工學博士學位論文

Access Time Control for TCP Fairness in WLANs

무선랜에서 공평성 향상을 위한 사용 시간 제어

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Access Time Control for TCP Fairness in WLANs

by

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Abstract

Wireless local area networks (WLANs) based on IEEE 802.11 standard have been widely used in infrastructure and ad hoc networks. Although the MAC layer of 802.11 WLAN provides fairness in accessing the wireless channel, the channel capacity is allocated unfairly among stations and flows due to effect of the upper layer protocols such as TCP and asymmetric network conditions. In this dissertation, this unfairness problem of TCP traffic in the IEEE 802.11 based infrastructure and ad hoc networks is studied.

In the first part of the dissertation, the per-station fairness in infrastructure WLANs is studied. Under the 802.11 distributed coordination function (DCF), there exists unfairness between uplink and downlink TCP flows in infrastructure WLANs. In addition, when each station has a different number of flows, the channel capacity is allocated unfairly among stations.

In order to resolve these problems, the Access Time Control (ATC) scheme and the Distributed Access Time Control (DATC) scheme are proposed. In the ATC scheme, an access point (AP) controls the rate of TCP flows based on the channel access time of each station so that it can provide per-station fairness regardless of the number and direction of flows in each station without sacrificing throughput. On the other hand, in the DATC scheme, each station controls the rate of its TCP flows based on the channel access time without coordination of the AP.

By applying these schemes, per-station fairness among stations can be achieved, and the QoS of traffic with high priority can be protected from the best-effort traffic in the 802.11e networks. Also, the network throughput decreases little even when there exist stations with low transmission rate. Since the proposed

schemes do not require any modification of the MAC and TCP layers, they can be easily implemented in current devices. The properties of the proposed schemes are investigated and their effectiveness is verified through ns2 simulations.

In the second part of the dissertation, the fairness in ad hoc networks is studied. In ad hoc networks, each station may forward some flows for other stations as well as sends its own flows. Therefore, the number of flows to deliver may be different among stations. However, since the 802.11 DCF provides an equal chance for sender stations to access the channel regardless of their number of flows, it can induce unfairness among flows. In addition, the hidden node problem degrades throughput and fairness of TCP flows in multihop ad hoc networks. In order to resolve these problems, the ATC scheme for ad hoc networks is proposed. In the proposed scheme, by fairly allocating the channel access time of a flow to each link and by accounting the interference level from hidden nodes, fairness and throughput are enhanced. Through ns2 simulations for various topologies, the effectiveness of the proposed scheme is shown.

Keywords: IEEE 802.11, TCP, Fairness, Virtual rate control, QoS, Infrastructure network, Ad hoc network

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Chapter 1

Introduction

1.1 Background and Motivations

The Internet has become an essential part of daily life of an ordinary man. People want to access the Internet at any time and at any place. To satisfy this public demand, wireless local area networks (WLANs) based on IEEE 802.11 standard [1] are widely deployed in public areas such as airports, cafes, hotels, etc. Deployment of these public WLAN hotspots, initiated by various network operators - cellular network operators (GSM and CDMA), Wireless Internet Service Providers (WISP), dial-up aggregators, and fixed broadband operators (xDSL, cable) - is growing rapidly across the globe. The Gartner predicts that by the year 2008 there will be more than 167 thousand public WLAN hotspots around the world, and there will be over 75 million users of public WLAN hotspots worldwide [2]. As the public WLAN is becoming more prevalently utilized and as a result, more people have access to the Internet [3], it is necessary to provide fair service among users.

The IEEE 802.11 MAC protocol is designed to provide equal chance to access

channel for each stations, but there exist some causes inducing unfairness: location dependency, a different number and direction of flows in each station, and interaction with higher layer protocols such as TCP. Especially, TCP fairness in WLANs is important because the majority of applications in today's Internet use the TCP protocol.

Many researches have been studied about TCP fairness in access networks using the 802.11 WLAN [4–7]. They showed that there exist unfairness between uplink and downlink flows because the packet drops in an access point (AP) penalizes downlink flows by the TCP congestion control mechanism. Though they have proposed solutions in order to resolve this unfairness, their solutions are required to modify the existing MAC protocol or TCP window size in the AP and thus, are impractical to apply to current infrastructure WLANs.

Moreover, since they did not consider the case when each station has a different number of flows, they cannot provide fairness among stations having a different number of flows. In the public access networks, users can make several flows simultaneously for various applications such as file transfer, e-mail, web browsing, etc. If there is a greedy user who generates many flows, the other users may use less network resource. Since the user is the entity with whom the contract is signed in the commercial access network, it is desirable that all users can utilize channel capacity equally regardless the number and direction of flows [8]. From this point of view, this dissertation proposes methods which can provide per-station fairness and can be implemented without any modification of the MAC and TCP layers.

In addition to the infrastructure mode, the 802.11 specification defines the ad hoc mode, in which each station can communicate with each other directly without

access points. Ad hoc networks are used in order to communicate in places such as battlefield, disaster area, and a conference room where network infrastructure such as access points does not exist. Recently, the 802.11 WLAN based ad hoc networks are widely used to set up temporary peer-to-peer networks for file share, network games, etc. The 802.11 WLAN has been considered as one of the most promising technologies for multihop ad hoc networks that can extend network coverage. In multihop ad hoc networks, TCP performance has been studied by many researchers [9–16], and they have found that TCP traffic in the 802.11 multihop ad hoc networks can suffer from degradation of throughput and fairness. In this dissertation, in order to enhance fairness and throughput of TCP traffic in ad hoc networks, a capacity allocation scheme based on access time of each station is proposed.

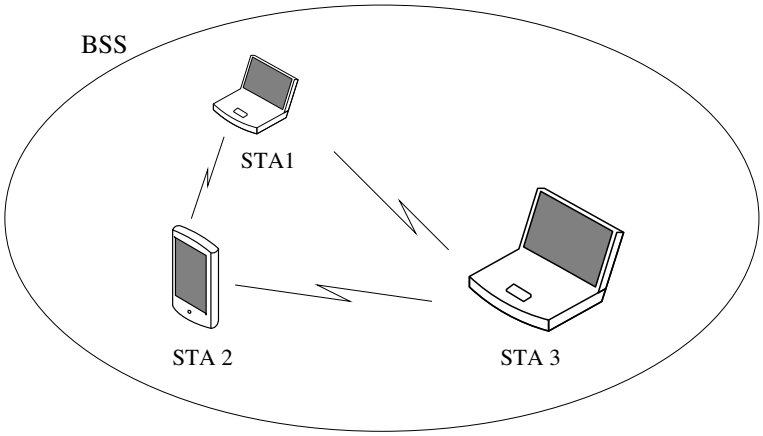
1.2 Preliminaries and Related Work

1.2.1 IEEE 802.11 WLAN Overview

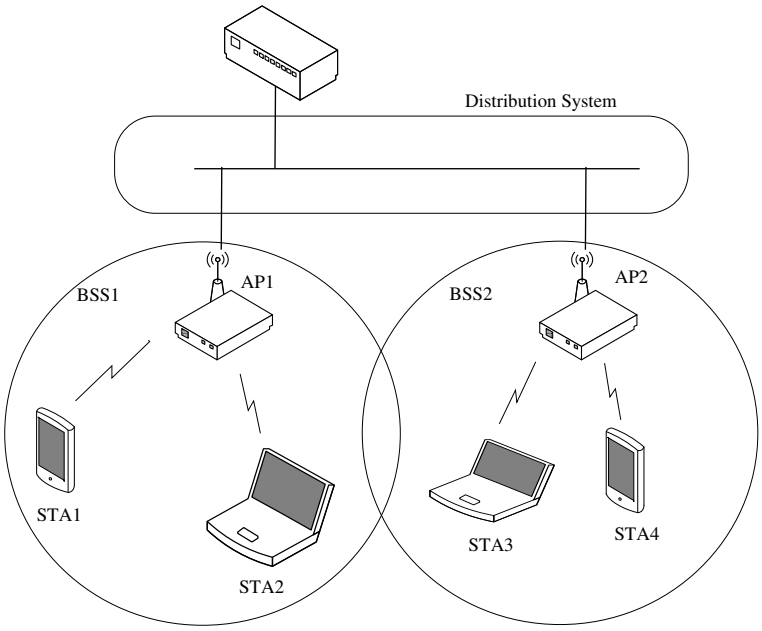
- **Network models in the IEEE 802.11 WLAN**

[1] IEEE 802.11 WLANs can be used either to replace wired LANs, or as an extension of the wired LAN infrastructure. The basic topology of an 802.11 network is shown in Fig. 1.1(a). This Basic Service Set (BSS) consists of two or more stations (STAs), which have recognized each other and have established communications.

The independent BSS (IBSS), shown in Fig. 1.1(a), is the most basic type of IEEE 802.11 LAN. This mode of operation is possible when IEEE 802.11 stations can communicate with each other directly. Because this type of IEEE 802.11 LAN is often formed without pre-planning, and operates as long as there is a



(a) BSS



(b) ESS

Figure 1.1: Basic service set and extended service set with APs

need, this type of operation is often referred to as an ad hoc network.

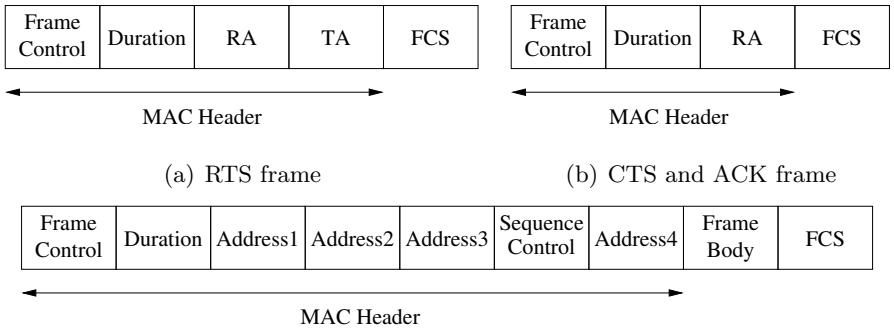
A BSS can contain an Access Point (AP). The main function of an AP is to form a bridge between wireless and wired LANs. The AP is analogous to a basestation used in cellular networks. When an AP is present, stations do not communicate on a peer-to-peer basis. All communications between stations or between a station and a wired network client go through the AP. The AP's are not mobile, and form a part of the wired network infrastructure. A BSS in such a configuration is said to be operating in infrastructure mode. The Extended Service Set (ESS) in Fig. 1.1(b) consists of a series of overlapping BSSs, connected together by means of a Distribution System (DS). Although the DS could be any type of network, it is almost an Ethernet LAN.

- **Multiple access in the IEEE 802.11 WLAN [1]**

In the IEEE 802.11 standard, access to wireless medium is controlled by means of rules, which are called coordination function. The fundamental MAC protocol defined in the 802.11, which must be implemented by every station, is called Distributed Coordination Function (DCF). DCF is a CSMA/CA (Carrier Sense Multiple Access / Collision Avoidance) channel access method, used both in ad hoc and infrastructure networks.

Carrier sense is performed using both physical and virtual mechanisms. In physical carrier sense, a stations senses the channel condition before attempting any transmission and checks that the wireless medium is idle for a certain period of time called the DCF interframe space (DIFS).

To avoid collision between two simultaneously transmitting stations, a backoff algorithm is used as well as deferral of transmission for DIFS time. When there are pending transmission requests and the wireless medium is detected busy, an



(a) RTS frame

(b) CTS and ACK frame

(c) Data frame

Figure 1.2: MAC frame format

STA would wait until the wireless medium is idle for DIFS time. Then it draws a random number within a contention window (CW), and this number is used to calculate an additional time period that it has to wait. If the wireless medium is detected to become busy during backoff, the countdown is suspended. The countdown is resumed once the wireless medium is detected to be idle subsequently for DIFS time.

The contention window varies depending on the number of previous retransmission attempts. The contention window is initially set to the minimum contention window (CW_{min}). Whenever a transmission fails, the contention window is doubled and grows up to the maximum contention window (CW_{max}). When a transmission succeeds or the number of retransmission exceeds a certain limit, the contention window is reset to CW_{min}.

Collision avoidance is also achieved by a virtual carrier sense mechanism. Each station maintains an internal timer, called the network allocation vector (NAV), which indicates when the wireless medium is busy. As can be seen in Fig. 1.2, a duration value is included in the MAC header of each frame that is transmitted by a station, which indicates how long the transmission lasts, including any

subsequent acknowledgements and interframe spaces. All stations in the vicinity receives the frame and uses the duration value to update its NAV. Then when a station is about to start a transmission, it first checks if its NAV is zero. If not, it must wait until the NAV is counted down to zero because NAV indicates that another station has access to the wireless medium.

On winning contention to the wireless medium a station can transmit one frame. Then the station waits for a time period called short interframe space (SIFS) for the ACK from the receiver. SIFS is shorter in duration than DIFS, which gives the ACK frame the highest priority for accessing the wireless medium. This ensures that any other STA would not start transmission while the ACK is expected. If the ACK is not received after SIFS, retransmission is attempted until the number of retransmission has exceeded a certain threshold. In this case, the frame is discarded.

In addition to this basic frame exchange sequence, an optional RTS/CTS mechanism is defined to increase the robustness of the protocol. Under this mechanism, a station sends a RTS frame to the destination before transmitting a data frame. The destination then responds with a CTS frame once it has correctly received a RTS. The sender can send the data frame after receiving the CTS response. All frames, including RTS and CTS frames, contain information about the duration of the transmission. Based on this information all surrounding stations can update their NAV and defer any transmission until this time expires. Even if a hidden station cannot hear the RTS from a source STA, it will be able to receive the CTS response from the destination station and update its NAV accordingly. Figure 1.3 shows the exchange of RTS/CTS, data, and ACK between source and destination stations, and NAV setting of other stations.

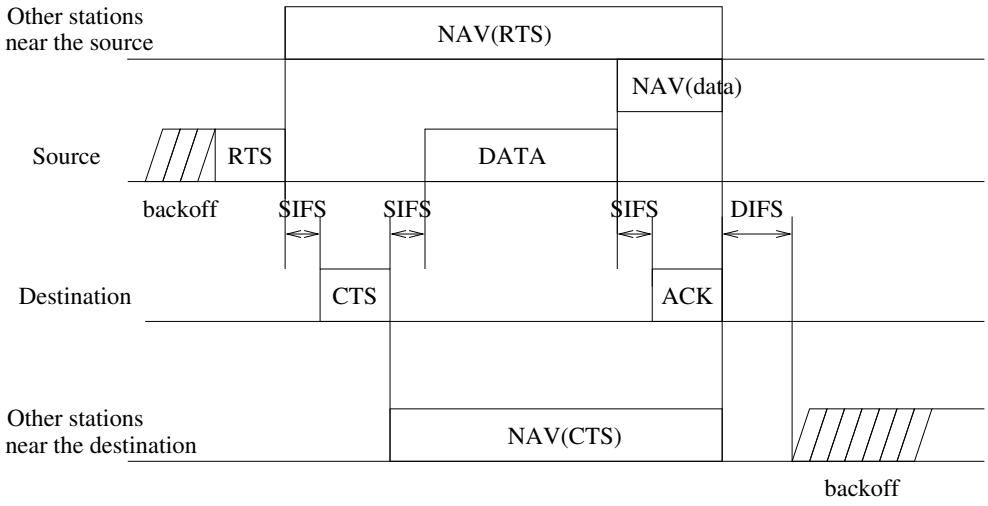


Figure 1.3: Example of DCF frame exchange sequence and NAV setting

Under the DCF, all stations contend for the wireless medium with the same priority. There is no differentiation between data flows to support traffic with QoS requirements. To solve this problem, the IEEE 802.11 committee has formed the Task Group E (802.11e) committee to define enhancements to the original 802.11 MAC. The 802.11e QoS facility defines a new coordination function called enhanced distributed channel access (EDCA) [17, 18].

