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Fairness and Transmission Opportunity Limit in IEEE 802.11e Enhanced Distributed Channel Access

Master's Thesis

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This thesis investigates the effect of transmission opportunity limit on fairness in IEEE802.11e enhanced distributed channel access. IEEE802.11e brings quality of service features into IEEE802.11 wireless local area networks. In stations operating with IEEE802.11e, traffic is divided into categories. Differentiation between these categories is achieved by using four parameters to control the channel access. This thesis investigates one of these parameters, the transmission opportunity limit, which controls the channel access duration. With the reference parameter values given in IEEE802.11e, as the network congestion level increases, low priority traffic suffers quickly to a point where none of it gets transmitted. This makes the network overall fairness poor.

To improve fairness while not disturbing high priority traffic, this thesis investigates the use of large transmission opportunity limit values. In the first set of simulations, the low priority traffic transmission opportunity limit values are set to infinite. This means that the low priority queue can send all its packets when it gains access to the channel. The results show that infinite transmission opportunity limit improves fairness when channel is getting congested. Also infinite transmission opportunity limit does not notably weaken high priority traffic performance.

Second set of simulations focuses on the network congestion level where the effect of the infinite transmission opportunity limit is the largest. In these simulations the transmission opportunity limit is set to static value ranging from zero to a maximum allowed value. The results from these simulations are similar to the results of the first simulation set.

Keywords: Transmission opportunity limit, IEEE 802.11e Enhanced distributed channel access, Fairness

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Tämä diplomityö tutkii lähetysaikarajan vaikutusta verkon reiluuteen IEEE802.11e tehostettuun ja hajautettuun kommunikaatiokanavaan pääsyyn. IEEE802.11e tuo palvelunlaatuominaisuuksia IEEE802.11 langattomiin verkkoihin. Asemat, jotka käyttävät IEEE802.11e-ominaisuuksia jakavat liikenteen neljään kategoriaan. Kategorioiden välinen erottelu saavutetaan neljällä parametrilla, jotka kontrolloivat kanavaan pääsyä. Tämä työ tutkii yhtä näistä parametreistä, lähetysaikarajaa, joka kontrolloi lähetyksen kestoa. IEEE802.11e antaa referenssiarvoja parametreille, mutta näillä arvoilla verkon kuormituksen lisääntyessä, alemman prioriteetin liikenne kärsii nopeasti. Hyvin pian kuormituksen lisääntyessä alemman prioriteetin liikenne ei pääse verkosta läpi lainkaan. Tällöin myös verkon reiluus on matala.

Reiluuden parantamiseksi, häiritsemättä korkean prioriteetin liikennettä, tämä työ tutkii ison lähetysaikarajan käyttöä. Ensimmäisessä simulaatiosarjassa alemman prioriteetin lähetysaikaraja on ääretön. Tämä tarkoitta sitä, että alemman prioriteetin jono voi lähettää kaikki pakettinsa kun se pääsee lähettämään. Tulokset osoittavat, että ääretön lähetysaikaraja parantaa reiluutta kun kanava on kuormittumassa. Tulokset osoittavat myös, että ääretön lähetysaikaraja ei merkittävästi heikennä korkean prioriteetin liikennettä.

Toinen simulaatiosarja keskittyy sellaiseen verkon kuormitustilaan, missä äärettömän lähetysaikarajan vaikutus on suurin. Näissä simulaatioissa lähetysaikarajan arvo on staattinen. Simulaatiosta toiseen lähetysaikarajan arvo muutetaan toiseen arvoon väliltä nolla-suurin sallittu arvo. Tulokset näistä simulaatioista ovat hyvin samanlaiset kuin ensimmäisen simulaatiosarjan tulokset.

Avainsanat: Lähetysaikaraja, IEEE802.11e tehostettu ja hajautettu kommunikaatiokanavaan pääsy, reiluus

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List of Acronyms

AC	Access Categories
ACK	Acknowledgement
AEDCF	Adaptive Enhanced Distributed Coordinating Function
AFEDCF	Adaptive Fair Enhanced Distributed Coordinating Function
AIFS	Arbitrary Interframe Space
AIFSN	Arbitrary Interframe Space Number
AP	Access Point
BSS	Basic Service Set
CA	Collision Avoidance
CBR	Constant Bit Rate
CFP	Contention Free Period
CP	Contention Period
CSMA	Carrier Sense Multiple Access
CTS	Clear to Send
CW	Contention Window
DCF	Distributes Coordination Function
DIFS	DCF Interframe Space
DLS	Direct Link Setup
DSSS	Direct Sequence Spread Spectrum
DWFQ	Distributed Weighted Fair Queuing
EDCA	Enhanced Distributed Channel Access
EHF	Extra High Frequency
FH	Frequency Hopping
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HCCA	Hybrid Coordination Function Controlled Channel Access
HCF	Hybrid Coordination Function
IEEE	Institute of Electrical and Electronics Engineering
IP	Internet Protocol
IR	Infrared
ITU	International Telecommunications Union
MAC	Media Access Control
MANET	Mobile Ad hoc Networks
MCCA	Multi-user Polling Controlled Channel Access
MPDU	MAC Protocol Data Unit
MSDU	MAC Service Data Unit
NS-2	Network Simulator 2
OFDM	Orthogonal Frequency Division Multiplexing
PC	Point Coordinator
PCF	Point Coordination Function
PHY	Physical
PIFS	PCF Interframe Space
QAP	QoS Enabled Access Point
QBSS	Quality of Service Capable Basic Service Set
QoS	Quality of Service

QSTA	Qos Enabled Stations
RTS	Request to Send
SHF	Super High Frequency
SI	Service Interval
SIFS	Short Interframe Space
STA	Station
TBTT	Target Beacon Transmission Time
Tcl	Tool Command Language
TCP	Transmission Control Protocol
TDMA	Time Division Multiplexing
TXOP	Transmission Opportunity Limit
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
UP	User Priority
VLF	Very Low Frequency
VMAC	Virtual MAC
VoIP	Voice over IP
WEP	Wireless Encryption Protocol
WiMax	Worldwide Interoperability of Microwave Access
WLAN	Wireless Local Area Network
WPA	Wi-fi Protected Access
WQF	Weighted Fair Queuing

1. Introduction

The first wireless network was developed in the University of Hawaii by Professor Norm Abramson and his team in the early 1970's. They wanted the Hawaiian Islands to be connected into a single wireless computer network. This they accomplished with seven computers distributed to four islands that connected with bidirectional links to a central computer in Oahu. This network was named AlohaNet and in early 1971 it was even connected to the ARPAnet in the continental USA.

From these humble but innovative beginnings wireless networks have taken over the world. In the past fifteen years wireless networks have spread all around the globe. Today it is even possible to send with an ordinary mobile phone a photo from 5000 meters at Mt. Everest instantly to Finland.

In addition to mobile phone networks that cover large areas, networks for smaller areas have become an everyday item. Wireless local area networks offer high data rates compared to GSM/GPRS/UMTS cost efficiently, their deployment is easy and their use convenient. There is a low barrier of entry into WLAN market due to unlicensed frequency bands and relative cheap hardware. This increases competition and drives down prices. In addition, personal user devices are getting cheaper and more powerful all the time and the usage of portable devices such as laptops has increased greatly. All these factors have made WLANs very popular.

In recent years also the popularity of applications with strict delay and throughput requirements has grown rapidly. Examples of such applications are voice over IP (VoIP) and video streaming. This increasing popularity places new demands to networks and devices used to connect to networks. In a perfect situation a network can handle all demands users make, but in reality this is not always the case. In situations where there are more demands than the network can meet, a decision must be made whether to prioritize some traffic over another. This means implementing some sort of a Quality of Service (QoS) scheme.

Bringing QoS features to WLANs is difficult due to their characteristics. Wireless networks are far more error prone than wired networks. The availability of radio frequency spectrum for WLAN use and the physical properties of waves propagating in the air create a theoretical limit for maximum channel capacity. Because these ultimate restrictions exist, it is important to use the channel as efficiently as possible. When bringing QoS features to WLANs, this need for maximum efficiency must be kept in mind. Also, because WLAN environment can rarely be totally controlled, true QoS guarantees are close to impossible. However, various levels of service are possible and they can be used to help meet the needs of demanding traffic types.

Institute of Electrical and Electronics Engineering (IEEE) develops a very popular wireless local area network (WLAN) standard family by the number of 802.11. This standard family uses the unlicensed spectrum. Originally it was not well equipped to handle situations where QoS features are needed. However, in 2005 IEEE approved an amendment named 802.11e. This amendment offers a new version of the media access control layer (MAC layer). It can be used to replace the existing MAC layer presented in the 802.11 standard approved in 1999. 802.11e aims to improve QoS features of IEEE802.11 wireless local area networks

IEEE802.11e is divided into two parts, the enhanced distributed channel access (EDCA) and the hybrid coordination function controlled channel access (HCCA). In the latter the channel access is centrally controlled, while in EDCA there is no central control. This thesis focuses on EDCA. In 802.11e EDCA, the differentiation between traffic is achieved by having four transmission queues instead of just one. Each queue has a different set of parameters that determine the frequency and duration of channel access. This creates differentiation between different traffic types, but efficiency and fairness issues remain.

In the 802.11e amendment parameter values are set statically. This does not take the overall network condition into account, which might lead to wasted bandwidth. Nor is there distinguishing between uplink and downlink flows, which creates unfairness between uplink and downlink flows. In addition, overall fairness in the network rapidly deteriorates as the amount of traffic increases. As the congestion increases the low priority flows start to suffer and very quickly no low priority traffic gets through at all. This total starvation of low priority flows makes the network unfair. Even though we want to prioritize high category flows we still want to be able to send some low priority traffic.

To improve fairness in EDCA, this thesis investigates modifying the transmission opportunity (TXOP) limit. TXOP limit is the parameter containing the channel access duration. In standard 802.11e, the low priority flows can only transmit one packet when and if they do get access to the channel. It is interesting to find out if sending suitable size packet clusters gives the low priority flows some chance of surviving. This thesis investigates modifying the TXOP limit. The aim of this thesis is to find out if modifying the TXOP limit improves 802.11e EDCA fairness, but in such a way that the delay sensitive traffic is not disturbed too much. Additionally, if changing the TXOP limit has an effect, the thesis studies how much and in which way the TXOP limit should be changed. In the thesis simulations are used to investigate the effect of varying the TXOP limit.

Results show that a large TXOP limit improves fairness. Additionally this improvement is not achieved at the expense of voice traffic. Voice traffic delay, throughput and packet delivery ratio do not suffer significantly compared to simulations with standard settings. However, the fairness improvement is not very large. The results also show that as number of transmitting stations increases, the network quickly becomes too congested for either kind of traffic and in such a situation modifying the TXOP limit is not useful.

This thesis is organized as follows. Chapter 2 describes the IEEE802.11 standard and the 802.11e amendment. It includes discussion about QoS in wireless local area networks in general and improvements IEEE802.11e amendment brings. Particular focus is on the details of IEEE802.11e EDCA. Chapter 3 discusses the concept of fairness in general and explains Jain’s fairness index, which is the metric used to measure fairness in this thesis. Chapter 4 presents earlier research in this area. Chapter 5 explains good simulation practices and the particulars of simulations for this thesis. Chapter 6 presents and discusses the simulations results. Finally Chapter 7 draws some final conclusions.

2. IEEE 802.11 Wireless local area networks

A key WLAN standard family is the IEEE 802.11 standard family. The 802.11 networks have become the de facto standard used around the world and are under constant development. This chapter describes this development and discusses some challenges that wireless networks face. The 802.11e amendment will be introduced and described in more detail.

2.1 Development of IEEE 802.11 standards

The IEEE 802.11 standards body was created in May 1989 motivated by regulations, which allowed for unlicensed transmissions in an 83 MHz band in the 2.4-GHz range. The progress was slow and careful, but finally in 1997 the first completed standard was ratified. This standard formed the basis for all later versions of 802.11 standards and amendments. It defined a common medium access control (MAC) and three physical access (PHY) methods. The PHY methods defined were: frequency hopping (FH), direct sequence spread spectrum (DSSS) and infrared (IR). Of these, IR has not been used commercially. The other two have been used in commercial applications with data rates of 1 and 2 Mbit/s [Bing2002].

The connection speeds reached initially were not satisfactory for the 802.11 group. They aimed to get at least the 10 Mbit/s data rate offered by the standard Ethernet at the time. So they continued to work and divided the research into two initiatives. The other considered the unlicensed 5-GHz band while the other focused on improving speed on the 2.4-GHz band; from the 5-GHz band research came the 802.11a standard and from the 2.4-GHz research came the 802.11b standard. Orthogonal frequency division multiplexing (OFDM) modulation scheme was incorporated into 802.11a while 802.11b had DSSS backward compatibility with two new data rates, 5.5 and 11 Mbit/s and two new coding forms [Bing2002].

After these standards there has been considerable activity in improving various aspects of the original standards. Notable improvements include 802.11g with 54

Mbit/s theoretical maximum speed and 802.11e, which attempts to improve QoS features. In 2007, IEEE created a revision 802.11-2007 to the 1999 version of 802.11 standard. This revision combines amendments a,b,d,e,g,h,i,j into one document. Table 1 shows a short history of the development of the 802.11 family. Years 2008-2009 depict planned development, since those amendments have not yet been ratified.

Table 1 Development of IEEE standards and amendments over the years.

1997	1999	2001	2003	2004
Legacy 802.11 2.4 GHz band Typ 1 Mbit/s Max 2 Mbit/s	**802.11a** 5.0 GHz band Typ 25 Mbit/s Max 54 Mbit/s Approx. 50m range **802.11b** 2.4 GHz band Typ 6.5 Mbit/s Max 11 Mbit/s Approx. 100m range	**802.11d** Roaming between regulatory domains	**802.11g** 2.4 GHz band Typ 25 Mbit/s Max 54 Mbit/s Approx. 100m range **802.11h** Spectrum management extension for 5GHz for Europe	**802.11i** MAC Security amendments **802.11j** Extensions for Japan
2005	**2007**	**2008** (Predicted)	**2009** (predicted)	
802.11e MAC leyel QoS enhancements	**802.11-2007** Revision to merge amendments a,b,d,e,g,h,i,j into 802.11 standard	**802.11r** Fast Roaming **802.11y** 3650-3700 MHz operation in USA **802.11k** Radio resource management	**802.11n** 2.4 or 5.0 GHz Typ 200 Mbit/s Max 540 Mbit/s Approx. 50m range **802.11p** Wireless access for the vehicular environment **802.11s** ESS mesh networking	**802.11u** Interworking with external networks **802.11z** Extensions to direct link setup **802.11w** Protected management frames **802.11v** Wireless network management

As wireless local area networks are being developed, researchers have to solve issues concerning security, range, speed and reliability. In addition to security issues common to wired networks, wireless networks face some unique challenges. Since the signal propagates freely in air, anyone can listen and catch it and try to break encryptions. The previously used wireless encryption protocol (WEP) is vulnerable and even wi-fi protected access (WPA) is not unbreakable. To address these security issues in 2004 IEEE approved a new amendment 802.11i that introduced WPA2, which offers stronger security than the two previously mentioned.

Range versus speed trade-off is a challenge in wireless transmissions. Figure 2 shows the range vs. speed of some wireless technologies. The current 802.11g offers speeds equivalent to those of worldwide interoperability of microwave access (WiMax) systems but with the cost of range, while 802.11n plans to offer speeds higher than WiMax. Cellular networks on the other hand have a range advantage over WLANs but not over WiMax. So far wireless technologies have not been very interoperable. This is changing however, for instance many cellular phones are now capable of connecting to WLANs.

