n) = (k - n)\lambda_s, \quad n = 0, 1, \dots, k, \quad (2.14)$$

Figure 2.5: State transition diagram of SD-M/M/1/k SSQ.

where λ_s is the arrival rate of the individual station, n is the queue size of the SSQ, and k is the total number of stations. The service rate of the SSQ, $\mu(n)$, is directly related to the throughput measure of the Bianchi analysis given in (2.11); let S_n denotes the saturation throughput of a network with n stations, the service rate is computed with

$$\mu(n) = \frac{S_n}{E[P]}, \quad n = 1, 2, \dots, k, \quad (2.15)$$

where $E[P]$ is the frame payload transmission time.

With a constant frame size, Foh and Zukerman show that the service process can be approximated with Erlang distribution. Modeling DCF protocol with an SSQ of type SD-M/E_j/1/k, we can efficiently compute the performance measures such as the throughput and delay performances. Let $p_{n,i}$ denote the probability that the system has n active stations and is in the i -th Erlang service stage, where $n = 0, 1, \dots, k$ and $i = 1, 2, \dots, j$. Note that $p_{0,i} = 0$ for $i > 1$. Assume that the invalid states, i.e. states with $n < 0$, $n > k$, $i < 1$, and $i > j$, have probabilities of zero. The steady state balance equations for this SSQ are

$$0 = -\lambda(n)p_{n,0} - j\mu(n)p_{n,1} + \lambda(n-1)p_{n-1,0} + j\mu(n+1)p_{n+1,j}, \quad (2.16)$$

$$0 = -\lambda(n)p_{n,i} - j\mu(n)p_{n,i} + \lambda(n-1)p_{n-1,i} + j\mu(n)p_{n,i-1}, \quad (2.17)$$

Chapter 2. Literature Review

and the probabilities normalization:

$$\sum_{n=0}^k \sum_{i=1}^j p_{n,i} = 1. \quad (2.18)$$

Having these, we can compute the mean arrival rate, $\bar{\lambda}$, of the system by

$$\bar{\lambda} = \sum_{n=0}^k \sum_{i=1}^j (p_{n,i} \lambda(n)). \quad (2.19)$$

Thus, the system throughput is given by

$$\rho = \bar{\lambda} E[P], \quad (2.20)$$

and the mean delay can be computed simply with

$$E[D] = \frac{1}{\bar{\lambda}}. \quad (2.21)$$

Chapter 3

Refinement of DCF Saturation Model

Chapter 2 has illustrated the progress made in wireless MAC protocols. The introduction of the IEEE 802.11 standards have popularized wireless networking to the current extent that all new laptops are equipped with a wireless networking device. This popularity has in turn generated much research interests which drive the development of more efficient protocols and support for QoS demanding applications.

Building upon past researches, this thesis is written to a goal of developing efficient MAC protocols for future high speed wireless networks. During the development of these protocols, we have made several contributions to the current effort on modeling the IEEE 802.11 DCF and EDCA protocols. This chapter serves as the opening of this thesis contents; in this chapter, we detail our contribution to the effort of modeling IEEE 802.11 DCF.

The bi-dimensional Markov Chain modeling introduced by Bianchi [8] for the

analysis of the IEEE 802.11 saturation throughput has become a common method to study the performance of the IEEE 802.11 Medium Control Access (MAC) protocol and its enhancements. The model was later refined to capture further details of the IEEE 802.11 protocol operations. Among the refinements, one is due to Ziouva and Antonakopoulos [13] aiming to capture the freezing of backoff counters when the broadcast channel is sensed busy by a station. Precisely, when a channel turns idle from busy due to, for example, a Distributed InterFrame Space (DIFS), Bianchi's model assumes that each station immediately reactivates and decrements its counter, whereas the IEEE 802.11 standard specifies that a backoff counter is decremented only after the channel continues to remain idle for a predefined slot time. The refinement reported in [13] was, however, introduced without realizing that the two key probabilities governing the performance, namely the channel access probability, τ , and the station collision probability, p , depend on the channel status. This inaccuracy in the model affects several important measures including the probabilities that a particular time period on the broadcast channel is an idle, a successful transmission, or a transmission collision period. Owing to the large difference in the duration of the different types of time periods, the inaccuracy does not reflect significantly in the final saturation throughput results in most common cases; however, fundamentally, the model is in error.

Here, we present a new model that corrects that of [13] by evaluating the channel access probabilities and the station collision probabilities conditioned upon the channel status. We then show the accuracy of our results via computer simulation, and demonstrate the errors if such details are ignored.

3.1 Saturation Throughput Model

The mechanism of the IEEE 802.11 Distributed Coordination Function (DCF) with basic access method is shown in Figure 3.1. It differs from the model presented in [8] in the decrement of the backoff counter. The IEEE 802.11 standard [1] specifies that a station freezes its backoff counter when it detects a transmission on the channel (note that the backoff counter is not decreased during the channel busy period). This backoff freezing procedure directly affects the probability that a station accesses the channel, and this probability also depends on whether the previous period is busy or idle.

To understand this, we first consider the channel access event after a busy period due to a collision. After a collision, since stations that did not participate in this collision had frozen their backoff counters, they will not access the channel after the busy period; only those suffered a collision may access the channel if their newly chosen backoff counter is zero. Hence, only a group of stations rather than all stations may access the channel after a busy period. In case of a successful transmission, only one station, which performed the successful transmission, may access the channel after the successful transmission period. As opposed to the case of a busy period, after an idle period, all stations whose backoff counters are decremented to zero will access the channel; hence, it is obvious that the channel access probability actually depends on whether the previous period is idle or busy. This is not modeled in [13] where the derived channel access probability is not conditioned upon the type of the previous time period.

The new model describing the backoff process of a station for the saturation network condition is presented in Figure 3.2. Here we consider a network of n

Nanyang Technological University

Figure 3.1: IEEE 802.11 basic access method.

Figure 3.2: The Markov chain representation of the new model.

Chapter 3. DCF Model Refinement

saturated stations. The state $\{i, j, k\}$ represents the state of a station at a particular time period, where i indicates the type of the previous period, either idle or busy ($i = 0$ or $i = 1$ respectively); j indicates the current backoff stage ($j = 0, 1, \dots, m$); and k indicates the current backoff counter ($k = 0, 1, \dots, W_j - 1$). Two important protocol parameters describing the backoff process are the minimum backoff window denoted by W_0 , and the maximum backoff stage denoted by m . The backoff window of a station at the j -th backoff stage is W_j , or precisely $2^j W_0$.

Two key probabilities governing the backoff process are first defined, they are, p_0 (p_1), the probability that, from a station point of view, at least one other station transmits during a slot after an idle (a busy) period. We further define q_0 (q_1) to be the probability that the broadcast channel remains idle after an idle (a busy) period.

Given that the channel is either an idle or a busy period, let P_i be the probability that a particular period on the channel is idle, then $P_i = q_0 P_i + q_1 (1 - P_i)$, which gives

$$P_i = \frac{q_1}{1 - q_0 + q_1}. \quad (3.1)$$

Let $b_{i,j,k}$ be the stationary distribution of the described Markov Chain. Each

Nanyang Technological University

of the state stationary probability can be expressed in terms of $b_{1,0,0}$ as

$$\begin{aligned}
 b_{1,j,0} &= \psi_j b_{1,0,0} && \text{for } j = 1, 2, \dots, m, \\
 b_{1,j,k} &= \frac{1 + p_0(W_j - 1 - k)}{1 - p_1} \psi_j b_{1,0,0} \\
 &&& \text{for } k = 1, 2, \dots, W_j - 1 \text{ and } j = 0, 1, \dots, m, \\
 b_{0,j,k} &= (W_j - 1 - k) \psi_j b_{1,0,0} \\
 &&& \text{for } k = 0, 1, \dots, W_j - 2 \text{ and } j = 0, 1, \dots, m,
 \end{aligned} \tag{3.2}$$

where

$$\psi_j = \begin{cases} 1 & \text{if } j = 0, \\ \frac{p_0(W_0 - 1)}{W_1} & \text{if } j = 1, \\ \frac{p_0(W_0 - 1)}{W_1} \pi_j & \text{if } j = 2, 3, \dots, m - 1, \\ \frac{p_0(W_0 - 1)}{W_1} \pi_j \frac{W_m}{W_m - p_1 - p_0(W_m - 1)} & \text{if } j = m, \end{cases}$$

and

$$\pi_j = \prod_{x=2}^j \left[\frac{p_1}{W_x} + \frac{p_0}{W_x} (W_{x-1} - 1) \right]$$

with

$$\sum_{j=0}^m \left[\sum_{k=0}^{W_j-2} b_{0,j,k} + \sum_{k=0}^{W_j-1} b_{1,j,k} \right] = 1.$$

Define τ_i and τ_b to be the probabilities that a station accesses the broadcast channel after an idle and a busy period respectively. Similar to [13] and [8], each of τ_i and τ_b can be expressed as a function of the stationary probabilities. They

Chapter 3. DCF Model Refinement

are given by

$$\begin{aligned}\tau_i &= \frac{\sum_{j=0}^m b_{0,j,0}}{\frac{q_1}{1-q_0+q_1}}, \\ \tau_b &= \frac{\sum_{j=0}^m b_{1,j,0}}{1 - \frac{q_1}{1-q_0+q_1}}.\end{aligned}\tag{3.3}$$

Having obtained τ_i and τ_b , q_0 (q_1) can be determined based on the fact that the broadcast channel remains idle after an idle (a busy) period if no station accesses the channel. Thus we get

$$\begin{aligned}q_0 &= (1 - \tau_i)^n, \\ q_1 &= (1 - \tau_b)^n.\end{aligned}\tag{3.4}$$

A particular station finds a slot to be busy if at least one of the other stations access the channel. Hence we have

$$\begin{aligned}p_0 &= 1 - (1 - \tau_i)^{n-1}, \\ p_1 &= 1 - (1 - \tau_b)^{n-1}.\end{aligned}\tag{3.5}$$

The system throughput S , the fraction of time used for successful payload transmission, can be expressed as

$$S = \frac{P_s E[P]}{P_i \sigma + P_s T_s + P_c T_c},\tag{3.6}$$

where $E[P]$ is the average payload length, σ is the duration of an empty slot time, T_s is the average time that the channel is sensed busy because of a successful transmission, and T_c is the average time that the channel is sensed busy due to a collision. These quantities for the basic and the RTS/CTS access methods are given by (14) and (17) in [8] (these quantities are also provided in Section 2.4).

The probabilities P_i , P_s and P_c describe the probabilities that a particular pe-

Table 3.1: Parameters used for numerical analysis and simulations.

Channel bit rate	1 Mbps
Propagation delay	1 μ s
Slot time	50 μ s
SIFS	28 μ s
DIFS	128 μ s
Frame payload	8184 bits

riod on the channel carries no transmission (idle), a successful data frame transmission, and two or more transmissions (collision), respectively. Probability P_i is given by (3.1) and probabilities P_s and P_c can be expressed as

$$\begin{aligned}
 P_s &= n\tau_i(1 - \tau_i)^{n-1}P_i + n\tau_b(1 - \tau_b)^{n-1}(1 - P_i), \\
 P_c &= 1 - P_i - P_s.
 \end{aligned} \tag{3.7}$$

3.2 Numerical Results

In Figures 3.3-3.4, numerical results for the saturation throughput along with several important probabilities obtained from our model are plotted and compared with that of [13]¹. We use the same protocol parameters as [8] for this comparison (see table 3.1).

The numerical results are obtained using the fixed point iteration technique [98]. In brief, initial guessing for $\tau_i^{(0)}$ and $\tau_b^{(0)}$ were first made, then p_0 and p_1 were computed by (3.5), and later used to calculate $b_{i,j,k}$ using (3.2). After that, new values for τ_i^* and τ_b^* were obtained by (3.3) and (3.4). We finally updated $\tau_i^{(1)}$

¹Two modifications were applied to the original model, which are (i) the special state, $\{-1, 0\}$, modeled in [13] is said to be inconsistency with the standard [61], it was removed; and (ii) equation (8) in [13] is corrected as follows: $p_b = 1 - (1 - \tau)^{n-1}$.

Chapter 3. DCF Model Refinement

Figure 3.3: Probabilities of an idle, a successful transmission, and a collision of various models for $W = 16$ and $m = 6$.

and $\tau_b^{(1)}$ for the next iteration by $\tau_i^{(1)} = 0.5\tau_i^{(0)} + 0.5\tau_i^*$ and $\tau_b^{(1)} = 0.5\tau_b^{(0)} + 0.5\tau_b^*$.

The simulation results (shown with symbols) are obtained with a 95% confidence interval of 0.1% above and below of the plotted mean. Our simulator is developed in C++ language and is verified wherever possible with the models in [8] and [13]. As in [8], our model assumes that after the completion of a transmission, each station detects a collision after a DIFS period. This assumption neglects the EIFS mechanism, which does not affect the operations of the protocol except for the length of the transmission period. The numerical computation takes less than one second on an Intel Pentium 4 machine, whereas a simulation run takes a few minutes on the same machine.

As can be seen in Figures 3.3-3.4, our model gives accurate results for the performance of IEEE 802.11 with freezing of backoff counters. For the model of [13], only small errors in saturation throughput are produced for the case of the

Figure 3.4: Saturation throughput of various models for $W = 16$ and $m = 6$.

RTS/CTS method as data frame transmission duration is significantly longer than other time periods, where the model of [13] appears to be accurate if not carefully studied.

3.3 Summary

In this chapter, we introduce a refinement to the existing saturation analysis for the IEEE 802.11 DCF which corrects the backoff freezing mechanism previously proposed by Ziouva and Antonakopoulos. The obvious benefit from this model is more accurate computation of the underlying governing probabilities: probabilities of idle slots, successful transmissions, and collisions. Accurate probabilities are important in the derivations of frame dropping probability and frame delay distribution.

Chapter 4

Saturation Analysis of EDCA

In this chapter, we first introduce a model for the EDCA mechanism under saturation condition, which conforms to the final version of the EDCA standard. This model captures the operation of the AIFS and contention window differentiation of the EDCA mechanism. We then develop a model to compute the delay distribution of the EDCA scheme. This delay model is based on the idea that the exponential backoff scheme employed by EDCA is equivalent to the p -persistent backoff scheme with a particular computable parameter p [8]. We verify our analytical models with simulations.

4.1 Analytical Models

4.1.1 Saturation Model

In EDCA, a station freezes its backoff counter when there is a transmission on the channel; it will continue its backoff count down after an empty AIFS period, which length depends on the priority of the station. EDCA has a minor difference

from DCF in the backoff process: in EDCA, the backoff counter is unfrozen a slot time before the empty AIFS period end in contrast with DCF, which backoff counter is unfrozen after the DIFS period end [14].

First, let slot denotes a time period during which a channel either contains a successful frame transmission, a collision, or no transmission. The length of a slot is not uniform. A slot that contains a successful transmission or a collision has a length that includes the frame transmission time, acknowledgment, and the physical layer overheads. The slot that contains no transmission (*idle slot*) has a length that is equal with the channel slot time specified in the standard (as explained before, the IEEE 802.11 divides the channel time into slots). We also define *busy slot* as the slot that either contains a successful transmission or a collision. The periods of successful slot T_S and collision slot T_C are given by

$$\begin{aligned}
 T_S^{bas} &= T_H + T_P + \delta + T_{SIFS} + T_{ACK} + \delta + T_{SIFS}, \\
 T_C^{bas} &= T_H + T_P^* + \delta + T_{DIFS}, \\
 T_S^{rts} &= T_{RTS} + \delta + T_{SIFS} + T_{CTS} + \delta + T_{SIFS} + \\
 &\quad + T_H + T_P + \delta + T_{SIFS} + T_{ACK} + \delta + T_{SIFS}, \\
 T_C^{rts} &= T_{RTS} + \delta + T_{DIFS}
 \end{aligned} \tag{4.1}$$

for basic access method and RTS/CTS method respectively. Note that we do not consider the full AIFS period in the busy slot lengths above; the AIFS period provides service differentiation, which needs a separate treatment below.