EDCA enhances the original DCF to provide prioritized QoS, i.e., QoS based on access priority to the wireless medium. Prioritized QoS is realized through the introduction of four access categories (AC), which provide delivery of frames associated with user priorities as defined in IEEE 802.1D. In the EDCA, each station has multiple queues for the prioritized service for different ACs, e.g., voice, video, best-effort, and background flows. The differentiation in priority between AC is realized by setting different values for the AC parameters such as arbitrary inter-frame space number (AIFSN), contention window (CW), and TXOP limit.

When a packet arrives, the 802.11e MAC first classifies the data with an appro-

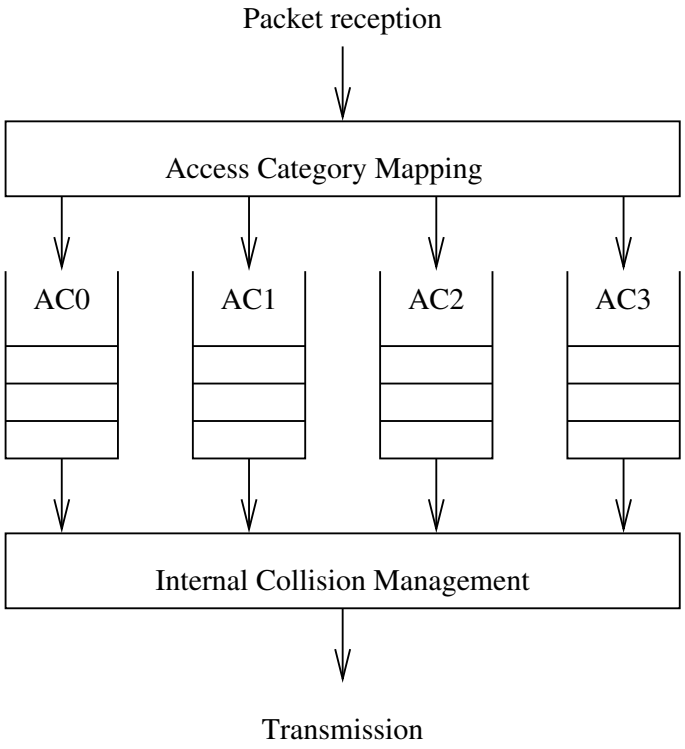


Figure 1.4: Four ACs in the 802.11e station

priate AC, and then pushes the packet into the appropriate AC transmit queue (see Fig. 1.4). Packets from different ACs contend internally within the station. The internal contention algorithm calculates the backoff, independently for each AC, based on AIFSN, contention window, and a random backoff count. The backoff procedure is similar to that in DCF, and the AC with the smallest backoff wins the internal contention. The winning AC would then contend externally for the wireless medium. The external contention algorithm has not changed significantly compared to DCF, except that in DCF the deferral and contention window were constant. 802.11e has changed the AIFS and contention window to be variable, and the values are set according to the appropriate AC [17, 18].

- **Hidden node problem**

It is a well known that hidden nodes induce the performance degradation with respect to the throughput and fairness in multihop ad hoc networks. Two nodes out of transmission range of each other are said to be hidden from one another when both attempt to send packets to the same receiving node, resulting in a collision of data at the receiver node.

To avoid this collision, all of the receiver's neighbor stations need to be informed that the channel will be occupied. This can be achieved by exchange of RTS/CTS message before transmitting data. By overhearing RTS and CTS, all neighbors of the sender and receiver will be informed that the medium will be busy, thus preventing them from transmitting. However, the RTS/CTS exchange cannot solve the hidden node problem completely. When collisions occur between RTS and CTS messages sent by different stations, it can induce collision in receiver stations during data transmission as illustrated in Fig. 1.5.

In addition, since the interference range of a station is larger than the transmission range in the 802.11 WLAN, there still exists a hidden node problem. Figure 1.6 shows that the transmission from C to D can interfere with the station B which has been receiving data from the station A. Since the station C is out of transmission range of B, the station C cannot hear the CTS message from the station B. Even when the station A transmits data to the station B, the station C senses the wireless channel idle and sends a RTS message to the station D. Since the station B is in the interference range of the station C, a collision occurs between the data from A and the RTS from C.

On the other hand, when the station B has received a RTS from A and the station C is transmitting data to the station D, the station B cannot respond to the RTS

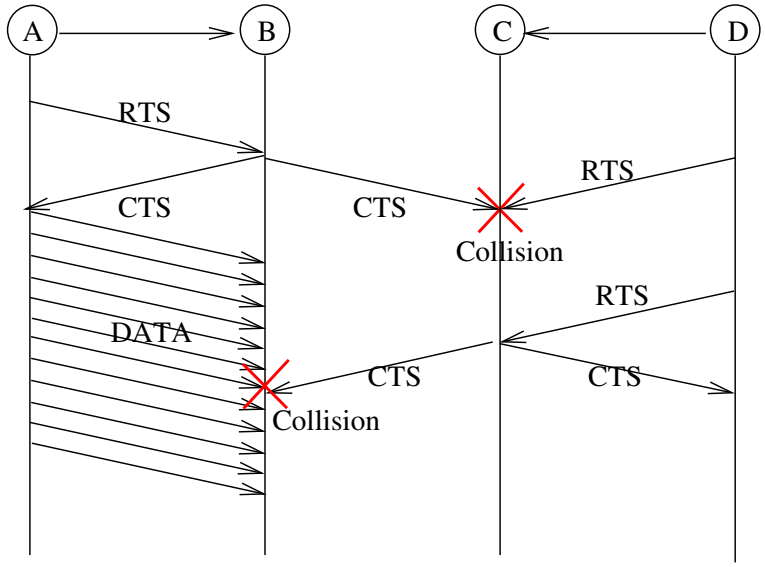


Figure 1.5: Incompleteness of RTS-CTS method

because the station B senses the wireless medium busy. As the station B does not respond with CTS, the station A keeps retransmitting RTS until it reaches the maximum count, and eventually drops the packet degrading the throughput. In addition to this, the interference from hidden nodes can induce unfairness between flows due to capture effect and the exponential backoff scheme of DCF [12, 19]. Figure 1.7 shows an example of unfairness between two flows due to the capture effect and the binary exponential backoff. In Fig 1.7, the station A wants to transmit to the station B, and the station D wants to transmit to the station C. During the transmission from the station A to the station B, the station C cannot reply to RTS from the station D because the station C, in the carrier sensing range of A, knows that a neighbor node is sending a packet. However, the station D, which is out of the carrier sensing range of B, can not hear any transmission, and it tries to retransmit RTS. Moreover, the station D must wait

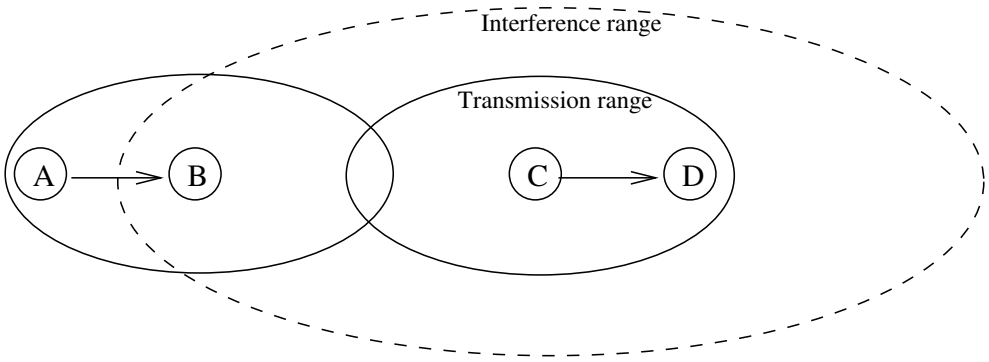


Figure 1.6: Hidden node problem due to a large interference range

longer due to the binary exponential backoff scheme. Even when the station D is in the carrier sensing range of B, the station B may successfully receive packets from A, because the signal from A may be strong enough to ignore the interference from D (capture effect). Since the station A can complete the transmission and the station D has to wait for a longer back off time, the station A will have more chance for the next transmission. Therefore, the flow from the station A to the station B becomes to have more bandwidth. When the flows use the TCP protocol, the sender station A increases its congestion window faster than the sender station D, and the unfairness becomes more significant.

1.2.2 Related Work on Fairness in WLANs

- **Fairness in infrastructure WLANs**

Fairness in wireless LANs have been studied in several papers, most of which have focused on the enhancement of the DCF for fairness at the MAC layer [20–23]. Although these MAC layer fairness mechanisms can provide fair access opportunity to each station, these solutions involve changes to the MAC layer and do not

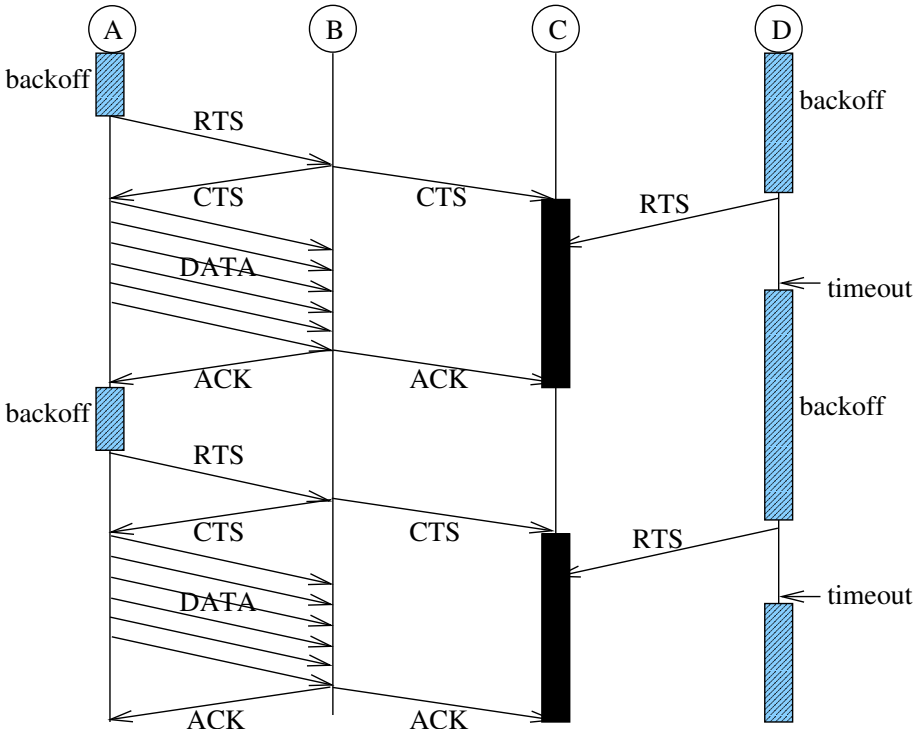


Figure 1.7: Unfairness in a multihop ad hoc network

consider effects of the upper layer protocol such as TCP.

Wang *et al.* [24] and Bottigliengo *et al.* [25] showed that the unfairness between uplink and downlink flows in infrastructure networks is due to contention at the MAC layer. When there are N mobile stations which have uplink flows, the AP has to contend with N stations to serve downlink flows. In [24], in order to provide more transmission opportunities for the AP, the contention window of AP is modified and the AP waits for PIFS instead of DIFS in the first defer period. In [25], the minimum contention window of mobile stations is increased as the number of mobile stations increases. However, when the mobile stations use TCP flows, the contention between AP and uplink stations does not necessarily

penalize the downlink flows because the uplink TCP flows are paced by TCP ACK packets in the downlink.

Pilosof *et al.* [4] showed that the buffer availability at the AP affects unfairness between uplink and downlink TCP flows, and proposed a solution which is based on the manipulation of the TCP receiver window at the AP. They set the value of the TCP receiver advertised windows to be at most $\left\lfloor \frac{buffersize}{Num.offlows} \right\rfloor$ so that the buffer never fills up ($\lfloor a \rfloor$ denotes the largest integer which is smaller than or equal to a.). This solution can provide an ideal performance of fairness, but it is very complex to implement since an AP needs to find out the number of active flows and manipulate the TCP header of all packets. In addition, when the IPsec that encrypts the packet is used, the AP cannot adjust the advertised window in the TCP header.

On the other hand, Detti *et al.* [5] proposed a "lossy rate control solution", which can be implemented with lower complexity compared to the solution of [4]. They considered a simple scenario in which the number of uplink stations is the same as that of downlink stations. In order to limit the rate of total uplink flows to one half of the AP's capacity, they proposed a method in which the AP drops the received packets of uplink TCP flows when the rate of uplink flows exceeds one half of its bandwidth. Thus, the bandwidth of uplink flows is limited and the rest bandwidth can be used of downlink flows. This solution can be simply implemented in the AP, but it did not support the case of that there are different number of flows in uplink and downlink.

Andrew *et al.* [6,26] proposed a modified access network, called CLAMP, that provides an opportunity to enforce local policies to TCP flows. CLAMP uses TCP advertisement window as in [4]. In CLAMP, the access router agent computes

its buffer availability periodically and sends this information to receiver stations. Then, each receiver station adjusts its advertisement window size based on the buffer availability. By this, it is possible to control the queue length of AP and the bandwidth of each mobile station. But, CLAMP requires some modification in the TCP layer at mobile stations and has to use an additional control message to control the queue length and bandwidth allocation.

Kim *et al.* [7] proposed a downlink compensation access (DCA) scheme for fair resource allocation. In DCA, an AP accesses the channel in advance before other stations start backoff procedure. Using DCA, the AP can allocate bandwidth for downlink flows so that it achieves the required utilization ratio of uplink and downlink traffic. Similarly, Fukuda *et al.* [27] proposed a prioritized AP scheme in which the AP use a shorter IFS than the DIFS used by the stations.

Wu *et al.* [28] proposed a per-flow queuing method in order to solve unfairness between uplink and downlink TCP flows and provided per-flow fairness. In their solution, there is a queue for each flow in AP, and a queue is selected for packet transmission by a weighted polling strategy. In this scheme, the AP applies a different polling weight to data queues for downlink TCP flows and ACK queues for uplink TCP flows considering the queue size and buffer overflow. However, when a station has several flows, these flows cannot be differentiated in the MAC layer because the port numbers are recorded in the TCP header. In addition, when the RTTs of flows are different, it is difficult to determine the polling weighted of each queue.

Malik *et al.* [29] proposed a wireless bandwidth management (WBM) framework to provide fair bandwidth distribution. In the WBM, an additional control connection between AP and mobile station, is required to regulate the amount of

data being transferred in the uplink direction, and this involves some modification in the transport layer of both AP and mobile stations.