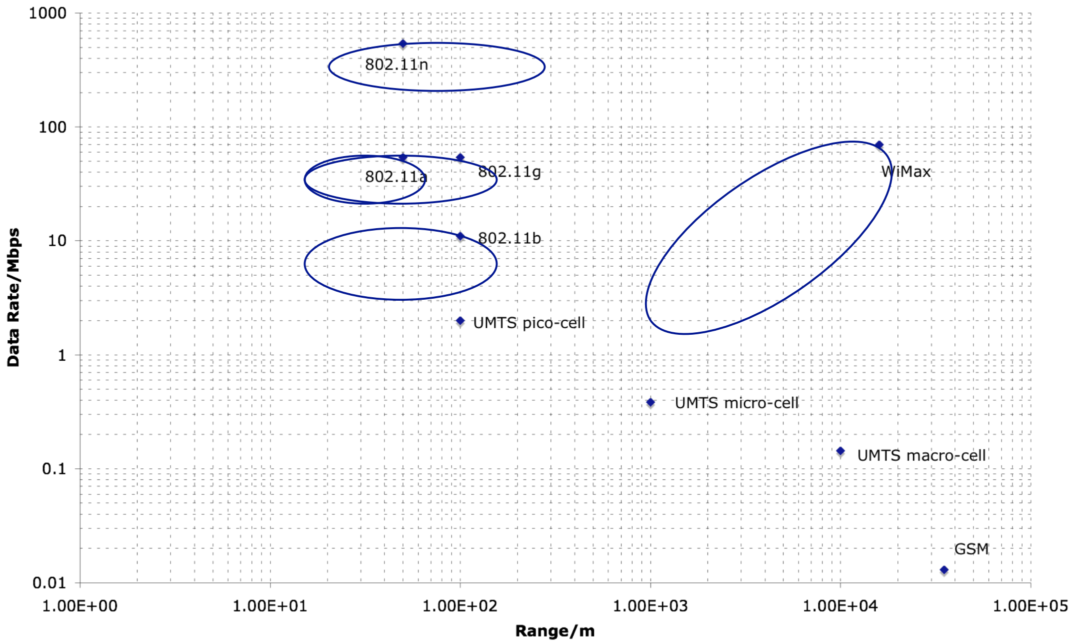

Figure 1 Range vs. Data Rate of some wireless technologies in logarithmic scale. The blue circles represent the area where a given technology typically operates.

Another issue in WLANs and in all wireless transmissions is the question of reliability. WLANs utilize the unlicensed spectrum in the super high frequency range (SHF) of 3 GHz -30 GHz. Figure 3 shows the radio wave spectrum ranging from very low frequencies (VLF) to extra high frequencies (EHF). Like any radio waves, WLAN signals encounter fading, shadowing, reflection, refraction, scattering and diffraction. In addition multipath propagation is possible, where the signal disperses over time. Neighboring transmission can also interfere with the signal [Lehto2006].

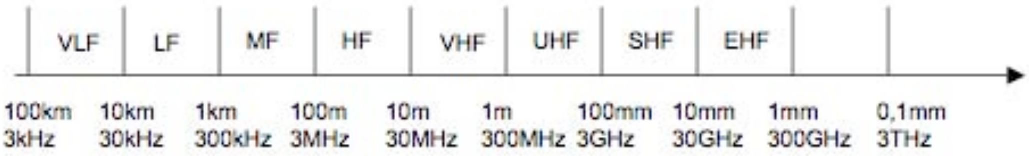

Figure 2 Radio wave spectrum

TCP traffic creates yet more issues to wireless environments. Data is lost in the volatile wireless environment also for reasons other than congestion. TCP on the other hand assumes losses are due congestion. When TCP assumes congestion, it proceeds to adjust its window size and retransmit packets. This leads to poor TCP performance in wireless networks. This is due to the fact that packets can be dropped because of errors in the wireless channel. Also, when TCP and UDP flows compete with each other the bandwidth distribution tends to favor UDP. This is because in case of congestion, TCP backs off due to its congestion control mechanism and UDP without any such mechanism consumes more aggressively the bandwidth left by TCP. Chapter 4 discusses some research done about TCP and IEEE802.11e.

2.1.1 Quality of Service

Wireless environments differ substantially from wired environments. The differences have to be taken into account when considering bringing QoS features to WLANs. In wireless networks, bandwidth tends to be scarce and channel conditions can vary greatly. Outside interference can be a burden. These issues lead to throughput limitations and increased loss, delay and jitter. QoS methods that work well in wired networks cannot necessarily be directly applied to a wireless network.

There are two opposite approaches for QoS support of Internet based services in wireless networks. The first is based on strict control, complex mechanisms and protocols and is similar to the Integrated Services [RFC1633]. Integrated services model focuses on providing per flow QoS. It can make strict bandwidth reservations for flows if every router on the way implements it. The model aims to integrate real-time services into best effort networks but is not very scalable. The other relies on the Internet design principle of simplicity and minimalism and is similar to Differentiated Services [RFC2475]. Differentiated Services allocates resources to a small number of traffic classes. Packets belong to one of these classes and receive service accordingly. It does not provide per flow QoS but instead it focuses on QoS for flow aggregates. This simple mechanism is more scalable than Integrated Services, but QoS cannot be guaranteed. IEEE802.11e has adopted both of these viewpoints. It alternates between tightly controlled and loosely controlled periods.

Radio link QoS is an important aspect in wireless network QoS. Phenomena such as propagation loss, multipath effects and interference degrade the channel quality and lead to retransmissions and dropped packets. This means increased latency and decreased throughput. This issue is unique to the wireless medium and has to be taken into account when designing QoS schemes for wireless networks.

It is possible for a WLAN device to change its PHY sending rate based on deteriorating channel quality. This is called link adaptation. Using link adaptation so that it is not a problem for wireless network performance is challenging. This is because even one user sending with a low rate can degrade the performance significantly. So to avoid this, when station negotiates QoS parameters, a minimum PHY sending rate should be specified and adhered to or no guarantees about QoS can be made. This is an important issue for 802.11e as well. Even one node transmitting at a low rate can degrade QoS available to all users.

Other components of QoS are admission control, scheduling, buffer management and policing. Admission control protects against resource overuse by comparing the service request with available resources. Scheduling algorithms handle packets at the network layer and decide which packets to forward. Both of these are of crucial importance to providing QoS in wireless networks. 802.11e amendment does not specify admission control or introduce an efficient scheduling algorithm, though both would be beneficial, and have been the subject of further research.

Ultimately it is the wireless medium that causes limitations to QoS service guarantees that can be made. True guarantees, especially in the unlicensed spectrum, are not necessarily possible.

2.2 From 802.11 to 802.11e

As IEEE standards grew in popularity there was a growing interest in improving the QoS properties in IEEE802.11 WLANs. In 802.11 standard, MAC layer was designed to support simple QoS features. However, it was never really implemented in actual hardware due to its limitations and problems.

2.2.1 802.11

In traditional 802.11, a set consisting of an access point (AP) and stations (STA) is called a basic service set (BSS). The basic MAC protocol in 802.11 is called the distributed coordination function (DCF). It uses carrier sense multiple access (CSMA) to listen to the channel before transmitting and collision avoidance (CA). Stations listen to the channel. When they sense the channel is not busy anymore, they have to wait a DCF interframe space (DIFS), which is the minimum waiting time after the channel is determined free. After DIFS the stations continue sensing the medium for an additional random time, the backoff time. The backoff time is derived from the contention window (CW) and it is a multiple of a slot time parameter. The number of slots is chosen randomly from an interval from 0 to CW. All stations have the same CW but choose their random backoff time by themselves, which reduces collisions. However, since all stations use the same CW_{min}, they have the same medium access priority. This does not result in a mechanism to differentiate between stations and their traffic, so QoS support in DCF is nonexistent [Mangold2003]. Figure 4 shows interframe space relationships and transmission process in the 802.11.

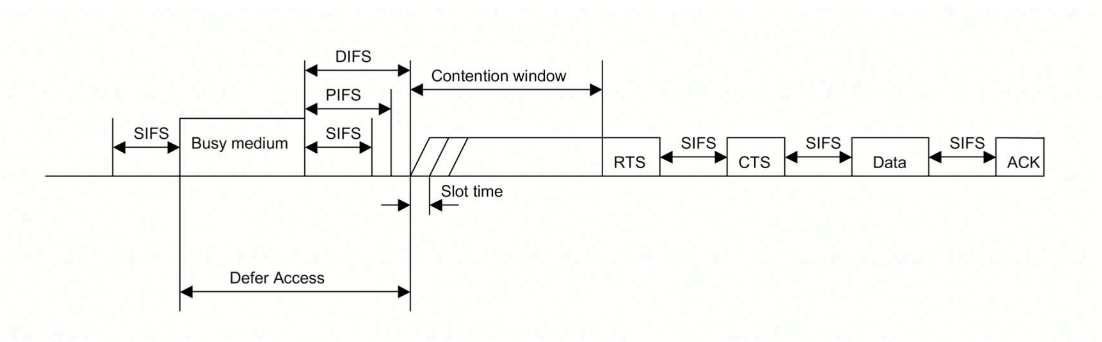

Figure 3. 802.11 standard transmission process and interframe time relationships

After each unsuccessful transmission the CW is doubled, which means the stations have to wait longer next time they attempt to transmit. The same happens during a random backoff performed after each successful transmission. Other mechanisms in use during DCF include requiring acknowledgement (ACK) messages for each transmitted MAC protocol data unit (MPDU). There is also an option of fragmenting MPDUs, which can reduce the need for retransmissions in high error situations. In addition, to help with the hidden terminal problem where two stations send at the same time because they cannot hear each other, a request-to-send/clear-to-send (RTS/CTS) mechanism can be used [Mangold2003].

In the 802.11 standard the point coordination function (PCF) was meant to provide some QoS support. In this mode a point coordinator (PC), normally the access point, takes control of the medium and decides who can transmit. Point coordinator polls the stations. If the polled station does not respond to the point coordinator’s poll in a PCF interframe space (PIFS), PC polls the next station. Because PIFS is longer than short interframe space (SIFS), the poll frame cannot interrupt an ongoing frame exchange, where SIFS is used.

In PCF the system alternates between a contention-free period (CFP) and a contention period (CP). During contention period DCF is used and during contention free period PCF is used. The AP also regularly transmits beacon frames, which help maintain synchronization of station timers and deliver other protocol related parameters. Beacon frames announce the change from CP to CFP.

There are some problems with PCF, which have lead to development enhancements in the form of 802.11e. Notable issues with PCF are unpredictable beacon delays due to poor cooperation between CP and CFP and unknown transmission durations of data transmission from the polled stations. Also the central polling scheme is inefficient so that it deteriorates PCF performance as traffic load increases [Qiang].

2.2.2 802.11e

802.11e was developed to improve the QoS features of the standard 802.11. 802.11e focuses on the MAC layer. It is not dependent on the physical layer chosen. 802.11a, 802.11b, 802.11g or any future standard can be used with it. In 802.11e a set of stations and an access point is called a quality of service capable basic service set (QBSS) and an access point is called QoS enabled access point (QAP). Stations are called QoS enabled stations (QSTA).

IEEE802.11e introduces a hybrid coordination function (HCF) for QoS provisioning. HCF is divided into contention and contention-free periods. Contention period is called enhanced distributed channel access (EDCA) and contention-free period is called HCF controlled channel access (HCCA). During EDCA, the stations compete for the medium according to preset parameters. In the HCCA mode the access point takes control of the medium and decides, based on a scheduling mechanism, how to distribute the transmission time [IEEE802.11e].

In addition to the main functions, IEEE802.11e introduces a few other improvements. Block acknowledgements allow several MAC service data units (MSDU) to be delivered without individual ACK frames. Only at the end of a block of frames the ACK is sent. Direct link setup (DLS) makes it possible for two QSTAs to communicate with each other without the QAP. After the setup procedure that still uses the QAP, the QSTAs can communicate directly with each other. In addition, each access category has an MSDU maximum lifetime to specify the time a frame may wait before being dropped. This helps in discarding frames of delay sensitive traffic that are no longer useful. Also it is possible for the QAP to poll stations even during EDCA. This way critical traffic can override all other traffic.

Ramos et al. [Ramos2005] identify three main challenges for QoS support in 802.11e networks. These are: handling time-varying network conditions, adapting to varying application profiles and managing link layer resources.

1. **Handling time-varying network conditions.** 802.11e does not take into account varying network conditions like channel condition and network load. Degrading channel condition can weaken the QoS differentiation mechanism of 802.11e so that it does not work as intended. Increasing amount of users in the network brings throughput degradation and even starvation; because of larger defer periods and higher collision probability.
2. **Adapting to varying application profiles.** The second problem area identified by Ramos et al. is the question of adapting to varying application profiles. The QoS requirements of a flow can vary significantly based on the application type. Requirements can also vary with time. Estimating these requirements correctly is crucial in designing and tuning the medium access mechanism. Poor estimation leads to unacceptable delays, buffer overflows and inefficiently used resources.
3. **Managing link layer resources.** Since 802.11e is a MAC layer enhancement, there remains a need for some kind of link layer cooperation, so that link layer resources can be optimally managed. General network goals for the QoS must be taken into account. Additionally, some kind of an overall admission control scheme should be designed. This admission control could also be used in EDCA, not just in HCCA like in the 802.11e amendment.

Addressing these three challenges has been a topic for several research papers but so far no all-encompassing solution has been proposed.

2.2.3 EDCA

To provide prioritized QoS, IEEE802.11 EDCA enhances the original IEEE802.11 DCF by introducing user priorities (UP) and access categories (AC). When traffic arrives to the MAC layer it has a user priority value that is mapped into an access category. Table 1 shows the mapping specified in the amendment. User priority zero is mapped between two and three because of IEEE802.1d bridge specification

[IEEE802.11e]. The highest AC is the voice category and lowest is the background category.

Table 2. IEEE802.11e user priorities to access categories mappings [IEE802.11e]

User priority (UP)	Access category (AC)	Designation
1	AC_BK	Background
2	AC_BK	Background
0	AC_BE	Best Effort
3	AC_BE	Best Effort
4	AC_VI	Video
5	AC_VI	Video
6	AC_VO	Voice
7	AC_VO	Voice

Each AC has its own transmission queue and an own set of parameters that determine channel access frequency and duration. These parameters are called the EDCA parameter set. Figure 5 shows a sketch of the new queue model. In addition to collisions between competing QSTAs, collisions can occur between queues in one QSTA. These are called virtual collision since packets don't actually collide. In such a situation the queue with higher priority gets the channel access while the lower priority queue backs off.

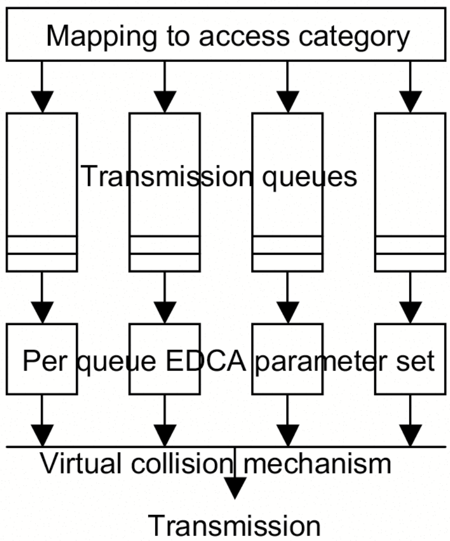

Figure 4. 802.11e channel access mechanism

Each AC has a different EDCA parameter set. The EDCA parameter set consists of the arbitrary interframe space number (AIFSN), contention window minimum and maximum (CW_{min}, CW_{max}) and transmission opportunity limit (TXOP limit). These will all be discussed in more detailed in later sections. QAP sets these parameters and transmits them to the QSTAs as part of periodic beacon frames. QAP broadcasts beacon frames at regular interval. The next target beacon transmission time (TBTT) is always announced in the previous beacon frame. As opposed to the 802.11, in 802.11e no QSTA may transmit across the TBTT. This way beacon delay is reduced and HC has a better control over the QBSS [IEEE802.11e].

Figure 6 shows the EDCA transmission process and interframe space relationships. The basic CW backoff mechanism remains the same as earlier, but with four different CW sizes and AIFS values there is differentiation between queues. High priority queues get to access the channel considerably more than low priority queues if they have packets to send. AIFS values should be selected so that earliest possible access time for QSTAs is DIFS.