Next, we define *frozen* and *unfrozen slots* as the slot when a station freezes its backoff counter and when a station continues its backoff process respectively (this view is from the channel point of view as channel time is slotted). The frozen slots

include the busy slots and AIFS slots, which is different for stations of different classes. We also define frozen and unfrozen state, which are the states where a station freezes and defrost its backoff counter as viewed by the station. Let η be the transition probability from a frozen state to the corresponding unfrozen state, and let θ be the transition probability from an unfrozen state to the next unfrozen state. Using similar notations with the notations in [8], we define p as the probability that a transmitted frame encounter a collision. Similar with [8], we assume that the probability p is constant and independent of the number of retransmission.

Figure 4.1 shows the Markov chain model of the backoff process of a station in EDCA. In this figure, state $\{i, j, k\}$ models a station in the i -th backoff stage with j as the current backoff counter, and the station is in frozen ($k = 1$) or unfrozen ($k = 0$) state. As defined before, the transition probability from a frozen state to the corresponding unfrozen state is given by η , and the transition probability from an unfrozen state to the next unfrozen state is given by θ .

The variable i ranges from 0 as the first backoff stage to r the retransmission limit. The value j ranges from 0 to $W_i - 1$, where W_i is the backoff window of stage i ; the backoff window W_i is given by

$$W_i = \begin{cases} 2^i \cdot CW_{min}, & 0 \leq i \leq m \\ 2^m \cdot CW_{min}, & m < i \leq r. \end{cases} \quad (4.2)$$

In this chapter, we also use a notation of $W_{c,i}$ to signify that the window size is of AC_c.

Let $b_{i,j,k}$ be the stationary distribution of the Markov chain. Owing to the chain

Figure 4.1: Markov chain model of the EDCA backoff process.

regularities, we have

$$\begin{aligned}
 b_{i,0,0} &= p \cdot b_{i-1,0,0}, & 0 < i \leq r \\
 b_{i,j,0} &= \frac{W_i - j}{W_i} \cdot b_{i,0,0}, & 0 \leq i \leq r, 0 < j < W_i \\
 b_{i,j,1} &= \frac{W_i - j - (W_i - j - 1)\theta}{W_i \eta} \cdot b_{i,0,0}, & 0 \leq i \leq r, 0 < j < W_i.
 \end{aligned} \tag{4.3}$$

With normalization condition $\sum_{i,j,k} b_{i,j,k} = 1$, we obtain

$$b_{0,0,0} = 2\eta(1-2p)(1-p) / \left((1+\eta+\theta)(1-2p)(1-p^{r+1}) + (1+\eta-\theta)W_0 \left[(1-(2p)^{m+1})(1-p) + 2^m p^{m+1}(1-2p)(1-p^{r-m}) \right] \right). \quad (4.4)$$

The unconditional probabilities τ_c that a station of AC_c transmits in a slot are given by

$$\begin{aligned} \tau_c &= \sum_{i=0}^{r_c} b_{i,0,0} \\ &= \frac{(1-p_c^{r_c+1})}{(1-p_c)} b_{0,0,0}. \end{aligned} \quad (4.5)$$

To compute τ , we still need to derive the probabilities η , θ , and p . Firstly, observe that the transition probability θ is related with the collision probability p . The probability $1-p$ that a transmitted frame does not encounter a collision is equal with the probability that no other stations transmit in that slot, which allows the continuation of the backoff counter count down. From this observation, we have the relation

$$\theta_c = 1 - p_c. \quad (4.6)$$

Also observe that the probability of transmission τ is not uniform across the slots. As mentioned before the frozen slots for each AC are different. For example, stations of AC_{BE} (with parameters in table 2.1) cannot transmit during the first three idle slots after a busy slot. Let B denotes the busy slot, and A_k denotes the k -th slot after a busy slot. Also define q_v as the probability that the $(j+1)$ -th slot following j idle slot is an idle slot; hence, q_0 is the transition probability

from B to A_1 , q_1 is the transition probability from A_1 to A_2 , until q_ω , the transition probability from $A_{\omega-1}$ to A_ω (we defer the explanation of the transition probability q for the sake of the flow). Here, ω denotes the maximum of the AIFS _{n} (or e_n) of all the ACs (we use e_n as a shorter notation for AIFS _{n}). We aggregate the slots A_ω to A_∞ as we do not require the individual information for the subsequent calculations. With these definitions, the probability P_{A_k} that a slot is an A_k slot can be computed by

$$P_{A_k} = \begin{cases} P_B \prod_{j < k} q_j, & k < \omega \\ \frac{P_B \prod_{j < \omega} q_j}{1 - q_\omega}, & k = \omega. \end{cases} \quad (4.7)$$

Applying the normalization condition $P_B + \sum_{k=0}^{\omega} P_{A_k} = 1$, the probability P_B that a slot is busy given by

$$P_B = \frac{1}{1 + \sum_{k < \omega} \prod_{j < k} q_j + \left(\prod_{j < \omega} q_j \right) / (1 - q_\omega)}. \quad (4.8)$$

Next, we condition τ as not all of the stations can transmit on some slots. We condition τ on whether the previous slot is the last frozen slot or is one of the unfrozen slot. Remember that a station can transmit after the last frozen slot as the backoff counter is decremented on the last frozen slot. The conditioned transmission probability κ is defined by

$$\kappa_c = \frac{\tau_c}{P_{T_c}}, \quad (4.9)$$

where P_{T_c} is the total probability of the last frozen slot and the unfrozen slots for

stations of AC_c given by

$$P_{T_c} = \sum_{k \geq e_c} P_{A_k}. \quad (4.10)$$

The transition probability q_v is actually the probability that the $v + 1$ -slot after a busy slot does not contain any transmission, which can be formulated by

$$q_v = \prod_{\forall c, e_c \leq v} (1 - \kappa_c)^{n_c}. \quad (4.11)$$

Note that the probability that the the $v + 1$ -slot after a busy slot is a busy slot is given by $1 - q_v$.

With the above definitions and derivations, we can now formulate the probabilities η and p . The transition probability η from a frozen state to the unfrozen state in the Markov chain is actually the transition probability from a busy slot to the first unfrozen slot. With the slot transition probability q defined, η can be computed by

$$\eta_c = \frac{\prod_{k < e_c} q_k}{1 + \sum_{k < e_c - 1} \prod_{j < k} q_j}. \quad (4.12)$$

The collision probability p_c is easily computed by proper conditioning and unconditioning, which is given by

$$p_c = \frac{\sum_{k \geq e_c} \left[P_{A_k} \left((1 - (1 - \kappa_c)^{n_c - 1} \prod_{\substack{\forall j, e_j \geq k \\ j \neq c}} (1 - \kappa_j)^{n_j} \right) \right]}{P_{T_c}}. \quad (4.13)$$

Equations (4.5), (4.12), and (4.13) form a set of non-linear equations that can

Figure 4.2: Probability that a slot contains a successful transmission by stations of each AC.

be computed numerically using, for example, fixed point iteration technique [98].

The probabilities P_{S_c} that a slot contains a successful transmission by a station of class AC_c are given by

$$P_{S_c} = \sum_{k \geq e_c} \left(P_{A_k} n_c \kappa_c (1 - \kappa_c)^{n_c - 1} \prod_{\substack{\forall j, e_j \geq k \\ j \neq c}} (1 - \kappa_j)^{n_j} \right). \quad (4.14)$$

The probability P_C that a slot contains a collision is $P_C = 1 - P_I - P_S$, where $P_S = \sum_{i=0}^3 P_{S_c}$. Figures 4.2 and 4.3 show the computed value of P_{S_c} , P_C , and P_I compared with the simulation results. We observe from the figures a good match between simulation results and analytical results, thus confirming the accuracy of our analytical framework

Figure 4.3: Probability that a slot is an idle slot and probability that a slot contains a colliding transmission.

4.1.2 Saturation Throughput

The saturation throughput for stations of AC_c is given by

$$S_c = \frac{P_{S_c} E[T_P]}{E[T_{slot}]}, \quad (4.15)$$

where P_{S_c} is the probability that a slot contains a successful transmission by stations of AC_c given by (4.14), $E[T_P]$ is the average length of frame payload, and $E[T_{slot}]$ is the average length of a slot. The value $E[T_{slot}]$ is computed by

$$E[T_{slot}] = P_I \sigma + P_S E[T_S] + P_C E[T_C], \quad (4.16)$$

where σ is the length of the channel slot time, T_S is the length of a successful slot, and T_C is the length of a collision slot. The value T_S and T_C are given by (4.1).

4.1.3 Frame Dropping Probability

A frame will be dropped by a station when the station is in the last backoff stage and the frame encounters a collision. This frame dropping occurs with probability P_{drop} , which is the probability that a frame encounters $r + 1$ collisions. The probability P_{drop} is given by

$$P_{drop,c} = p_c^{r_c+1}. \quad (4.17)$$

4.1.4 Saturation Delay

Here, we use z-transform to derive the saturation delay performance of EDCA. Previously, Zanella and Pellegrini [64] also use z-transform analysis to derive the delay distribution of DCF. Z-transform is used here to reduce the complexity of the model and to ease the burden of computing the numerical results especially the probability mass function. The usefulness of working in z-transform domain is apparent in the easeness of the manipulation and the compounding of the probabilities. As z-transform is a discrete transform, we need to define a discrete step for the transform. The choice of the discrete step is arbitrary depending on the required granularity of the results.

In EDCA, stations encounter multiple backoff stages until a frame is successfully transmitted or dropped after a number of retransmissions. Let $T_B(v)$ be the frame transmission delay if the frame is successfully transmitted in backoff stage v . The overall frame transmission delay can be computed by summing these delays adjusted with the probability to successfully transmit in those stages. Written mathematically in z-transform domain, the frame transmission delay for stations

of AC_c is given by

$$G_{D_c}(z) = \sum_{v=0}^{r_c} P\{C = v\} E[z^{T_{B_c}(v)}], \quad (4.18)$$

where $P\{C = v\}$ is the probability that a frame encounters v collisions before it is successfully transmitted. Substituting $P\{C = v\}$ gives

$$\begin{aligned} G_{D_c}(z) &= \sum_{v=0}^{r_c} \left(\frac{p_c^v (1 - p_c)}{(1 - p_c^{r_c+1})} E[z^{T_{B_c}(v)}] \right) \\ &= \frac{1 - p_c}{(1 - p_c^{r_c+1})} \sum_{v=0}^{r_c} (p_c^v E[z^{T_{B_c}(v)}]), \end{aligned} \quad (4.19)$$

where the probability p_c is the frame collision probability for AC_c given in (4.13).

The frame transmission delay in a backoff stage depends on the number of encountered slots in that stage. The number of encountered slots itself depends on the station's window size in the backoff stage and the number of encountered backoff freezing. Lets define $N_{S_c(v)}$ as the random variable of the number of backoff slots a station encounter until backoff stage v . Let T_{slot} denotes the length of a slot¹. The frame transmission delay until backoff stage v , $T_B(v)$, is then can be computed by

$$T_{B_c}(v) = \begin{cases} 0 & \text{if } N_{S_c(v)} = 0 \\ \sum_{j=1}^{N_{S_c(v)}} T_{slot} & \text{if } N_{S_c(v)} > 0 \end{cases}. \quad (4.20)$$

¹As mentioned earlier, backoff process in EDCA is slotted. However, a slot might be an idle slot, a successful transmission, or a colliding transmission. The mean of T_{slot} is given in (4.16). We defer the derivation of the distribution T_{slot} to the later part.

Nanyang Technological University

The transform representation of $T_B(v)$ is

$$\begin{aligned} G_{T_B(v)} &= E[z^{\sum_{j=1}^{N_{S_c(v)}} T_{slot}}] \\ &= G_{N_{S_c(v)}}(G_{T_{slot}}). \end{aligned} \quad (4.21)$$

To compute the number of encountered slots, firstly, note that from the Markov chain model, we have the conditional collision probability, p , and the station's slot access probability, τ . With exponential backoff scheme, different backoff stage has different slot access probability, with the final slot access probability computed from each stage's probability by unconditioning on the probability to reach that stage. Here, we denote the slot access probability of stations from the AC_c in the backoff stage i as $\tau_{c,i}$, which is computed by

$$\tau_{c,i} = \frac{2\eta_c}{(\eta_c + 1 - \theta_c)(W_{c,i} + 1) + 2\theta_c} \quad (4.22)$$

Secondly, with the parameters p and τ , we can construct an equivalent p-persistent CSMA process for each stage of the exponential backoff scheme. Let $X_{c,i}$ be the random variable of the number of slots that a station of AC_c encounters during the backoff period in backoff stage i including the the station's transmission slot. In the p-persistent model, $X_{c,i}$ is the geometric sequence of $\tau_{c,i}$ with a density function given by

$$P\{X_{c,i} = s\} = (1 - \tau_{c,i})^s \tau_{c,i}. \quad (4.23)$$

Figure 4.4: Illustration of the backoff process for delay computation.

The z -transform of $X_{c,i}$ is then given by

$$G_{X_{c,i}}(z) = \frac{\tau_{c,i}z}{1 - (1 - \tau_{c,i})z}. \quad (4.24)$$

Having $X_{c,i}$, the total number of backoff slots encountered until backoff stages v is

$$N_{S_c}(v) = \sum_{i=0}^v X_{c,i}. \quad (4.25)$$

The number of slots a station has encountered when it is in the v -th backoff stage

Nanyang Technological University

then can be computed in z-transform domain by

$$\begin{aligned} G_{N_{S_c(v)}}(z) &= \prod_{i=0}^v G_{X_{c,i}}(z) \\ &= \prod_{i=0}^v \frac{\tau_{c,i}z}{1 - (1 - \tau_{c,i})z} \end{aligned} \quad (4.26)$$

Figure 4.4 illustrates the backoff process, whereby we compute the total number of backoff slots $N_{S_c(v)}$ with regard to $\tau_{c,i}$. In the figure, we use BS_i as a short form of backoff stage i .

Substituting (4.21) and (4.26) to (4.19), we thus have

$$G_{D_c}(z) = \frac{1 - p_c}{(1 - p_c)^{r_c+1}} \sum_{v=0}^{r_c} \left(p_c^v \prod_{i=0}^v \frac{\tau_{c,i} G_{T_{slot}}(z)}{1 - (1 - \tau_{c,i}) G_{T_{slot}}(z)} \right) \quad (4.27)$$

We are left with the derivation of T_{slot} , which is simply computed in transform domain by

$$G_{T_{slot}}(z) = P_S G_{T_S}(z) + P_C G_{T_C}(z) + P_I z^\sigma. \quad (4.28)$$

4.2 Results and Discussions

4.2.1 Saturation Throughput

Saturation throughput of the four ACs are shown in Figures 4.5 and 4.6 for basic access method and RTS/CTS method respectively. The number of stations on the x-axis is the number of stations of each AC; hence, the total number of stations is four times of those number of stations. We use the parameters of IEEE 802.11b standard [3] for both the analysis and simulations. Table 4.1 summarizes the pa-

Table 4.1: Parameters used for numerical analysis and simulations.

Channel bit rate	11 Mbps
Propagation delay	1 μ s
Slot time	20 μ s
SIFS	10 μ s
DIFS	50 μ s
CW_{min}	32
CW_{max}	1024
Frame payload	8000 bits

Figure 4.5: Saturation throughput for EDCA basic access methods.

rameters that we use for both the analysis and simulations. The figure shows a close match between analytical and simulation results. The results indicate that EDCA can provide rate differentiation among stations of various ACs. We can observe excellent matches between analytical and simulation results for the EDCA saturation throughput. We see that EDCA basic access method can provide higher throughput with smaller number of stations. The numerical throughput computation takes less than one second on an Intel Pentium 4 machine, whereas a simula-

Figure 4.6: Saturation throughput for EDCA RTS/CTS methods.

Figure 4.7: Probability of dropping a frame.

tion run takes a few minutes on the same machine.