In addition to the 802.11 access networks, several researchers have studied the 802.11e access networks with respect to effect of uplink traffic. Casetti *et al.* [30] showed that uplink voice and TCP traffic penalizes downlink voice and TCP traffic in the 802.11e networks. They proposed a method that modifies the QoS parameter such as AIFS and contention window so that the downlink traffic has shorter AIFS and smaller contention window than uplink traffic.

Leith *et al.* [31] [32] showed unfairness among TCP flows in an 802.11e testbed. They also used AIFS and contention window of the 802.11e MAC EDCF to address transport layer unfairness among uplink flows, and in order to solve the asymmetry between uplink and downlink TCP flows, they prioritized the downlink TCP data packets by setting TXOP on the AP to allow as many transmissions as the number of downlink flows.

Yang *et al.* [33] showed that uplink TCP traffic degrades the QoS of high priority traffic in downlink, and proposed "Fast-Backoff" and "Dynamic Adjustment when Fail or Successful (DAFS)", in which the QoS parameters (CW and AIFS) of best-effort flows are adjusted dynamically.

Since the previous solutions focused on fairness among flows or considered only single flow per station, they cannot guarantee per-station fairness when each station has a different number of flows. Moreover, most previous solutions required modification of the MAC or TCP layers for providing fairness. Unlike the previous work, this dissertation proposes a method which can provide per-station fairness regardless of the number and direction of TCP flows in each station without any modification of the MAC and TCP layers.

- **Fairness in ad hoc networks**

Recently, several researchers have studied the performance of WLAN on ad hoc networks. Bae *et al.* [9] investigate how effective is the RTS/CTS handshake in terms of reducing interference in multihop ad hoc networks. They showed that RTS/CTS cannot resolve the hidden nodes problem when the interference range is much larger than transmission range. In order to resolve interference caused by large interference range, they proposed a simple MAC layer scheme where a receiver reply CTS when the power of received RTS is larger than a certain threshold. Ye *et al.* [10] presented an analysis of the spatial reuse characteristics of RTS/CTS based ad hoc networks, and proposed an Aggressive Virtual Carrier Sensing (AVCS) scheme in order to improve spatial reuse.

Chaudet *et al.* [34] summarized performance issues in the IEEE 802.11 ad hoc network. They presented that some flows may suffer from starvation while other flows capture the whole channel bandwidth in ad hoc networks. This unfairness is more severe when the network load is heavy. Liew *et al.* [35] showed that controlling the offered load at the sources can eliminate high packet loss rate, re-routing instability and unfairness problems.

In [12], the performance of TCP flows in multihop ad hoc networks was investigated. When TCP is used as transport layer protocol in multihop ad hoc networks, it causes significant performance degradation in terms of throughput and fairness. Since a TCP session is a two-way connection to maintain reliability and it tries to use as much bandwidth as possible in the network, it enlarges the hidden/exposed node problem and collisions in the MAC layer. Thus, TCP traffic in multihop networks suffers from severe unfairness.

Gerla *et al.* [11, 36] also showed that TCP over multihop wireless connection

has serious performance limitations. A single TCP connections provides good throughput only if the window is properly tuned. Several TCP connections in multihop ad hoc networks tend to lock each other out over long periods of time. Kanth *et al.* [13] proposed a modified backoff scheme in order to enhance TCP fairness in ad hoc networks. When a transmission fails, they inserted a delay for before retransmitting an RTS with the intention of improving throughput and fairness. Similarly, ElRakabawy *et al.* [37] proposed a pacing scheme in which the sending rate of TCP packets is adjusted by inserting a delay.

Chen *et al.* [38, 39] proposed an adaptive congestion window limit strategy to achieve optimal performance of TCP flows in ad hoc networks. They adjusted TCP's congestion window limit according to the round trip hop count (RTHC) of its path. By setting the congestion window limit as 1/5 RTHC, the TCP throughput can be improved.

Papanastasiou *et al.* [40, 41] proposed a TCP variant which slowly increases the sending rate in the congestion avoidance phase, which leads to improvement of TCP performance reducing the negative effects of interference from hidden nodes.

Gerla *et al.* [42] proposed a Neighborhood RED (NRED) scheme, which extends the random early detection (RED) concept to the distributed neighborhood queue. In the NRED, a mobile station drops TCP packets as the RED when the station received a congestion notification from its neighborhood. The NRED scheme can improve TCP unfairness in ad hoc networks without MAC protocol modifications. However, the NRED scheme needs a control message for a congestion notification, and it reduces the aggregate throughput of the network for the sake of enhancing fairness.

Fu *et al.* [14, 15] showed that TCP achieves best throughput with a certain TCP

window size under a specific network topology and flow patterns. They also showed that TCP does not operate around the optimal point in multihop networks and this leads to packet drops and reduces throughput. In order to keep the TCP window size around the optimal size, they proposed Link RED (LRED) and adaptive pacing. The LRED tunes the drop probability at the link level by marking packets according to the average number of retransmissions experienced in the previous packets. The LRED thus provides TCP with an early sign of overload at link level. Adaptive spacing is introduced to improve spatial channel reuse, reduce the risk of stations' starvation, and mitigate the exposed node problem. With LRED and adaptive pacing, TCP throughput increased by 5 to 30 percent.

Nahm *et al.* [16] showed that throughput degradation is more severe when a dynamic routing protocol such as Dynamic Source Routing (DSR) [43] is used. When a packet is dropped due to hidden node problem, the routing failure message is delivered to the routing agent. Then, the DSR floods the network with a route request packet to find a new routing path, which makes the network more loaded and induces more packet drops. This vicious circle degrades the throughput of TCP flows in multihop ad hoc networks. In order to resolve this problem, they proposed a method in which TCP sender slowly increases its congestion window similarly with a method in [40, 41].

Most previous studies presented that a large congestion window of TCP flow makes network overloaded degrading the throughput and fairness in multihop ad hoc network, and most of the previous solutions keep the TCP congestion window small in order to enhance throughput and fairness. Considering this property of ad hoc network, this dissertation proposes a method that enhances the fairness

and throughput in ad hoc networks.

1.3 Organization of the Dissertation

This dissertation is organized as follows. In Chapter 2, the problems related to TCP flows in 802.11 access networks are discussed, and two schemes for per-station fairness in infrastructure WLANs are proposed. In order to provide per-station fairness regardless of the number and direction of flows, the rate of each mobile station is controlled by either dropping or marking TCP packets. The drop probability is calculated based on the channel access time of each mobile station so that each station can have fair access time. The proposed schemes are applied to the IEEE 802.11 and 802.11e access networks, and their properties and effectiveness is evaluated through ns2 simulations.

Chapter 3 is about fairness in ad hoc networks. The unfairness problem of TCP flows in ad hoc networks are presented, and in order to resolve unfairness and reduce the effect of interference from hidden nodes, a capacity allocation scheme based on channel access time is proposed. The effectiveness of the proposed method is shown through simulations under various topologies.

Finally, Chapter 4 contains concluding remarks and directions for future research.

Chapter 2

Fairness in Infrastructure WLANs

In this chapter, fairness among stations, each of which has TCP flows in an infrastructure WLAN, is discussed. In infrastructure WLANs, the packet drops in an AP induce unfairness between uplink and downlink flows. In addition, there exists unfairness among stations which have a different number of flows. These problems are explained in Section 2.1. In Section 2.2 and Section 2.3 the Access Time Control (ATC) scheme is proposed for IEEE 802.11 and 802.11e access network respectively. In Section 2.4, the Distributed Access Time Control (DATC) scheme, which can be implemented in mobile stations without coordination of the AP, is proposed.

2.1 Problem Statement

Figure 2.1 shows an infrastructure WLAN composed of wired and wireless networks. Each mobile station (s_i) makes TCP connections with a corresponding

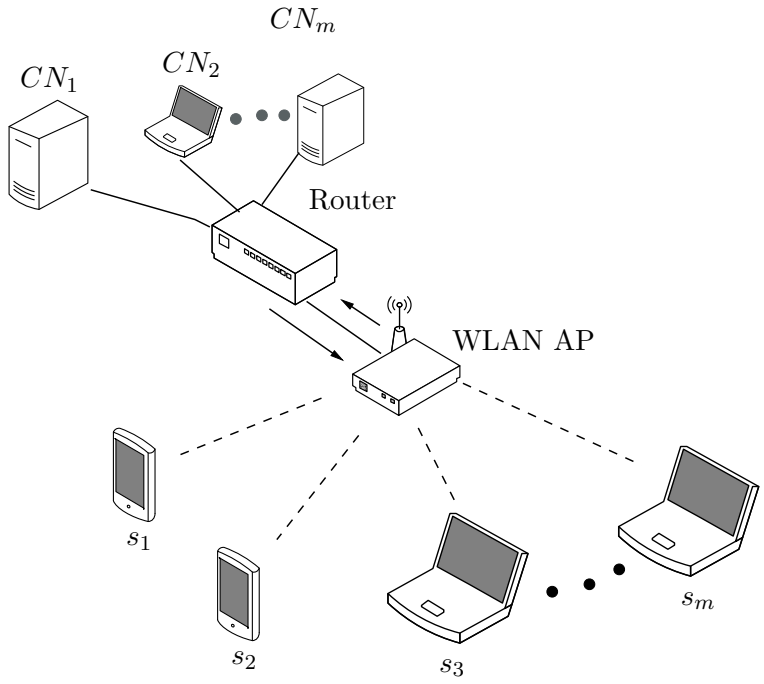


Figure 2.1: WLAN access network

node (CN_i) via a WLAN AP. A WLAN access point receives TCP data frames for downlink flows and TCP ACK packets for uplink TCP flows from a wired network, and these packets are sent to the AP queue. The IEEE 802.11 WLAN can support a bandwidth of up to 11 Mb/s, which is much smaller than that of wired networks, and has to be shared by multiple stations that are served by the same access point. Therefore, traffic congestion tends to occur frequently at the access point, resulting in packet loss due to buffer overflow.

When mobile stations have downlink TCP flows and congestion occurs in the AP queue, their TCP data packets are dropped at the queue, inducing a reduction of the sending rate of TCP senders in the wired network. On the other hand, when the senders are mobile stations, an overflow in the AP queue does not help to reduce the sending rate of TCP flows. The mobile stations do not reduce their

window sizes due to the mechanism of cumulative acknowledgement even when some of the TCP ACK packets from the corresponding nodes are dropped at the AP queue. Therefore, if uplink and downlink TCP flows coexist, the stations having uplink TCP flows tend to use most of bandwidth.

In order to examine this unfairness, a simulation was performed with a WLAN access network, which is composed with ten downlink stations and the various number of uplink stations. In the simulation, each station has a TCP flow under the same condition, and a WLAN AP has a drop-tail queue with a size of 100 packets. Figure 2.2 shows the aggregated throughputs of uplink and downlink. As the number of uplink stations increases from 0 to 10, the downlink throughput decreases drastically. Even when there are four uplink stations, the throughput of ten downlink stations is almost zero. This is because the packet drops in the AP shrink the window size of downlink TCP flows while it does not affect the uplink flows. Since the maximum window size of TCP flow can be 42 packets of 1500 byte each, TCP ACK packets for the uplink flows make the AP queue full when there are three or more uplink flows. Thus, there are few slots for downlink flows in the AP queue, and the incomming packets are dropped.

In addition, even though the directions of flows in mobile stations are the same, fairness among stations is not guaranteed when each station has a different number of flows. Figure 2.3 shows the throughput of six stations when the station s_i has i downlink TCP flows. As a mobile station has more TCP flows, it tends to use more bandwidth. Therefore, in order to assure fairness among mobile stations, it is necessary to take the number as well as the direction of flows into account.

Now, let's consider the fairness in IEEE 802.11e. IEEE 802.11e can provide QoS

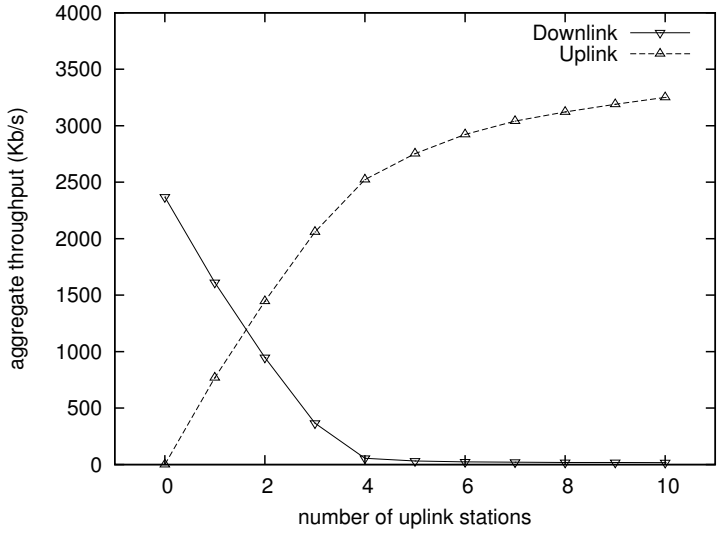


Figure 2.2: Average throughput of uplink and downlink stations under DCF as the number of uplink station increases.

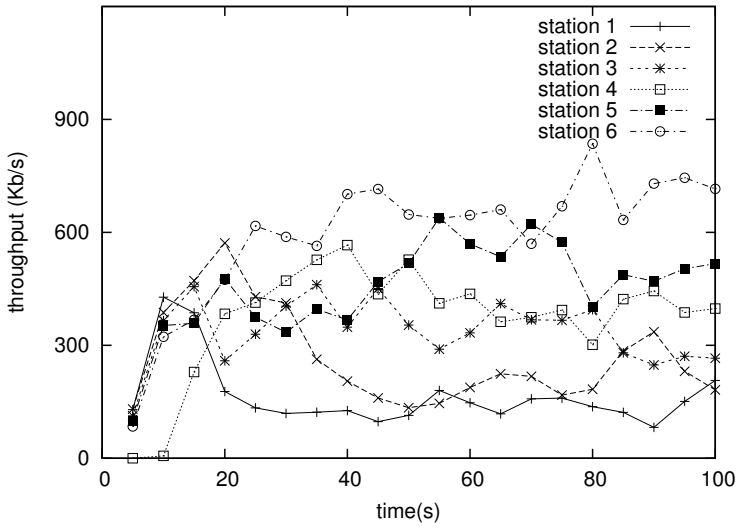


Figure 2.3: Throughput of downlink stations having a different number of TCP flows

for real-time applications such as voice and video. Each station must negotiate with its AP for these QoS traffic and the AP assigns higher priorities to them compared with best-effort traffic. Since best-effort flows can access the networks without any negotiation with the AP, it can induce unfairness among stations as in the 802.11 networks.

Moreover, best-effort traffic may even affect QoS of high priority traffic. Although stations having best-effort traffic use a larger backoff window than the AP having high priority traffic, the chances that the AP will transmit the QoS traffic decreases as the number of stations having best-effort uplink traffic increases.

Therefore, in order to ensure per-station fairness and protect the high priority traffic, it is necessary to control the best effort traffic in the 802.11e networks.