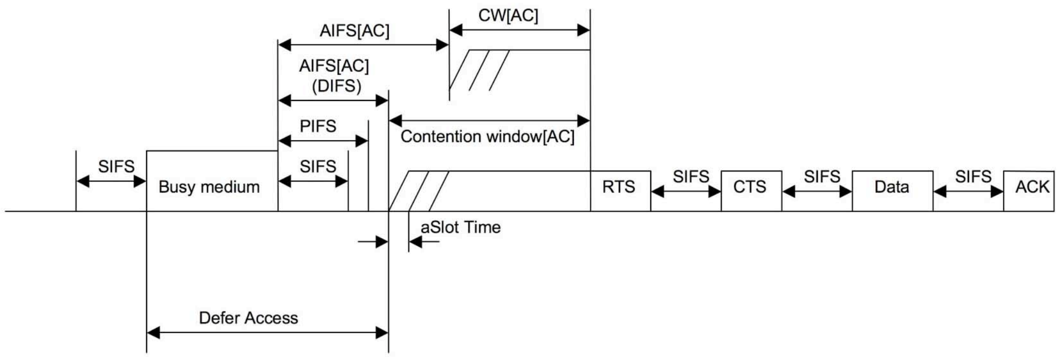

Figure 5. 802.11e transmission process and interframe time relationships

2.2.4 HCCA

The other half of the HCF is the contention-free period, HCCA. The hybrid coordinator (HC), in practice the QAP, has the highest priority to access the channel. This is accomplished by setting the stations' AIFS so that it is at least DIFS, which is longer than PIFS. At the same time QAP uses PIFS without backoff to access the channel. This way the QAP gains control of the channel and announces the beginning and the end of the contention free period. During CFP the stations wait to be polled, except when they are sending reservation requests. These requests contain flow information like mean data rate, mean packet size and maximum tolerable delay. QAP determines the polling cycle according to the flow information and the algorithm it is using. After determining the cycle, the QAP starts to issue QoS contention-free polls (QoS CF-Polls) to QSTAs that have requested parameterized services. QAP sends the QSTA in question a TXOP limit, which is also called polled TXOP or HCCA TXOP. During a polled TXOP a QSTA can transmit multiple frames with SIFS in between, provided that the total given TXOP limit is not exceeded.

The standard provides a simple scheduler algorithm as a reference scheduler for CFP. With the information QSTAs send, the QAP determines the maximum service interval (SI) to be used for all of the QSTAs. The selected SI should satisfy the delay requirements of all the flows. The QAP also determines TXOP durations for each of the flows based on mean application data rates. This simple scheduler is, however, quite inefficient. Each time a new flow is added or terminates, the QAP needs to recalculate the SI. In addition to recalculation issue, if two or more WLAN cells are overlapping they interfere with each other. When this happens the traffic suffers from unpredictable delays and throughput degradation. In such a situation, a coordinated resource sharing between the QAPs of overlapping cells needs to take place in order to provide QoS guarantees.

2.3 EDCA parameter set

In IEEE802.11e EDCA the parameter set selected determines the actual traffic differentiation. Therefore it is crucial that the parameter set reflects the differentiation required. Modifying these parameters also provides possibilities to improve network performance. Table 2 shows the basic EDCA parameter set that is provided in the amendment [IEEE802.11e]. In the amendment, the contention window is called aCW.

Table 3. Standard EDCA parameter set.

Access Category	AIFSN	CW_{min}	CW_{max}	TXOP limit 802.11a PHY	TXOP limit 802.11b PHY
Priority 0 AC_VO	2	$(aCW_{min}+1)/4 - 1$	$(aCW_{min}+1)/2 -1$	1.504 ms	3.264 ms
Priority 1 AC_VI	2	$(aCW_{min}+1)/2 -1$	aCW_{min}	3.008 ms	6.016 ms
Priority 2 AC_BE	3	aCW_{min}	aCW_{max}	0	0
Priority 3 AC_BK	7	aCW_{min}	aCW_{max}	0	0

2.3.1 Contention Window

802.11e EDCA contention window is similar to the distributed coordination function (DCF) contention window, except in the backoff countdown rules. In 802.11e EDCA, the first backoff countdown occurs at the end of the AIFS, not DIFS. Also, each AC has a different size CW to create further differentiation. The CW sizes relative to each other are important in determining the relative channel access frequency of an individual AC.

For the higher two ACs, voice and video, the CW_{max}-CW_{min} difference shouldn't be very large, otherwise the delay this traffic experiences will be too big. In heavy congestion it can be better to just drop the packet than wait indefinitely for a transmission opportunity. This is especially true with delay sensitive traffic. When the CW is small there are more opportunities for transmission and smaller delay. However, a small CW causes a bigger collision probability. On the other hand, if CW_{min} is increased, the overall throughput in the network decreases.

As the number of high priority traffic streams increases, the differentiation effect of the CW becomes smaller. This is because there are more collisions among the high priority flows. Also, the smaller the CW size, the more significant the impact of the AIFS value on differentiation. Setting a small CW_{min} is a good way to give a flow more throughput, but this will starve other flows, especially low priority ones.

2.3.2 AIFSN

AIFS is a new interframe space time that varies in length depending on the AC. Each AC has its own AIFS. It is the minimum time interval for the medium to remain idle before starting a backoff. AIFS helps to differentiate between different priority streams.

The arbitrary interframe space number (AIFSN) is used to calculate AIFS. It specifies how many times a slot time should be multiplied by. The formula for AIFS is as follows [IEE802.11e]:

$$AIFS[AC] = SIFS + AIFSN[AC] * aSlotTime, \quad AIFSN[AC] \geq 2$$

Here *aSlotTime* means the duration of a slot. It is a MAC variable, which is set to a predefined value. The smaller the AIFSN, the smaller the AIFS and higher the medium access priority [IEEE802.11e].

Increasing AIFS decreases the overall system throughput because stations must wait longer to access the medium. This effect is stronger when network load increases, because AIFS occurs after every transmission. Thus large AIFS can have a dramatic negative effect on the network under heavy load. AIFS should be kept as small as possible and focus on relative AIFS difference between queues to create differentiation. However, if difference is large, low priority might not be able to access medium at all.

2.3.3 TXOP limit

The fourth parameter the QAP sets is the TXOP limit. There are two kinds of TXOP limits. The TXOP limit used during EDCA is called an EDCA TXOP limit. EDCA TXOP limit is sent in the beacon frame and it has a same value for one access category across the QBSS. The TXOP limit used during HCCA is called the HCCA

TXOP limit. HCCA TXOP is unique for a QSTA and it is based on the QSTAs requirements. This work focuses on EDCA TXOP limit.

In EDCA, for each transmission opportunity the AC wins, it may initiate multiple frame-exchange sequences. These sequences are separated by SIFS. The total duration of frame-exchange sequences must not exceed the TXOP limit. The duration of the frame-exchange sequence can of course be shorter than the maximum allowed. In such a case, the QSTA releases the media and normal contention resumes. The value of TXOP limit is a multiple of 32μs up to the maximum of 8160μs [IEEE802.11e].

If TXOP limit is zero, QSTA can transmit a single MSDU, irrespective of its length or PHY sending rate. In many research papers zero is the value used because it is part of the standard EDCA parameter set. Some of the papers point out however, that zero should not be used at all. This is because QSTAs can perform link adaptation leading to a lower PHY, when they determine degradation in the connection. If such a QSTA then has a TXOP limit value of zero, it will send the one packet allowed considerably slower than before. This degrades the network performance. Also using zero leads to lower class starvation as the network load increases because of the minimal amount of packet it can send [del Prado Pavon2004].

The overall system throughput increases as TXOP is increased because overhead is reduced. However, if TXOP limit is too large for one category, other traffic categories experiences delays. This way TXOP limit has direct effect on network fairness.

2.4 Summary

WLANs have unique challenges compared to wired networks. Issues such as security, range vs. speed and reliability need to be considered when developing WLANs. Reliablity issues are especially challenging where quality of service is concerned. IEEE 802.11 standard family has developed over the years to address these unique issues and a multitude of amendments to the standard have been approved.

To improve QoS features of IEEE 802.11 WLANs, a MAC layer amendment 802.11e was approved in 2005. IEE802.11e bases QoS provisioning on a hybrid coordination function that is divided into contention and contention-free periods. The focus of this thesis is the contention period called the enhanced distributed channel access (EDCA).

EDCA is based on a new queue mechanism, where each station has four queues instead of one. Traffic is divided into these queues based on traffic's requirements and each queue has a different set of channel access parameters. There are four channel access parameters that control the frequency and duration of channel access. The parameter for channel access duration is called the transmission opportunity (TXOP) limit. TXOP limit is a multiple of 32μs up to the maximum of 8160μs. With TXOP limit value zero however, a station may transmit one packet irrespective of its length of physical sending rate. With larger TXOP limit values, the system throughput increases but can cause delay to other traffic.

3 Fairness

Without a need to favor one kind of traffic, fairness in computer networks is generally a good thing. Ideally, everyone would get the service they want without disturbing others. However, the real world is not ideal and network congestion does occur, particularly in wireless local area networks where network capacity has a strict upper limit. When congestion occurs, different traffic streams might not get what they want, in terms of throughput or delay. This is when a decision needs to be made whether to prefer some traffic over another. This chapter first discusses different ways to look at fairness. Secondly, some key fairness schemes are introduced.

3.1 Different Kinds of Fairness

Fairness is a broad concept and its roots are in philosophy and social sciences. Each individual has a sense of what is fair generally but the outcome of a person's though on what is fair might be different depending on circumstances and preferences. Similarly fairness in computer networks is seen generally as a good thing but what is perceived as fair in congestion situation varies. It is also important to distinguish what kind of fairness is looked at and how it is measured. For instance fairness can be considered between flows, between same protocols or between two different protocols. Fairness can also be looked at between sessions, users or other entities.

Fairness can be absolute or relative. Absolute fairness means that each user gets the exact same amount of time, throughput or any other desired measure of resources. However, this is often not a very useful measure, since different traffic types have different requirements. Relative fairness is a better way of measuring fairness. Relative fairness takes into account how much of your individual requirements are being fulfilled. The overall relative fairness can be calculated by comparing how much of individual requirements are being fulfilled.

Fairness that uses time as a measurement unit is called temporal fairness. However, even in a network where each user gets to send the same amount of time, the transmissions can use different rates. This is a very typical situation in 802.11 WLANs since stations are allowed to decrease their transmission rate if channel conditions worsen. Hence the amount of bits a station is able to send can be different. Instead of temporal fairness, the focus could be on cost fairness, throughput fairness, access probability fairness, delay fairness, packet delivery fairness or any other metric.

3.2 Fairness Schemes

Some notion of fairness is incorporated in many network mechanisms used today. They mostly consider fairness between flows but recently cost fairness has also been proposed. This section presents some well-known fairness schemes.

3.2.1 TCP

A familiar example of incorporated fairness scheme is TCP. It utilizes congestion avoidance mechanism to avoid congestion collapse in the network. The congestion avoidance mechanism was first introduces by Jacobson et al. [Jacobson1988]. It tries to create fairness between flows with the assumption that it is fair if flow rates through a bottleneck ling converged on equality. However, it cannot take into account history or the flows as a whole. This means that it can be cheated by starting new flows or splitting flows. TCP aims for absolute fairness since there is no differentiation between flows.

In addition to TCP, an algorithm can be TCP-friendly [RFC3448]. TCP-friendliness is based on the fairness notion that TCP-friendly flows should get the same rate as TCP compatible flows. TCP-friendly flows converge at the same rate as TCP flows and they need to have the same dynamics as well. TCP-friendly flows face the same problems as TCP.

3.2.2 Utility Based Fairness

Utility based fairness criteria defines a utility function that describes the utility a flow gets from the network with a certain capacity share. It aims to maximize the total utility of all users. Max-min fairness is a special case of utility fairness. Other special cases include maximizing the overall throughput, proportional fairness and minimizing the potential delay [LeBoudec2005].

3.2.3 Max-min Fairness

A famous fairness scheme in networking is max-min fairness. It proposes that fair service means that the service of the entity receiving the worst service is maximized. In practice this means that small flows receive all they demand while large flows have to share the remainder of the capacity equally. Starting from the smallest flow, the bandwidth is distributed so that all flows receive what they need until bandwidth is exhausted. In the case the flows that are not receiving all they require, they have to divide the capacity [LeBoudec2005]. Max-min fairness guides the user to appreciate a very low bit rate, which is unnatural. If a user wants to cheat max-min fairness algorithm, the flows are split into small flows so that everyone else's allocation is reduced.

3.2.3 Proportional Fairness

Proportional fairness tries to maintain a balance between maximizing the network throughput and allowing users to have at least a minimal level of service. Each flow is given a data rate or a scheduling priority which is inversely proportional to anticipated resource consumption. This criterion also favors small flows, but not as much as max-min fairness [LeBoudec2005].

A case of proportionally fair scheduling is weighted fair queuing (WFQ) that was introduced by Demers et al. [Demers1989]. It aims to ensure that a router's capacity is fully utilized. Low volume traffic is scheduled first and high volume traffic shares the remaining bandwidth according to weights assigned.

3.2.4 Jain's Fairness Index

A well-known index of fairness was proposed by Jain et al. [Jain1984]. It is a very general definition and suitable for many situations. If the amount of contending users is n and i^{th} user receives an allocation x_i then Jain's fairness index $f(x)$ is

$$f(x) = \frac{\left[\sum_{i=1}^{n} x_i\right]^2}{n\sum_{i=1}^{n} x_i^2}$$

The result is the measure of equality of the allocation of values. The index gets values between 0 and 1. When all the users receive an equal share i.e. the system is completely fair, the index gets the value 1. As fairness decreases the index value decreases until it reaches 0. This index is dimensionless, independent of scale and continuous with respect to the allocation variable x_i. It can be used on any number of users. Additionally because of continuity, even slight changes in the allocation of values change the value of the index [Jain1984].

Section 3.1 explained the concept of relative fairness. When using Jain's fairness index, relative fairness can be calculated by

$$x_i = \begin{cases} \dfrac{a_i}{d_i} & if\ a_i < d_i \\ 1 & Otherwise, \end{cases}$$

where d_i is the total demand of user i and a_i is the amount it is actually given [Jain1984]. In later calculation in this thesis both a and d are throughputs.

This thesis uses Jain's fairness index in estimating fairness because of its generality. It would not be sensible to use for instance max-min fairness because this thesis is not looking into just maximizing the throughput of small flows.

3.2.5 Cost Based Fairness

The above mentioned schemes are mostly focused on flow rate fairness. Briscoe [Briscoe2007] criticizes this view and says that it is myopic. He claims that since schemes based on flow rate do not take into account how many flows users create or how long flows last it would be better to focus on cost fairness. By cost fairness he means sharing out the cost of one user's actions on others. He says that in order to arbitrate cost fairness only the volume of congestion is needed. This is calculated by multiplying the congestion with bit rate of each user causing it. In his paper he goes further into details of how a cost fairness scheme could be achieved while vigorously criticizing flow rate based fairness schemes.

3.3 Summary

Fairness is a complex concept but it has an integral part in computer network design. The question is what is a fair way to allocate scarce resources and how are you going to measure it? Some schemes such as max-min fairness prioritize based on flow size, while other let weights be assigned. An interesting new proposal is cost fairness, which takes a step to another direction. An important matter to consider in a fairness scheme is its complexity. Complex algorithms take a lot of processing time. This decreases link capacity since time is spent in choosing the next packet. With a decreased link capacity low priority traffic is more likely to suffer starvation.

In evaluation of fairness this thesis uses Jain's fairness index. It is a simple and general definition to see how far a set of shares is from equality. Additionally in this thesis fairness is calculated in a relative sense. This means that the calculations take into account the throughput need of each flow and not just pure equality. With Jain's fairness index this is also easy to calculate.

4. Related Work

This chapter shortly describes earlier work on IEEE802.11e enhanced distributed channel access period. First it introduces research that has been done to prior to EDCA. Next it presents research done to evaluate EDCA as it is presented in the standard. Then it discusses work done to improve EDCA and finally introduces work specifically focusing on EDCA fairness.

4.1 Pre-EDCA Research

Prior to EDCA several papers were published about bringing better QoS features to 802.11 WLANs. Although 802.11 MAC already has QoS features they were deemed not sufficient. Ni et al. [Ni2002] list the QoS limitations of 802.11. They say that the DCF period of 802.11 can only support best-effort services and no guarantees to high priority flows can be made. All flows have to share just one queue and thus they all experience the same delay. They say that only by using admission control quality of service in the DFC period can be improved. PCF on the other hand was designed to support time-bounded multimedia applications. However there are three main problems associated with it. The first is that the central polling scheme forces all communications to go through the access point, which wastes channel bandwidth. Secondly, the operations between DCF and PCF modes can lead to unpredictable beacon delays. Thirdly, the transmission time of a polled station is difficult to control. This section only presents research related to DCF since it is the predecessor or EDCA.

Banchs et al. [Banchs2002] propose a distributed weighted fair queuing (DWFQ) algorithm to be used in improving DCF. The DWFQ mechanism gives a flow an average bandwidth proportional to its weight by dynamically changing the contention window. Their simulations show that the scheme is able to provide the desired bandwidth distribution regardless of the aggressiveness of the flows or their willingness to transmit. However, using a contention window means that there is always certain randomness, which leads to variability in throughput and delay.