4.2.2 Frame Dropping Probability

Figure 4.7 shows the probability of dropping a frame by stations of AC_{VO} and AC_{VI} . We can also observe a close match between analytical and simulation results in these figures. Note that in saturation condition, EDCA can only support a few stations with acceptable frame dropping. This is in contrast with DCF that

can support higher number of stations. This reduction on the number of stations supported is caused by aggressive window size parameters of the stations of AC_{VO} and AC_{VI} . In real environments, stations transmitting real-time frames are unlikely to be saturated. Nevertheless, this saturation model is still useful for many applications, even to derive the performance of EDCA under non-saturation condition, such as the method in [99].

4.2.3 Delay Distribution

We obtain the delay distribution of EDCA by first computing the probability mass function. We compute the probability mass function by inverting (4.27) using the numerical inverse discrete time Fourier transform with $z = e^{j2\pi ft}$. We show the delay distribution of EDCA in Figure 4.8 for EDCA basic access method and in Figure 4.9 for EDCA RTS/CTS method. We observe a close match between simulation results and the analytical results, confirming the accuracy of our methods. The speed of the numerical computation for the delay distribution depends on the desired granularity. The numerical results in Figures 4.8 and 4.9 take less than 30 seconds to compute on an Intel Pentium 4 machine. The simulation results are collected from a two hours run on the same machine to achieve reliable simulation results.

4.2.4 Mean Frame Delay

We calculated the mean frame delay by computing the derivative of the z-transform equation (4.27) with $z = 1^-$. Figure 4.10 and 4.11 show the average delay of AC_{VO} and AC_{VI} for basic access method and RTS/CTS method respectively. The

Figure 4.8: Delay distribution of EDCA basic access method with five stations of each AC.

Figure 4.9: Delay distribution of EDCA RTS/CTS method with five stations of each AC.

Chapter 4. Analysis of EDCA

Figure 4.10: Saturation delay for basic access methods.

Figure 4.11: Saturation delay for RTS/CTS methods.

evidence of delay differentiation by EDCA is shown in the figure. Observe that RTS/CTS method can provide better delay performance with higher number of stations. With higher number of stations, there are more transmission collisions on the channel; hence, RTS/CTS method, which has lower collision overheads, exhibits better performance than that of basic access method.

4.3 Summary

In this chapter, we have introduced a model of the EDCA mechanism under saturation condition. This model captures the operation of the AIFS and contention window differentiation of the EDCA mechanism. Using this model, we analyzed the throughput and delay performances of the EDCA mechanism. Using z-transform analysis, we derive the delay distribution of EDCA. The results of our analytical model are then verified by simulations, which show the accuracy of our model.

Chapter 5

Out-of-Band Signaling Scheme

This chapter covers the *out-of-band signaling scheme* (OBS) that we proposed to improve the throughput and delay performance of WLANs. OBS separates the channel reservation process to a low bit rate channel. The basic idea behind the OBS scheme is that in DCF, idle periods and transmission collisions are the main overheads of data transmissions. These overheads become significant as the bit rate of the data channel increases [7]. By separating the contention and the actual data transmission onto two different channels, and using the low bit rate channel for the contention whereby the high bit rate channel for data transmissions, the costly idle periods and transmission collisions can be avoided entirely on the high bit rate channel. This will in turn improve the overall performance of WLANs.

5.1 Description of Out-of-Band Signaling Scheme

The OBS scheme can be briefly described as follows. OBS utilizes two channels in a WLAN, one of which operates at a low bit rate for channel assignment pur-

Figure 5.1: The OBS scheme illustrated.

poses, and another operates at a high bit rate for the actual data transmissions. We shall call the low and the high bit rate channels the signaling and the data channels, respectively. In our design, we may consider the use of the IEEE 802.11a/b/g channel as the data channel for OBS, and the signaling channel operating at a low bit rate¹ may be allocated from a free spectrum.

When a station is ready for a data transmission, it transmits a *Request for Transmission* (RFT) frame using IEEE 802.11 basic access method on the signaling channel. When the *access point* (AP) receives the RFT frame, it acknowledges the request and schedules the data transmission to the next PCF period on the data channel. The RFT frame is basically an IEEE 802.11 MAC control frame that uses one of the unallocated frame type in the IEEE 802.11 standard. We set the length of the RFT frame to be equal to the length of RTS frame in the RTS/CTS method. Figure 5.1 illustrates the operations of the OBS scheme.

The use of the OBS scheme also brings other advantages to WLANs. The

¹The low bit rate channel has a maximum bit rate that is lower than the data channel bit rate; Considering the same modulation technique for wireless transmission for both channels, a 12 Mbps signaling channel requires about 20% of the spectrum allocated for a 54 Mbps channel in IEEE 802.11a. Similar to the data channel, the bit rate for the signaling channel can be lowered by link adaptation algorithm because of distance or noise.

immediate benefit of OBS is its flexibility of transmission scheduling. The AP collects requests from stations on the signaling channel. The AP may perform service differentiation based on requested QoS and the network setting. A possible mechanism to achieve such service differentiation is the introduction of fair queuing algorithm such as *Deficit Round Robin* (DRR) [100] to ensure fairness or provide rate differentiation among stations.

Additionally, any improvement to the PCF protocol is applicable to OBS. Some examples include the combination of *ACK* or *Poll* frames with data frames, and use of superpoll [51] which further reduce transmission overheads.

One important characteristics of OBS is its backward compatibility with the current IEEE 802.11 standard. While OBS introduces two channels, each of these channel operates the standard DCF and PCF respectively. Therefore, in WLANs implementing OBS, the OBS-enabled AP can accommodate the existing IEEE 802.11 stations by sharing the data channel, which is similar with the current super frame mechanism described in the IEEE 802.11 standard [1] for the coexistence of DCF and PCF. The only drawback for coexistence of the existing IEEE 802.11 and OBS-enabled stations is that the existing IEEE 802.11 stations will not enjoy the performance benefits of OBS which introduces unfairness between the two groups.

5.2 Saturation Analysis of Out-of-Band Signaling Scheme

In this section, we study the saturation performance of the OBS scheme. In brief, we first model the operation of the protocol using the Markovian Framework where an equivalent continuous time Markov Chain is developed for the study of the protocol. We then study the protocol performance by analyzing the developed Markov Chain.

5.2.1 Markovian Framework for Saturation Analysis

We define a *ready station* to be a station that is currently scheduling its RFT transmission on the signaling channel, a *backlogged station* to be a station that has exchanged its RFT/ACK message and currently waiting for a poll message from the AP on the data channel, and an *idle station* to be a station that has no frame for transmissions. An idle station switches to a ready station when it generates a frame into its local buffer. A ready station switches to a backlogged station when it has successfully exchanged the RFT/ACK message with the AP. After a successful frame transmission, a backlogged station switches to either an idle station if there is no more frame to transmit, or a ready station if there is at least one frame awaiting in its local buffer for transmissions.

In this section, we consider a finite number of stations under saturation load condition [8]. With this load condition, a station will never become an idle station; a station simply alternates itself between a ready station and a backlogged station.

Applying the Markovian Framework modeling technique presented in [12],

we model the OBS scheme into a single server queue (SSQ). Stations implementing the OBS scheme exchange RFT/ACK messages with the AP on the signaling channel, and the actual frame transmission is performed on the data channel using PCF protocol, which may not follow the RFT/ACK message exchange immediately (see Figure 5.1). To describe such a system, we model the arrival process of the SSQ into a process that stations switch from ready stations to backlogged stations, and the service process of the SSQ into the process that stations switch from backlogged stations to ready stations. Hence, the queue size of the SSQ describes the number of backlogged stations in the network.

On the signaling channel, the exchange of the RFT/ACK frames is the basic access method of the IEEE 802.11 MAC protocol. A success of such an exchange generates a backlogged station indicating an arrival to the SSQ. Hence the arrival process of the SSQ is the service process of the signaling channel. Knowing that the signaling channel operates the basic access method of the IEEE 802.11 MAC protocol, and according to [12], the service time distribution of the IEEE 802.11 MAC protocol can be accurately described by an appropriate Erlang distribution under saturation load condition, then the interarrival time of the SSQ can be accurately modeled by an Erlang distribution.

We adopt the usual assumption in the MAC protocol performance analysis that the frame size is constant. As a result, the protocol service time distribution on the data channel is deterministic. In the Markovian Framework, this distribution can be approximated by an Erlang distribution due to the small variance characteristics of the Erlang distribution [12]. Consequently, we construct a state dependent $E_j/E_k/1/n$ (SD- $E_j/E_k/1/n$) queue to model the OBS scheme where E_j indicates the Erlang distribution with j stages. The arrival rate of our SSQ is state depen-

dent because different number of ready stations requires different time duration to obtain a successful RFT/ACK message exchange on the signaling channel².

Assume that the total number of stations in a network is n . Let m be the number of backlogged stations, then the number of ready stations is $n - m$. The arrival rate λ_{n-m} is the rate of a successful RFT/ACK message exchange when there are $n - m$ ready stations, and the service rate μ is the frame transmission rate. The arrival rate λ_{n-m} is the IEEE 802.11 MAC protocol service rate using basic access method [8, 14]. We use the approach in Section 3.1 [14], which is

$$\lambda_{n-m} = \frac{P_s}{P_i\sigma + P_sT_s + P_cT_c}, \quad (5.1)$$

where σ is the length of a slot time, P_s is the probability that there is a successful transmission during a slot time given $n - m$ ready stations, P_c is the probability that a transmission is colliding with other transmissions, and P_i is the probability of an idle slot. The formulation of the probabilities P_s , P_c , and P_i are provided in Section 3.1. The values of T_s and T_c , the average time that the channel is sensed busy because of a successful transmission and the average time that the channel is sensed busy during a collision respectively, are given by

$$\begin{cases} T_s = RFT + SIFS + \delta + ACK + DIFS + \delta \\ T_c = RFT + DIFS + \delta \end{cases}. \quad (5.2)$$

The service rate of the SSQ, μ , is given by [17]

$$\mu = 1/T_{cycle}, \quad (5.3)$$

²The duration difference is caused by the employed contention protocol; generally higher number of stations requires longer time to resolve the contention as more collisions will occur.

where the T_{cycle} is the period of a polling cycle, which can be expressed as

$$T_{cycle} = POLL + SIFS + \delta + T_{DATA} + SIFS + \delta. \quad (5.4)$$

Let the Markov Chain state $\{x, y, z\}$ denotes the situation that the SSQ has x backlogged stations, and the SSQ is in the Erlang arrival stage y and Erlang service stage z . For simplicity of formulation, let the invalid states of the Markov Chain model, i.e. state $\{x, y, z\}$ outside the range of $0 \leq x \leq n$, $0 \leq y < j$, and $0 \leq z < k$, has stationary probability of zero. We also assume that the arrival and service rates are zeros for out-of-range parameters i.e. the arrival rate is non-zero if $0 \leq x < n$ and the service rate is non-zero if $0 < x \leq n$. The balance equations set for the SSQ is given by

$$\begin{aligned} 0 &= -\left(j\lambda_{n-x} + k\mu(x)\right)P_{x,y,z} + j\lambda_{n-x}P_{x,y-1,z} + k\mu(x)P_{x,y,z-1} \\ 0 &= -\left(j\lambda_{n-x} + k\mu(x)\right)P_{x,0,z} + j\lambda_{n-x-1}P_{x-1,j-1,z} + k\mu(x)P_{x,0,z-1} \\ 0 &= -\left(j\lambda_{n-x} + k\mu(x)\right)P_{x,y,0} + j\lambda_{n-x}P_{x,y-1,0} + k\mu(x+1)P_{x+1,y,k-1} \\ 0 &= -\left(j\lambda_{n-x} + k\mu(x)\right)P_{x,0,0} + j\lambda_{n-x-1}P_{x-1,j-1,0} + k\mu(x+1)P_{x+1,0,k-1}. \end{aligned} \quad (5.5)$$

We compute the stationary probabilities of the SD- $E_j/E_k/1/n$ queue with $j = 16$ and $k = 32$. These settings are chosen based on the study given in [12].

Let p_i be the probability that there are i backlogged stations in the network, where $i = 0, 1, \dots, n$. Relating to the SSQ, $p_i = \sum_{y,z} \pi_{i,y,z}$, where $\pi_{x,y,z}$ is the steady state probability of the SSQ being in state $\{x, y, z\}$. The mean arrival rate

Nanyang Technological University

of the SSQ, $\bar{\lambda}$, can be computed by

$$\bar{\lambda} = \sum_{i=0}^n (\lambda_{n-i} \cdot p_i), \quad (5.6)$$

and the system throughput, γ , can be obtained by

$$\gamma = \bar{\lambda} \cdot d_p, \quad (5.7)$$

where d_p is the average size of the payload in a frame.

Let $\bar{m}$ be the average number of backlogged stations. Knowing p_i , the value $\bar{m}$ can be computed by

$$\bar{m} = \sum_{i=0}^n (i \cdot p_i). \quad (5.8)$$

Using Little's formula, we calculate the average queuing delay of a frame on the data channel, W , as

$$W = \bar{m} / \bar{\lambda}. \quad (5.9)$$

The average queuing delay corresponds to the time period between a successful RFT/ACK message exchange and the frame transmission time. This queuing delay time does not include the time period of the contention and RFT/ACK message transmission on the signaling channel. Including that for the mean MAC transmission delay, we simply need

$$W_s = n / \bar{\lambda}. \quad (5.10)$$

The above result is derived based on the concept that in the saturation condition, there are always n ready stations in the network either competing on the signaling

channel or waiting for frame transmission on the data channel. An arrival of the network is the generation of a new frame from any of the n stations, which occurs when a backlogged station switches to a ready station. Moreover, the rate for a backlogged station switching to a ready station is the same as the rate for a ready station switching to a backlogged station in steady state under saturation condition, where the latter is simply $\bar{\lambda}$ according to our SSQ model. Applying Little's formula, we obtain the mean MAC transmission delay, W_s , of the protocol implementing the OBS scheme.

5.2.2 Performance Comparison

To illustrate the performance advantages of our OBS scheme, performance of OBS is compared with the existing IEEE 802.11 MAC protocol. In particular, we focus on the saturation throughput and delay measures, which are indications of the high load performance of protocols. Here, we assume that the OBS's signaling channel is an overhead of the protocol, which is not considered in the performance comparison³. However, it should be obvious from the comparisons provided in this section and the subsequent sections that the performance improvement gained by using OBS is higher than the performance of in-band signaling protocols running on the larger combined channel.

Figure 5.2 shows the saturation throughput versus data channel bit rate of the OBS scheme compared with the IEEE 802.11 schemes. We rely on the physical layer parameters of the IEEE 802.11a [2] (reported for the convenience of the reader in table 5.1). The frame size is a constant of 1500 bytes. The symbols shown in the figures represent the simulation results, while the solid lines show

³We opted to provide this comparison to show the future potential of OBS.

Table 5.1: Parameters used for numerical analysis and simulations.

Slot Time	$9\mu s$
SIFS	$16\mu s$
DIFS	$34\mu s$
PIFS	$25\mu s$
Propagation Delay	$1\mu s$
CW_{min}	16
CW_{max}	1024

the analytical results. The analytical results for the IEEE 802.11 schemes are computed using the model in [8]. The figure shows close match between our employed analytical framework and the simulation results for the OBS scheme confirming the accuracy of our analysis.

Let OBS_x denote OBS scheme with x Mbps signaling channel. According to Figure 5.2, we see that OBS_{12} can provide a maximum saturation throughput of almost 75 Mbps from a 150 Mbps data channel. For comparison, the IEEE 802.11 basic access method only achieves 50 Mbps throughput from a 150 Mbps data channel. Since OBS requires an additional of 12 Mbps signaling channel, the total usage of the bandwidth is the bandwidth combination of the two OBS channels, which gives 162 Mbps. If this was the bit rate of the IEEE 802.11 basic access method, as can be seen from Figure 5.2, its throughput would still be just around 50 Mbps, which remains below the performance of the OBS scheme. This shows that the use of the signaling channel in OBS allows much higher utilization in the data channel, and comparing to the existing IEEE 802.11 schemes, the throughput gain in the data channel for OBS exceeds that the additional bandwidth required for the signaling channel.