2.2 Access Time Control (ATC) in IEEE 802.11 infrastructure networks

In order to guarantee fairness among station, an AP should control the rate of TCP flows in each station. TCP flow is designed to adjust the flow rate in order to fully utilize the network bandwidth. When a packet loss occurs or a TCP sender receives an ACK packet marked with the ECN bit, it reduces its sending rate by halving the congestion window [44]. By using this congestion control mechanism, an AP can control the incoming rates of TCP packets for downlink flows by either marking or dropping the packets with a certain probability at the AP queue. Likewise, the AP can control the rate of uplink TCP flows by either marking or dropping the outgoing TCP packets. When the corresponding nodes

do not support the ECN mechanism, the ECN marking cannot be used in both uplink and downlink. In this case, dropping both the incoming and outgoing packets can be used to adjust the rates of downlink and uplink flows.

In order to determine the marking or dropping probability of packets at the AP queue, active queue management (AQM) can be used. Among various AQM schemes, the VRC [45, 46] algorithm is used in this dissertation. The VRC algorithm can effectively regulate the queue length $q(t)$ to a desired queue length q_{ref} with a small variation, while maintaining high throughput and small loss rate of packets. The performance of the VRC is robust to changes in the network parameters such as the number of connections, RTT, link capacity, and the buffer size, which is desirable for WLAN environment.

If an AP controls the rate of each TCP flows in both directions, by either dropping or marking packets with the probability calculated by the VRC algorithm, the queue length at AP can be regulated. By dropping or marking the incoming TCP data packets for downlink flows, the number of TCP data packets in the AP queue can be regulated. Similarly, by dropping or marking the outgoing TCP data packets for uplink flows, the number of TCP ACK packets in the AP queue can be adjusted. Therefore, by applying the VRC algorithm, the queue length at the AP can be controlled to prevent overflows which penalize downlink flows, and then fairness between uplink and downlink flows can be guaranteed. However, when stations have a different number of flows, per-station fairness cannot be achieved by applying the same marking probability to all the packets that come to an AP. In this case, all the flows are served at the same rate and a station having a larger number of flows uses more channel capacity.

In order to resolve this problem, the drop or marking probability needs to be

adjusted for each station, i.e., a larger probability should be applied for a station having the more flows. The dropping probability p_i for the packet which comes from or to the station s_i is determined by applying the VRC algorithm.

When the average sending rate $r_i(t)$ of the station s_i exceeds its target rate $r_{t,i}(t)$ for fair share, the station should reduce its flow rate. Thus, the drop probability p_i for the mobile station s_i can be adjusted in order to keep the rate $r_i(t)$ to a given target rate $r_{t,i}$ as

$$p_i(t) = [\alpha(r_i(t) - r_{t,i}(t))]^+, \;\; \alpha > 0, \tag{2.1}$$

where $[\cdot]^+ = max(min(\cdot, 1), 0)$.

However, the rate control mechanism (2.1) cannot ensure that the rate r_i converges to the target rate $r_{t,i}$ at equilibrium when r_i is a decreasing function $f(\cdot)$ of p_i. Figure 2.4 shows the discrepancy when (2.1) is applied to control the rate. The equilibrium rate r_i^* is greater than the target rate $r_{t,i}$, and consequently there is a rate error at equilibrium. In order to compensate for this rate error at equilibrium, a virtual target rate $r_{v,i}$ is used as in [45].

In the VRC, the virtual target rate $r_{v,i}$ is updated to minimize the difference between the average rate r_i and the target rate $r_{t,i}$. Then, the VRC can make the rate r_i converge to the target rate $r_{t,i}$ at equilibrium.

At every sampling period kT, the AP calculates the target rate $r_{t,i}(kT)$ of the station s_i based on the number of active stations at the time kT. When there are $N(kT)$ active stations at time kT and the channel capacity is C, the target rate $r_{t,i}(kT)$ for the station s_i is set as

$$r_{t,i}(kT) = \frac{C}{N(kT)} + \gamma(\frac{q_{ref}}{N(kT)} - q_i(kT)), \; \gamma > 0 \tag{2.2}$$

in order to keep the rate $r_i(kT)$ near its fair bandwidth $C/N(kT)$ and to keep the queue occupancy $q_i(kT)$ of the station s_i close to $\frac{q_{ref}}{N(kT)}$, which is the fair share of

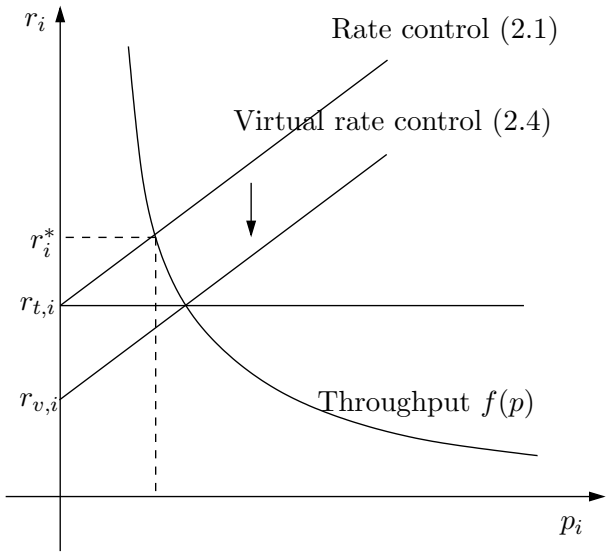


Figure 2.4: Equilibrium points for rate control

the station s_i for the reference queue length q_{ref}. The virtual target rate $r_{v,i}(kT)$ of the station s_i is updated to minimize the difference between $r_i(kT)$ and the target rate $r_{t,i}(kT)$ as

$$\begin{aligned} \Delta r_{v,i}(kT) &= \Delta r_{v,i}((k-1)T) + \beta T(r_i(kT) - r_{t,i}(kT)), \\ r_{v,i}(kT) &= r_{t,i}(kT) - \Delta r_{v,i}(kT), \\ &\text{where, } \beta > 0,\ \Delta r_{v,i}(0) = 0. \end{aligned} \tag{2.3}$$

Using $r_{v,i}(kT)$, at every period, the AP calculates the marking probability $p_i(kT)$ as

$$p_i(kT) = [\alpha(r_i(kT) - r_{v,i}(kT))]^+, \qquad \alpha > 0. \tag{2.4}$$

When $N(kT)$ is constant, it can be proven that the $r_i(kT)$ converges to the target rate by marking or dropping the packets for the station s_i with $p_i(kT)$ as in [45]. Let r_i^* and q_i^* be the target rate and the queue length at equilibrium for

the station s_i, respectively. Then, $r_i^* = \frac{C}{N} + \gamma(\frac{q_{ref}}{N} - q_i^*)$. Let us consider the rate error $e_i(kT) = r_i(kT) - r_i^*$. If the rate of TCP flow can be expressed as $r_i(t) = f(p_i(t))$, we have

$$r_i((k+1)T) - r_i(kT) = \Delta f(p_i(kT)(p_i((k+1)T) - p_i(kT)),$$
$$\Delta f(p_i(kT)) = \frac{f(p_i((k+1)T)) - f(p_i(kT))}{p_i((k+1)T) - p_i(kT)}.$$

Therefore,

$$e_i((k+1)T) - e_i(kT) = \Delta f(p_i(kT))(p_i((k+1)T) - p_i(kT)). \tag{2.5}$$

Using (2.3) and (2.4), $p_i(kT)$ can be written as

$$p_i(kT) = \alpha(r_i(kT) - r_{t,i}(kT)) + \alpha\beta T \sum_{m=1}^{k}(r_i(mT) - r_{t,i}(mT)). \tag{2.6}$$

By substituting (2.6) into (2.5), we have

$$e_i((k+1)T) = \frac{1 - \alpha\Delta f(p_i(kT))}{1 - \alpha(1+\beta T)\Delta f(p_i(kT))} e_i(kT)$$

Since $\Delta f(p_i(kT)) < 0$, $\alpha > 0$, and $\beta > 0$, it can be seen that $0 < \frac{1-\alpha\Delta f(p_i(kT))}{1-\alpha(1+\beta T)\Delta f(p_i(kT))} < 1$ and $e_i(kT)$ converges to zero. Consequently $r_i(kT)$ converges to the target rate r_i^*. Also, from the queue dynamics $q_i((k+1)T) = q_i(kT) + T(r_i(kT) - C/N)$, $r_i(kT)$ becomes C/N at equilibrium and the queue length q_i^* becomes the target queue length q_{ref}/N from (2.2). This scheme makes the AQM system stable, and so, by either dropping or marking with the probability calculated by (2.4), the AP can provide fair bandwidth allocation and can regulate the queue length for each station. We will call this scheme the Access Time Control (ATC).

Now let us study the implementation of the ATC scheme. In an infrastructure WLAN, an AP can easily monitor how long each station has occupied the wireless

channel during each period T. When the AP sends or receives a packet for the station s_i, it calculates the time spent for transmitting the packet and updates the channel access time $t_i(kT)$ during $[(k-1)T, kT]$. Here, $t_i(kT)$ includes the overhead time spent for transmitting data packets such as RTS, CTS, and ACK packets.

Instead of using the target rate in (2.2), let us use the target access time $t_t(kT)$ as

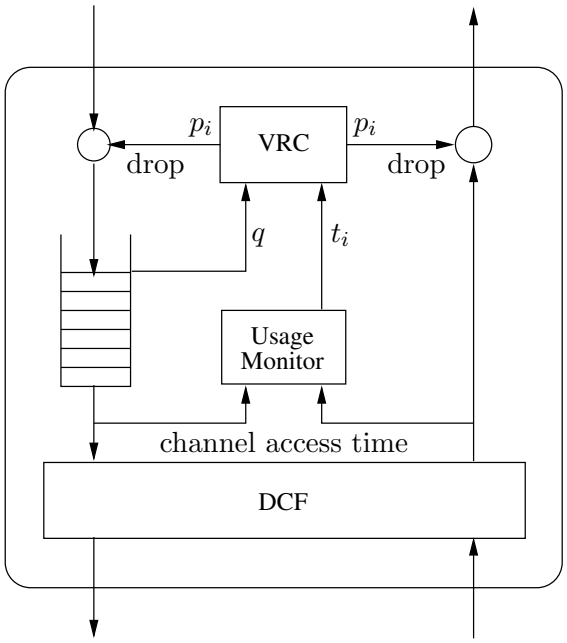
$$t_{t,i}(kT) = \frac{T}{N(kT)} + \frac{\gamma T}{C \cdot N(kT)}(q_{ref} - q(kT))), \tag{2.7}$$

which can be obtained by multiplying (2.2) with $\frac{T}{C}$. Since $q_i(kT)$ becomes $q(kT)/N(kT)$ in steady state, $q(kT)/N(kT)$ is used instead of $q_i(kT)$ for simplicity of implementation. Then, the drop probability $p_i(kT)$ can be rewritten as

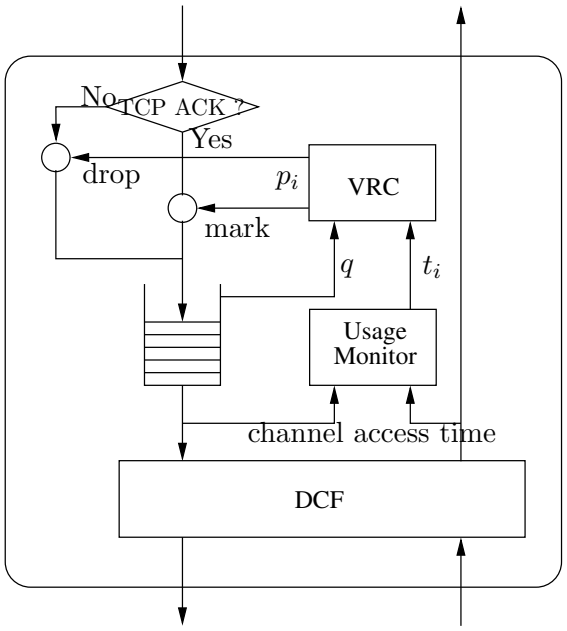
$$\begin{aligned} \Delta t_{v,i}(kT) &= \Delta t_{v,i}((k-1)T) + \beta T(t_i(kT) - t_{t,i}(kT)), \\ t_{v,i}(kT) &= t_{t,i}(kT) - \Delta t_{v,i}(kT), \\ p_i(kT) &= [\alpha \frac{C}{T}(t_i(kT) - t_{v,i}(kT))]^+. \end{aligned} \tag{2.8}$$

Every sampling period T, the AP updates the virtual target rate $t_{v,i}(kT)$ of each station based on the queue length of the AP and the channel access time of the station. With this virtual target rate, the AP calculates the drop probability $p_i(kT)$ for each station s_i, and drops or marks the incoming TCP packets in both uplink and downlink.

Let us consider two structures, ATC-I and ATC-II, of the ATC scheme in Fig. 2.5. Figure 2.5(a) shows the behavior of an AP dropping packets in both directions to adjust the rate of flows in ATC-I scheme. The usage monitor in the AP measures the channel access time t_i that the station s_i uses during each T. When a packet arrives at the AP, the AP looks for the index i of the station



(a) Dropping in both directions (ATC-I)



(b) Dropping for downlink flows and marking for uplink flows (ATC-II)

Figure 2.5: ATC scheme in IEEE 802.11 AP

which the packet is sent to or received from, and then drops the packet with the probability p_i. The information of the index can be easily obtained because all the packets are transmitted or received through the AP, and the AP can identify the station that accesses the channel by observing either the destination or source address of the MAC frame header.

The method of dropping packets in both directions can be simply implemented in an AP and it does not require any modification to the mobile stations. However, dropping outgoing packets results in waste of capacity of wireless channel and may reduce throughput. Thus, in order to control the rates of uplink TCP flows without reducing throughput, the ECN marking for downlink TCP ACK packets can be used locally as shown in Fig. 2.5(b). When a packet for the station s_i arrives, the AP in ATC-II scheme examines whether the packet is a TCP data packet or a TCP ACK packet. If the packet is a TCP data packet, the AP either drops the packet or marks the ECN bit in the IP header of the packet with the probability p_i. If the packet is a TCP ACK packet, the AP marks the ECN bit in the IP header of the TCP ACK packet. The mobile station performs congestion control when it receives ECN-marked TCP ACK packets. It requires some modification in the mobile station, but it can be simply implemented in the protocol stack.

In order to support differentiated service for each station, $t_{t,i}(kT)$ in (2.7) can be modified with the weight W_i for the station s_i as

$$t_{t,i}(kT) = \frac{T \cdot W_i}{\sum_i W_i} + \frac{\gamma T}{C} \frac{W_i}{\sum_i W_i}(q_{ref} - q(kT))).$$

Since the target of each station is proportional to the weight, the channel access time of stations can be allocated proportionally to the weight W_i. The weight can be determined based on the user's profile that is predetermined by the con-

tract between a user and a service provider. When a user connects to an AP, the AP can obtain the user's profile through Authentication, Authorization, and Account (AAA) procedures.

2.3 ATC Scheme in IEEE 802.11e

In this section, the ATC scheme is applied to IEEE 802.11e networks in order to provide QoS for priority traffic and fairness among mobile stations simultaneously. In order to support applications with different QoS requirements, IEEE 802.11e provides the Enhanced Distributed Channel Access (EDCA) mechanism and the concept of Access Category (AC) [17]. In the EDCA, each station has multiple queues for the prioritized service for different ACs, e.g., voice, video, and best-effort flows. Each AC has different QoS parameters such as Arbitration InterFrame Space (AIFS), and minimum/maximum of the contention window (CWmin/CWmax). Using these parameters, APs can support the QoS of voice and video traffic through admission control.