Vaidya et al. [Vaidya2000] propose a distributed fair scheduling algorithm. In their scheme packets with smallest ratio between its packet length and weight are transmitted first. The weight is higher for a higher throughput class. With the combination of packet length and throughput need, differentiation of service can be achieved with backoff calculations. In this scheme though, mapping QoS requirements to a weight is complicated.

Campbell et al. [Campbell2001] propose a virtual MAC (VMAC) algorithm for distributed service differentiation. VMAC monitors the radio channel and estimates service levels that can be achieved locally. VMAC does not handle real packet transmissions. The goal of VMAC is to estimate QoS parameters in the radio channel accurately. The scheme then uses different contention window values for delay differentiation of different kinds of traffic. The drawback of this algorithm is the processing capacity needed in each device.

Sobrinho et al. [Sobrinho1996] propose a Blackburst scheme to minimize the delay of real-time traffic. In this scheme low priority stations use CSMA/CA for channel access while the high priority stations use the Blackburst scheme. High priority stations send bursts called black bursts to jam the channel if the medium is busy. The length of the black burst is determined by the time the station has waited to access the medium. After transmitting the burst, the station listens to the channel to find out if someone else is sending a black burst. If so, that other station has waited longer and should access the channel first. Once a station does get to transmit, it schedules its next frame transmission. This way real-time flows synchronize and share the medium in time division multiplexing (TDMA) fashion. This means that unless low priority flows disturb the situation, very few blackburst periods need to occur. The main drawback of the scheme is that high priority traffic needs to arrive at constant intervals or else the performance degrades considerably.

In addition to work suggesting improvement to 802.11 QoS features, research on fairness and 802.11 has been conducted. Pong et al. [Pong2004] have investigated the trade-off between fairness and capacity in the 802.11, especially in the presence of channel errors. They compare throughput fairness and temporal fairness and come to the conclusion that in error situations, when link adaptation takes place and stations transmit with different rates, maintaining temporal fairness leads to higher capacity. They also suggest that admission control should be used to maintain fairness. Also, if possible, stations should transmit only at high PHY rates during congestion so that the network has high efficiency.

Jiang et al. [Jiang2005] investigate proportional fairness in WLANs and ad-hoc networks. By proportional fairness they mean finding a balance between fairness and throughput. They point out that in multi-rate environments, throughput fairness can lead to degrading network performance. They do not consider 802.11 specifically but WLANs in general. They come to the conclusion that in multi-rate WLANs, fairness deriving from time allocations rather than throughput is more natural and would lead to better network performance.

4.2 Evaluations of EDCA

There are quite a few evaluations made of EDCA, mainly with simulations. Practical testing has been somewhat limited. Simulations by Qiang et al. [Qiang] show that EDCA supports better QoS than DCF or PCF when load conditions are low or medium. In their simulations they increased the number of stations from 2 to 50 and notice that the total goodput increases between 2 to 15 stations, but after 15 stations it decreases rapidly. They also notice that the average delay increases as the number of stations increase. Another observation they make is that EDCA-based ad-hoc networks saturate very fast. In addition they mention that finding optimal EDCA parameters is difficult as they are static and not adjusted to the network conditions. Finally they remark that strict service guarantees can only be made when admission control is used together with EDCA to stop the network from becoming too congested.

Similar results are reported by Choi et al. [Choi2003]. They say that EDCA works better than the legacy 802.11 in providing differentiated channel access to different priority traffic. The researchers did not optimize the network by tuning of EDCA parameters, which they say would be important to research. They also mention that an admission control scheme would be needed for the QoS provisioning to work acceptably.

Del Prado Pavon et al. [Del Prado Pavon2004] evaluate the effect of frame size, number of stations and mobility on EDCA. Generally they found that EDCA offers 5-20% throughput efficiency improvement over the legacy DCF. They note that small frame sizes mean that overhead consumes a significant amount of the channel capacity. Also as the number of transmitting stations increase so does the collision probability. At that point lower priority traffic starts to experience significant packet loss. The authors also notice that a bad link penalizes all other links as well. They say that it is important to use TXOP values other than zero. If TXOP is zero the station is allowed to transmit one packet irrespective of its length or the physical transmission rate. However, it is possible for a station to independently reduce their physical transmission rate, if it for example moves further away from the AP. If they then are allowed to send one packet regardless of the time it takes to send it, all other traffic has to wait longer than normally.

In their study, Xi et al. [Xi2005] also investigate the 802.11e effectiveness. Their focus is on different traffic types. They agree that 802.11e is an improvement over the legacy 802.11 but say that the improvement comes at the cost of decreased quality for the lower priority traffic. The higher priority is able to acquire the channel very effectively, which makes the lower priority traffic suffer up to a point of starvation. They also found that 802.11e has a much higher collision rate than the legacy system and hence suffers from increased retransmissions and packet loss. This has a negative effect on channel efficiency.

Tinnirello et al. [Tinnirello, May2005] investigate the performance of new channel utilization mechanisms in 802.11e via an analytical model. They prove that the block ACK mechanism is not useful for low date rates and low TXOP values, but it is very attractive for high data rates. Also they conclude that the optimal selection between an immediate ACK and a block ACK does not depend on the number of stations.

Banchs et al. [Banchs2005] are one of the few to report results from practical testing. They investigate EDCA mechanism's ability to support traffic engineering and service guarantees. The results show that with UDP traffic the system works well. With TCP traffic the results were also promising, and only slight deviations from the desired was noticed. Overall, EDCA worked better than DCF. Service guarantees were harder to satisfy and more work needs to be done in developing optimal EDCA configuration. They note that the inherent uncertainties of a mobile environment make creating service guarantees very difficult. They think that monitoring WLAN traffic situation in real time to help an admission control algorithm could be a solution to providing service guarantees.

4.3 Proposed Enhancements to EDCA

Early on it was clear that albeit 802.11e was a better than the legacy 802.11 there were still adjustments and improvements that could be done. This section presents some of the most interesting ones.

Ni et al. [Ni2002] propose a scheme called adaptive enhanced distribution coordination function (AEDCF). They investigate resetting CW values more slowly to adaptive values while considering CW current sizes and collision rate in the network. The factor for CW update is calculated so that flows with high collision rate have better chance to transmit the next time. CW of high priority traffic increases slower than CW of low priority traffic. This dynamic varying of the CW for each class of service achieved better throughput, delay and jitter performance in an ad-hoc 802.11e network. Even though the main focus of their research was ad-hoc networks, they say that this scheme could be extended to access point controlled networks as well. The problem with this scheme is that performance of low priority streams degrades with high network load.

Zheng et al. [Zheng2005] investigate using arbitrary interframe space number (AIFSN) to improve the performance of real-time traffic. In their proposition real-time traffic has no backoff period and has the smallest AIFSN. Hence real-time traffic gets to transmit before any other traffic and only collisions between real-time traffic are possible. To avoid these collisions, real-time queues are assigned a different AIFSN based on the time packets have been waiting. This scheme naturally decreases latencies of the high priority flows but is not fair by any means. It is be useful if real-time traffic needs to get a strong priority but otherwise it is not be the best solution.

Also their simulated with an ad-hoc network, so further testing is need to be done to investigate the behavior of this scheme in the infrastructure mode.

Kim et al. [Kim2005] propose a new MAC scheme called multi-user polling controlled channel access (MCCA), which is based on EDCA multi-user polling. It also uses two-level frame aggregation, on MAC and PHY layers. Their scheme can aggregate frames with different QoS requirements and different destinations but needs good scheduling to work properly. Therefore they have created a scheduling mechanism to do this. Through simulations they are able to show that MCCA improves system throughput quite a lot while delay remains reasonable. However this scheme needs to be tested or simulated in more realistic channel environment.

Gu et al. [Gu2003] present a measurement-based distributed admission control method in their paper. Their scheme is aimed to protect high priority flows and improve network performance in heavily loaded 802.11e networks. They propose that each station measures the existing traffic load in the network and has an admission controller, which decides if more packets can have the right to access the medium. Each station measures either relative occupied bandwidth or average collision ratio. Measurement-based admission control can be a viable solution, but any strong conclusions cannot be drawn based to on this paper.

Naoum-Sawaya et al. [Naoum-Sawaya2005] propose a scheme to adapt the CW according to the channel congestion level. CW is set directly to a value close to a required one for transmission thus eliminating the time spent on try, fail and wait. In They demonstrate the effectiveness of their scheme compared to the standard model especially in high congestion situations.

Approaches to improve EDCA are quite numerous, ranging from adaptive CW to frame aggregation. However, there is still a lot of work to be done to optimize the tradeoff between channel efficiency, priority and fairness. Adapting EDCA parameters according to the traffic load sounds easy but is in fact a very difficult problem. Researchers have pointed out that adapting all four parameters dynamically at the same might improve network performance. However, such a complex dynamic scenario is very difficult to mathematically model or simulate or to test in any other way. Nevertheless, it seems that using dynamic parameters to some extent as in many of the above-mentioned research, gives better performance than the static model provided in the standard.

4.4 Research on Fairness in IEEE802.11e WLANs

To address the fairness issue in AEDCF described in Section 4.2, Malli et al. [Malli2004] propose an adaptive fair EDCF scheme (AFEDCF). This scheme aims to decrease collision rate and idle time. In it, CW increases not only when there is a collision but also when the channel is sensed busy during deferring periods. The backoff timer can decrease linearly or exponentially. Backoff threshold is the boundary between these two. When a collision occurs or the station is deferring, it doubles the CW, randomly chooses a new backoff time and reduces the backoff threshold. After a successful transmission, the station resets the CW to minimum, chooses a backoff time randomly and increases the backoff threshold. The adaptive CW of the AEDCF scheme is not used. AFEDCF achieves higher absolute throughput fairness than AEDCF. The fairness in high traffic loads is due to the fact that contention windows of each queue are at their maximum value and they will transmit almost at the same time with the same CW. The issue here in contrast to the AEDCF is that high priority traffic can suffer and sometimes it can even have a bigger CW than low priority traffic.

Leith et al. [Leith2005] are interested in TCP fairness in 802.11e networks. TCP dominates current network traffic. However, because of cross-layer interaction between 802.11 MAC and TCP flow/congestion control used, TCP and 802.11 WLANs do not work optimally together. The result is gross unfairness between individual flows. They identify two issues to be solved in order to improve fairness. First, the asymmetry between TCP data and TCP ACK paths disrupts TCP congestion control. Second, the network level asymmetry between TCP upload and download flows. To solve these issues they propose that the MAC should be configured so that TCP ACKs have unrestricted access to the wireless medium. This way the volume of TCP ACKs is matched to TCP data packets, which restores path symmetry at transport layer. To restore fairness in the network level they suggest prioritizing the downlink data packets at the AP so that downlink traffic gets an equal share of the wireless channel. This approach would no doubt increase the fairness of TCP traffic. This is an interesting topic to be investigated further especially with voice and other real-time traffic. This way the effect MAC modifications has on real-time traffic can be investigated.

Tinnirello et al. [Tinnirello, June 2005] investigate temporal fairness provisioning in multi-rate contention-based 802.11e WLANs. They show that equalization of the channel access times allows stations to obtain throughputs proportional to their transmission rates but independent of frame lengths. They focus especially on the effect of fragmentation on the system. Through their model they prove that the system throughput is optimized if MPDUs exceeding the TXOP limit are divided into equal-sized fragments. Also depending on the network condition, it can be more efficient to release the channel before the TXOP limit expiration without activating fragmentation to fully exploit the TXOP limit. Additionally, they show that there exists some tradeoff between fairness and system efficiency since equalization of channel holding time generally requires fragmentation, which introduces a lot of overhead. This is a very interesting research but it does not consider different traffic types.

Inan et al. [Inan2007] focus on the uplink-downlink fairness issue in 802.11e EDCA when default EDCA parameter set is used. They note that the QAP is a huge bottleneck in the system because according to the standard, it is mostly treated like any other node. According to them, this effect is even more catastrophic in the case of TCP flows. They propose a model-assisted measurement-based dynamic EDCA parameter adaptation algorithm. They claim that their algorithm achieves a predetermined utilization ratio between uplink and downlink flows of the same AC while maintaining the AC prioritization. The algorithm also differentiates the adaptation depending on whether the traffic is TCP or UDP. They also propose that QAPs should use any value of the CW, instead of using exponents of two. QSTAs on the other hand should still use values that are exponents of two. This way the QAP can satisfy any required utilization ratio. Additionally, their results show that using constant ECDA parameters does not result in high fairness.

4.5 Summary

There has been a considerable amount of research on quality of service and 802.11 and 802.11e. Early on 802.11 standard's deficiencies in this area were identified and attempts to improve its QoS capabilities were proposed. Once 802.11e started to take form, research focus turned into comparing it to 802.11 and suggesting further improvements. Of the EDCA parameter set, especially contention window modifications have been a popular research topic. The effect of modifying the TXOP limit on the other hand has not been researched very much. There has also been research with specific focus on fairness but fairness together with TXOP limit has not been investigated thoroughly.

5. Simulations

Simulations are a popular way to investigate computer networks. There are several different simulators available but these often only include current features of networks. Thus any new ideas require tweaking of existing simulators or even programming a new simulator. Simulations also require a lot of details and assumptions about the network that is simulated. Even so, they are never as realistic as monitoring a real network. However, when keeping in mind pitfalls and shortcomings, simulations do have their use. This chapter first discusses general principles of simulations, then the simulation setup and finally the tools used.

5.1 Simulation Goals

The purpose of the simulations is to investigate the effect of transmission opportunity limit on fairness during EDCA. The simulations investigate the effect by changing the TXOP limit of the lower category traffic and compare the results to simulations done with EDCA standard parameter set. The purpose is to find out whether modifying the TXOP limit improves 802.11e EDCA fairness, but in such a way that the delay sensitive traffic is not overly disturbed. Additionally, if changing the TXOP limit has an effect, the purpose is to find out how the TXOP limit should be changed.

5.2 Introduction to Simulations

There are basically three methods of doing computer network research: mathematical modeling, simulations and real-life measurements. Each has merits and each has drawbacks. Mathematical analysis is an exact method, but because of this exactness it falls short in depicting the complex real world. Simulations do a better job in describing the real world, especially if real traffic traces are used, but they still are not close to perfect accuracy. Measuring real networks would be the most realistic way to study networks but it expensive and time consuming. Moreover exactly the same measurement conditions can be difficult to re-create for repetitions and random factors can influence the situations. This can make it hard to see underlying principles factors.

In order to create usable and valid results via simulations it is important to follow good simulation practices. Kurkowski et al. [Kurkowski2005] list four key aspects that should be taken into account when doing simulations; otherwise the credibility of the simulations can be called to question. According to the article simulations should be repeatable, unbiased, rigorous and statistically sound. More specifically, the researcher should explain the simulation type used, validate the simulation model, explain random number generation, define variables, develop scenarios, determine steady-state and provide good statistical analysis of the results, including confidence intervals. In their study of MANET papers they come to the staggering conclusion that none of the papers with simulation studies provided enough information for the reader to determine if they fill all these criteria.

5.3 Simulation Setup

This section presents the specifics of the simulations used in this thesis. First the simulation topology is explained followed by the traffic models used. Lastly this section presents scenarios used in the simulations.

5.3.1 Simulation Topology

Figure 7 shows the simulation topology. QSTAs are evenly distributed in a circle around the QAP. During the simulations the QSTAs do not move and can always hear each other. The radio links are free of errors. The physical layer the simulations use is IEEE802.11b.