Figure 5.2: Saturation throughput with various data channel bit rate.

Figure 5.2 also reports an interesting performance trend for the OBS scheme at high data channel bit rate. The performance of the OBS scheme increases steadily at low data channel bit rate region, then stays relatively flat at high data channel bit rate. In the low data channel bit rate region, the data channel is always fully utilized with data transmission, and the limit of the throughput is mainly bounded by the transmission overhead caused by the physical preamble and positive acknowledgment, hence with higher bit rate on the data channel, higher throughput is achieved. At the turning point from the increasing trend to the flat trend, the successful requests from the signaling channel are just enough to saturate the data channel, thus the increment of the data channel bit rate beyond the turning point does not elevate the throughput performance further.

In Figure 5.3, we illustrate the saturation throughput of the OBS scheme with various number of stations compared with the IEEE 802.11 schemes. For this

Figure 5.3: Saturation throughput with 108 Mbps channel bit rate.

comparison, we consider the data channel bit rate to be 108 Mbps, double of the current maximum channel bit rate of IEEE 802.11a and g. Although OBS requires an additional of 12 Mbps bandwidth for the signaling channel, the comparison presented in Figure 5.3 shows that OBS performs over 12 Mbps higher in terms of throughput than the existing IEEE 802.11a protocol. This again confirms that the separation of the channel acquisition process and the actual data transmission reduces the bandwidth wastage caused by collisions in the existing IEEE 802.11 schemes.

We plot the mean MAC transmission delay of the OBS scheme compared with the IEEE 802.11 schemes with various number of stations in Figure 5.4. With a frame size of 1000 bytes, OBS maintains MAC transmission delay of 8.9 ms even with 50 stations under saturation load; IEEE 802.11, however, gives delays of 15 ms and 13.4 ms for RTS/CTS and the basic access method, respectively. The

Figure 5.4: Mean saturation transmission delay with 108 Mbps channel bit rate.

immediate benefit of lower transmission delay is better support for delay-sensitive applications such as voice over Internet Protocol (VoIP) or video conferencing.

Each numerical result presented in this subsection takes less than 10 seconds⁴ to compute on an Intel Pentium 4 machine. Each simulation run takes a few minutes to complete.

5.2.3 Signaling Channel Optimization

Previously, we have reported an interesting characteristics of OBS in Figure 5.2 where there exists a turning point in the OBS throughput performance such that OBS operates at its maximum throughput given a pair of bit rates of signaling and data channels. At this turning point, the operation of OBS is considered optimized. The appropriate choice of signaling channel bit rate for a given data channel bit

⁴This timing depends on the desired error threshold.

rate ensures that the data channel maintains full utilization. A lower signaling channel bit rate will cause lengthy RFT/ACK message exchange, hence under utilizing the data channel, whereas, a high bit rate on the signaling channel will cause spectrum wastage. Our study in this subsection deals with the analysis of such pairs of bit rates that optimize the OBS operation.

To optimize the OBS operation, we need $\frac{\bar{\lambda}}{\mu} = 1$, that is, when the rate of a successful RFT/ACK message exchange equals the frame transmission rate, which keeps the data channel fully utilized. The variable $\bar{\lambda}$ depends on the bit rate of the signaling channel and the number of stations contending on the signaling channel, whereby the variable μ depends on the bit rate of the data channel and the frame size. To be precise, signaling channel bit rate affects RFT in (5.2) which in turn affects (5.1) and $\bar{\lambda}$ in (5.6). Data channel bit rate influences T_{DATA} which in turn influences μ in (5.3). Given a particular data channel bit rate, we find the signaling channel bit rate such that $\frac{\bar{\lambda}}{\mu} = 1$, and this pair of signaling and data channels achieves the optimized throughput in OBS operation.

In Figure 5.5, we show the bit rate pairs of signaling and data channels required to optimize the OBS operation. As can be seen, for the considered frame sizes, OBS requires relatively low bit rate signaling channel to fully utilize the data channel which is below 100 Mbps. Our particular interest here is the data channel of 108 Mbps bit rate. It is found that the signaling channel bit rate between 6 Mbps and 20 Mbps appears to be a good choice for commonly used frame sizes, and hence we consider 12 Mbps for the signaling channel bit rate for studies conducted in the previous subsections.

Figure 5.5: Minimum signaling channel bit rate required to achieve full utilization on the data channel.

5.3 Analysis of the Out-of-Band Signaling Scheme under Statistical Traffic

To investigate the performance advantage of OBS under statistical traffic condition, we further extend the Markovian Framework presented in the previous section to capture the characteristics of statistical arrivals. Then, we compute and analyze the throughput and delay performance of OBS under the considered traffic. All analytical results are compared and validated by simulation.

5.3.1 Markovian Framework for Out-of-Band Signaling Scheme under Statistical Traffic

We consider Poisson process with one frame buffer as arrival process for each station. In other words, each station holds only a frame in its local MAC buffer. Fur-

thermore, we assume that the stations are statistically identical and independent. We reuse the definitions of *ready station*, *backlogged station*, and *idle station*, which are defined in the previous section. Under Poisson arrival, an *idle station* switches to a *ready station* when it generates a frame into its local buffer according to a Poisson process. A *ready station* switches to a *backlogged station* when it has successfully exchanges the RTS/CTS message with the AP. Under statistical traffic and one buffer assumption, a backlogged station switches to an idle station after its successful frame transmission on the data channel.

Applying Markovian Framework, we characterize the system with two queues in tandem. The first queue models the stages that a station switches from an idle station to a ready station indicating an arrival to the queue, and from a ready station to a backlogged station indicating a departure from the queue. The arrivals to the queue is an aggregation of arrivals from all idle stations, where the inter-arrival time has exponential distribution, and the aggregated arrival rate depends on the number of idle stations. The service process of the queue is the RFT/ACK frame exchange process, which, as described in the previous section, can be modeled by an Erlang distribution.

The departure of the first queue goes into the second queue. The second queue models the actual data transmission stage, that is, a backlogged station switching back to a ready station after its successful frame transmission. The arrival process of the second queue is the output process of the first queue, which has Erlang distribution. The service process of the second queue depends on the data frame size. For the constant frame size assumption, an Erlang distribution for service time is used as in the previous section.

Let the total number of stations in the network be n . Define v to be the number

of ready stations, then the number of idle stations is $n - v$. The arrival rate λ_{n-v} is the state-dependent Poisson arrival process, which is given by

$$\lambda_{n-v} = (n - v) \cdot \hat{\lambda}, \quad (5.11)$$

where $\hat{\lambda}$ is the individual arrival rate of a station.

The service rate of the first queue, μ_1 , is the rate of a successful RFT/ACK message exchange when there are v ready stations, which is given by (5.1). The service rate of the second queue, μ_2 , is the rate of frame transmission with polling scheme and constant frame size given by (5.3).

Let the Markov Chain state $\{v, x, y, z\}$ denotes the situation that the system has v ready stations and x backlogged stations, and the system is in the Erlang service stages y and z of the first and second queues respectively. Let the invalid states of the Markov Chain model, i.e. state $\{v, x, y, z\}$ outside the range of $0 \leq v \leq n$, $0 \leq x \leq v$, $0 \leq y < j$, and $0 \leq z < k$, has stationary probability of zero. Assume that the arrival and service rates are zeros for out-of-range parameters i.e. the arrival rate λ is non-zero if $0 \leq v < n$, the service rate μ_1 is non-zero if $0 \leq x < v$, and the service rate μ_2 is non-zero if $0 < x \leq v$. The balance equations set for the system is given by

$$\begin{aligned} 0 = & - \left(\lambda_{n-v} + j\mu_1(v-x) + k\mu_2(x) \right) P_{v,x,y,z} + \lambda_{v-1} P_{v-1,x,y,z} \\ & + j\mu_1(v-x) P_{v,x,y-1,z} + k\mu_2(x) P_{v,x,y,z-1} \\ 0 = & - \left(\lambda_{n-v} + j\mu_1(v-x) + k\mu_2(x) \right) P_{v,x,0,z} + \lambda_{v-1} P_{v-1,x,0,z} \\ & + j\mu_1(v-x-1) P_{v,x-1,j-1,z} + k\mu_2(x) P_{v,x,0,z-1} \end{aligned} \quad (5.12)$$

Nanyang Technological University

$$\begin{aligned}
 0 &= - \left(\lambda_{n-v} + j\mu_1(v-x) + k\mu_2(x) \right) P_{v,x,y,0} + \lambda_{v-1} P_{v-1,x,y,0} \\
 &\quad + j\mu_1(v-x) P_{v,x,y-1,0} + k\mu_2(x+1) P_{v+1,x+1,y,k-1} \\
 0 &= - \left(\lambda_{n-v} + j\mu_1(v-x) + k\mu_2(x) \right) P_{v,x,0,0} + \lambda_{v-1} P_{v-1,x,0,0} \\
 &\quad + j\mu_1(v-x-1) P_{v,x-1,j-1,0} + k\mu_2(x+1) P_{v+1,x+1,0,k-1}.
 \end{aligned}$$

Similar with the previous section, we compute the stationary probability of the queue with Erlang stages of the first queue $j = 16$ and Erlang stages of the second queue $k = 32$.

Let p_i be the probability that there are i ready stations in the network, where $i = 0, 1, \dots, n$. Relating to the system, $p_i = \sum_{x,y,z} \pi_{i,x,y,z}$, where $\pi_{v,x,y,z}$ is the steady state probability of the system being in state $\{v, x, y, z\}$. The mean arrival rate of the system, $\bar{\lambda}$, can be computed by

$$\bar{\lambda} = \sum_{i=0}^n (\lambda_{n-i} \cdot p_i), \tag{5.13}$$

and the system throughput, γ , can be obtained by

$$\gamma = \bar{\lambda} \cdot d_p, \tag{5.14}$$

where d_p is the average size of the payload in a frame.

Let $\bar{v}$ be the average number of ready stations. The value $\bar{v}$ can be computed by

$$\bar{v} = \sum_{i=0}^n (i \cdot p_i). \tag{5.15}$$

Similar to (5.10), given the number of ready stations, $\bar{v}$, and the mean arrival

Figure 5.6: Mean transmission delay under statistical traffic.

rate, $\bar{\lambda}$, by Little's formula, we calculate the average delay of a frame, W_s , as

$$W_s = \bar{v} / \bar{\lambda}. \quad (5.16)$$

5.3.2 Performance Comparison

Figure 5.6 shows the mean transmission delay of the OBS scheme compared with the IEEE 802.11 schemes under statistical traffic. We use 12 Mbps signaling channel for the OBS scheme as justified in the previous section. Similarly, we use 108 Mbps data channel for the comparison. The symbols shown in the figures represent the simulation results, while the solid lines show the analytical results. Analytical results for the IEEE 802.11 schemes are computed using the model in [12]. The results show that OBS has lower delay under medium to high load compared with the IEEE 802.11 schemes. Under very light load condition, all

Figure 5.7: Throughput versus mean transmission delay under statistical traffic.

schemes performs similarly well where they offer mean MAC transmission delay of below 0.5 ms.

The throughput versus mean transmission delay of the OBS scheme is plotted in Figure 5.7 and compared with the IEEE 802.11 schemes. OBS can achieve 57% throughput level, whereby IEEE 802.11 with basic access and RTS/CTS method achieve 41% and 34% throughput level respectively. OBS also maintains delays lower than 1.6 ms, whereby IEEE 802.11 has delays up to 2.4 ms and 2.9 ms for basic access and RTS/CTS method respectively.

The results reported in the two figures verify the accuracy of our analytical framework. The figures show close match between the analysis and the simulation results. This analytical framework is useful for determining the number of application streams that can be supported by an access point (see for example [73]). It is also useful for admission control for quality of service demanding

applications such as VoIP and video streaming.

The numerical results presented in this subsection were computed on an Intel Pentium 4 machine; each result takes less than 30 seconds to compute. Each simulation sample consumes less than 10 minutes to produce on the same machine.

5.4 Summary

In this chapter, we have proposed and analyzed the OBS scheme for high speed WLANs. OBS uses a low bit rate channel for signaling and a high bit rate channel for the actual data transmission. We showed that the use of out-of-band signaling technique achieves higher overall throughput despite the need for an additional low bit rate channel for signaling. Besides, OBS maintains backward compatibility with the existing users of the IEEE 802.11 standards, where users of the IEEE 802.11 standards can access to OBS WLANs, however, they will not enjoy the performance benefit.

To illustrate the performance advantage of OBS, we applied Markovian Framework to study the throughput and delay performance under the saturation load and the statistical traffic conditions. The analytical results, validated by simulation, indicated performance advantages of OBS compared with the current the IEEE 802.11 schemes. We also investigated the channel bit rate settings for optimal performance in OBS, where we focus our study to the issue of protocol parameter design for the justification of OBS parameter selection.

Chapter 6

Out-of-Band Polling Scheme

In this chapter, we discuss the *out-of-band polling scheme* (OBPS). OBPS improves the performance of high speed WLANs by using out-of-band signaling technique with an efficient polling scheme. Specifically, the MAC protocol maintains two channels of which one is a low speed and one is a high speed channel. The low speed channel is used mainly for channel assignment purpose, henceforth called a signaling channel, and the high speed channel is used for the actual data transmissions, henceforth called a data channel. Stations that have packets to transmit must first make a request on the signaling channel. The request process is based on the proposed time division registration mechanism where each station registers its request for bandwidth at its minislot during a scheduled time division multiple access (TDMA) period announced by the *access point* (AP). This registration scheme also supports requests for priority transmission.

Figure 6.1: Operations of out-of-band polling.

6.1 Scheme Overview

In this section, we outline the operations of OBPS on the signaling and data channel. We also describe an extension of OBPS to support transmissions of both upstream and downstream real-time packets. We call a data stream from a station to a AP as an upstream data transmission and a data stream from a AP to a station as a downstream data transmission.

6.1.1 Signaling Channel

OBPS moves transmission requests from the data channel to a separate low speed signaling channel leaving the data channel solely for data transmissions. We use a time division registration approach for signaling. We divide the signaling channel into slotted time. After being polled by the AP, stations jam the channel during their assigned time slots to signify that they have data to send. We use a group polling scheme to minimize the overhead of polling (see Figure 6.1).

The group polling scheme uses an ordered poll-list. This list is generated and broadcast by the AP. The poll-list contains identity of stations that are given a chance to make transmission requests. After receiving the poll-list, each station

waits for its turn to send a jamming signal on the signaling channel if it has data to transmit to the AP. The jamming signal represents a one bit information (a jam signifies that a station has data to transmit to the AP). After sending the jamming signal, the station waits for a data transmission poll on the data channel. After receiving a jamming signal, the AP queues the transmission request from the corresponding station in its upstream transmission queue. As stations will listen to the data channel after a transmission request, the AP will only include stations that are not in its transmission queue when generating a poll-list. This scheme reduces the overheads of polling cycles compared to a mechanism that polls a station and waits for an empty packet sent by the station to signify that it has no data to send.

We use a two-byte field to represent a station identity. Each jamming signal takes $9\ \mu s$, which is the length of a slot-time specified in the IEEE 802.11a standard [2]. A slot-time includes clear channel assessment (CCA) time, which allows the AP to detect the transmission of a jamming signal. We use a short inter-frame space (SIFS) as the separator of two jamming signals. Using our scheme, a signaling cycle time is 1.832 ms for 64 stations on a 6 Mbps signaling channel (calculated using (6.1) provided in Section 6.2).

6.1.2 Data Channel

On the data channel, OBPS polls stations based on its upstream transmission queue. To further minimize the signaling cycle time on the signaling channel, we append a request-to-be-queued indicator (RI) on each packet. This indicator enables a station currently being polled for data to request for another transmission if it has more data to send, thus reducing the signaling cycle time as the station

Figure 6.2: Downstream and upstream transmissions for station A.

Figure 6.3: Station A downstream packet transmission preempted by station B.

can be excluded from the next polling list.