In IEEE 802.11e, an AP performs admission control for the QoS traffic such as voice and video, but it is recommended for the access categories AC_BE and AC_BK not to request admission control. The Admission Control Mandatory Bit (ACM) for these categories should be set to "0". Therefore, mobile stations can produce many best-effort flows without any restriction and these flows can induce unfairness among stations. For example, a greedy user on viewing a video stream, may simultaneously download files through several FTP sessions, and utilizes more capacity than other users. In order to provide per-station fairness in IEEE 802.11e networks, it is necessary to control the access of best-effort traffic.

Since an AP can check the access categories of frames through the QoS control field in the MAC header, it can monitor the channel access time t_i^{QoS} used by the station s_i for the QoS traffic of the access categories AC_VI and AC_VO during each period T. It is assumed that QoS flows with the access categories AC_VI and AC_VO are admitted as long as $t_i^{QoS} < \frac{T}{N}$ is satisfied for N stations. Then, for per-station fairness, the allowable channel access time of the station s_i for best-effort traffic can be derived as

$$t_{t,i}^{BE}(kT) = \begin{cases} 0 & \text{if the station } s_i \text{ has no best-effort traffic,} \\ \frac{T}{N(kT)} - t_i^{QoS}(kT) + \tau + \frac{\gamma T}{C \cdot N(kT)}(q_{ref} - q(kT)) & \\ & \text{if the station } s_i \text{ has best-effort traffic,} \end{cases} \tag{2.9}$$

$$\tau = \frac{\sum_{i \in S_{BE-inactive}} (\frac{T}{N(kT)} - t_i^{QoS}(kT))}{N_{BE}(kT)},$$

where τ is the time which is left by stations having no best-effort traffic and can be shared by the $N_{BE}(t)$ active stations which have active best-effort traffic. $S_{BE-inactive}$ is the index set of the stations that do not have any best-effort traffic.

Even though the ATC scheme in IEEE 802.11e can be implemented in two ways as in IEEE 802.11b, only the implementation that corresponds to ATC-II will be considered. The other scheme can be implemented similarly. Figure 2.6 shows the operation of an AP under the ATC-II scheme in IEEE 802.11e. By either dropping or marking packets in the queue for best-effort traffic with the probability calculated with the target access time in (2.9), the AP controls the rate of each station so that each station has equal channel access time.

Moreover, the ATC scheme can guarantee QoS of traffic in higher access categories without being affected by best-effort traffic. When there are many uplink

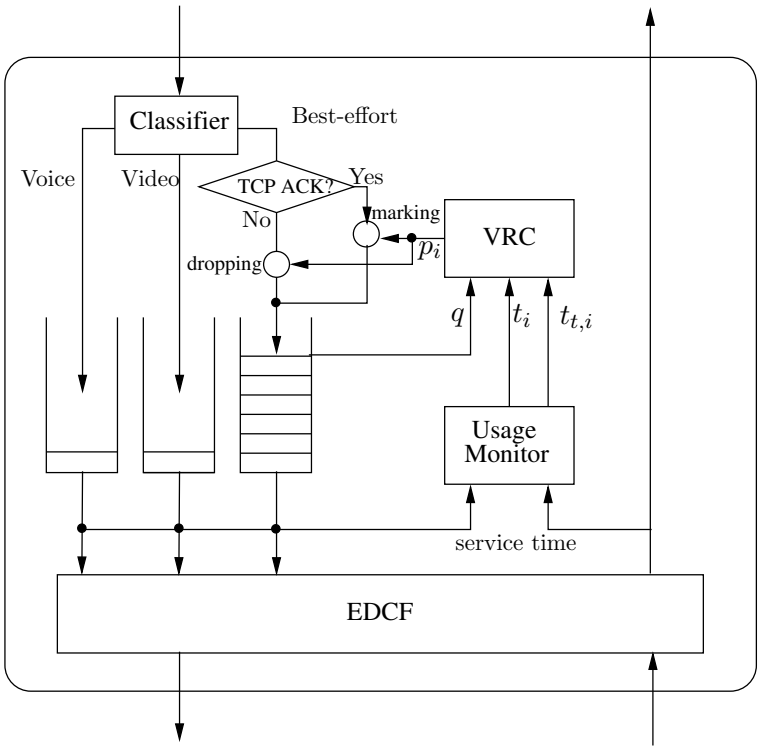


Figure 2.6: ATC scheme in IEEE 802.11e AP (ATC-II)

best-effort traffic, the AP in IEEE 802.11e may fail to support QoS of downlink flows with high priority, because the AP has to contend with other stations having uplink flows. In EDCA of IEEE 802.11e, the QoS of downlink flows having higher priorities can be protected from uplink best-effort flows by adjusting the QoS parameters. When there are many uplink best-effort flows, the AP can increase AIFS and CW_{min} for the access category of best-effort traffic so that the QoS of high priority traffic can be gauranteed. However, when there are non-QoS stations (non-QSTAs) in the 802.11e access network, these stations do not adjust their AIFS and CW_{min} and the QoS of high priority flows can be affected by non-QSTAs. Since the ATC scheme controls the rates of best-effort flows such

that the total amount of channel access time used by these traffic is no more than $\sum t_i^{BE} = T - \sum t_i^{QoS}$ in each T, QoS for the access categories AC_VI and AC_VO can be guaranteed even when there are many non-QSTAs having uplink best-effort traffic.

2.4 Distributed Access Time Control (DATC) Scheme

In the ATC scheme, the AP controls the rate of each station in order to allocate the access time of each station fairly. In this section, a distributed access time control (DATC) scheme, in which each mobile station calculate its drop probability and control its rate by dropping packets instead of the AP, is proposed.

The drop probability p_i which is calculated by the mobile station s_i, is adjusted by the virtual rate control (2.4) similar to the ATC scheme. Since each station cannot know the occupancy of the AP queue, the target access time $t_{t,i}$ of each station is determined as

$$t_{t,i}(kT) = \frac{T}{N(kT)}. \tag{2.10}$$

Except this change, the drop probability for each station is calculated by the same way as in the ATC scheme. In order to calculate the target access time $t_{t,i}$, each station has to measure the number $N(kT)$ of active stations during $[(n-1)T, nT]$, which can be counted by looking at the receiver address of MAC frames transmitted from the AP to mobile stations. Since all stations are within the transmission range of the AP, stations can overhear all packets transmitted from the AP. Most of WLAN cards based on Intersil PrismII chipset provide the

monitor mode, in which a station can examine this information easily [47]. Then, by examining the receiver address field in the MAC frame header, each station can know how many stations are in the network.

There are transmitter and receiver addresses in a RTS message and a receiver addres in a CTS message. And RTS and CTS messages are transmitted at a basic rate, all stations can hear RTS and CTS messages from AP to stations. Therefore, when RTS/CTS messages are exchanged, all the stations can find which station transmits a packet and can count the number of stations correctly. Also there is a receiver address in an ACK message. Even when the RTS/CTS messages are not exchanged, stations can overhear ACK messages from AP to stations because ACK is transmitted at the basic rate. Although there may be stations with different data rates, all the station can find which station transmits a packet by overhearing the ACK messages. For uplink TCP flows, stations can overhear the ACK messages for uplink TCP data packets. Similarly, for downlink TCP flows, stations can overhear the ACK messages for uplink TCP ACK packets. If an AP allows more than one transmission rate as its basic rate, i.e., 1Mb/s and 2Mb/s are used as the basic rates in BSS, stations using lower rate may not overhear ACK frames transmitted from AP to other stations with high rate [1, 48]. Therefore, in order to insure that all stations can overhear ACK frames transmitted from AP to stations, the AP should be configured to use only one basic rate.

In the DATC scheme, each station drops its packet both in downlink and uplink. Figure 2.5 shows the behavior of a mobile station in the DATC scheme. When a mobile station sends or receives a TCP data packet during $[kT, (k+1)T]$, its access time controller drops the packet with the probability $p_i(kT)$ calculated

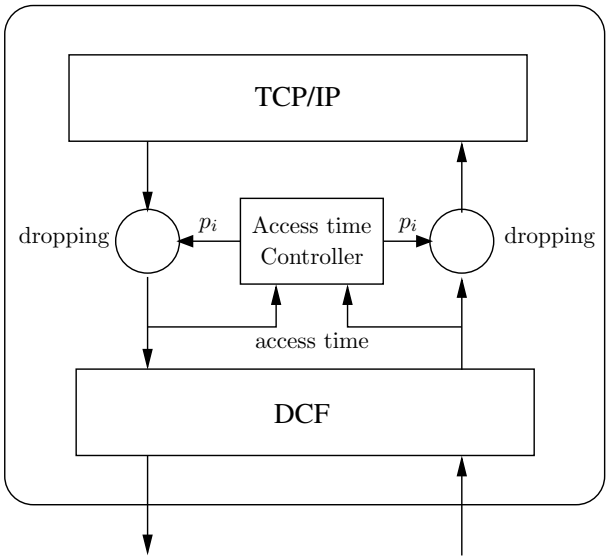


Figure 2.7: Behavior of a mobile station in the DATC scheme

by (2.8). Then, the channel access time of each station converges to the target $t_{t,i}(kT) = \frac{T}{N(kT)}$ so that the channel access time of each station keeps at the same level. Therefore, each station achieves the same throughput and fairness among stations can be assured.

In the DATC scheme, there has been no assumptions on the packet drop mechanisms at the AP. Since the DATC scheme tries to provide the channel access time fairly to each mobile station, most of the packet drop mechanisms at the AP will not affect the performance of DATC. In this dissertation, the AP in the DATC scheme use a drop-tail queue.

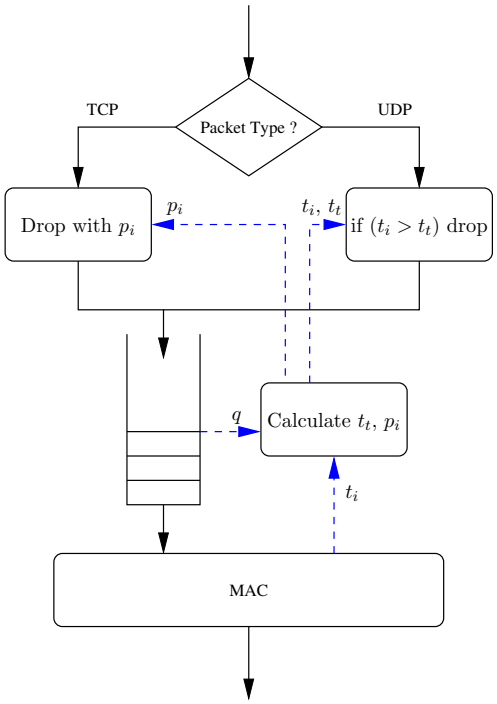
2.5 UDP Flow Control in the ATC and DATC Schemes

Until now, we have considered TCP flows in infrastructure WLAN, and use the TCP congestion control mechanism in order to control the access time of each station. Therefore, when stations have UDP flows, the access time for UDP flows cannot be controlled.

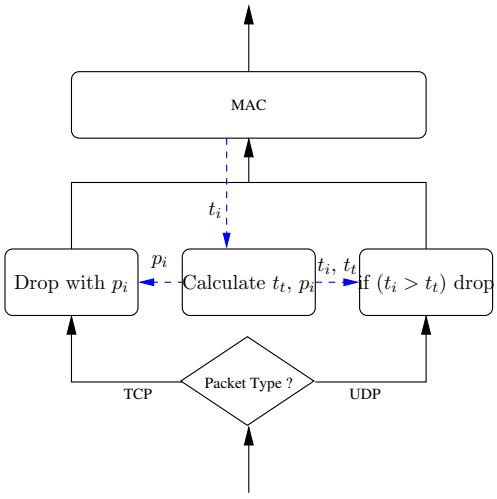
Since an UDP flow does not have a mechanism to control its data rate, the AP and stations have to drop UDP packets in order to control the access time of UDP flows. Figure 2.8 shows methods to control access time of UDP flows in the ATC and DATC schemes. In the ATC scheme, the AP can control downlink UDP flows by dropping incoming UDP packets when the access time t_i of the station using UDP flows exceeds the target access time t_t. Similarly, in the DATC scheme, each station can control its uplink UDP flows by dropping (or delaying) its outgoing UDP packets when its access time t_i exceeds the target access time t_t. Therefore, by applying both the ATC and DATC schemes in an infrastructure WLAN, the access time of each station having UDP flows as well as TCP flows can be allocated fairly.

2.6 Performance Evaluation

In this section, the performance of the ATC and DATC schemes is investigated under various scenarios via the *ns*-2 network simulator [49], and also compare with the schemes proposed in [4] and [5] with respect to throughput and fairness. The latter two schemes will be referred to as Fixed Advertisement Window (FAW) and Limited Uplink Bandwidth (LUB), respectively.



(a) Downlink flow control in an AP (ATC scheme)



(b) Uplink flow control in a station (DATC scheme)

Figure 2.8: UDP flow control in the ATC and DATC schemes

For implementing the ATC scheme, a VRC queue object is implemented and used as an interface queue of the AP in the *ns*-2 network simulator. In addition, the *Usage_Monitor* object is added into the MAC class to measure the channel access time of each mobile station. Using this information, the AP calculates the drop probability p_i for each mobile station s_i. In order to simulate the DATC scheme, the access time controller for each mobile station was implemented in the MAC class of *ns*-2 network simulator. In order to evaluate the performance in IEEE 802.11e, IEEE 802.11e extension for *ns*-2 in [50] was used.

As shown in Fig. 2.1, an access network is composed of one AP and a number of mobile stations, each of which has either an uplink or a downlink TCP session with a corresponding node in a wired network. The AP is connected to a router which is a gateway to fixed nodes. The capacity of each wired link was set to 100 Mb/s, which is much higher than 11 Mb/s, the capacity of IEEE 802.11. Therefore, the wireless channel link became the bottleneck link for the downlink flows. The propagation delay between the router and the AP was set to 20 ms, and those of the other wired links were set to be uniformly distributed between 40 ms and 60 ms.

For the AP, the buffer size of the queue was set to 100 packets, and the target queue length of the ATC scheme was set to 50 packets. For the TCP protocol, the TCP-NewReno was used. The TCP data frame length was set to 1460 bytes, and the advertised window size was set to 42. The parameters of ATC scheme, α, β and γ were set to 0.015, 0.66, and 3.35 respectively, and the parameters of DATC scheme, α and β were set to 0.5, 0.3 respectively, which satisfy the stability condition [46]. The sampling period T was set to 300ms. During this period, either the AP or mobile stations measure the access time t_i of each station and

the number of active stations to calculate the target access time $t_{t,i}$. RTS/CTS was used and the rest of the parameters of IEEE 802.11 and 802.11e were set to the values specified in the *ns*-2 simulator. These parameters and configuration are not changed unless otherwise stated.