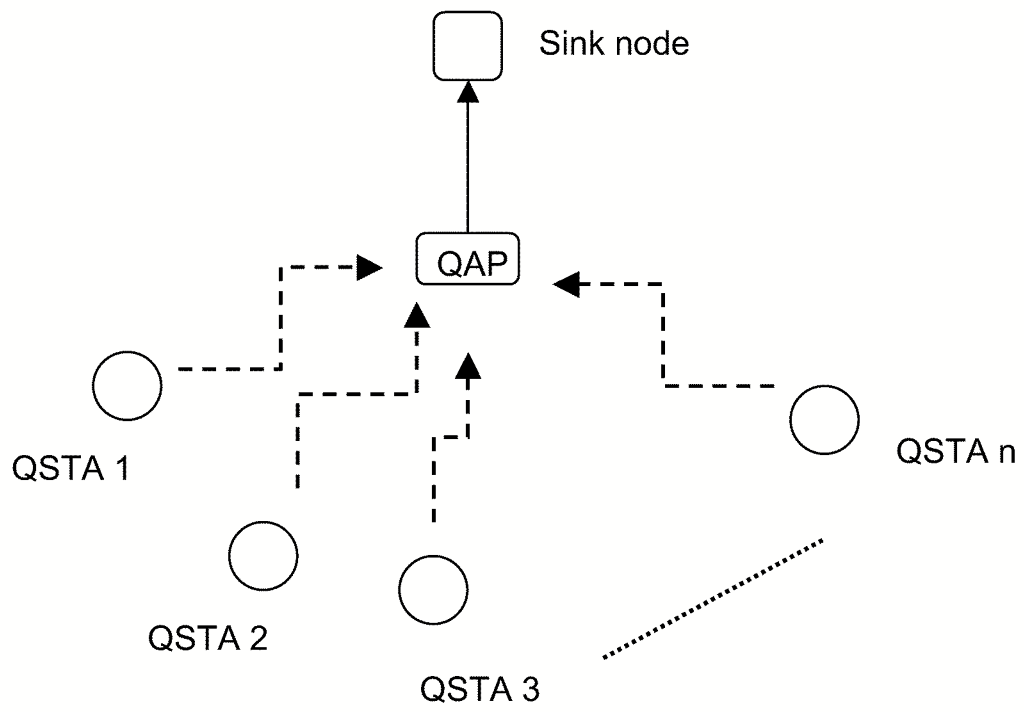

Figure 6. A simulation topology sketch.

All simulations have one QAP and n QSTAs. The amount of QSTAs can be changed in the simulation. Each node transmits two kinds of traffic, voice and data, and the transmissions are one-way. The transmission is sent to the sink node, which is an ordinary wired network node. The wired node and the AP are connected via a fast connection of 500 Mbits/s to minimize its effect to the results. There is no interference, channel errors or noise.

5.3.2 Simulation Traffic

In order to simplify the situation in the network, there are only two kinds of traffic, voice and data.

In an attempt to be as realistic as possible, network simulations often use heavy-tailed distributions to model Internet data traffic. Heavy-tailed distributions are distributions whose tails are not exponentially bounded. These distributions produce mostly small values with occasional large values. However, these random large values can be very large. Research such as by Mah et al. [Mah1997] and Park et al. [Park1996] points out that HTTP traffic file size distribution is heavy-tailed.

Most often used heavy-tailed distributions come from the Pareto family. Weigle [Weigle2006] investigates the use of heavy-tailed distributions in networks simulations. She uses Pareto distribution as example and comes to the conclusion that using Pareto distributed traffic creates high variability. Single large values the distribution generates greatly affect the network load. For this reason it is very hard for a Pareto traffic simulation to reach steady state in a reasonable amount of time. To achieve two-digit accuracy for the mean over 10^{10} samples were needed. Even then the mean is unstable. She comes to the conclusion that heavy-tailed distributions, although theoretically great for traffic modeling, do not create meaningful simulation results. The author presents three possible solutions for dealing with this issue. The first is to use bounded Pareto, the second is to approximate Pareto with Lognormal distribution and the third is to treat the result as transient.

To make it possible for simulations to reach steady state they use bounded Pareto in data traffic modeling. In the modified code bounded Pareto distribution generates values normally, but all values larger than a threshold value are discarded and the distribution is asked to generate a new value. Packet sizes are drawn from the Pareto distribution with average packet size of 140 bytes, shape parameter 1.2 and maximum size of 200 bytes. Data traffic rate is 10 kbits/s and it is UDP traffic. Data traffic is generated during random bursts that are also drawn from the bounded Pareto distribution with a shape parameter of 1.2. Burst average time is 1.35 seconds and average idle time is 1.5 seconds. Maximum burst time is 5 seconds and minimum is zero. Since packet size is not constant only an estimate of packets per second is possible. With the average packet size of 140 bytes packet throughput rate during bursts is 8,9 packets per second.

To model voice traffic the simulations use ITU-T G.729 standard. G.729 is supported widely in VoIP products. In G.729, the voice is encoded at the rate of 8 kbps and with 20 or 40 bytes payload size in a packet. The voice quality can be degraded compared to another widely used standard, G.711, because the compression in G.729 can be lossy. However G.729 requires less bandwidth. The payload size is 20 bytes. With packet overhead, the rate required is 26.4 kbits/s [Cisco]. In the simulations voice traffic is generated in bursts with the average duration of 1.35 seconds and average idle time duration of 1.5 seconds. The burst length is drawn from the exponential distribution and it also has a maximum duration of 5 seconds and minimum of 0 seconds. During bursts packet throughput rate is 165 packets per second.

5.3.3 Simulation Scenarios

There are three simulation scenarios. The first is the reference scenario that uses the standard EDCA parameter set. Other two modify the TXOP limit. In each case only the TXOP limit of data traffic is modified. The voice traffic TXOP limit remains at the standard static value in all simulations. Voice traffic uses the AC_VO access category standard values. Data traffic uses the AC_BK access category standard values for other parameters than the TXOP limit.

The seconds simulation scenario uses an infinite TXOP limit. Each station sets the data traffic TXOP limit independently based on the data queue's length. Each time a queue in a QSTA needs a TXOP limit value, it calculates how long it would take to send all the packets currently in the queue and uses this as TXOP limit. If that queue then wins access to the channel it can send all its packets. This represents the maximum TXOP limit value that is useful, because any values larger than this do not create extra benefit for the queue. This way the effect of maximizing the data traffic TXOP limit has to voice traffic can be observed.

The third scenario investigates the effect of different size static TXOP values for data traffic. According to the standard the value should be zero. In this scenario however, the data traffic TXOP limit is set to a non-zero value that remains the same through one simulation set. By changing the data traffic TXOP limit from one simulation set to another, a range of values is covered.

5.4 Simulation Tools

This section describes the tools used in the simulations. First the simulator used is explained followed by details about execution. Then statistical analysis discussed and finally simulation metrics that are used in interpreting the results are explained.

5.4.1 Simulator

In the simulations Network Simulator 2 (NS-2) is used. NS-2 is a discrete event simulator written in C++ for many different kinds of computer network simulations [NS-2]. In NS-2, the actual simulations are written in Tool Command Language (Tcl). The basic NS-2 itself has built in IEEE802.11 WLAN capabilities but it does not have 802.11e features. However, there are extensions to NS-2 to make it possible to simulate 802.11e networks as well. The simulations use the module developed by Wiethölter et al. [Wiethölter] of the Telecommunication Networks Group of Technical University of Berlin. Their extension module to NS-2 focuses on EDCA and it does not have HCCA functionalities. Because their module only runs with NS version 2.28 that version of NS-2 is used.

Wiethölter's extension module simulates in accordance of the IEEE802.11e standard and uses by default the EDCA parameter set provided in the standard, which is provided in Table 4. Reference simulations in this thesis use these standard values. The C++ source code related to the TXOP limit is changed so that it is possible to modify the TXOP limit. With the modified code it is also possible to change the TXOP limit dynamically during a simulation, which was not possible before. Also, the changes in the TXOP limit are local to each QSTA. The values for other parameters like the AIFS and the CW size remain the same as in the amendment and they are controlled by the QAP. CW and AIFSN values or any other settings of NS-2 are not changed at all in the simulations. This way the effect of the TXOP limit can be isolated.

Table 4. Standard EDCA parameter set values

Access Category	AIFSN	CW_{min}	CW_{max}	TXOP limit 802.11b PHY
AC_VO	2	7	15	3.264 ms
AC_VI	2	15	31	6.016 ms
AC_BE	3	31	1023	0
AC_BK	7	31	1023	0

5.4.2 Simulation execution

In order to collect a sufficient amount of samples each simulation is run 20 times with exactly the same parameters. The duration of one simulation is 1000 seconds. During the simulation NS-2 creates a trace file, which then can be processed to abstract information about the simulation.

In order to determine when the simulation has reached a steady state, a script calculates the mean of a metric, such as delay or fairness, every second. Then with the help of the script, data is removed from the beginning of the file as much as is appropriate for the simulation to have reached steady state.

For random number generation the instructions in a paper by Weigle [Weigle2006] are followed. She notes the importance of using random generator sub streams correctly.

5.4.3 Statistical Analysis

The built in statistical features of Matlab are used to perform statistical analysis. Each one steady-state simulation produces only one value for throughput, delay, packet delivery ratio or fairness. This value is the mean value for that one simulation run. After 20 runs there are 20 individual results from which the final mean value is calculated. From these values the standard deviations and the confidence intervals are calculated.

5.4.4 Simulation Metrics

To understand the overall behavior of a network it is helpful to look at several different metrics. In addition, for simulations where the TXOP limit changes, the program creates an output file detailing the change of the TXOP limit.

Delay, or latency, can be classified depending on what is causing it. Propagation delay is the time it takes for a signal to travel in the wire or in air, and it can only be improved by improving the wires for example from copper to fiber. The second type of delay is processing delay, which is the delay caused by packetization, compression and routing. The third is queuing delay that is caused by congestion. In addition there is a serialization delay, which is the amount of time it takes to place a bit onto an interface. This delay can be reduced by faster link speed and smaller packet sizes. [Davidson2006] One notable issue with delay measurements in the simulations is that it can only be measured of packets that are actually delivered. Packets dropped or lost have an infinite delay and they are not taken into account in the calculations.

Delay can be one way or two-way depending on what we want to calculate. One-way delay means the time it takes for a packet to reach its destination, two-way is the roundtrip time. In these simulations only one-way delay is measured.

Packet delivery ratio simply means packets that are delivered and not dropped at some point. It is a ratio of packets received divided by packets sent. Throughput is measured in bits/second and is a capacity measure for the network. Fairness in this case means relative throughput fairness as discussed in Chapter 3.

In these simulation scenarios the lower limit for fairness is 0.5 instead of 0, because the actual fairness calculation only uses two figures, one from data traffic and one from voice traffic. The fairness $f(a)$ is calculated as follows:

$$f(a) = \frac{\left[\sum_{i=1}^{2} a_i\right]^2}{2\sum_{i=1}^{2} a_i^2} \qquad \textbf{(1)}$$

Where a_i is

$$a_i = \frac{\sum ReceivedBytes_i}{\sum ReceivedBytes_i + \sum DroppedBytes_i} \qquad \textbf{(2)}$$

In each calculation a_1 = voice traffic and a_2 = data traffic.

The Equation (2) is used when calculating fairness because in this situation, the interest is in fairness related to the closeness of the allocations of respective demands.

In EDCA, there are four categories of traffic and each gets to use the WLAN resources differently. The network is completely fair if all traffic categories get to send all the packet they want. On the other hand, the network can be totally unfair, if one category of traffic gets its needs fulfilled and others get nothing. It should be noted that when no traffic gets through at all the network is also completely fair.

6. Results

This section presents the simulation results. First the section presents simulation results with infinite TXOP limit. The section concludes with the results from static TXOP limit simulations.

6.1 Infinite TXOP Limit

In this section simulations with standard settings are compared to simulations where the TXOP limit changes. Standard setting simulations use the static values given in the 802.11e amendment. This means that in standard setting simulations data traffic TXOP limit is always zero. In the infinite TXOP limit changing simulations, data traffic TXOP limit is always set to a value that allows all packets in a queue to be sent. This is the maximum useful TXOP limit because queues release the channel when they have nothing to send. With this scheme, the maximum disturbing effect of data traffic has on voice traffic can be observed.

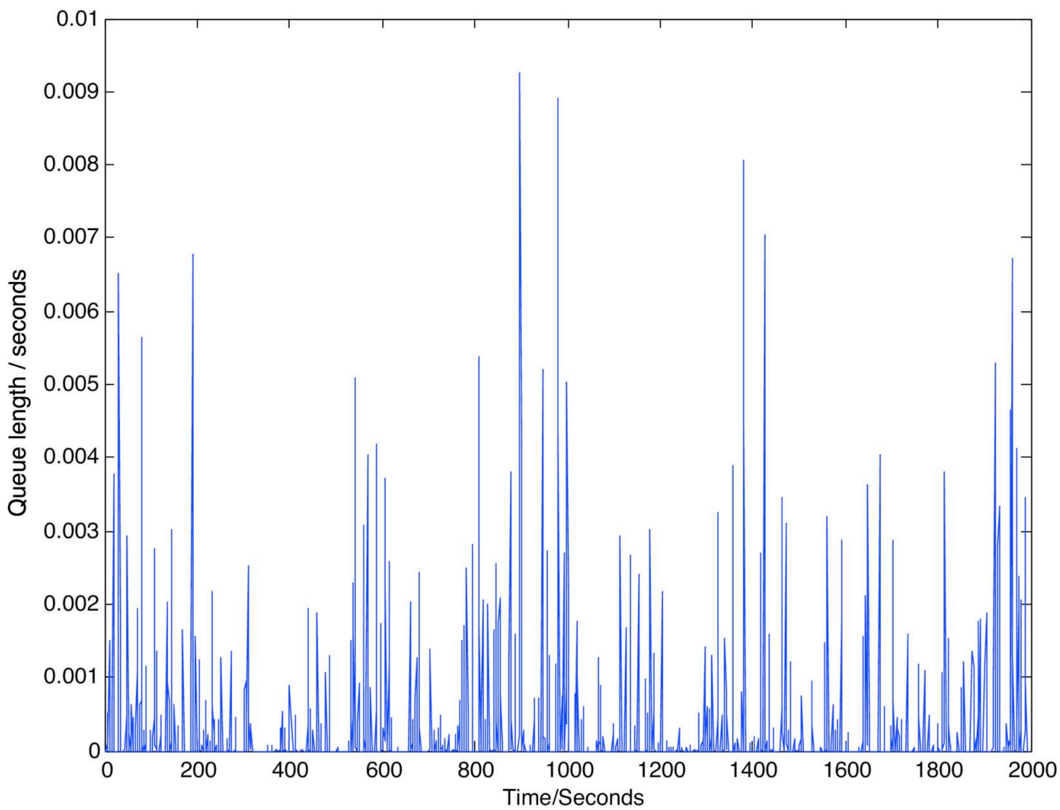

Figure 7. Data traffic queue length during one simulation.

Figure 7 shows an example of how data traffic queue length changes during one infinite TXOP limit simulation run. The figure shows that data traffic packets have to wait a random time for channel access, which can at times be quite long. When the data traffic queue gets to transmit the queue empties.

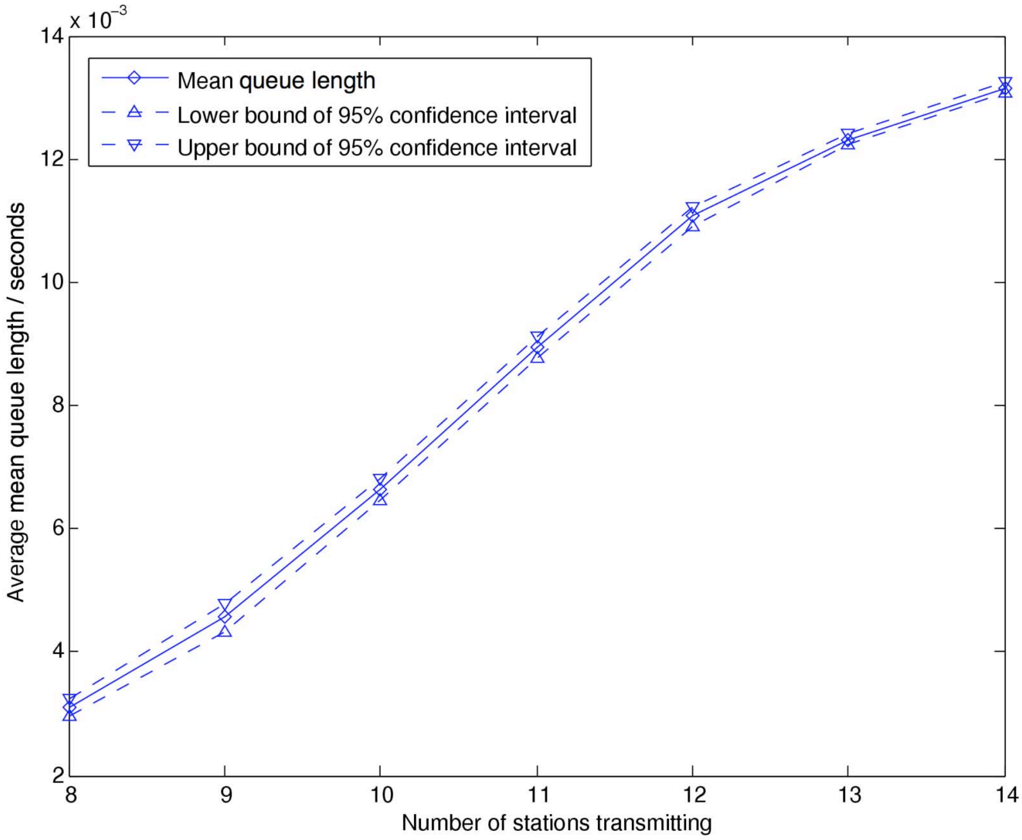

Figure 8. Average non-zero data traffic queue length when number of stations with infinite TXOP limit. The average queue length also shows the average data traffic queue length.