Additionally, our protocol merges the upstream and downstream transmission periods, hence reducing the overheads of using separate DCF and PCF periods for upstream and downstream transmissions respectively. For example, a station, *Station-A*, can get a consecutive downstream and upstream transmission sequence (see Figure 6.2).

6.1.3 Quality-of-Service in OBPS

The decoupling of transmission requests and data transmissions allows OBPS to use efficient scheduling schemes for transmission requests. To expedite real-time packet transmissions, OBPS uses a priority based signaling scheme. Data packets are classified as priority or non-priority packets. Priority packets usually refer to packets generated by real-time applications. A AP generates an ordered *Priority Signaling List* (PSL) and an ordered *Non-priority Signaling List* (NSL). A signaling cycle consists of a PSL poll and an NSL poll. The AP keeps separate

transmission request queues for PSL and NSL, where the PSL queue will be serviced with priority over the NSL queue. To cater for priority transmission, we use two bits for RI, which signify whether a station has more priority or non-priority packets or both.

With this signaling scheme, the AP can fully utilize the data channel to efficiently support upstream and downstream data transmissions. With only upstream packets, the AP polls a station, which transmits its data and receives an acknowledgment from the AP. We call this as “Transmission Sequence A” (TSA). As WLAN channels are half-duplex channels, the utilization of a channel can be improved by transmitting a downstream packet for a station being polled for upstream data transmission. In other words, a AP polls and sends a downstream packet to a station that transmits its upstream data and receives an acknowledgment from the AP (see Figure 6.2). We call this “Transmission Sequence B” (TSB). However, TSB is insufficient to support priority service for downstream packets as the most time critical downstream priority packet (taking into account that a deadline based scheduling algorithm is used for downstream traffic) may not belong to the station being polled for upstream transmission. We introduce “Transmission Sequence C” (TSC) where a AP preempts the transmission of downstream packet from the station to be polled. The AP sends the most time critical packet and waits for an acknowledgment for the transmission. After that, the AP polls the preempted station for upstream transmission (see Figure 6.3). The downstream packet being preempted may either be delivered to its station on the next poll for the station (using TSB) or preempt the downstream packet on the next poll if it is the next most time critical downstream packet to be serviced by the AP (using TSC).

In sections 6.2 and 6.3, we analyze and show that the worst-case delay is sufficient to make delay guarantees for common real-time applications, and the attainable saturation throughput is higher than that of the IEEE 802.11 schemes. Additionally, we also show that OBPS is scalable even when the data rate of the WLAN channel and the number of supported stations are doubled.

In a general sense, with the delay bounds derived, OBPS may adopt any scheduling algorithms to provide service differentiation to real-time packets. The derived worst-case delay bounds for OBPS provides scheduling algorithms with transmission sequences' cycle times for delivery of upstream and downstream traffic. These cycle times are important parameters for analyzing scheduling algorithms that can be deployed to achieve QoS guarantees for QoS-sensitive real-time packets.

Specifically, we adopted DRR to provide service differentiation for our scheme. For downstream transmissions, each station is allocated a queue. Each queue is allocated a weighted quantum using a proportional weighted function. Each queue, queue i , is allocated ω_i worth of bits in each round. Through a prior bilateral agreement between the user and the service provider or reservation protocol [101], a AP can assign quantum for each station. Define $\omega^{min} = \min_i(\omega_i)$, the quantum allocated to queue i is simply $\frac{\omega_i}{\omega^{min}}$. DRR works in rounds. A round exhausts when all queues are either empty or their instantaneous quantum - Deficit Counter (DC) is less than the length of the packet to be serviced. DC is decreased by the size of packet transmitted. Unused DC is carried over to the next round only if its queue is not empty.

To enforce rate differentiation and to prevent starvation of non-priority stations for upstream transmissions, OBPS adopts the DRR scheduling algorithm. For

upstream DRR control, OBPS buffers two transmission requests (one priority and one non-priority) for each station. This is because stations that deploy OBPS are allowed to signal for one priority and one non-priority upstream transmissions at a time. Each DC is allocated a quantum worth of bits in each round of service. A round ends only when there is a deficit of quantum (negative quantum) in every DC or there is no more traffic to transmit. We allow negative DC in our scheme to simplify the design as the AP does not need to transmit the DC value to the station. DC is decreased by the size of transmitted packets. The value of DC is carried over to the next round. A station with a negative DC will not be polled for data although the station has requested for transmission.

At a cost of incurring additional delay in transmitting priority packet, a DCF period may be used on the data channel of OBPS to maintain compatibility with the existing IEEE 802.11 standard. Stations that do not support OBPS may content for upstream transmission during this period and wait for its downstream data in both the PCF and DCF periods.

6.2 MAC Delay Analysis

We analyze the *worst-case delay* of high priority traffic for delay sensitive services in OBPS, the worst-case delay of a packet in PCF and the average delay of a packet in DCF with basic access method and RTS/CTS method. We also provide simulation results of a scenario where packets are generated in each station with exponential inter-arrival time. We define the worst-case delay as the longest waiting time, if any, for a station to successfully acquire transmission right on the data channel when a packet is generated by the station. In this chapter, sim-

ilar to the previous chapter, we consider the signaling channel as an overhead of OBPS; hence, comparisons are done with a same size data channel. The performance advantage of OBPS is shown by the large gain of performance, which is unachievable with in-band signaling protocols.

In the following subsections, we derive the OBPS worst-case delay for priority packets ($D_{upstream}^{OBPS}$), the PCF worst-case delay (D^{PCF}) and the DCF average delay (D^{DCF}). Additionally, we show that the OBPS worst-case delay for priority upstream traffic with downstream traffic is derived using $D_{upstream}^{OBPS}$ with TSC transmission sequence and the OBPS worst-case delay for downstream traffic with upstream traffic ($D_{downstream}^{OBPS}$) is simply the time required to transmit the transmission sequence TSC.

6.2.1 OBPS Worst-case Delay for Priority Packets

The signaling poll-list of OBPS uses two bytes to identify each station. Thus, a poll-list containing k registered stations is $16k$ bits long and the time required to complete this signaling poll cycle is

$$T_k^{Signal} = T_{Poll-List} + SIFS + k \times (T_{Jam} + SIFS) \quad (6.1)$$

where T_{Jam} is the period of a jamming signal (we use a slot-time period as the period of the jamming signal).

The worst-case delay of a high priority packet occurs when a station, which requires to make a new reservation for transmission of high priority packets, has just missed a PSL poll in the signaling channel and it is scheduled to send its jamming signal on the last slot in the PSL. Therefore, this station will have to

wait for the current PSL and an NSL cycle before it gets a chance to be the last station to send a jamming signal in the next PSL cycle (assuming no re-shuffling of stations position in the poll-list). Additionally, we assume that RI is not used and the data channel does not turn idle as there is always a backlog of non-priority traffic.

The worst-case delay of OBPS is given by

$$D_{upstream}^{OBPS} = SIFS + T_{Jam} + SIFS + T_{TS*} \times \alpha + T_{TS*} - (T_{Data} + SIFS + \delta) \quad (6.2)$$

$$\alpha = \begin{cases} \lceil \phi \rceil + m - 1, & \text{if } \lfloor \tau \rfloor \geq (m - 1) \\ \lceil \phi \rceil + \lfloor \tau \rfloor, & \text{if } \lfloor \tau \rfloor < (m - 1) < \lfloor \phi \rfloor \\ \lfloor \tau \rfloor + m - 1, & \text{if } \lfloor \phi \rfloor \leq (m - 1) \end{cases}$$

$$\tau = \frac{2 \times T_k^{Signal} - T_{Poll-List} - SIFS - T_{Jam}}{T_{TS*}}$$

$$\phi = \frac{2 \times T_k^{Signal} - T_{Poll-List}}{T_{TS*}}$$

where T_{TS*} is the time required to transmit a sequence of frames that corresponds to TSA, TSB or TSC. The value α is the total number of TS* transmissions that occur from the time after the first jam period of a priority signaling poll to the time before the start of the TS* transmission that the worst-case packet belong to. The value τ is the total number of TS* transmissions that occur from the time after the

Figure 6.4: An illustration for the value τ and ϕ .

first jam period of a priority signaling poll to the time just before the next priority signaling poll. The value ϕ is the total number of TS^* transmissions that occur from the time after the first jam period of priority signaling poll to the time after the first jam period of the next priority signaling poll (see Figure 6.4). The value m is the number of active priority stations.

TSA, TSB and TSC are given by

$$\left\{ \begin{array}{l} TSA = T_{Poll} + SIFS + \delta + T_{Data} + SIFS + \delta \\ TSB = T_{Poll+Data} + SIFS + \delta + T_{Data+Ack} + SIFS + \delta \\ TSC = T_{Data} + SIFS + \delta + T_{Ack} + SIFS + \delta + \\ \quad T_{Poll} + SIFS + \delta + T_{Data} + SIFS + \delta. \end{array} \right. \quad (6.3)$$

6.2.2 IEEE 802.11 PCF Worst-case Delay

For PCF, the worst-case delay of a packet occurs when a station has a packet to transmit just after the station has been polled. The packet will be transmitted on the next polling cycle, which occurs after all the other registered stations are polled.

The worst-case delay of PCF is given by

$$\begin{aligned}
 D^{PCF} = & (n-1) \times (T_{Poll} + SIFS + \delta + T_{Data} + SIFS + \delta) \\
 & + (k-n) \times (T_{Poll} + SIFS + \delta + T_{Ack} + SIFS + \delta) \\
 & + T_{Poll} + SIFS + \delta
 \end{aligned} \tag{6.4}$$

where k is the total number of registered stations and n is the number of active stations.

6.2.3 IEEE 802.11 DCF Average Delay

With non-zero probability, the worst-case delay of DCF is infinite. To provide a delay comparison, we study the average delay of DCF. In the next subsection, we show that the worst-case delay of OBPS is even lower than the average delay of DCF.

We derive the average delay for a station using the Little's formula. We model the system as a queuing system where the service represents a successful transmission and the elements of the queue refers to contending stations. The arrival rate of the system is equal to the departure rate as all the stations are in saturation condition.

The arrival rate of the system is the service rate derived in [8, 14] and Section 3.1. We use the more standard conforming model in [14] and Section 3.1, which is given by

$$\lambda_n = \frac{P_s}{P_i\sigma + P_sT_s + P_cT_c} \tag{6.5}$$

where σ is the slot time and n is the number of active stations. The value T_s is

the average time the channel is sensed busy because of a successful transmission and T_c is the average time the channel is sensed busy during a collision. P_s is the probability that there is a successful transmission during a slot time given $n - m$ ready stations, P_c is the probability that a transmission is colliding with other transmissions, and P_i is the probability of an idle slot. The formulations of T_s and T_c are given in [8] and in Section 2.4; the formulations for P_s , P_c , and P_i are provided in Section 3.1 [14].

The average delay of the DCF is given by

$$D^{DCF} = W - (T_{Data} + SIFS + \delta + T_{Ack} + DIFS + \delta). \quad (6.6)$$

The value W is given by the Little's formula, precisely $W = n/\lambda_n$.

6.2.4 Upstream Transmission Delay Comparison

We plot a scenario where there is a total of 60 registered stations with increasing number of priority stations and active stations where the proportion of priority stations over active stations is fixed at $\frac{2}{5}$ (see Figure 6.5). We have calculated the worst-case delay with a 54 Mbps channel and the length of each packet is 1500 bytes. We use 6 Mbps signaling channel for OBPS. We use the IEEE 802.11a channel parameters for both the analysis and the simulations (see table 5.1 in Chapter 5). TSA is used for this comparison as our analysis is based on upstream traffic. OBPS can maintain much lower delay for high priority packets as compared with the IEEE 802.11 schemes. This is because a station, S , with high priority packets using DCF, has to compete with the other active stations for transmission and the same station S , using PCF, has to wait for all other registered

Figure 6.5: Worst-case delay for upstream transmissions with fixed packet size of 1500 bytes.

stations to be polled before it gets a chance to transmit. With OBPS, station S only has to wait for other stations that have made priority transmission requests.

The results also show low delay experienced by stations using OBPS compared with others. For as many as 24 active priority stations out of 60 total active stations, a transmission of high priority traffic takes just above 10 ms from the time a packet is generated to the time this packet is transmitted successfully (assuming that there is no transmission error). These results indicate the feasibility of providing delay sensitive services in WLANs implementing the OBPS scheme.

Figure 6.6: Worst-case delay for bidirectional transmissions with fixed packet size of 1500 bytes.

6.2.5 OBPS Worst-case Delay for Priority Upstream Traffic with Downstream Traffic

WLAN uses a half duplex transmission channel where a transmission is either an upstream or a downstream transmission at one time. This greatly affects the delay for both types of transmission. Assuming that all registered stations are in the priority stations list, the worst-case delay of an upstream high priority packet with a downstream packet transmitted when a station is polled at the data channel can be computed using (6.2) and TSC. We plot this delay (see Figure 6.6) with an increasing number of active priority stations over 32 and 64 registered stations using a 54 Mbps data channel. This plot shows the worst-case delay, which may assist AP in deciding the number of priority stations to support in a network environment for a given QoS addressable by worst-case delay or vice versa.

Table 6.1: OBPS worst-case delay for downstream traffic on a 54 Mbps data channel.

Packet size (bytes)	$D_{wc,downstream}^{OBPS} (\mu s)$
500	320
1000	472
1500	616

Additionally, we also plot the worst-case delay of 128 registered stations using a 108 Mbps data channel. Figure 6.6 shows that OBPS can maintain low worst-case delay even when the bandwidth and the number of stations are doubled.

6.2.6 OBPS Worst-case Delay for Downstream Traffic with Upstream Traffic

We look at the worst-case delay of the head-of-line packet at the downstream buffer of the AP. The worst-case delay occurs when a packet arrived at the start of a polling cycle on the data channel. This packet has to wait for the completion of a transmission sequence. The worst-case delay for downstream traffic is given by

$$D_{downstream}^{OBPS} = T_{TSC}. \quad (6.7)$$

Table 6.1 shows the value of $D_{downstream}^{OBPS}$ with various packet sizes. The values show that OBPS can provide low delay for downstream traffic, which is necessary for multimedia traffic.

Figure 6.7: Average delay with fixed packet size of 1500 bytes.

6.2.7 Simulated MAC Queuing Delay

We have simulated the average delay of packets with various number of active stations sharing a 54 Mbps data channel (see Figure 6.7). Each active station generated packets, whereby the inter-arrival time between packets is exponentially distributed with a mean of 25 ms. Each simulation run consumes a few minutes to complete on average.

In Figure 6.7, PCF shows a rapid increase in average delay as the number of active stations increases. DCF with basic access method provides lower average delay as compared with RTS/CTS method. This is because the basic access method has less overhead as compared with the RTS/CTS method. However, the bandwidth wasted for contention using DCF causes a lack of bandwidth in provisioning network requests from about 50 stations. On a special condition when all registered stations are active, the average delay using OBPS is comparable to

the DCF schemes. Although OBPS makes use of the polling scheme, Figure 6.7 shows that OBPS provides lower delay compared with the PCF scheme. OBPS can also maintain low delay for more active stations than the DCF schemes.

6.3 Throughput Analysis

In this section, we evaluate the saturation throughput of OBPS using analytical methods and simulations. We compare OBPS with OBS [17], PCF, DCF with basic access method and DCF with RTS/CTS method. Additionally, we show that OBPS can enforce rate differentiation among stations.