2.6.1 Fairness in the IEEE 802.11 infrastructure network

• Fairness between uplink and downlink TCP flows

In order to investigate how the ATC and DATC schemes solve the unfairness problem among uplink and downlink TCP flows, a simulation was performed with ten mobile stations, each of which has either one downlink or one uplink TCP flow. The average throughput per station was examined as the number of stations having an uplink TCP flow increased from 1 to 9. For the ATC scheme, the ATC-II was used only, because the ATC-I scheme gives very similar results. Figure 2.9 shows that IEEE 802.11 induces severe unfairness when some of the stations have uplink TCP flows. In IEEE 802.11 network, frequent packet dropping at the AP keeps the congestion window corresponding to downlink flows small while it has little influence on the uplink flows due to the cumulative acknowledgement. However, in the ATC and DATC schemes, mobile stations share the bandwidth fairly regardless of the number of uplink stations.

• Fairness among stations having a different number of flows

As can be seen in Fig. 2.3, the station having more TCP flows tends to use more bandwidth under the 802.11 DCF. However, the ATC and DATC scheme can provide per-station fairness even when each station has a different number

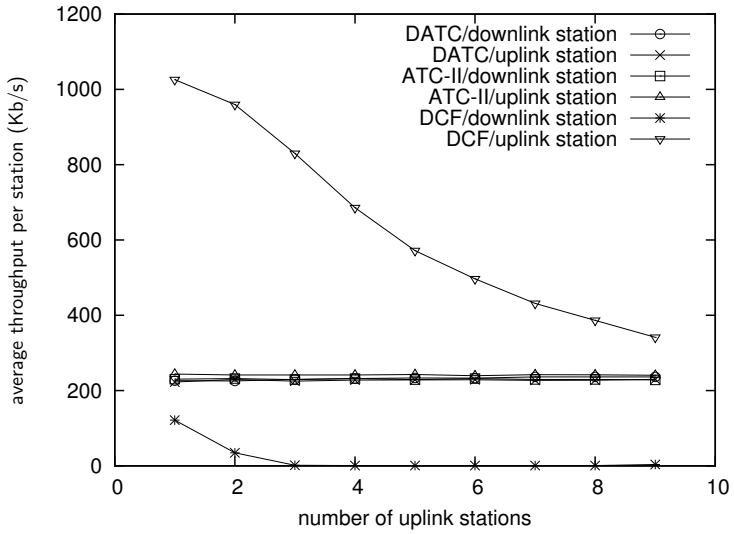
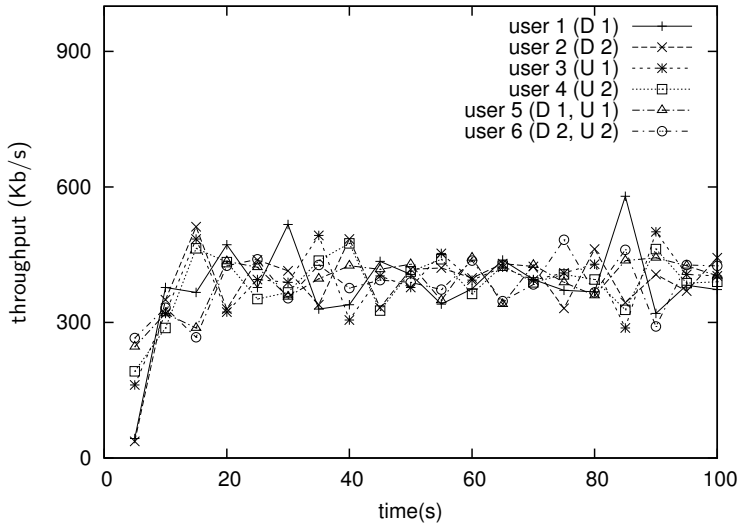


Figure 2.9: Average throughput of uplink and downlink stations

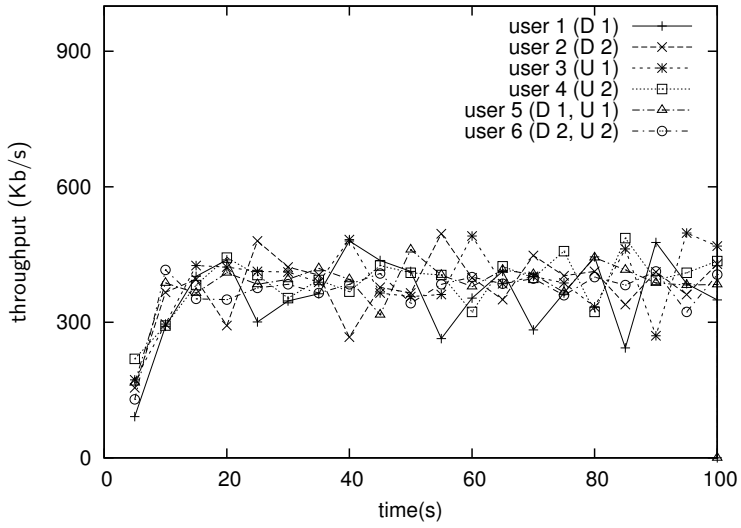
of flows. Figures 2.10 shows the throughput of six stations each of which has a different number of flows in both directions. User 1 and user 2 have one and two downlink flows respectively, while user 3 and user 4 have one and two uplink flows respectively. User 5 has an uplink flow and a downlink flow, and user 6 has two uplink flows and two downlink flows. The ATC-II and DATC schemes can keep the bandwidth of each user at nearly the same level as shown in Fig. 2.10.

- **Weighted bandwidth allocation**

Now, let us check the bandwidth allocation in the ATC scheme when the users have different weights. It is assumed that there are three user classes each of which has the weight W of 1, 2, and 3, respectively. When there are four users in each class, Fig. 2.11 shows the throughput of each user and the Jain's fairness index [51] in every three seconds. The users share the bandwidth nearly proportional to their weights. Also the fairness index with the weight taken into

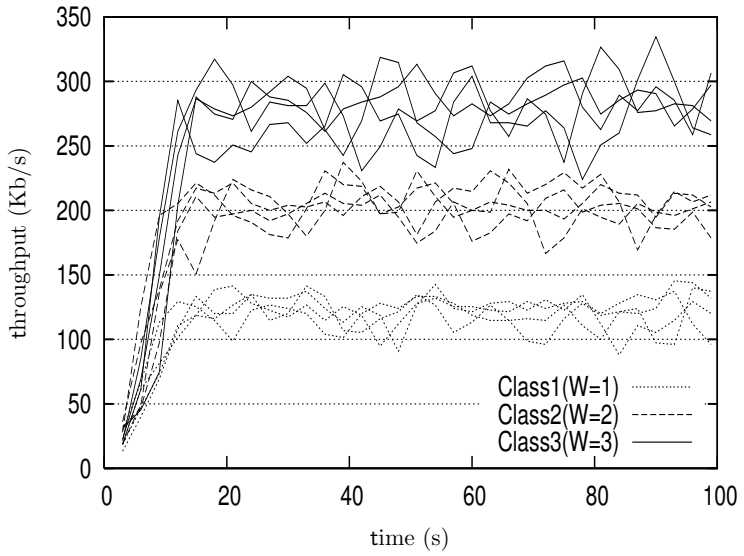


(a) ATC-II

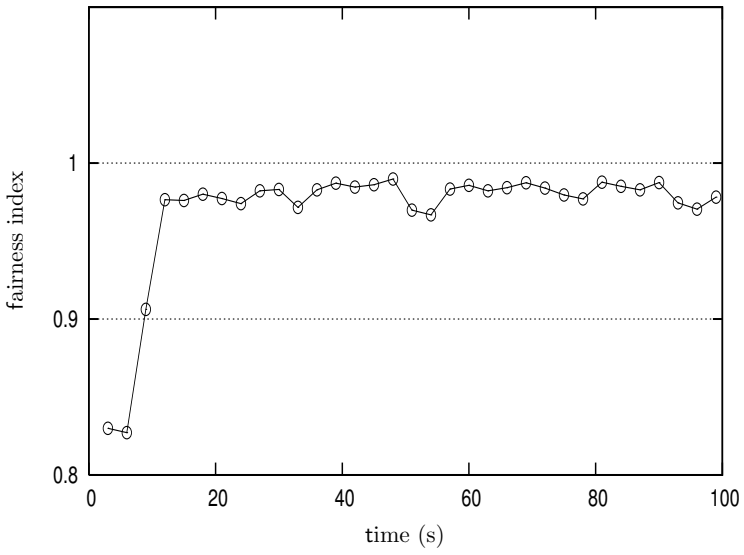


(b) DATC

Figure 2.10: Throughputs when each user has a different number of flows



(a) Throughput of each station



(b) Fairness index in every three seconds

Figure 2.11: Weighted bandwidth allocation

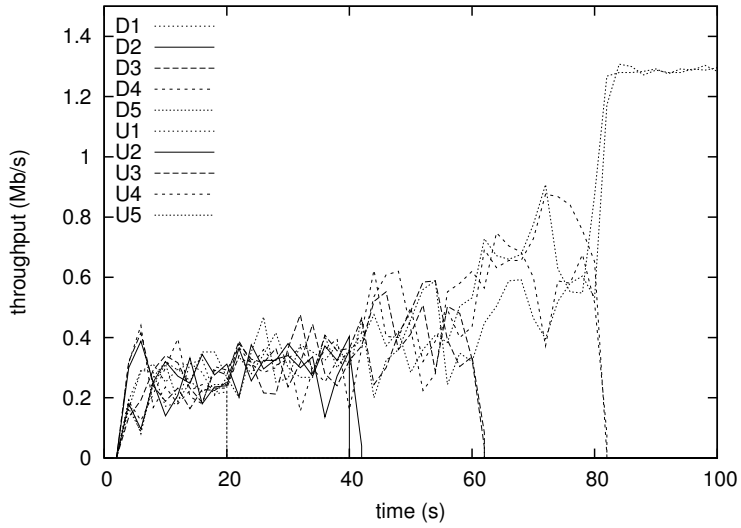
consideration is seen to be close to 1.

- **Adaptation to the change of the number of active station**

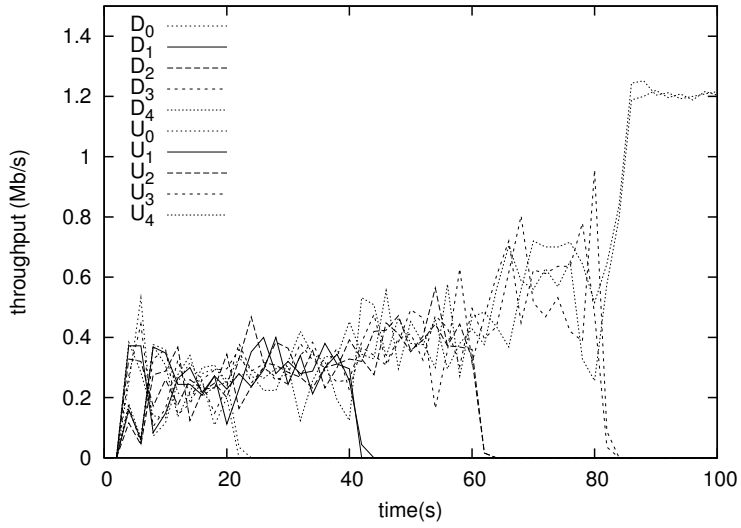
In the ATC and DATC schemes, the target access time of each station is calculated based on the number of active stations. Figure 2.12(a) shows the throughput change of stations as one downlink and one uplink stations are deactivated every twenty seconds after ten stations start TCP sessions, half of which is uplink and the other half is downlink. As the number of active stations decreases, the remaining active stations increases their bandwidth to match the new target so that the bandwidth is shared fairly and is fully utilized.

- **Effect of a large number of stations**

In order to examine the performance under heavy contentions, a simulation was conducted with a large number of stations having an uplink TCP flow each. In the IEEE 802.11, as the number of stations increases from 20 to 120, frequent collisions induces packet losses resulting in retransmission timeout of TCP flows. Therefore, the throughput for some TCP flows which experience several retransmission timeouts, is extremely small in IEEE 802.11, and this causes unfairness among stations as demonstrated by the Jain's fairness index in Fig. 2.13. When there are 120 stations, about 80 stations have very low throughput while the other stations use the most bandwidth, and the aggregated throughput of all stations is reduced by 6%. The aggregated throughput in the ATC-II and DATC schemes is reduced similar to IEEE 802.11. However, the ATC-II and DATC schemes provide good fairness because these schemes control the channel access time of each station, keeping the throughput of each station at the same level (see Fig.



(a) Throughput of each station in ATC-II



(b) Throughput of each station in DATC

Figure 2.12: Adaptation to the change of the number of active stations

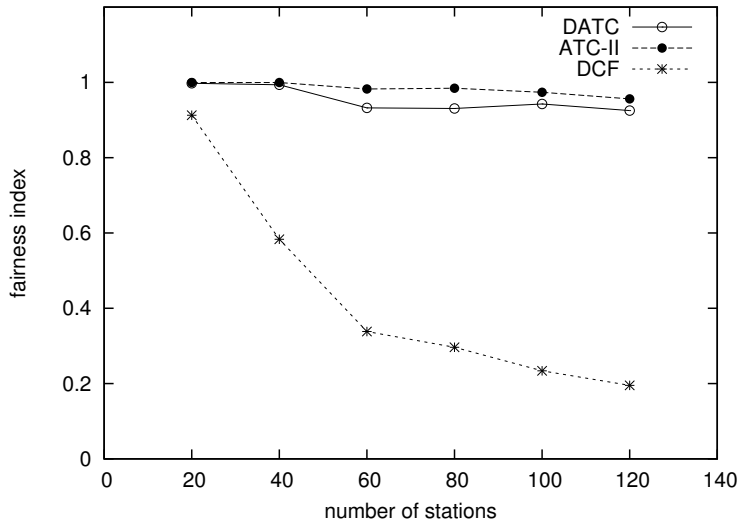


Figure 2.13: Fairness index when there are a large number of uplink stations

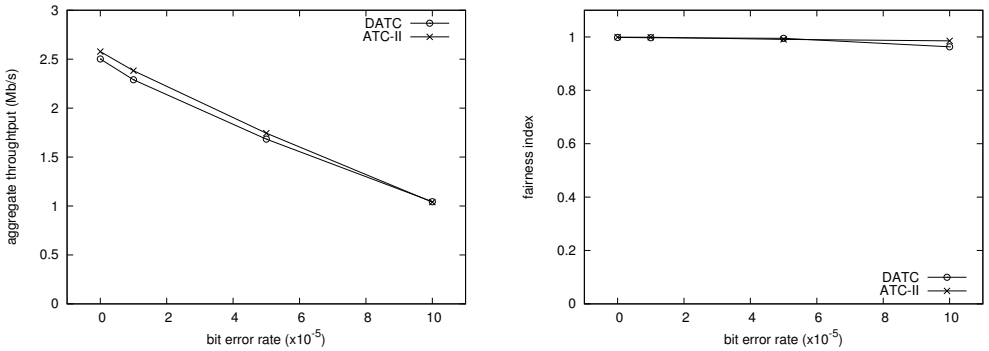
2.13).

- **Effect of wireless transmission errors**

In order to investigate the effects of wireless transmission error on the performance of the ATC-II and DATC schemes, a simulation was performed with ten uplink stations and ten downlink stations. Figure 2.14(a) shows that the total throughput decreases as the bit error rate (BER) increases as expected, but the fairness changes little over a wide range of bit error rates as shown in Fig. 2.14(b).