Figure 8 shows how data traffic TXOP limit increases as more stations are transmitting. The figure also shows the average data traffic queue length. When there are 1-7 stations transmitting the average TXOP limit for data traffic is zero. This means that all traffic is transmitted without waiting in the queue. When over 7 stations are transmitting the queue length and queuing delay increase.

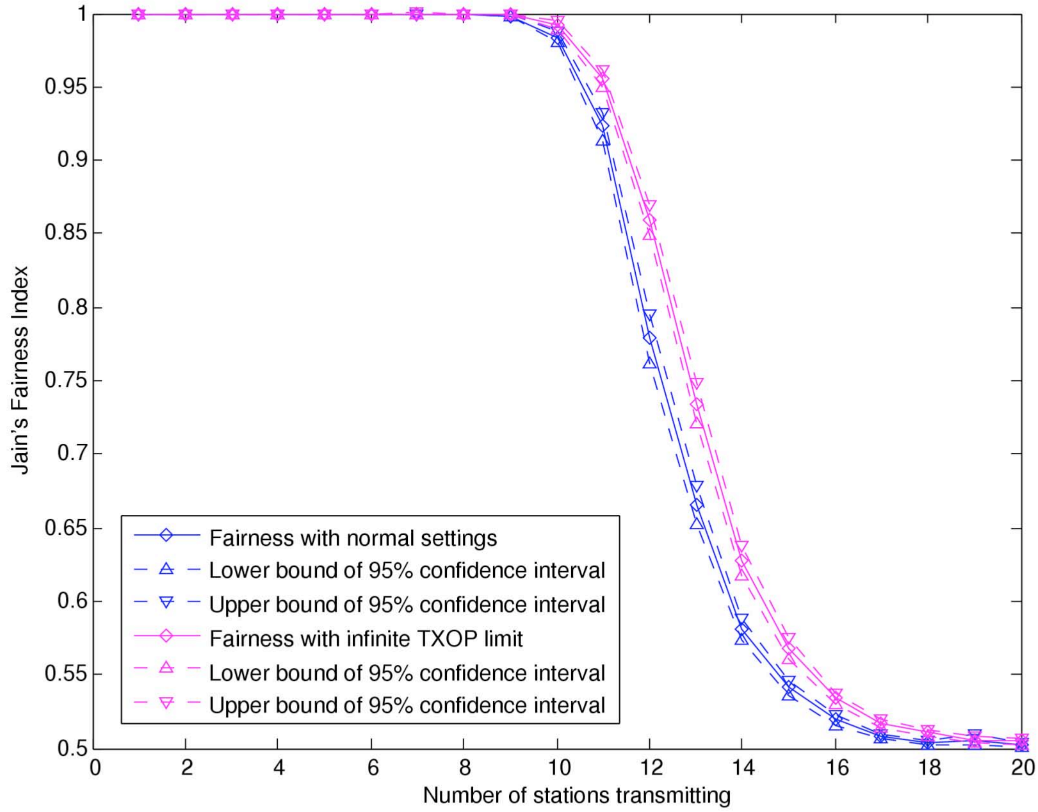

Figure 9. Fairness, standard and infinite TXOP limit settings

Figure 9 shows the network fairness with standard and infinite settings. When more than eight nodes are transmitting the network fairness starts to suffer. When 20 nodes are transmitting the system is totally unfair. Comparing the standard settings to the infinite TXOP limit settings, it seems that the latter settings produce better fairness. This applies only to the situation where the network is getting congested; when the network is already congested the infinite settings are of no further help. The maximum difference in fairness index is 8% and it occurs when 12 stations are transmitting. At that point, the fairness index in the infinite case is 0.859 and 0.7788 in the standard case. This difference means about 10% improvement. The improvement percentage is even higher as number of stations increase beyond 12, but because at the same time traffic increases the significance of the improvement lessens.

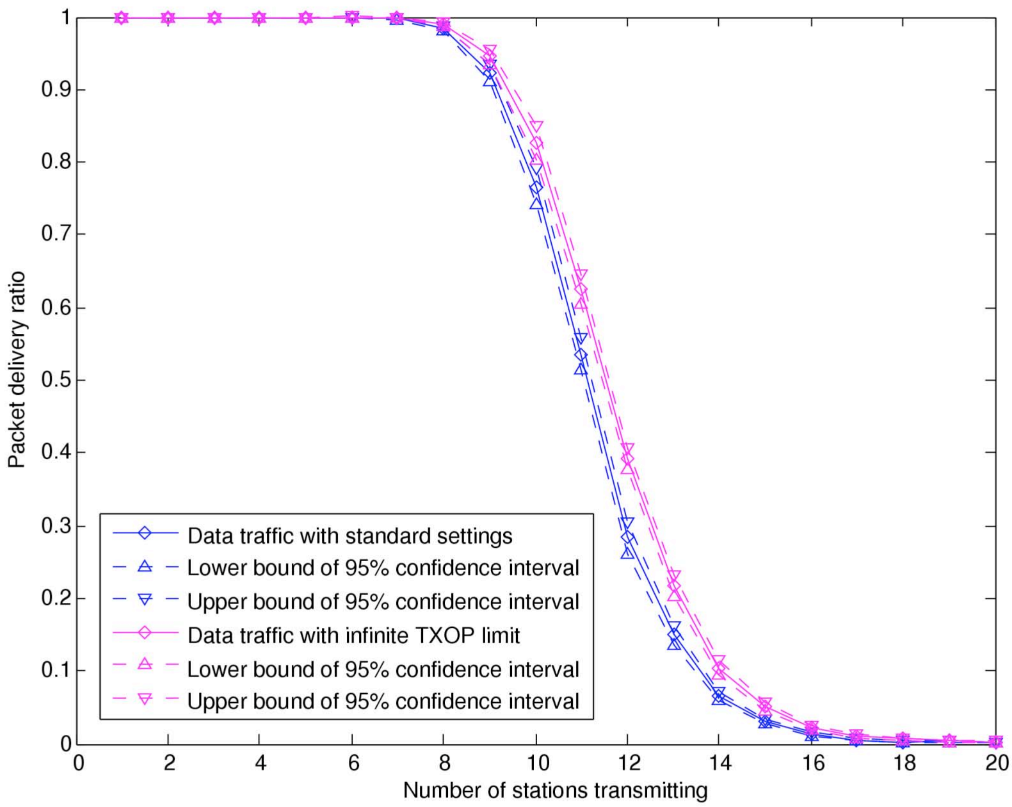

Figure 10. Data traffic packet delivery ratio, standard and infinite TXOP limit settings

Figure 10 shows the data traffic packet delivery ratio. Results are similar to Figure 10. The figure shows that infinite data traffic TXOP limit improves the packet delivery ratio. When 12 stations are transmitting the improvement is 38%. The result shows that more data packets are getting through with infinite TXOP limit settings than with standard settings. The improvement occurs when transmitting stations increase so that the network slowly becomes congested. However, after more than eight stations are transmitting the network starts to become congested. As the amount of stations increases, packet delivery rate drops rapidly and the network quickly becomes unusable.

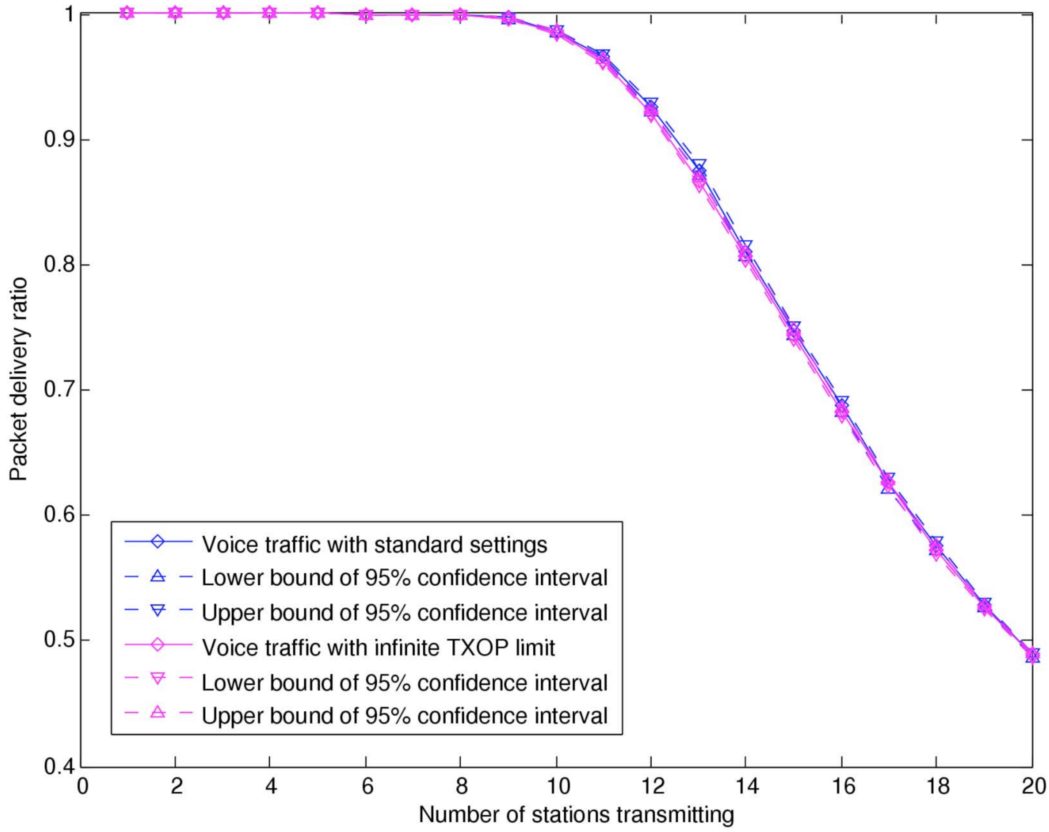

Figure 11. Voice traffic packet delivery ratio, standard and infinite TXOP limit settings

Figure 11 shows the voice traffic packet delivery ratio as number of stations transmitting increases. The key point of interest is whether infinite data traffic TXOP limit disturbs the voice traffic packet delivery ratio. The results show that when 12 stations are transmitting the decrease in data traffic packet delivery ratio is around 5%. As with data traffic in Figure 10, after more than eight stations are transmitting the packet delivery ratio of voice traffic starts to drop. As the number of transmitting stations increases, the network becomes unusable and any voice conversation unintelligible.

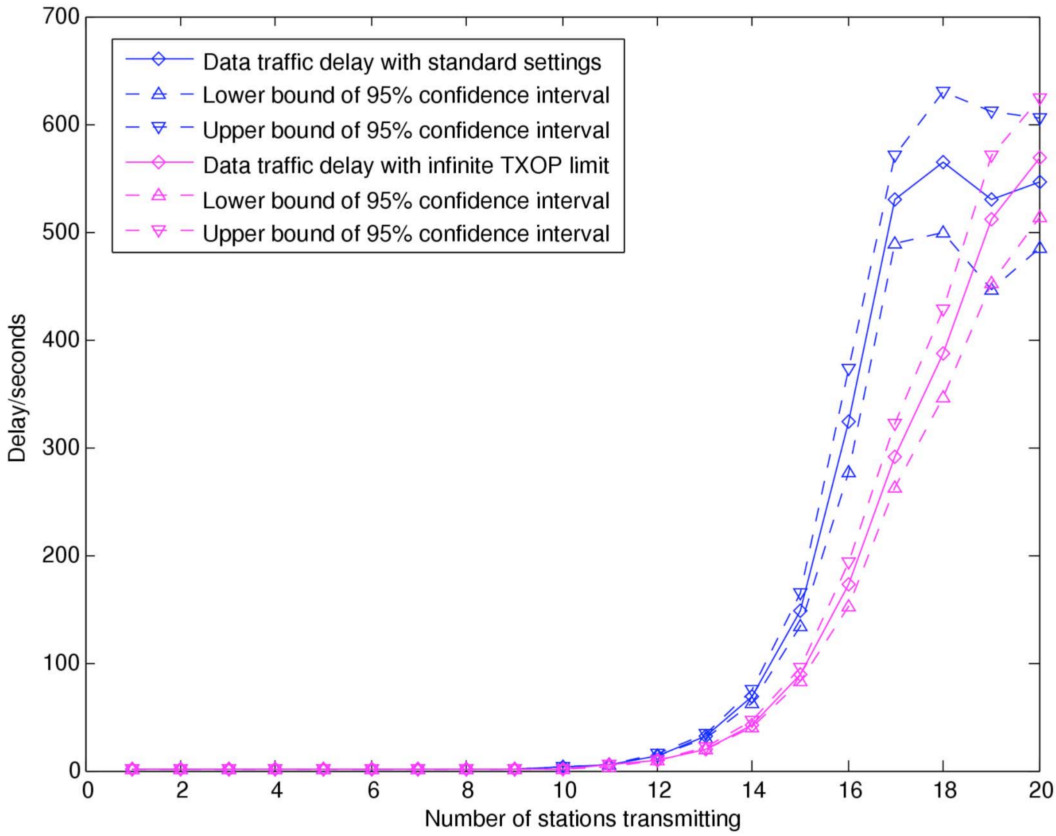

Figure 12. Data traffic delay, standard and infinite TXOP limit settings

Another crucial metric to look at is delay. Figure 12 shows data traffic delay as more and more stations are transmitting. The figure shows that when there are between 14 and 18 stations transmitting, data traffic delay is significantly larger with standard settings than with infinite TXOP limit settings. With a very large number of nodes the delay is similar. However, since delay can only be measured of traffic that actually gets through, dropped packets are ignored. So in case of a congested network, the sink node receives very few data traffic packets and the mean is calculated from significantly smaller amount of data. Hence delay values when a large number of stations are transmitting only indicate that practically no data traffic gets through in the network. When 12 stations are transmitting data traffic delay decreases about 32%.

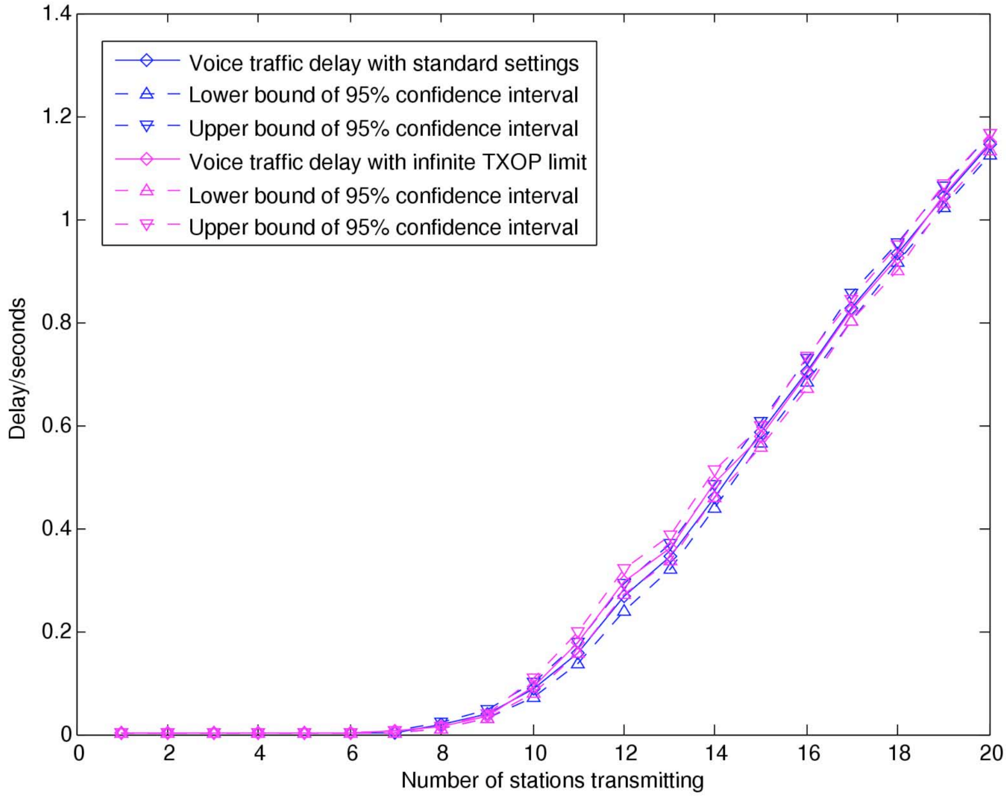

Figure 13. Voice traffic delay, standard and infinite TXOP limit settings

Figure 13 shows the voice traffic delay. This is a crucial metric for voice traffic because it cannot tolerate large delays. The figure shows that using standard or infinite TXOP limit settings the difference is not very largest. The biggest increase in voice traffic delay, 11.7%, occurs when 12 stations are transmitting. It's worth noticing that when more than 10 stations are transmitting voice traffic delay becomes substantial. It is so large that voice quality suffers. Delays beyond 12 stations transmitting are so large that voice conversation cannot be sustained.