6.3.1 Saturation Throughput without Priority Stations

The saturation throughput, γ , is given by

$$\gamma = \begin{cases} \frac{T_{Payload}}{T_{TS*}} & \text{if } \theta \leq n \\ \frac{n \times T_{Payload}}{T_k^{Signal}} & \text{if } \theta > n \end{cases} \quad (6.8)$$

where k is the total number of registered stations and n is the number of active stations. The value T_k^{Signal} is given by (6.1). The value θ , which is the number of TS* in a signaling cycle, is given by

$$\theta = \frac{T_k^{Signal}}{T_{TS*}}. \quad (6.9)$$

Figure 6.8 plots the numerical results (using solid lines) against their simulated results (using symbols). This figure shows the saturation throughput of the IEEE 802.11 schemes and OBPS using a 54 Mbps channel with each station generates

Figure 6.8: Saturation throughput with fixed packet size of 1500 bytes (the solid lines represent numerical results).

endless network packets of size 1500 bytes each. The analytical results for DCF are computed using the equations in Section 3.1 [14]. The saturation throughput of PCF grows gradually as the number of active stations increases. Both of the DCF schemes have higher saturation throughput when the number of active stations is small. Their saturation throughput decreases as the number of active stations increases. The maximum throughput attainable is about 32 Mbps for basic access method and 27 Mbps for RTS/CTS method. OBS, which uses DCF for signaling, can maintain high throughput as packets transmissions time is longer than the time required to resolve contention; hence, the data channel is fully utilized.

Figure 6.9 plots the simulated results with fixed packet size of 500 bytes. This figure shows that OBPS can maintain high throughput with small packet sizes. OBPS with a smaller packet size will reach the saturation throughput with a higher number of active stations as it needs more packets to fill the data channel. Both

Figure 6.9: Saturation throughput with fixed packet size of 500 bytes.

of the DCF schemes have low saturation throughput when the number of active stations is high. Although OBS can maintain higher throughput than that of the DCF schemes, it has lower throughput than that of the OBPS with a high number of active stations. PCF shows similar throughput growth as with a larger packet size.

Using these results as a context for a comparison, OBPS has a maximum attainable throughput of about 39 Mbps. OBPS without RI hits its maximum throughput with eight active stations out of 64 registered stations. In a special case when all the stations are active, OBPS without RI hits its maximum throughput with just two active stations. With RI, OBPS hits its maximum throughput immediately when there is one active station. Additionally, OBPS attains higher maximum throughput compared with the IEEE 802.11 schemes. OBPS also attains higher maximum throughput compared with the OBS scheme with small

packet sizes.

6.3.2 Saturation Throughput for OBPS with Bidirectional Traffic and Priority Stations

Assuming that all registered stations are in the priority stations list, the saturation throughput bound for bidirectional traffic, $\gamma^{bidirectional}$, is given by

$$\gamma^{bidirectional} = \begin{cases} \frac{2 \times T_{Payload}}{T_{TS*}} & \text{if } \theta \leq n \\ \frac{n \times T_{Payload}}{T_k^{Signal}} & \text{if } \theta > n \end{cases} \quad (6.10)$$

where k is the total number of registered stations, n is the number of active stations and TS^* refers to either TSB or TSC. The value T_k^{Signal} is given by (6.1). The value θ , which is the number of TS^* in a signaling cycle, is given by

$$\theta = \frac{2 \times T_k^{Signal}}{T_{TS*}}. \quad (6.11)$$

Figure 6.10 plots the numerical results for the upper bound (using TSB) and lower bound (using TSC) saturation throughput for upstream and downstream transmissions with 64 and 128 stations. The plot uses a 54 Mbps data channel for 64 stations and a 108 Mbps data channel for 128 stations. For the 54 Mbps with 64 stations, OBPS achieves a maximum saturation throughput of about 45 Mbps. With an increase in the channel data rate to 108 Mbps and total number of stations to 128, OBPS can achieve maximum saturation throughput of about 78 Mbps.

Chapter 6. OBPS

Figure 6.10: Saturation throughput of OBPS with various number of stations.

Figure 6.11: Throughput attained by each station with fixed packet size of 1500 bytes.

6.3.3 Simulated Rate Differentiation

In this simulation, each station generates higher network traffic than the reserved bandwidth. We assigned a DC value for each station with an increasing value of interval 1500 from station 1 to station 10. Each station obtains a proportional share ($station_id / \sum_{i=1}^{10} i$) of the attainable bandwidth.

Figure 6.11 shows that RTS/CTS method shares the bandwidth equally among all stations while OBPS provides stations with service rate differentiation. Each station is given its share of assigned bandwidth. This feature also ensures that malicious stations will not be given extra bandwidth, which prevents other stations from using the channel.

6.4 Summary

OBPS uses an efficient signaling mechanism on a low data rate channel for stations to request for packet transmissions, leaving the high data rate channel for efficient scheduling of upstream and downstream packet transmissions. OBPS allows priority signaling for upstream transmission of real-time packets. It also provides transmission sequence for time-critical downstream packet to preempt the transmission of downstream packet for the station being polled in order to achieve a priority service. OBPS provides higher throughput compared with the existing schemes and low worst-case delay for priority packets. These characteristics make OBPS suitable for applications that require a certain level of QoS such as multimedia streaming applications.

Chapter 7

Contention-Tone Protocol

The previous two chapters have highlighted the benefits of using out-of-band signaling to improve system performances. In this chapter, we propose a method that efficiently avoids collisions among stations and make the channel assignment work conserving even under heavy load. The protocol makes use of *contention-tone* (CT) on a separate narrow band signaling channel to resolve the station contention concurrently during an ongoing frame transmission. In our design, the busy tone (BT) [10] may be used as the CT. However, instead of using the tone to indicate busy in BT-based protocols, we use the tone for channel assignment. With our design, the proposed *CT protocol* (CTP) is capable of resolving more than 96% of the contentions within an ongoing frame transmission period; hence, the station that is assigned to access the medium in the next sequence can transmit its frame immediately after the ongoing transmission. This efficient channel assignment allows CTP to operate close to the theoretical maximum throughput of a WLAN MAC protocol.

Figure 7.1: Illustration of contention-tone protocol.

7.1 Protocol Detail

Firstly, we define a tone as a slot length transmission burst identifiable by all stations. The proposed protocol uses a separate signaling channel with a separate circuitry for the transmission and detection of CTs; hence, a station can perform the CT contention resolution operation even when the station is currently transmitting data on the data channel.

CTP is implemented as follows. Stations access the medium using DCF basic access method [2]. When a station starts a data frame transmission on the data channel, it also starts a *CT contention procedure* on the signaling channel concurrently. The goal of the CT contention procedure is to distributively assign a *winner* among a group of contending stations. The duration of the CT contention procedure is always shorter than the DCF data transmission period, which is inclusive of all overheads such as the acknowledgment reply, so that a winner can be decided by the end of the data transmission period. An illustration of the CTP operation is shown in Figure 7.1.

CTP divides the duration of the CT contention period into two consecutive time segments, which are called the *first segment* and the *second segment* for the earlier and the later time segments respectively. The first segment is accessible by stations which are currently transmitting data on the data channel. In most

circumstances, there is only one station accessing this segment. However, if a transmission collision occurs on the data channel, the number of stations accessing the first segment is the number of collided stations on the data channel. The purpose of the first segment is to resolve such an unlikely but inevitable collision on the data channel.

The second segment is accessible for stations that do not participate in the data transmission but has a backlogged data frame in their local buffer. CT contention procedures for the two segments are similar. As shown in Figure 7.1, each time segment is subdivided into several mini-slots named *contention slots*. The duration of a contention slot must be at least the sum of a CT detection time and the listen-to-transmit turnaround time. The number of contention slots in each time segment is a protocol parameter. Typically, the second segment consists of more contention slots due to the fact that the number of stations involved in the second segment is usually higher than that in the first segment, and with more contention slots, the probability of generating a single winner is higher. We shall describe in details the CT contention procedure as follows.

In CT contention procedure, the time segment starts with a CT that must be transmitted by all the involved stations. This CT signifies the existence of the segment. For each contention slot after the first CT, each involved station probabilistically transmits a CT. A station can either transmit or listen to the signaling channel, but not performing both during a contention slot. When listening to the channel, if a station detects a CT, it loses the contention and stops transmitting CTs for the rest of the segment period. As the procedure progresses, more stations will lose the contention leaving fewer stations transmitting CTs towards the end of the segment period. The winner of the contention shall be the last station

that transmits a CT.

Obviously, it is possible that more than one winner is resulted by the end of a time segment. According to CTP, where winners transmit immediately after the ongoing transmission, a data transmission collision occurs if the number of winners is more than one. However, with a proper protocol design in practical situations, such a collision can be rare. In the next section, we show that it is possible to design a practical system such that the success rate to obtain a single winner is more than 96%.

With each segment produces a winner, the priority is given to the winner of the second segment if the second segment of the CTP exists. Hence, the winner of the second segment will start its transmission immediately after the current transmission is completed (successfully or not). If the second segment does not exist, the winner of the first segment will be the ultimate winner. It is possible that this winner is also the sender of the data transmission during the CT contention. If its data transmission is successful and it has no more data to transmit, it simply turns itself to idle and let the medium be available for future access from any station generating new data frames. With this CT contention procedure, the data channel can be fully utilized as long as there is a station with a data frame to transmit.

7.2 Success Rate of CTP

The performance of CTP depends on the efficiency of the CT contention procedure. We first derive the probability that a single winner is resulted from the CT contention procedure. This probability determines the number of successful

transmissions from the total number of possible transmissions, i.e. the success rate of the system. Let N denote the total number of stations in the network and K_j denote a random variable of the number of stations contending on the j -th contention slot. Here, we assume that the stations are in saturation condition, i.e. they always have frames to transmit [8]. The number of stations contending on the j -th slot depends only on the number of stations contending on the previous slot, i.e. $(j - 1)$ -th slot. This relationship forms a recursive equation which defines the probability density of K_j conditioned on K_{j-1} :

$$\begin{aligned} Pr\{K_j = k_j | K_{j-1} = k_{j-1}\} = & \quad (7.1) \\ & \begin{cases} \binom{k_{j-1}}{k_j} \theta^{k_j} (1 - \theta)^{k_{j-1} - k_j}, & \text{if } k_j = 1, 2, \dots, k_{j-1} - 1 \\ \theta^{k_{j-1}} + (1 - \theta)^{k_{j-1}}, & \text{if } k_j = k_{j-1}, \end{cases} \end{aligned}$$

where θ is the slot access probability, i.e. the probability to transmit a CT on a contention slot. Unconditioning (7.1), we get a recursive equation describing the probability density of K_j :

$$\begin{aligned} Pr\{K_j = k_j\} = & [\theta^{k_j} + (1 - \theta)^{k_j}] Pr\{K_{j-1} = k_j\} \\ & + \sum_{m=k_j+1}^n \binom{m}{k_j} \theta^{k_j} (1 - \theta)^{m - k_j} Pr\{K_{j-1} = m\}. \end{aligned} \quad (7.2)$$

The probability of successfully resolving contentions among N stations, $P_S(N)$, is the probability that the number of contending stations on the last slot is one, i.e.

Nanyang Technological University

$Pr\{K_\omega = 1\}$:

$$P_S(N) = Pr\{K_{\omega-1} = 1\} + \sum_{m=2}^N m\theta(1-\theta)^{m-1}Pr\{K_{\omega-1} = m\}. \quad (7.3)$$

Given the number of stations, the number of contention slots, and the slot access probability, we can compute the probability of a successful transmission resulted from the CT contention using (7.3). We found that with nine contention slots and slot access probability between 0.3 and 0.4, we can achieve over 96% success rate for up to 100 stations. With three contention slots and slot access probability of 0.35, we can successfully resolve more than 70% of the contentions among up to 20 stations. Non-uniform access probability can yield better success rate. However, in favor of simplicity, we implement uniform access probability. Our argument is that a few percent improvement on the success rate will not affect the performance much, as even with uniform probability, CT contention efficiency is good enough to drive the protocol operating close to the theoretical maximum throughput.

With CTP, we expect that most of the contentions occur in the second segment; the second segment CT contention usually involves high number of stations, hence, we allocate more contention slots to the second segment, specifically 9 contention slots which gives 96% success rate. Moreover, priority is given to the winner of the second segment which gives incentive to promote the second segment's success rate.

On the other hand, we expect that the number of participating stations in the first segment is small. These stations may be those accessing an idle medium

at the same time or those claiming to be winners in the previous CT contention. Either of the above scenarios typically involves a small number of stations. Thus, we allocate only 3 contention slots to the first segment.

Adding the first compulsory CT, we design 4 contention slots for the first segment and 10 contention slots for the second segment. For protocol robustness, we enforce an empty mini-slot between the first and the second segments as a guard time. This results in 15 slots for the duration of the CT contention period. Considering the IEEE 802.11a standard [2] which specifies duration of $9 \mu s$ for the slot time, the duration of the CT contention period will be $135 \mu s$. This duration is still shorter than a DCF data frame transmission period even for a zero sized data frame¹.

7.3 Throughput Performance of CTP

Wireless communications impose high synchronization overhead at the physical layer on every transmitted frame. Besides, the necessity of an explicit positive acknowledgment further reduces the maximum achievable throughput at the MAC layer. With these consideration, let P denote the payload size, the maximum achievable throughput of a WLAN MAC protocol is

$$S_{max} = \frac{E[P]}{T_H + E[P] + T_{SIFS} + T_{ACK} + T_{DIFS}}, \quad (7.4)$$

¹A DCF data frame transmission period includes the overheads of header and an acknowledgment sent with the basic bit rate. For future high bit rate WLAN implementation, we have the option of either enforcing frame padding for very short data frames or reducing the CT contention slots. The former option is favorable as it will have imperceptible impact on the system performance.

where T_H and T_{ACK} are the time durations required to transmit the header and acknowledgment respectively; T_{SIFS} and T_{DIFS} denote the inter-spacing spaces between transmitted frames [2].

The throughput of CTP can be easily computed as follows. From (7.3), we determine the success rate for the channel assignment. Precisely, it is the probability that a data transmission is free from a collision. Assuming N saturated stations, for a successful data transmission, the success rate for the next channel assignment is $P_S(N - 1)$ as there are $N - 1$ participants in the second segment of the CT contention. If the CT contention results in c winners where $c > 1$, a collision occurs and the success rate of the following CT contention will be $P_S(N - c)$. Our numerical results suggest that $P_S(N - c)$ is not sensitive to c for our design². Hence, we assume that regardless of the outcome of the data frame transmission, the success rate for the next channel assignment is always $P_S(N - 1)$. With this approximation, the throughput of CTP can be determined by

$$\begin{aligned} S_{CTP} &= P_S(N - 1) \cdot S_{max} \\ &= \frac{P_S(N - 1)E[P]}{T_H + E[P] + T_{SIFS} + T_{ACK} + T_{DIFS}}. \end{aligned} \quad (7.5)$$

Figure 7.2 shows the saturation throughput performance of CTP compared with the DCF protocols. Similar to the previous two chapters, we consider CTP's signaling channel as an overhead, which we omit in the comparisons. CTP requires only a narrow contention tone channel, thus, it should be obvious from

²Except when $c = N$ which is a very rare event. This event occurs when all stations claim to be the winner of a CT contention. Even when this event occurs, it is expected that the recurrence of $c = N$ is unlikely due to the CT contention procedure in the first segment, and hence normal operation resumes immediately.

Figure 7.2: Saturation throughput performance of the proposed CTP compared with the DCF standard.

Figure 7.2 that CTP's performance gain is much higher than the performance of in-band signaling protocols with a larger combined channel. Symbols represent simulations results, whereas solid lines represent analytical results (we use the equations in Section 3.1 [14] for DCF). We use IEEE 802.11a parameters [2] (see table 5.1 in Chapter 5) with a bit rate of 54 Mbps and a frame size of 1000 bytes. We also plot the theoretical maximum throughput of the MAC protocol in the figure. Each point of the numerical results takes less than one second to compute, whereas each simulation run consumes a few minutes to complete. The figure shows that CTP can achieve throughput results that are very close to the theoretical maximum. The performance advantage is obvious when it is compared with the existing DCF standard protocols. For comparison, with 50 stations, CTP offers 61.7% additional goodput than that of the DCF basic access method.