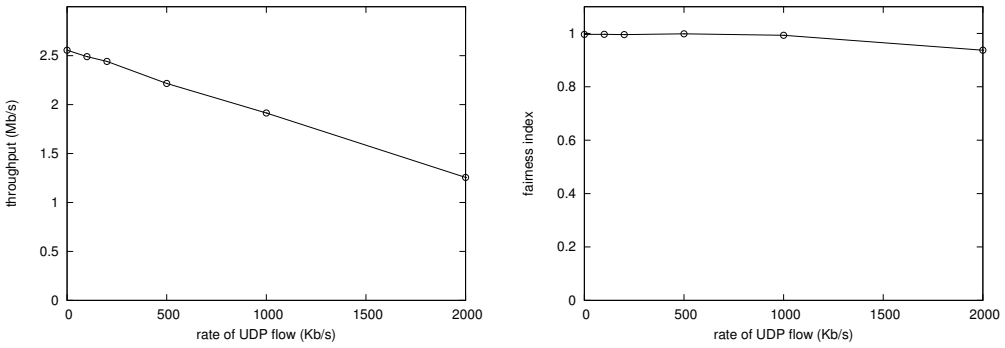
- **Effect of UDP flows**

Here, the effect of UDP flow to the fairness of TCP flows is investigated in the ATC scheme. Unlike the TCP flows, we implemented a policy in which the UDP packets are not dropped except when the buffer overflows. In this simulation, the AP serves thirty stations having one TCP flow each and another station having a



(a) Throughput vs. BER (b) Fairness index vs. BER

Figure 2.14: Effect of bit error rate in the ATC-II and DATC schemes



(a) Aggregate throughput of TCP flows (b) Fairness index of TCP flows

Figure 2.15: Effect of an UDP flow

downlink UDP flow with constant rate. Half of the TCP flows were uplink while the other half were downlink. Figure 2.15 shows the aggregated throughput and fairness index for the thirty stations having TCP flows. As the rate of the UDP flow increases, the throughput of TCP flows decreases as expected. Nevertheless, the ATC-II scheme shows to provide good fairness among stations having TCP flows regardless the rate of the UDP flow. In the ATC scheme, the drop probabilities for both downlink and uplink flows are increased as the UDP pack-

ets fill up the AP queue. Therefore, as the UDP flow rate increases, throughput of both uplink and downlink TCP flows decreases but fairness among stations having TCP flow is kept.

In order to provide per-station fairness for stations having UDP flows as well as TCP flows, the methods shown in Fig. 2.8 can be used. The simulation was performed with 12 stations, the first 5 stations have a downlink TCP flow each, the other 5 stations have an uplink TCP flow each, and the remaining 2 stations have an UDP flow of 500 Kb/s in uplink and downlink respectively. Figure 2.16 shows the access time of each station. Since the uplink UDP flow is controlled by the DATC scheme and the downlink UDP flow is controlled by the ATC scheme, the access time of each station is allocated fairly.

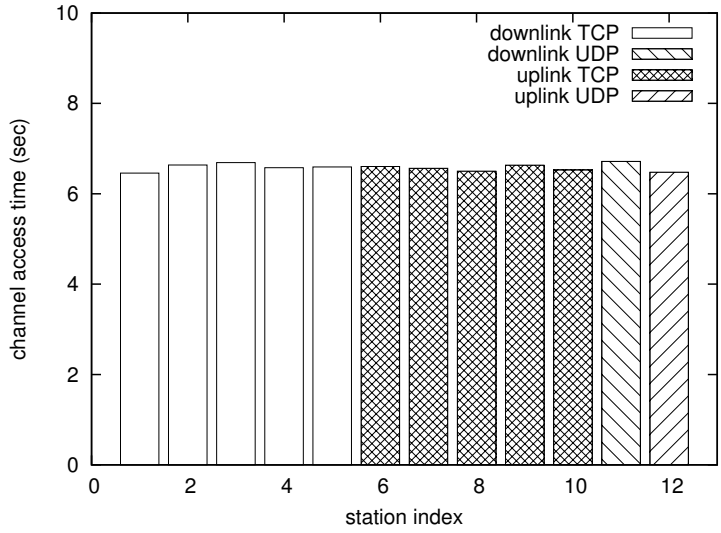


Figure 2.16: Access time of each station when there are UDP flows

- **Effect of multi-rate stations**

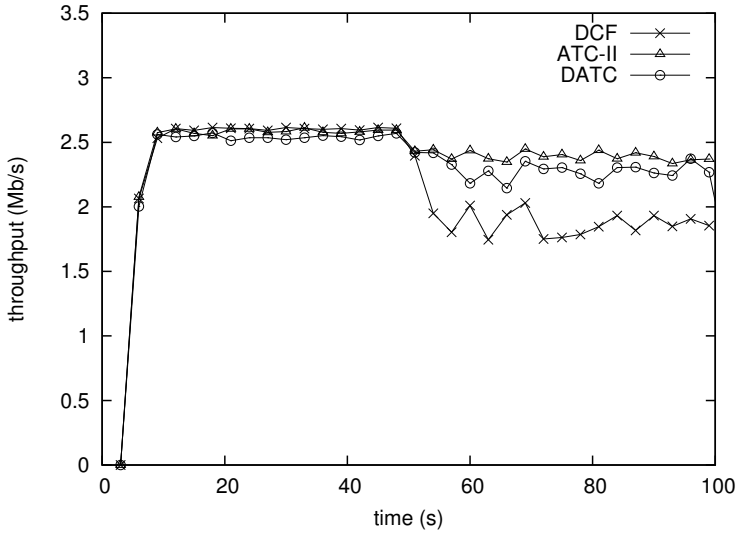
As the proposed schemes allocate fair channel access time to each station, they

can provide per-station fairness regardless of the number and direction of flows. And it can also solve the multi-rate problem that degrades overall throughput when stations have different data rates [52]. Since stations have equal chances to access the channel by the DCF, the station having a lower data rate uses channel longer and penalizes other stations that have a higher data rate. Tan *et al.* [53] proposed a time-based scheduling method using separate queues to resolve the multi-rate problem. Since the proposed schemes provide fair channel access time for each station, they improves the total throughput as Tan [53] without using separate queues.

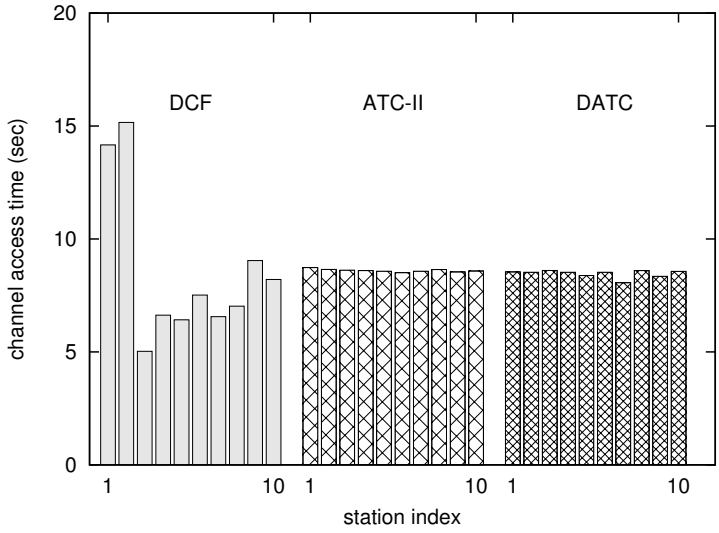
The throughput and access time of stations were examined with ten stations having different data rates coexist. Each station had a downlink TCP flow during the simulation time of 100 seconds. The data rate of two stations changed from 11Mb/s to 2Mb/s at 50 seconds while the data rates of the other stations remained constant at 11Mb/s. Figure 2.17(a) shows the aggregated throughput of all stations. When the two stations send data at a lower rate, the throughput is reduced by 27% in IEEE 802.11 network, while the throughput of the ATC-II and DATC schemes is reduced by 9 to 10%. This is because the access time of each station is kept at the same level in the ATC-II and DATC schemes as shown in Fig. 2.17(b) so that the stations with higher data rate are not penalized by the stations with a lower data rate.

- **Comparison of ATC and DATC with schemes in [4] and [5]**

Now, the performance of ATC and DATC schemes is compared with the schemes of [4] and [5] with respect to throughput and fairness. Although the FAW scheme [4] is impractical to implement, it provides an ideal performance and hence a good



(a) Aggregate throughput



(b) Channel access time of each station

Figure 2.17: Effect of multi-rate stations

Table 2.1: Performances of ATC, DATC, FAW and LUB schemes

Scheme	Total throughput (Mb/s)	Downlink throughput (Mb/s)	Uplink throughput (Mb/s)	Drop rate (%)	Fairness index
FAW scheme	2.56	1.28	1.28	0	0.9999
LUB scheme	2.41	1.20	1.21	5.7	0.9615
ATC-I scheme	2.50	1.29	1.21	4.1	0.9978
ATC-II scheme	2.55	1.27	1.28	2.0	0.9995
DATC scheme	2.49	1.22	1.27	4.4	0.9967

reference for other schemes to compare their performances. Since the ATC and DATC schemes are per-station based scheme while these two solutions are per-flow based schemes, each station was set to have one flow in order to make the comparison fair.

In the simulation, the number of mobile stations was set to 30, and half of these stations had one uplink flow each and the other half had one downlink flow each. The buffer size of the AP was set to 100 packets as in [4] and [5], and q_{ref} was set to 50 for the ATC scheme. For the FAW scheme, the advertise window was set to $\left\lfloor \frac{buffersize}{Num.offlows} \right\rfloor = 3$. The LUB scheme [5] needs to limit the uplink bandwidth to a half of the channel capacity and thus we set this limit to 1.28 Mb/s, which was obtained experimentally. Table 2.1 shows the steady performances of the four schemes based on the data collected for the period that lasts from 20 seconds to 120 seconds in the simulation. Considering that the FAW scheme provides ideal performance, the ATC-II scheme shows good performance. Since both the ATC-I and DATC schemes drop uplink and downlink packets transmitted through wireless channel, each throughput of the two schemes are a little smaller than that

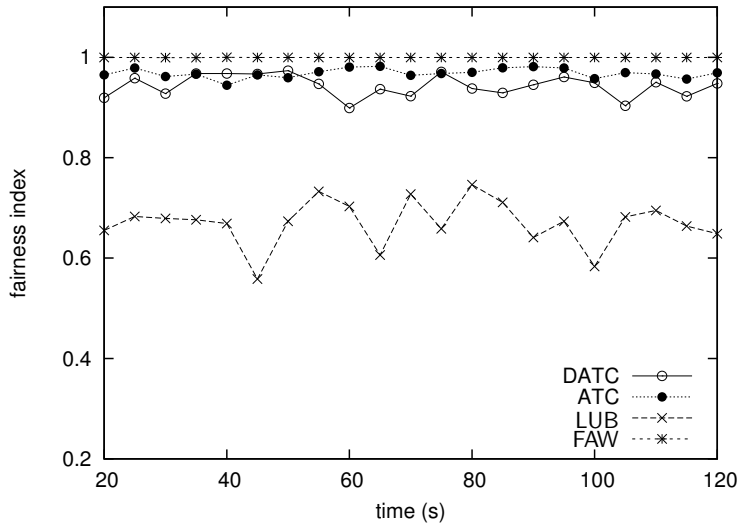


Figure 2.18: Fairness index computed in every five seconds

of the ATC-II scheme. On the other hand, the performance of the LUB scheme falls a little bit behind in terms of throughput and drop rate. Though the ATC-I scheme has a slightly smaller (6%) uplink throughput compared to the ATC-II scheme, both schemes show very similar performance in downlink throughput and fairness. Therefore, only the results of the ATC-II scheme is shown hereafter. The fairness index in Table 2.1 is based on the packets delivered during the simulation time of 100 sec, while the fairness index at time t (sec) in Fig. 2.18 is calculated from the data on the interval $[t-5, t]$. This short term fairness of the ATC-II and DATC schemes is also very close to 1, while the LUB scheme is found to work very poorly.

In order to study the throughput of flows having different RTTs, the network was configured with ten uplink stations and ten downlink stations, and the delays between the corresponding nodes and the router were adjusted. The link delays between the corresponding nodes and the router increased from 20 ms to 200 ms

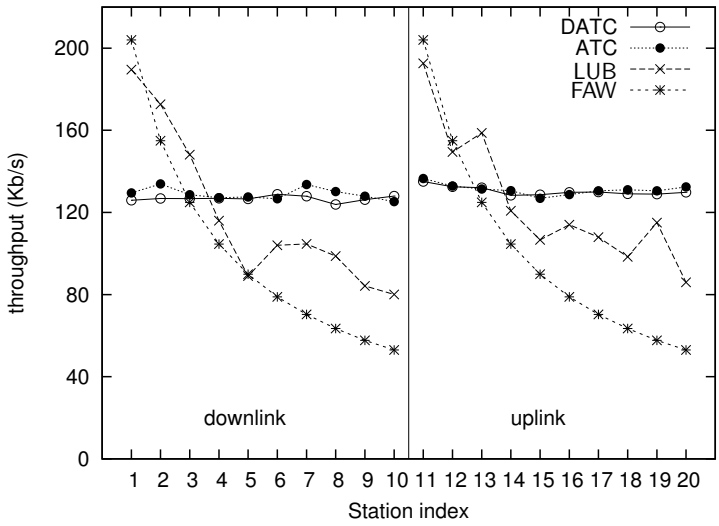


Figure 2.19: Throughput of stations having different RTTs

with a step of 20 ms, so that the difference between the largest and the smallest RTT was 360 ms. Figure 2.19 shows the throughput of the station s_i having a downlink flow with the RTT of $20 \times i + 40$ ms, $i = 1, .., 10$, and uplink flow with the RTT of $20 \times (i - 10) + 40$ ms, i=11,..,20. The throughput of the TCP flows decreases in general as the RTT increases in the FAW and LUB schemes. However, the DATC and ATC schemes ensure fairness among stations with different RTTs because the disadvantage of the flows with longer RTT is compensated by the adjustment of the drop probability so that stations become to have the same channel access time.

In addition, the DATC and ATC schemes keep the same throughput even though the RTTs of the flows become larger. Figure 2.20 shows that the total throughput of different schemes as the RTT is increased from 140ms to 440ms by adjusting delay between the router and the AP. It is seen that the DATC and ATC-II schemes provide high throughput regardless of the delay, while the

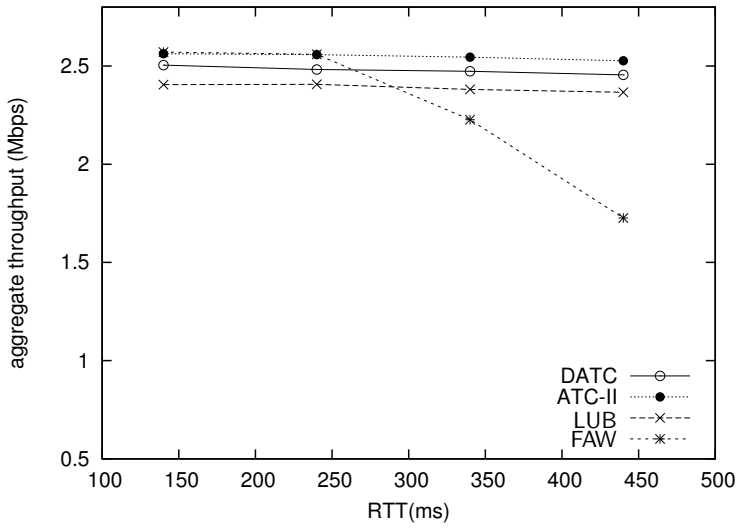


Figure 2.20: Effect of RTT on the total throughput

throughput of the FAW scheme decreases as the RTT exceeds 250 msec. This is due to the fact that the FAW scheme does not fully utilize the wireless channel, because this scheme limits the TCP window size to 3, and the queue becomes empty quite often when the RTT is long.