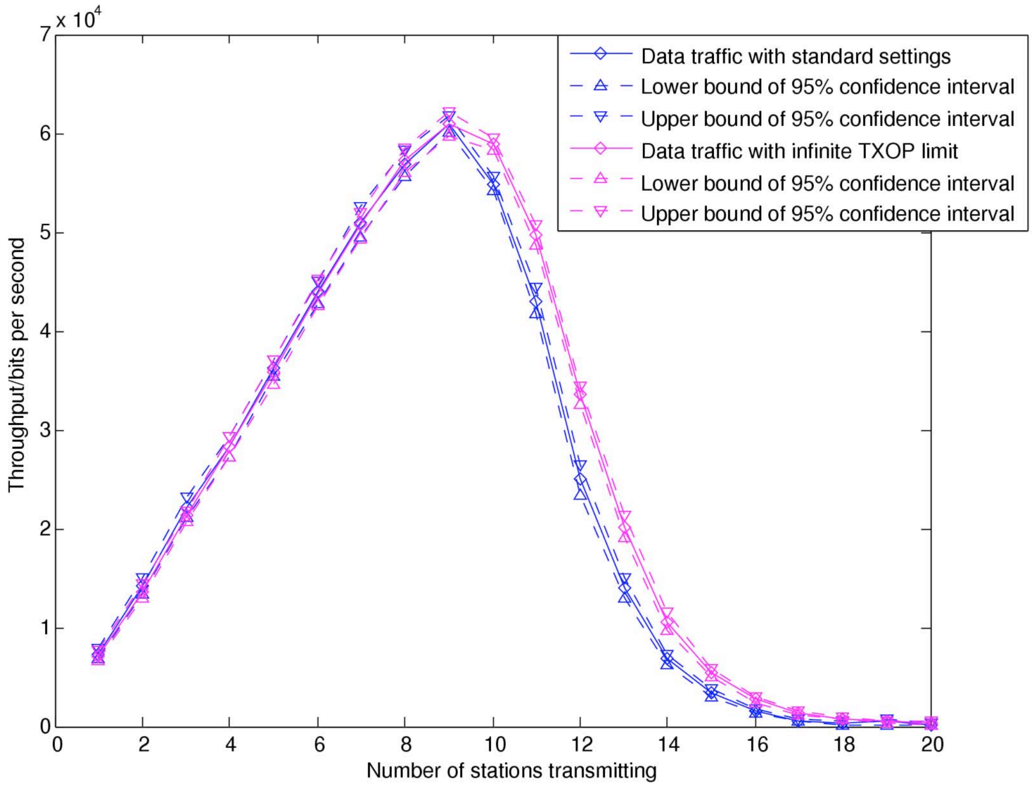

Figure 14. Data traffic throughput, standard and infinite TXOP limit settings

Figure 14 shows the data traffic throughput. The throughput reaches its peak when nine nodes are transmitting. After nine nodes data traffic throughput starts to deteriorate. This means that more packets are dropped. When comparing standard and infinite TXOP limit settings the figure shows that dynamic settings allow more data traffic to pass through than the standard settings when amount of stations transmitting increases from 10 to 16. When 12 stations are transmitting the improvement in throughput is about 35%.

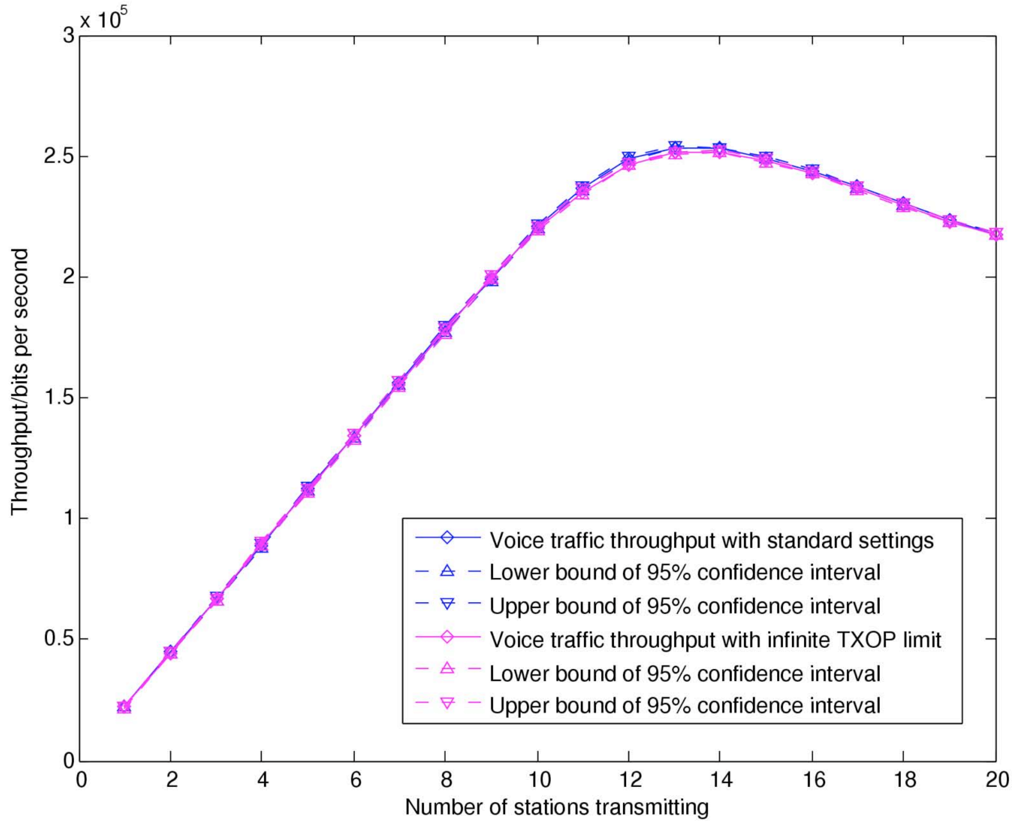

Figure 15. Voice traffic throughput, standard and infinite TXOP limit settings

Figure 15 shows the voice traffic throughput. The figure shows that standard and infinite TXOP limit settings produce nearly identical results. This means that infinite data traffic TXOP limit settings do not disturb the voice traffic throughput much. When 12 stations are transmitting the decrease in voice traffic throughput is about 0.85%. The network maximum capacity causes voice traffic throughput to deteriorate when more than 12 stations are transmitting.

6.2 Static TXOP Limit

The purpose of these simulations is to investigate how different static TXOP limit values for data traffic affect the system. In one simulation set the data traffic TXOP limit is preset to a fixed value. In the next simulation set, the data traffic TXOP limit is preset again to a different value. TXOP limit values are multiples of 32µs like the 802.11e specifies.

In all the simulations of this section 12 stations are transmitting. 12 transmitting stations is an interesting moment to look at the network. This is because with 12 transmitting stations the difference in the results between standard TXOP limit values and infinite TXOP limit values is the largest. From one simulation set to another, only the TXOP limit value changes. All other parameters remain the same.

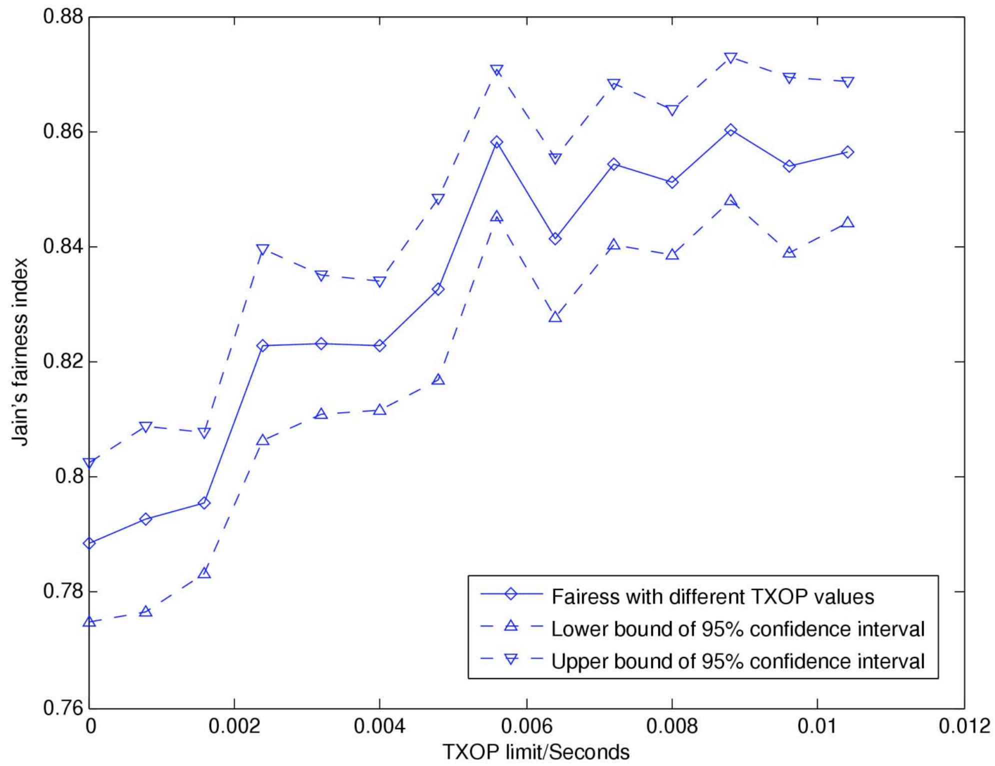

Figure 16. Fairness with different static TXOP limit values.

Figure 16 shows the system fairness. Compared to the data traffic TXOP limit value zero recommended in the amendment, it seems that there is improvement in fairness when data traffic TXOP limit is increased to larger values. Maximum improvement is about 9%. This occurs when TXOP limit is 0.0088 seconds. At that point fairness index is 0.86. Depending on the packets in the queue the actual used TXOP limit time can be shorter that what would be allowed. This is because the station releases any left-over time it cannot use. Stations are also allowed to fragment MSDUs in order to increase the probability of successful transmission or maximize the use of their allowed TXOP limit.

More and more data traffic gets through when its TXOP limit is set to a larger value. This is seems to happen up to a point when the TXOP limit is 0.0056 seconds. After that, the fairness seems to level off and the effect of increasing the data traffic TXOP limit is not so large anymore. According to the standard, the largest TXOP limit should be 0.00816 seconds. Interestingly, maximum fairness improvement occurs with 0.0088 seconds TXOP limit. Values larger than this seem not to provide much more benefit in fairness. Additionally, if the data traffic TXOP limit is set to a very large value, even larger than 802.11e recommends, it is likely that high priority traffic suffers too much. A large static data traffic TXOP limit might not be useful to data traffic either, if it has very few packets to send it will not need all the time a large TXOP limit allows.

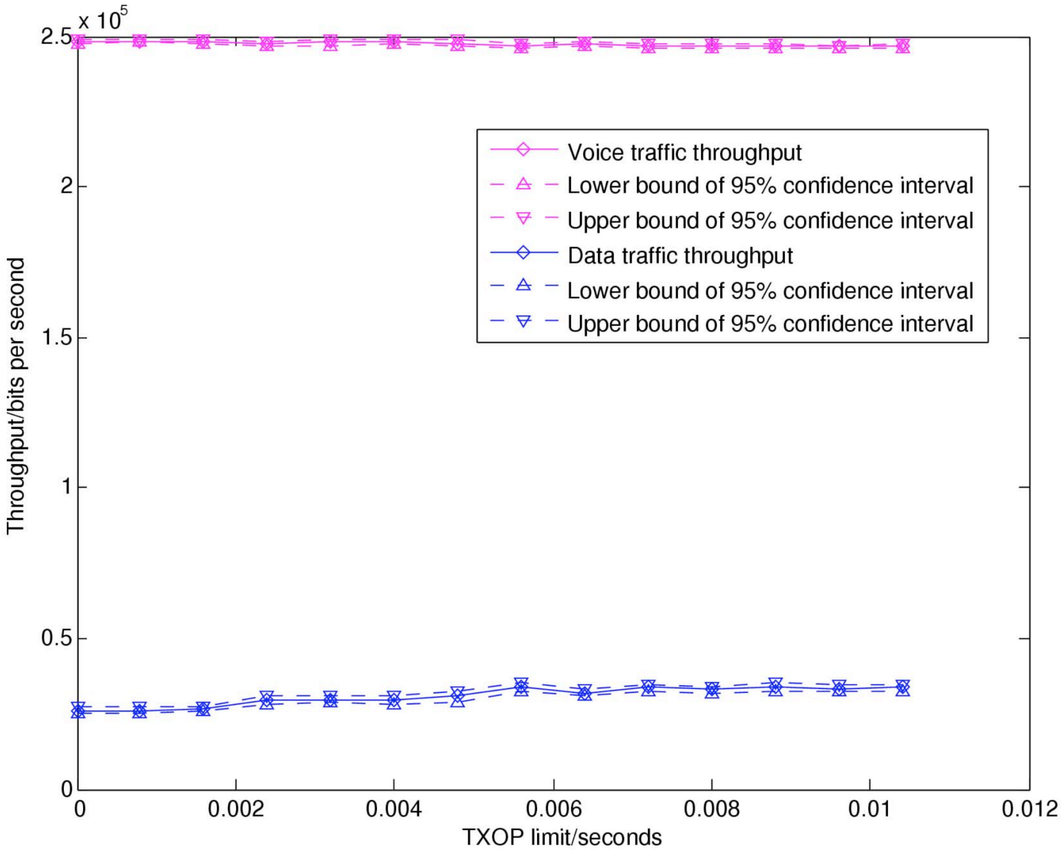

Figure 17. Voice and data traffic throughput

Figure 17 shows the voice and data traffic throughput with different TXOP limit values. The figure shows that data traffic throughput increases when TXOP limit increases. At the same time voice traffic throughput decreases. The change is around 30 % improvement for data traffic and 0,78% decrease for voice traffic. Positive effect on data traffic throughput is larger than negative effect on voice traffic throughput. However, at this traffic level data traffic packet delivery ratio is around 30-40% as Figure 18 shows. This means that most data traffic is not delivered. 30% improvement is not enough for all data traffic to get through.