7.4 Delay Performance of CTP

The saturation delay of CTP can be easily derived using Little's Theorem: $W = N/\lambda$. The arrival rate, λ , can be computed using a method similar with (7.5), which gives

$$S_{CTP} = \frac{P_S(N-1)}{T_H + E[P] + T_{SIFS} + T_{ACK} + T_{DIFS}}. \quad (7.6)$$

Having this, the final equation for the saturation delay is

$$D_{CTP} = \frac{N \cdot (T_H + E[P] + T_{SIFS} + T_{ACK} + T_{DIFS})}{P_S(N-1)}. \quad (7.7)$$

We show the saturation delay performance of CTP in Figure 7.3. We compare the delay performance of CTP with those of DCF basic access method and RTS/CTS method. We use the same parameters to compute the delay as those in the previous section. From the figure, we can observe that CTP can consistently achieve lower delays than what DCF can achieve.

7.5 Summary

In this chapter, we proposed a contention-tone protocol, which uses a separate narrow band signaling channel to resolve contentions among stations. Since the contention resolution occurs concurrently with the data transmission period, the channel assignment exhibits work conserving characteristic. This allows the protocol to operate at near maximum throughput. Our analysis confirmed that CTP resolves over 96% of the contentions successfully. Further analysis and simula-

Figure 7.3: Saturation delay performance of CTP compared with the DCF standard.

tion provide evidence of the performance advantage of CTP over the existing DCF standard. Combining DCF basic access method as the method to access an idle medium with our proposed CT contention to resolve collisions, CTP offers not only low transmission delay in light load, but also high efficiency in heavy load.

Chapter 8

Conclusion

This thesis covers both performance modeling and protocol designs, in which we have three contributions. On performance modeling, we have two main contributions: a refinement to the Bianchi's saturation model, and a saturation model for IEEE 802.11e EDCA. The last contribution is on protocol designs; we propose and analyze three WLANs protocols, which have a common theme: the use of out-of-band signaling.

In developing network protocols, there are specific goals to be achieved depending on the applications of the protocols. For example, in WLANs, some considerations for MAC protocols are throughput and delay performances, power consumption, implementation complexity, and hardware cost. Throughput performance is one of the most important criteria for network protocols, as the purpose of all communication systems is conveying information, and throughput is the measure of how fast an amount of information can be conveyed. Delay is another important criteria to measure the quality of a connection especially for real-time applications such as voice and video. The other considerations that we

mentioned earlier are engineering constraints that depends on the specific applications of the protocols. Most of the time, it is possible to reduce complexity and cost by reducing the throughput and delay qualities of a protocol. Owing to the importance of performance measurements of protocols, protocol modeling has an important research value. Protocol modeling allows protocol designers to capture and mathematically analyze details of the protocol; hence, with a protocol model, optimizing the protocol will be much easier.

For IEEE 802.11 DCF, Bianchi's saturation model [9, 8] proves to be accurate and tractable. Nevertheless, there are some missing details in this model that causes it not standard conforming. There are some research papers that improve Bianchi's model and claim to be more standard conforming such as the paper by Ziouva and Antonakopoulos [13] and the paper by Xiao [61]. However, we found that there is a detail that has been improperly modeled by those papers, specifically the detail of backoff freezing. We feel that it is important to notify the community of this issue, thus, the refinement presented in Chapter 3, which is published in [14]. This refinement will be useful for further researches for DCF protocols optimization, and it should also clear some doubts on the protocol details.

The other modeling contribution that we have introduced is a model to analyze IEEE 802.11 EDCA throughput and delay performances in saturation condition. This model captures both the contention window and inter-frame space differentiations. Using this model, we can compute both the saturation throughput and delay for stations from each *access category*. We also presented a method to derive the delay distribution of EDCA using transform analysis. This model should be useful especially for admission control design. The delay distribution derivation is useful for analyzing the performance of the system under admission control, as

the delay distribution measure is important for real-time traffic.

In this thesis, we have also presented three protocols for WLANs that make use of the out-of-band signaling technique. One important characteristic of the protocols is their compatibility with the existing IEEE 802.11 MAC protocol. We have presented the performance analysis of the protocols and have shown that out-of-band signaling technique performs better compared with the current in-band signaling technique.

The first protocol, OBS, uses a contention scheme on the signaling channel for transmission request purposes. We have studied the performances of OBS analytically, and we use simulations to verify the analysis. The results assert that the OBS technique performs better than the in-band signaling technique used by the existing scheme. The results also suggest that our OBS scheme is more effective compared to the existing scheme for higher data rate WLANs. Furthermore, increasing the signaling channel's bit rate allows the saturation throughput of the data channel to grow further for higher bit rate WLANs.

The second protocol, OBPS, uses an efficient polling mechanism on a low data rate channel for stations to request for frame transmissions. OBPS allows priority signaling for upstream transmission of real-time frames. It also provides transmission sequence for time-critical downstream frame to preempt the transmission of downstream frame for the station being polled in order to achieve a priority service. Specifically, we have simulated OBPS with a frame scheduling algorithm - DRR for upstream and downstream transmission. This frame scheduling algorithm imposes rate control for each station and prevents starvation of non-priority frame transmissions.

With its ability to interleave upstream and downstream frames on the data

channel, OBPS can provide good worst-case delay guarantees for upstream and downstream traffic on a half-duplex channel. We have shown the performances of OBPS through analytical and simulation results. We have shown that OBPS utilizes the wireless channel efficiently from our throughput analysis. The derived delay bounds for OBPS provide scheduling algorithms the cycle times for delivery of upstream and downstream traffic. These cycle times are important parameters for analyzing scheduling algorithms that can be deployed to achieve QoS guarantees for QoS-sensitive real-time frames. Our delay bounds are realistic as it is based on upstream and downstream frame transmissions on a half-duplex channel. Additionally, the efficiency of OBPS allows it to be deployed on higher data rate channels.

The third protocol, CTP, uses contention tone on a separate low bit rate signaling channel to reserve a transmission slot on the high bit rate data channel. Shortly explained, in CTP, stations transmit a sequence of short contention tone, i.e. a sequence of either a short transmission burst that can be identified by the other stations or an idle period. During the contention tone sequence transmission, if a station detects a contention tone from other stations, it will stop the transmission and retry the contention tone sequence transmission again on the next chance. We show that this mechanism can resolve close to 100% of the contention successfully; hence, it can achieve close to the maximum achievable MAC layer throughput.

Compared with in-band protocols, implementation wise, these out-of-band signaling protocols require an extra radio transceiver to operate. This transceiver incurs additional cost to the out-of-band signaling devices. However, this additional cost is minimal considering that the transceiver transmits on much lower

Table 8.1: Comparison of OBS, OBPS and CTP.

	QoS support	Client complexity	Access point complexity
OBS	probabilistic (soft)	medium	medium
OBPS	deterministic (hard)	low	high
CTP	probabilistic (soft)	high	low

speed or only transmits contention tone.

These three protocols have different implementation complexities. OBPS requires complex access point, but it has low complexity for the stations. OBS requires medium complexity of both the access point and the stations. On the other hand, CTP has high complexity stations implementation. These three protocols have their own uses for the specific applications. Table 8.1 summarizes the comparison of OBS, OBPS and CTP in terms of QoS support and implementation complexity. The choice of protocol depends on the design trade-off required, which is outside the scope of this thesis considering the vast amount of possible applications.

8.1 Future Considerations

I myself would like to end this thesis without any issue left. However, research is a continuation process; every completed project introduces many more interesting topics to tackle. Therefore, here, I list out some issues that I think are interesting topics for future works.

Admittedly, the protocols and the analysis presented in this thesis can still be improved. One part missing from the protocols is the interconnection with the physical layer. There are some considerations and tradeoffs that must be done on

both the MAC layer and physical layer for the protocols to be efficient. This will be one interesting future research to be done. One particular interesting focus is the effect of the use of two channels on the power consumption of the WLANs device. The consideration of power consumption has become more prominent with the emergence of sensor network applications. Another interesting idea is on using more than one signaling channel to develop protocol based on CTP or OBS.

Applying our EDCA model to develop an admission control is also an interesting issue to work on. The issue of QoS support has been popular recently, but, like we mentioned in the introduction, it still lacks researches on the implementation issues. Admission control, bandwidth reservation and management, and QoS improvements are some applications that can benefit from the EDCA model.

Acronyms and Abbreviations

AC	Access category
ACK	Acknowledgment
AIFS	Arbitrary inter frame space
AP	Access point
BT	Busy tone
BTMA	Busy tone multiple access
CCA	Clear channel assessment
CFB	Contention free bursting
CSMA/CA	Carrier sense multiple access with collision avoidance
CSMA/CD	Carrier sense multiple access with collision detection
CT	Contention tone
CTP	Contention tone protocol
CTS	Clear-to-send
CW	Contention window
DBTMA	Dual busy tone multiple access
DC	Deficit counter
DCF	Distributed coordination function
DIFS	Distributed inter frame space
DRR	Deficit round robin
EDCA	Enhanced distributed channel access
EIFS	Extended inter frame space
HCF	Hybrid coordination function
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet protocol
MAC	Medium access control

Nanyang Technological University

MACA Multiple access with collision avoidance
MACAW Multiple access with collision avoidance for wireless
NAV Network allocation vector
NSL Non-priority signaling list
OBPS Out-of-band polling scheme
OBS Out-of-band signaling scheme
PCF Point coordination function
PIFS Point coordination inter frame space
PSL Priority signaling list
QoS Quality of service
RFT Request for transmission
RI Request-to-be-queued indicator
RTS Request-to-send
SRMA Split-channel reservation multiple access
SIFS Short inter frame space
SSQ Single server queue
TDMA Time division multiple access
TSA Transmission sequence A
TXOP Transmission opportunity
VoIP Voice over IP
WLANs Wireless local area networks

Bibliography

- [1] *Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications*, IEEE Std. 802.11, 1997.
- [2] *Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: High-Speed Physical Layer in The 5 GHz Band*, IEEE Std. 802.11a, Sept. 1999.
- [3] *Part 11: Wireless LAN Medium Access Control (MAC) And Physical Layer (PHY) Specifications: High-speed Physical Layer Extension In The 2.4 GHz Band*, IEEE Std. 802.11b, Sept. 1999.
- [4] *Part 11: Wireless LAN Medium Access Control (MAC) And Physical Layer (PHY) Specifications Amendment 4: Further Higher Data Rate Extension in the 2.4 GHz Band*, IEEE Std. 802.11g, June 2003.
- [5] A. S. Tanenbaum, *Computed Networks*, 4th ed. Upper Saddle River, New Jersey: Prentice Hall PTR, 2003.
- [6] *Draft Supplement to Part 11: Wireless Medium Access Control (MAC) and Physical Layer Specification: Medium Access Control (MAC) Enhancements for Quality of Service (QoS)*, IEEE 802.11 WG Std. 802.11e, 2002.
- [7] Y. Xiao and J. Rosdahl, "Throughput and delay limits of IEEE 802.11," *IEEE Communications Letters*, vol. 6, no. 8, pp. 355–357, Aug. 2002.
- [8] G. Bianchi, "Performance analysis of the IEEE 802.11 distributed coordination function," *IEEE Journal on Selected Areas in Communications*, vol. 18, no. 3, pp. 535–547, Mar. 2000.

- [9] ———, “IEEE 802.11—saturation throughput analysis,” *IEEE Communications Letters*, vol. 2, no. 12, pp. 318–320, Dec. 1998.
- [10] F. A. Tobagi and L. Kleinrock, “Packet switching in radio channels: Part II—the hidden terminal problem in carrier sense multiple-access and the busy-tone solution,” *IEEE Transactions on Communications*, vol. COM-23, no. 12, pp. 1417–1433, Dec. 1975.
- [11] Z. J. Haas and J. Deng, “Dual busy tone multiple access (DBTMA)—a multiple access control scheme for ad hoc networks,” *IEEE Transactions on Communications*, vol. 50, no. 6, pp. 975–984, June 2002.
- [12] C. H. Foh and M. Zukerman, “Performance analysis of the IEEE 802.11 MAC protocol,” in *Proc. European Wireless 2002*, Florence, Italy, Feb. 2002.
- [13] E. Ziouva and T. Antonakopoulos, “CSMA/CA performance under high traffic conditions: throughput and delay analysis,” *Computer Communications*, vol. 25, no. 3, pp. 313–321, 2002.
- [14] C. H. Foh and J. W. Tantra, “Comments on IEEE 802.11 saturation throughput analysis with freezing of backoff counters,” *IEEE Communications Letters*, vol. 9, no. 2, pp. 130–132, Feb. 2005.
- [15] J. W. Tantra, C. H. Foh, and A. B. Mnaouer, “Throughput and delay analysis of the IEEE 802.11e EDCA saturation,” in *Proc. IEEE ICC*, Seoul, Korea, May 2005.
- [16] ———, “Delay analysis of the IEEE 802.11e EDCA,” in preparation for journal submission.
- [17] J. W. Tantra, C. H. Foh, and B. S. Lee, “An efficient scheduling scheme for high speed IEEE 802.11 WLANs,” in *Proc. IEEE VTC*, Orlando, Florida, USA, Oct. 2003.
- [18] J. W. Tantra, C. H. Foh, G. Bianchi, and I. Tinnirello, “Performance analysis of the out-of-band signaling scheme for high speed wireless LANs,” in *Proc. IEEE Globecom*, Dallas, Texas, USA, Nov.-Dec. 2004.

Bibliography

- [19] ———, “Out-of-band signaling scheme for high speed wireless LANs,” *IEEE Transactions on Wireless Communications*, to be published.
- [20] T. M. Lim, J. W. Tantra, C. H. Foh, and B. S. Lee, “Out-of-band polling scheme for QoS support in wireless LANs,” *International Journal of Communication Systems, Special Issue on QoS Support and Service Differentiation in Wireless Networks*, vol. 17, no. 6, pp. 643–661, Aug. 2004.
- [21] J. W. Tantra and C. H. Foh, “Achieving near maximum throughput in IEEE 802.11 WLANs with contention tone,” *IEEE Communications Letters*, vol. 10, no. 9, pp. 658–660, Sept. 2006.
- [22] C. Xu and Z. Yang, “Non-saturated throughput analysis of IEEE 802.11 ad hoc networks,” *IEICE Transactions on Information and Systems*, vol. E89-D, no. 5, pp. 1676 – 1678, May 2006.
- [23] G. S. Paschos, I. Papapanagiotou, S. A. Kotsopoulos, and G. K. Karagiannis, “A new MAC protocol with pseudo-TDMA behavior for supporting quality of service in 802.11 wireless LANs,” *EURASIP Journal on Wireless Communications and Networking*, vol. 2006, pp. 1 – 9, 2006.
- [24] T. Issariyakul, D. Niyato, E. Hossain, and A. S. Alfa, “Exact distribution of access delay in IEEE 802.11 DCF MAC,” in *Proc. IEEE Globecom*, St. Louis, Missouri, USA, Nov.-Dec. 2005.
- [25] N. Abramson, “The ALOHA system - another alternative for computer communications,” in *Proc. of Fall Joint Computer Conference, AFIPS Conference*, 1970.
- [26] L. G. Roberts, “ALOHA packet systems with and without slots and capture,” *ASS Note 8, Stanford Research Institute, Advanced Research Projects Agency, Network Information Center*, 1972.
- [27] P. Karn, “MACA — a new channel access protocol for packet radio,” in *Proc. ARRL/CRRL Amateur Radio Ninth Computer Networking Conference*, 1990.

- [28] V. Bharghavan, A. Demers, S. Shenker, and L. Zhang, “MACAW: A media access protocols for wireless lans,” in *Proc. ACM SIGCOMM '94*, 1994.
- [29] F. A. Tobagi and L. Kleinrock, “Packet switching in radio channels: Part III—polling and (dynamic) split-channel reservation multiple access,” *IEEE Transactions on Communications*, vol. COM-24, no. 8, pp. 832–845, Aug. 1976.
- [30] —, “Packet switching in radio channels: Part I—carrier sense multiple-access modes and their throughput-delay characteristics,” *IEEE Transactions on Communications*, vol. COM-23, no. 12, pp. 1400–1416, Dec. 1975.
- [31] L. Kleinrock and S. S. Lam, “Packet switching in a a multiaccess broadcast channel: Performance evaluation,” *IEEE Transactions on Communications*, vol. COM-23, no. 4, pp. 410–423, Apr. 1975.
- [32] F. A. Tobagi and V. B. Hunt, “Performance analysis of carrier sense multiple access with collision detection,” *Computer Networks*, vol. 4, pp. 245–259, 1980.
- [33] H. Takagi and L. Kleinrock, “Throughput analysis for persistent CSMA systems,” *IEEE Transactions on Communications*, vol. COM-33, no. 7, pp. 627 – 638, July 1985.
- [34] —, “Correction to ‘Throughput analysis for persistent CSMA systems’,” *IEEE Transactions on Communications*, vol. COM-35, no. 2, pp. 243 – 245, Feb. 1987.
- [35] K. Sohraby, M. L. Molle, and A. N. Venetsanopoulos, “Comments on ‘Throughput analysis for persistent CSMA systems’,” *IEEE Transactions on Communications*, vol. COM-35, no. 2, pp. 240 – 243, Feb. 1987.
- [36] G. Bianchi, L. Fratta, and M. Oliveri, “Performance evaluation and enhancement of the CSMA/CA MAC protocol for 802.11 wireless LANs,” in *Proc. IEEE PIMRC*, Taipei, Taiwan, R.O.C., Oct. 1996.

Bibliography

-
- [37] F. Cali, M. Conti, and E. Gregori, "IEEE 802.11 protocol: Design and performance evaluation of an adaptive backoff mechanism," *IEEE Journal on Selected Areas in Communications*, vol. 18, no. 19, pp. 1774–1786, Sept. 2000.
 - [38] —, "Dynamic tuning of the IEEE 802.11 protocol to achieve a theoretical throughput limit," *IEEE/ACM Transactions on Networking*, vol. 8, no. 6, pp. 785–790, Dec. 2000.
 - [39] C. Wang, B. Li, and L. Li, "A new collision resolution mechanism to enhance the performance of IEEE 802.11 DCF," *IEEE Transactions on Vehicular Technology*, vol. 53, no. 4, pp. 1235 – 1246, July 2004.
 - [40] Y. Kwon, Y. Fang, and H. Latchman, "A novel MAC protocol with fast collision resolution for wireless LANs," in *Proc. IEEE Infocomm*, San Francisco, USA, March-April 2003.
 - [41] —, "Design of MAC protocols with fast collision resolution for wireless local area networks," *IEEE Transactions on Wireless Communications*, vol. 3, no. 3, pp. 793 – 807, May 2004.
 - [42] J. Choi, J. Yoo, S. Choi, and C. Kim, "EBA: an enhancement of the IEEE 802.11 DCF via distributed reservation," *IEEE Transactions on Mobile Computing*, vol. 4, no. 4, pp. 378–390, July-Aug. 2005.
 - [43] Z. ning Kong, D. H. Tsang, and B. Bensaou, "Adaptive RTS/CTS mechanism for IEEE 802.11 WLANs to achieve optimal performance," in *Proc. IEEE ICC*, Paris, France, June 2004.
 - [44] P. Chevillat, J. Jelitto, A. N. Barreto, and H. Truong, "A dynamic link adaptation algorithm for IEEE 802.11a wireless LANs," in *Proc. IEEE International Conference on Communications*, Anchorage, Alaska, USA, May 2003.
 - [45] D. Qiao and S. Choi, "Goodput enhancement of IEEE 802.11a wireless LAN via link adaptation," in *Proc. IEEE International Conference on Communications*, Helsinki, Finland, June 2001.
-

- [46] D. Qiao, S. Choi, and K. G. Shin, “Goodput analysis and link adaptation for IEEE 802.11a wireless LANs,” *IEEE Transactions on Mobile Computing*, vol. 1, no. 4, pp. 278–292, 2002.
- [47] D. Qiao and S. Choi, “Fast-responsive link adaptation for IEEE 802.11 WLANs,” in *Proc. IEEE ICC*, Seoul, Korea, May 2005.
- [48] L. Y. Hung and B. Bensaou, “Improving wireless local area networks goodput through link adaptation and scheduling,” in *Proc. IEEE WCNC*, New Orleans, LA, USA, Mar. 2005.
- [49] B.-S. Kim, Y. Fang, T. F. Wong, and Y. Kwon, “Throughput enhancement through dynamic fragmentation in wireless LANs,” *IEEE Transactions on Vehicular Technology*, vol. 54, no. 4, pp. 1415 – 1425, July 2005.
- [50] O. Sharon and E. Altman, “An efficient polling MAC for wireless LANs,” *IEEE/ACM Transactions on Networking*, vol. 9, no. 4, pp. 439–451, Aug. 2001.
- [51] A. Ganz, A. Phonphoem, and Z. Ganz, “Robust superpoll with chaining protocol for IEEE 802.11 wireless LANs in support of multimedia applications,” *Wireless Networks*, vol. 7, pp. 65–73, 2001.
- [52] S.-C. Lo, G. Lee, and W.-T. Chen, “An efficient multipolling mechanism for IEEE 802.11 wireless LANs,” *IEEE Transactions on Computers*, vol. 52, no. 6, pp. 764–778, June 2003.
- [53] J. W. Tantra, T. M. Lim, C. H. Foh, and B. S. Lee, “An enhanced superpoll protocol for IEEE 802.11 pcf with rate differentiation,” in *Proc. of the Fifth IFIP TC6 International Conference on Mobile and Wireless Communications Networks (to appear)*, Singapore, Oct. 2003.
- [54] B.-S. Kim, S. W. Kim, Y. Fang, and T. F. Wong, “Two-step multipolling MAC protocol for wireless LANs,” *IEEE Journal on Selected Areas in Communications*, vol. 23, no. 5, pp. 1276 – 1286, June 2005.
- [55] Y. Xiao, “Packing mechanisms for the IEEE 802.11n wireless LANs,” in *Proc. IEEE Globecom*, Dallas, Texas, USA, Nov. 2004.

Bibliography

- [56] —, “IEEE 802.11 performance enhancement via concatenation and piggyback mechanisms,” *IEEE Transactions on Wireless Communications*, vol. 4, no. 5, pp. 2182 – 2192, Sept. 2005.
- [57] —, “IEEE 802.11n: enhancements for higher throughput in wireless LANS,” *IEEE Wireless Communications Magazine*, vol. 12, no. 6, pp. 82 – 91, Dec. 2005.
- [58] T. S. Ho and K. C. Chen, “Performance evaluation and enhancement of the CSMA/CA MAC protocol for 802.11 wireless LANs,” in *Proc. IEEE PIMRC*, Taipei, Taiwan, R.O.C., Oct. 1996.
- [59] H. S. Chhaya and S. Gupta, “Performance modeling of asynchronous data transfer methods of IEEE 802.11 MAC protocol,” *Wireless Networks*, vol. 3, pp. 217–234, 1997.
- [60] F. Cali, M. Conti, and E. Gregori, “IEEE 802.11 wireless LAN: Capacity analysis and protocol enhancement,” in *Proc. IEEE Infocomm*, Kobe, Japan, Apr. 1997.
- [61] Y. Xiao, “Performance analysis of IEEE 802.11e EDCF under saturation condition,” in *Proc. IEEE ICC*, Paris, France, June 2004.
- [62] H. Wu, Y. Peng, K. Long, S. Cheng, and J. Ma, “Performance of reliable transport protocol over IEEE 802.11 wireless LAN: analysis and enhancement,” in *Proc. IEEE INFOCOM*, New York, USA, June 2002.
- [63] P. Chatzimisios, A. C. Boucouvalas, and V. Vitsas, “IEEE 802.11 packet delay - a finite retry limit analysis,” in *Proc. IEEE Globecom*, San Francisco, USA, Dec. 2003.
- [64] A. Zanella and F. D. Pellegrini, “Statistical characterization of the service time in saturated IEEE 802.11 networks,” *IEEE Communications Letters*, vol. 9, no. 3, pp. 225–227, Mar. 2005.
- [65] R. German and A. Heindl, “Performance evaluation of IEEE 802.11 wireless LANs with stochastic petri nets,” in *Proc. International Workshop on Petri Nets and Performance Models*, Zaragosa, Spain, Sept. 1999.

- [66] V. Vishnevsky and A. Lyakhov, "802.11 LANs: Saturation throughput in the presence of noise," in *Proc. IFIP-TC6 Networking Conference*, Pisa, Italy, May 2002.
- [67] Z. Hadzi-Velkov and B. Spasenovski, "Saturation throughput - delay analysis of IEEE 802.11 DCF in fading channel," in *Proc. IEEE ICC*, Anchorage, Alaska, May 2003.
- [68] M. M. Carvalho and J. J. Garcia-Luna-Aceves, "Delay analysis of IEEE 802.11 in single-hop networks," in *Proc. of the IEEE International Conference on Network Protocols*, Atlanta, Georgia, USA, Nov. 2003.
- [69] M. K. Panda, A. Kumar, and S. H. Srinivasan, "Saturation throughput analysis of a system of interfering IEEE 802.11 WLANs," in *Proc. IEEE International Symposium on a World of Wireless Mobile and Multimedia Networks (WoWMoM'05)*, Taormina, Italy, June 2005.
- [70] O. Tickoo and B. Sikdar, "A queueing model for finite load IEEE 802.11 random access MAC," in *Proc. IEEE ICC*, Paris, France, June 2004.
- [71] —, "Queueing analysis and delay mitigation in IEEE 802.11 random access MAC based wireless networks," in *Proc. IEEE Infocomm*, Hong Kong, China, Mar. 2004.
- [72] G. Bianchi and I. Tinnirello, "Kalman filter estimation of the number of competing terminals in an IEEE 802.11 network," in *Proc. IEEE Infocomm*, San Francisco, USA, March-April 2003.
- [73] P. Patil and V. Apte, "Sizing of IEEE 802.11 wireless LANs," in *Proc. ACM WMASH*, Cologne, Germany, Sept. 2005.
- [74] W. Pattara-atikom, P. Krishnamurthy, and S. Banerjee, "Distributed mechanisms for quality of service in wireless LANs," *IEEE Wireless Communications Magazine*, vol. 4, no. 3, pp. 26–34, June 2003.
- [75] J. Deng and R. S. Chang, "A priority scheme for IEEE 802.11 DCF access method," *IEICE Transaction on Communications*, vol. E82-B, no. 1, Jan. 1999.

Bibliography

- [76] I. Aad and C. Castelluccia, "Differentiation mechanisms for IEEE 802.11," in *Proc. of the IEEE INFOCOM*, Anchorage, Alaska, USA, Apr. 2001.
- [77] N. H. Vaidya, P. Bahl, and S. Gupta, "Distributed fair scheduling in a wireless LAN," *Sixth Annual International Conference on Mobile Computing and Networking, Boston*, Aug. 2000.
- [78] W. Pattara-atikom, S. Banerjee, and P. Krishnamurthy, "Starvation prevention and quality of service in wireless LANs," in *Proc. IEEE WPMC*, Oct. 2002.
- [79] A. Veres, A. T. Campbell, M. Barry, and L. H. Sun, "Supporting service differentiation in wireless packet networks using distributed control," *IEEE Journal on Selected Areas in Communications*, vol. 19, no. 10, Oct. 2001.
- [80] T. Suzuki and S. Tasaka, "Performance evaluation of priority-based multimedia transmission with the PCF in an IEEE 802.11 standard wireless LAN," in *Proc. IEEE PIMRC*, San Diego, USA, September-October 2001.
- [81] A. Lindgren, A. Almquist, and O. Schelén, "Evaluation of quality of service schemes for IEEE 802.11 wireless LANs," in *Proc. of the IEEE Conference on Local Computer Networks*, Tampa, Florida, USA, Nov. 2001.
- [82] J. L. Sobrinho and A. S. Krishnakumar, "Quality-of-service in ad hoc carrier sense multiple access wireless networks," *IEEE Journal on Selected Areas in Communications*, vol. 17, no. 8, Aug. 1999.
- [83] D. Gu and J. Zhang, "QoS enhancement in IEEE 802.11 wireless local area networks," *IEEE Communications Magazine*, vol. 41, no. 6, pp. 120–124, June 2003.
- [84] S. Mangold, S. Choi, G. R. Hiertz, O. Klein, and B. Walke, "The evolution of wireless LANs and PANs - analysis of IEEE 802.11e for qos support in wireless LANs," *IEEE Wireless Communications Magazine*, vol. 10, no. 6, pp. 40–50, Dec. 2003.

- [85] R. S. Ranasinghe, L. L. H. Andrew, and D. Everitt, "Distributed contention-free traffic scheduling in IEEE 802.11 multimedia networks," *10th IEEE Workshop on Local and Metropolitan Area Networks*, pp. 18–28, 1999.
- [86] M. Shreedhar and G. Varghese, "Efficient fair queuing using deficit round robin," *IEEE/ACM Transactions on Networking*, vol. 4, no. 3, pp. 375–385, June 1996.
- [87] M. Ergen, D. Lee, R. Sengupta, and P. Varaiya, "WTRP — wireless token ring protocol," *IEEE Transactions on Vehicular Technology*, vol. 53, no. 6, pp. 1863 – 1881, Nov. 2004.
- [88] Y. Xiao, "Backoff-based priority schemes for IEEE 802.11," in *Proc. IEEE ICC*, Anchorage, Alaska, May 2003.
- [89] —, "A simple and effective priority scheme for IEEE 802.11," *IEEE Communications Letters*, vol. 7, no. 2, pp. 70–72, Feb. 2003.
- [90] —, "Performance analysis of priority schemes for IEEE 802.11 and IEEE 802.11e wireless LANs," *IEEE Transactions on Wireless Communications*, vol. 4, no. 4, pp. 1506 – 1515, July 2005.
- [91] J. W. Robinson and T. S. Randhawa, "Saturation throughput analysis of IEEE 802.11e enhanced distributed coordination function," *IEEE Journal on Selected Areas in Communications*, vol. 22, no. 5, pp. 917 – 928, June 2004.
- [92] Z. ning Kong, D. H. Tsang, B. Bensaou, and D. Gao, "Performance analysis of IEEE 802.11e contention-based channel access," *IEEE Journal on Selected Areas in Communications*, vol. 22, no. 10, pp. 2095 – 2106, Dec. 2004.
- [93] G. Bianchi and I. Tinnirello, "Analysis of priority mechanisms based on differentiated inter frame spacing in CSMA/CA," in *Proc. of the IEEE VTC*, Orlando, Florida, USA, Oct. 2003.

Bibliography

- [94] H. Zhu and I. Chlamtac, "An analytical model for IEEE 802.11e EDCF differential services," in *Proc. International Conference on Computer Communications and Networks*, Dallas, Texas, USA, Oct. 2003.
- [95] —, "Performance analysis for IEEE 802.11e EDCF service differentiation," *IEEE Transactions on Wireless Communications*, vol. 4, no. 4, pp. 1779 – 1788, July 2005.
- [96] J. Hui and M. Devetsikiotis, "Performance analysis of IEEE 802.11e EDCA by a unified model," in *Proc. IEEE Globecom*, Dallas, Texas, USA, November-December 2004.
- [97] —, "A unified model for the performance analysis of IEEE 802.11e EDCA," *IEEE Transactions on Communications*, vol. 53, no. 9, pp. 1498 – 1510, Sept. 2005.
- [98] L. V. Fausett, *Numerical Methods Algorithm and Applications*. Prentice Hall, 2003.
- [99] J. W. Tantra, C. H. Foh, I. Tinnirello, and G. Bianchi, "Analysis of the IEEE 802.11e EDCA under statistical traffic," in *Proc. IEEE ICC*, Istanbul, Turkey, June 2006.
- [100] M. Shreedhar and G. Varghese, "Efficient fair queuing using deficit round-robin," *IEEE/ACM Transactions on Networking*, vol. 4, no. 3, pp. 375–385, June 1996.
- [101] J. Ho and M. Fischer, "Signaling for parameterized QoS support," *IEEE 802.11-02/241r0*, Nov. 2002.