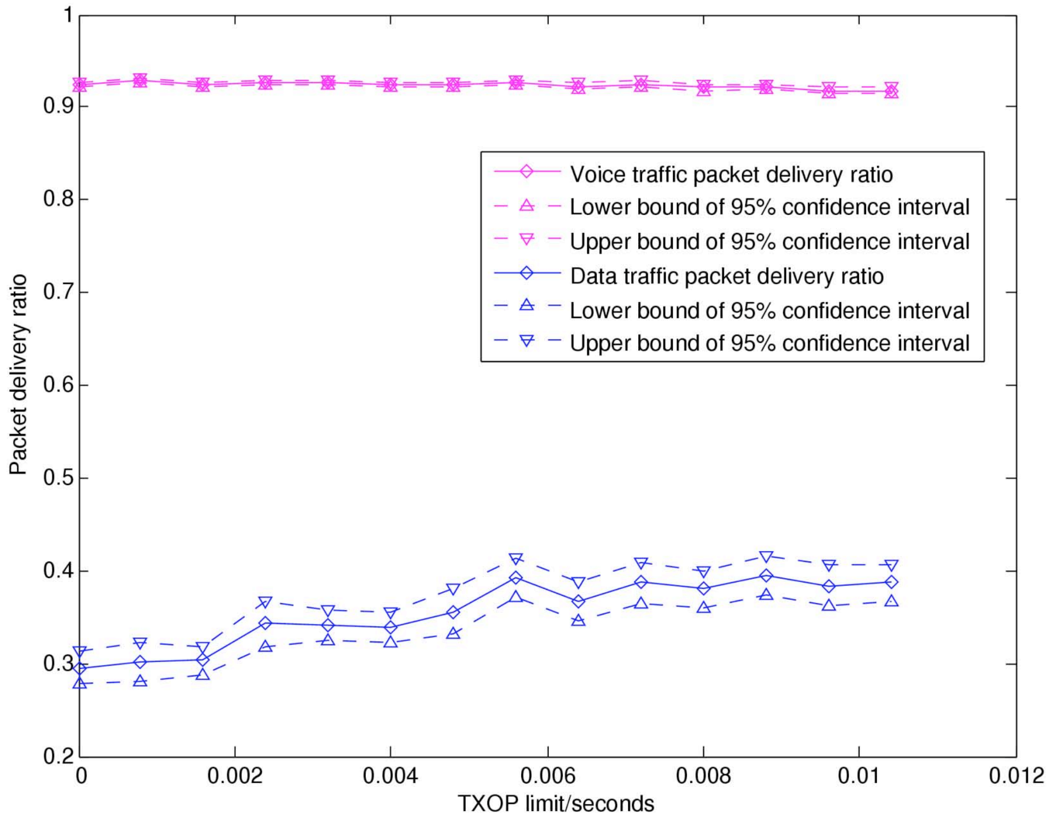

Figure 18. Voice and data traffic packet delivery ratio

Figure 19 shows the packet delivery ratio of voice and data traffic with different TXOP limit values. The figure shows that voice traffic packet delivery ratio suffers slightly with large data traffic TXOP limit values. However the change is only approximately 0,78%. The figure shows that with large data traffic TXOP limit values, data traffic packet delivery ratio is improved by approximately 34% from 30% to 40%. The improvement in data traffic packet delivery ratio is larger than the decrease in voice traffic packet delivery ratio. As with fairness presented in Figure 16, it seems that data traffic packet delivery ratio improves with larger TXOP limit values up to 0.0056 seconds. There does not seem to be much significant improvement with values larger than that. It is worth noticing that although voice traffic packet delivery ratio only suffers slightly, at just above 90% delivery rate, voice conversation is not very usable. When comparing the result to those of Figures 10 and 11 at twelve stations transmitting, it seems that results are very similar.

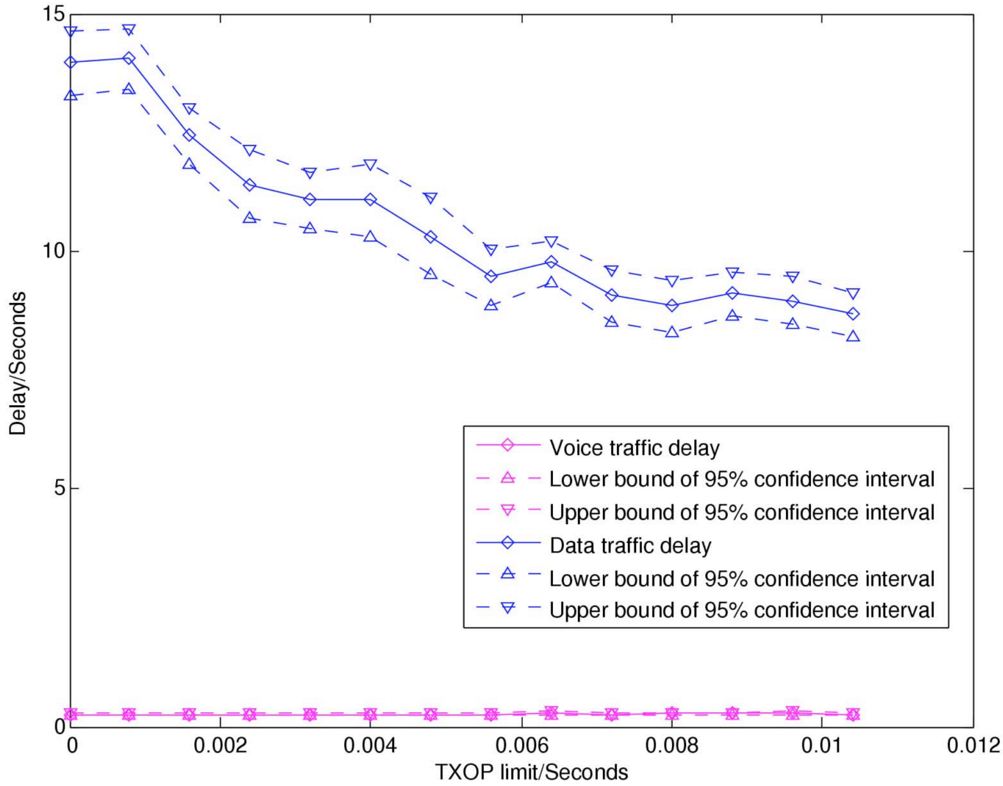

Figure 19. Voice and data traffic delay

Figure 19 shows the voice and data traffic delay when TXOP limit increases. The figure shows that voice traffic delay increases by approximately 12% as data traffic is sent in bigger bursts. With data traffic, there seems to be a much larger improvement in delay. Data traffic delay seems to decrease about 38%. However, even with the improvement the delay is very large. Again results seem similar to those of Figure 12 and 13 with twelve stations transmitting. As with infinite TXOP limit simulations, dropped packets are not included in the delay calculations.

7. Conclusions

Both static and infinite TXOP limit simulations show an improvement in fairness as data traffic TXOP limit increases. The effect of the increase depends on the network congestion level; in highly congested networks, large data traffic TXOP limit does not improve fairness. This is because the network does not have any capacity left to accommodate more traffic. Conversely, when the network has very little traffic, large data traffic TXOP limit is not needed. In such a situation all traffic can be transmitted without much delay in any case.

Most interesting phase in the network occurs when the network moves from not congested to fully congested. During this change data traffic starts to be dropped because of its low priority. Data is always dropped sooner than voice. Figures 11 and 12 showed the data and voice traffic packet delivery ratio respectively. These figures show that data traffic packet delivery ratio drops to almost zero in a relative short time, while voice traffic throughput only suffers. When there are 16 stations transmitting almost zero data packets are being delivered. At the same time approximately 77% of voice traffic is still delivered. When there are 8 stations transmitting, the packet delivery ratio for both traffic types is approximately 100%. In this situation all traffic can be delivered.

The transition phase from not congested to fully congested is the period when TXOP limit change can have an impact on the network fairness. Figure 10 shows this impact. When infinite TXOP limit is used, network fairness improves.

7.1 Comparing Static TXOP and Infinite TXOP

In the simulations of Section 6.1 an infinite TXOP limit was used. This means that the data traffic TXOP limit is always set to the maximum value where all the packets in the queue can be sent. The results show that using an infinite TXOP limit the network fairness improves as network becomes congested. The biggest effect was noticed when 12 stations are transmitting. This same network situation with 12 transmitting stations was then looked at more carefully by using static TXOP limit values. The results of these simulations are shown in Section 6.2. The static TXOP limit simulations showed an increase in the network fairness with larger TXOP limit values. It seems that the larger the TXOP limit the more the fairness improves. However, after a certain TXOP limit value the fairness does not significantly improve anymore. This means that bigger TXOP limit values are not useful to data traffic of the kind used in these simulations.

When comparing the maximum improvement in fairness in both infinite and static TXOP limit case, the difference in the maximum fairness value is only 0.1%. In the static case the maximum fairness occurs with a large TXOP limit of 0.0088 seconds. Static TXOP limit values larger than this do not lead to better fairness. This indicates that the biggest TXOP limit value 802.11e amendment suggests to be used, 0.00816 seconds, is a good maximum value for a static TXOP limit. In case of the infinite TXOP limit simulations, the average of the TXOP limit increases as network becomes more congested. When 12 stations are transmitting it is 0.011 seconds. This average is larger than the static TXOP limit value needed to produce almost the same fairness results. This indicates that a smaller static TXOP limit is enough to produce the same results as using an infinite TXOP limit.

The effect fairness improvement had on voice traffic was varied. Voice traffic delay increased by 12% in the static simulations. In the infinite TXOP limit simulation voice traffic delay overall remained almost same with standard simulations. Voice traffic packet delivery ratio remained approximately the same as well as. Voice traffic throughput also did not seem to suffer.

From the data traffic's point of view, in addition to fairness, also delay, packet delivery ratio and throughput improved with a non-zero TXOP limit. The biggest change was in delay, which improved more than the others metrics.

Overall it seems that having a high static or infinite data traffic TXOP limit improves fairness in the network, as well as data traffic throughput, delay and packet delivery ratio. However no one static TXOP value seems to produce better results than the infinite TXOP limit.

7.2 Evaluation of results and improvements

The results show improvement in fairness with non-zero data traffic TXOP limit values. However, there were only two kinds of traffic in the network in the simulations. In real life traffic would be far more diverse. It is difficult to create traffic models for simulations. Simulation traffic models, like the kind used in this thesis, are simplifications of real traffic. So results from live networks can be different.

In order to improve simulation traffic modeling, real traffic traces can be used. Realistic traffic models create spikes and other random behavior that can make it harder to interpret the results. For further realism, noise and channel error should also be used. Real life situation is not error free like the simulations in this thesis. Also the traffic model in this thesis is heavily VoIP laden, which might not be realistic.

Where VoIP is concerned it is crucial to pay attention to delay. The rule of thumb for VoIP delay is that one-way delay should not exceed 150 ms and anything over 250 ms will make the conversations difficult. Also the encoding scheme chosen can have a delay effect. G.711 does not compress and hence adds little delay but G.729 compress voice and adds a delay of 25 ms [Cisco]. It is possible to reduce bandwidth requirements either by using a technique called RTP header compression or with voice activity detection (VAD) technique, which prevents packets of silence in the conversation from being sent.

In the simulations of this thesis, VoIP delay quickly exceeds 150 ms when more than 10 stations are transmitting. After more than 10 stations are transmitting VoIP quality suffers. This excessive delay occurs in both standard simulations and infinite TXOP limit simulations and is due to increasing overall congestion in the network. Changing the TXOP limit did not seem to have a big further effect on VoIP delay.

Of the absolute numbers of how many voice or data traffic streams the network can sustain definite conclusions cannot be made based on the simulations. This is because the network used in the simulations is not realistic enough. Also based on the simulations it cannot be determined what would generally be the best static TXOP limit value. In real life, traffic is much more varied than in the simulations of this thesis. The high TXOP limit value determined to give the best fairness results in the simulations is the best result for the kind of traffic modeled here, but it might not be best for all kinds of traffic and all kinds of traffic combinations.

7.3 Topics for Further Study

This thesis only focused on two access categories, voice and background. It would be interesting to use all four access categories in simulations. Simulation model in such a case would be more complicated to construct, especially in terms of modeling different kinds of traffic. Such simulations would show the effect TXOP limit changes have on all four categories together. In addition, changing TXOP limit in all the access categories, either simultaneously or separately would make it possible to observe TXOP limit effect on the entire system.

TXOP limit affects the transmission duration, while AIFS and contention window define the channel access frequency. In this thesis the relationship between transmission duration and channel access frequency was not studied. This relationship and its effect on fairness would be interesting to research. This could be researched for example by looking at changing TXOP limit effect on best effort and background access categories.

Besides TXOP limit, the access point can change the contention window maximum and minimum and AIFS per access category. 802.11e gives a basic standard parameter set of static values as a reference. The standard parameter set is just one example of how the parameters can be used. It would be interesting to investigate other possibilities for the entire set, either static or dynamic. Dynamic tuning of all the parameters would likely be the most optimal solution, but it is very difficult to create an algorithm for dynamic change. It is also possible to set EDCA parameter set differently inside an access category, which would create even more differentiation. Whether this brings any benefits or not would be interesting to investigate. However, finding an optimal way to do this is also very complicated.

Another point that needs to be considered in optimizing the 802.11e WLAN is uplink-downlink fairness. The access point can use a different set of EDCA parameters than those it advertises to the QSTAs. If an access point is not prioritized in the parameter set, it can become a bottleneck that diminishes the overall network capacity.

In 802.11e there are also other parameters that could be changed. These are: RTS threshold, fragmentation threshold, long and short retry limit. However, it is likely that sufficient traffic differentiation can be accomplished without changing these parameters. Fragmentation threshold is useful when there are a lot of channel errors and in these situations changing it dynamically could be beneficial. For real-time multimedia, retransmitted frames may be too late to be useful, so smaller retry limit is appropriate and for non real-time a larger retry limit is needed for reliable transmissions.

In addition to modifying the TXOP limit/EDCA parameter set it is possible to improve the system fairness in other ways. One simple way to improve the efficiency and reduce overhead is to use block acknowledgments introduced in 802.11e. Another would be using admission control. This is mentioned in several research papers as way to assure that the network never becomes so congested that QoS suffers. Research papers suggest admission control as an interesting topic to study further.

Admission control can be centralized or decentralized. In the centralized model, solely the access point handles admission control. In the decentralized model, the QSTA's or applications in need decide if there is enough capacity available to fulfill their requirements. If an application realizes that the channel is unable to meet its QoS requirements it can refrain from loading the channel further or reduce its demands for example by increasing compression. Application based access control assumes that there are no greedy applications, but in real life preparations for the occurrence of such applications would need to be made. Admission control could also be tentative. First all streams would be accepted into the network and delay and throughput is measured to see whether there actually is any capacity for the new traffic.

In order for admission control to work, the access point or the station implementing it needs to measure the network situation in some way. Network measurement is an intriguing area to study and would benefit 802.11e networks as well. The measurements could be used not only for admission control but also as a trigger for changes in EDCA parameter set. Changing the parameters in accordance to the network situation is an interesting topic to study, although also complicated. One possibility for the AP to know when the EDCA parameter set should be changed is to monitor the network and periodically measure its situation. For different access categories different parameters could be monitored. The channel usually becomes delay limited before throughput limited, so delay should be the focus of higher category traffic measurements.

The downside in using network monitoring to perform admission control or general controlling is the fact that it is hard to do. Especially measuring delay in a live situation is very complicated since clocks in each end need to be synchronized to a high degree. For traffic load a simple way to measure it, is to look at the relative occupied bandwidth, which means channel busy time divided by total time.

To synchronize clocks for delay measurements in 802.11e it might be possible to use beacon frames. Then the timestamp in the packets could be used to estimate the delay. Also it is possible for each individual station to measure delay in its queues and in ACK receiving. Then if AP needs the results of the measurements, the stations would need to periodically transmit the results to the access point but this would increase overhead. Besides these methods, using some specialized measuring protocol could be studied further.

7.4 Summary

Overall the effects of TXOP limit changes seem positive. Fairness in the network increased, but not at a big expense to voice traffic. Voice traffic delay, throughput and packet delivery ratio remained almost the same while data traffic experienced improvement. It seems that sending data traffic in bigger bursts improves the network fairness. However, the improvement is not very large. Also when the network is heavily congested, data traffic queues almost never gain access to the channel. In such heavy congestion situations this method does not improve fairness.

The results indicated that having a large static TXOP limit gives equal results as letting the TXOP limit set itself according to the queue length. In the latter case, TXOP limit average is very high. This means that large data traffic TXOP limit of any kind improves fairness in the network. Whether it is enough just to use a static TXOP limit in a more realistic network needs to be researched further.

In the end everything comes down to trade-offs. The most important question to ask is what kind of traffic we want favor and how much? If we are only interested in the highest category of traffic then we should be willing to sacrifice all other traffic. However, it might not be optimal to favor one kind of traffic that much. It is also important that the lower priority traffic gets transmitted. To improve data traffic throughput, TXOP limit modifications are beneficial. Most importantly, modifications do not seem to disturb high priority traffic very much. These results indicate that low priority TXOP limit changes should be used if network fairness improvement is wanted. However, TXOP modifications are likely to be just a part of the answer. For fully optimized network the entire EDCA parameter set should robustly respond to network conditions.

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