2.6.2 Fairness in the IEEE 802.11e infrastructure network

In the previous simulations, the performance of the ATC and DATC schemes was investigated in IEEE 802.11 network. Here, the performance of the ATC and DATC schemes in IEEE 802.11e network is studied. In the simulations, voice traffic was generated with burst and idle times that have the Pareto distributions with the mean of 3 seconds and the shape parameter of 1.5. During the burst time, a packet with a size of 92 bytes was generated at 64 Kb/s. For video traffic, a packet was generated at the constant rate of 250 Kb/s.

- **Fairness between uplink and downlink best-effort traffic**

A simulation was conducted using voice, video and data traffic in the 802.11e network configured as follows. The AP served ten stations (s_i, i = 0, 1, .., 9) each of which had one TCP flow for data traffic. Each of the stations, $i = 0, .., 4$, had a downlink flow and each of the stations, $i = 5, .., 9$ had an uplink flow. Each of s_0 and s_5 had an additional bi-directional voice traffic, and each of s_1 and s_6 had an additional downlink video traffic. Figure 2.21 shows the channel access time of each station during the simulation time of 100 seconds. The IEEE 802.11e, the ATC-II, and DATC networks were observed to provide QoS for the voice and video traffic. However, for the data traffic in IEEE 802.11e networks, the stations having downlink flows cannot access the channel while the stations having uplink flows monopolize the bandwidth for their TCP flows. On the other hand, in the ATC-II and DATC networks, the stations have the same channel access time regardless of the direction of data flows. Since the two schemes control the rate of the best-effort traffic so that the stations have the same channel access time, the stations having QoS flows allocate less time for their best-effort data traffic than the other stations having no QoS traffic. The total access time of the stations is about 75 seconds, and the other 25 seconds are used in collisions and backoffs.

- **Effect of uplink best-effort traffic on throughput of video traffic**

In order to investigate the effect of a number of uplink TCP flows on the QoS traffic such as video and voice, a simulation was conducted with ten stations, each of which had an uplink TCP flow. In addition, each of the two stations (s_0, s_1) had a bidirectional voice traffic of 64 Kb/s and each of the eight stations (s_2, .., s_9) had a downlink video traffic of 250 Kb/s. Though not presented in

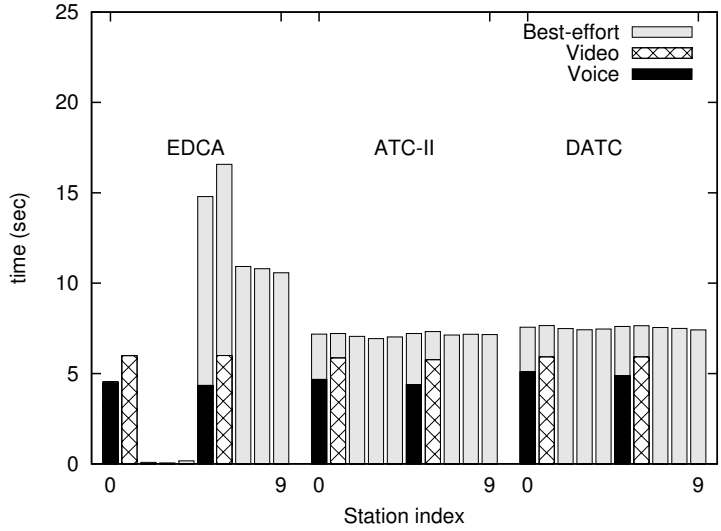
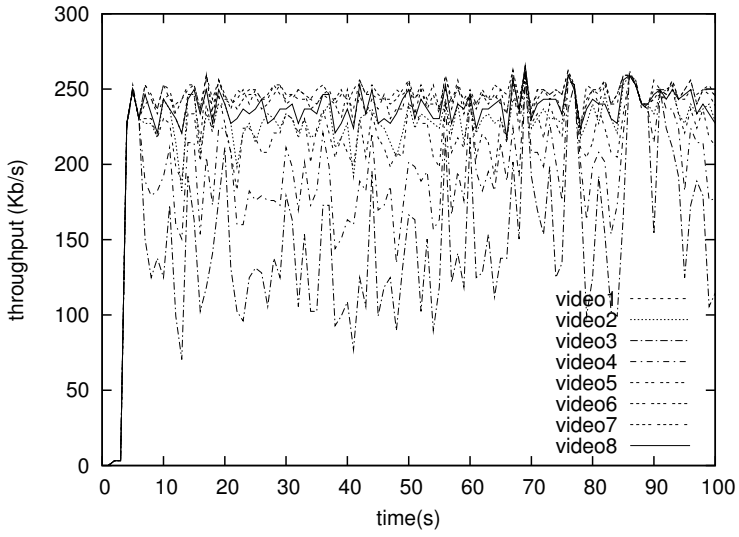


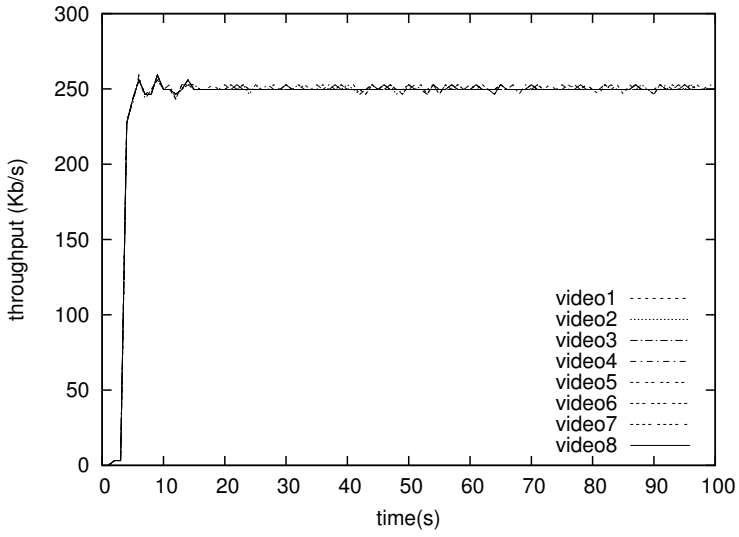
Figure 2.21: Access time used by each station in the 802.11e infrastructure network

the dissertation, the performance of the voice traffic is not affected in both IEEE 802.11e and the ATC scheme. But, the performance of the video traffic, which has a lower priority than the voice traffic, is degraded in IEEE 802.11e network. Figure 2.22 shows that the throughputs of some video traffic are smaller than 250kb/s frequently in IEEE 802.11e, while the ATC-II scheme shows to maintain a near constant throughput level for each video traffic at 250 Kb/s during the simulation time. Although the video traffic has a higher priority than the data traffic, some stations do not allocate sufficient time for the video traffic as shown in Fig. 2.23(a). Since the AP serving the video traffic has to contend with the stations having uplink data flows, the QoS of video traffic tends to be compromised when there are a large number of stations having uplink data flows. Nevertheless, the ATC-II scheme guarantees the QoS of the voice and video traffic by controlling the rate of best-effort data traffic in uplink. (Since the DATC

scheme shows similar results as the ATC-II scheme, the results are omitted.)

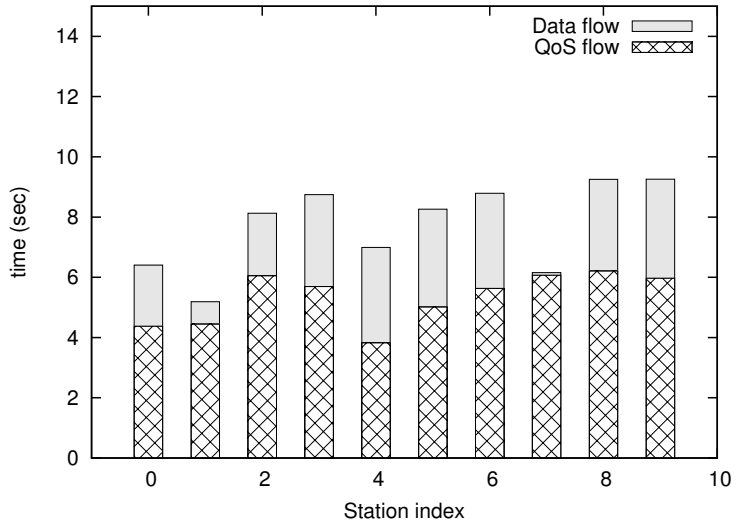


(a) Throughput of each video traffic in IEEE 802.11e network

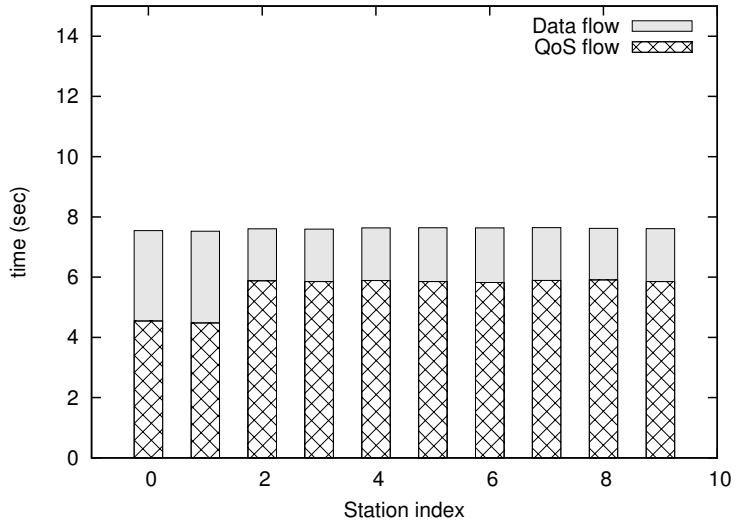


(b) Throughput of each video traffic in the ATC-II scheme

Figure 2.22: Effect of uplink TCP flows on the throughput of video traffic



(a) Access time in IEEE 802.11e network



(b) Access time in the ATC-II scheme

Figure 2.23: Access time of each stations using uplink flows

- **Effect of uplink best-effort traffic on delay of video traffic**

Here, the average end-to-end delay of video traffic was examined as the number of stations having an uplink TCP flow increases. When each of eight stations each of which had a downlink video traffic, the number of stations having an uplink TCP flows was increased from 0 to 6. Figure 2.24 shows that the average delay of video traffic increases as the number of uplink stations increases in IEEE 802.11e network while the ATC-II and DATC schemes keep the delay at the same level regardless of the number of uplink stations.

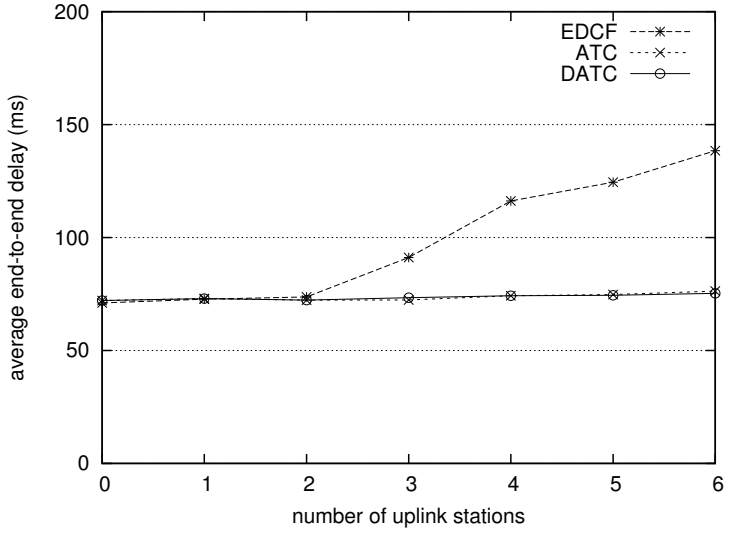


Figure 2.24: Average end-to-end delay of video traffic vs. the number of uplink best-effort flows

Chapter 3

Fairness in Ad Hoc Networks

In the IEEE 802.11 ad hoc networks, stations are able to communicate with each other directly without support of network infrastructure [1]. Therefore, ad hoc networks are used in order to communicate in places where network infrastructure does not exist. However, there is no central device that manages the network, and it is difficult to share network resources fairly in ad hoc networks.

Fairness issues in IEEE 802.11 ad hoc network has been studied in many research studies, most of which have attempted to modify the distributed coordination function (DCF) of IEEE 802.11 at the MAC layer [20–23]. However, these studies did not take the effects of the upper layer protocols (e.g., TCP) on fairness when adjusting priority to access channel. Moreover, they focused only on the fair access among sender stations without taking the number of flows and links into consideration. Since a sender station can communicate with numerous receiver stations, fair access among sender stations can induce unfair bandwidth allocation among the receivers.

In addition, several researchers showed that TCP flows in multihop ad hoc net-

works suffer from severe performance degradation and unfairness due to hidden node problem [9, 11, 12]. A large TCP window can aggravate the hidden node problem and induces more collisions in the MAC layer degrading throughput and fairness. In order to resolve this problem, many researchers have proposed solutions, most of which keep the TCP congestion window small [14–16, 38–40, 42].

In this chapter, a method that can fairly allocate channel access time over each link to enhance fairness among TCP flows, is proposed. Although fairness in a network can be measured based on many different terms such as station, flow, link, and connection, the present study is mainly focused on per-flow fairness. Since it is assumed that there is only one flow for a given source-destination pair throughout the paper, the per-flow fairness is the same as the per source-destination pair fairness.

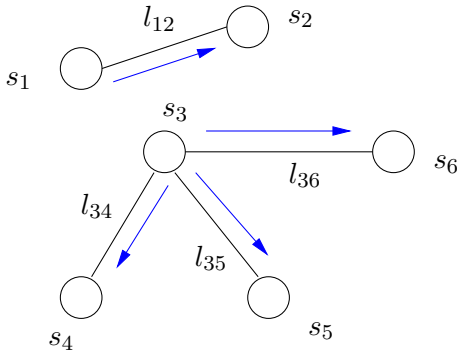
In ad hoc networks, a station can measure the channel access time of its neighborhood stations, and also monitor the interference level by counting the number of transmission failure. Based on these information, each station adjusts its target access time and controls the rate of TCP flows by marking TCP packets in a distributed manner. We assume that the RTS/CTS mechanism is used and it is reasonable for ad hoc networks where collisions are apt to occur due to interference from hidden nodes. Since the proposed scheme does not require any modification to be made to the 802.11 MAC and TCP protocols, it can be easily implemented on the existing devices.

3.1 ATC Scheme in Single-hop Ad Hoc Networks

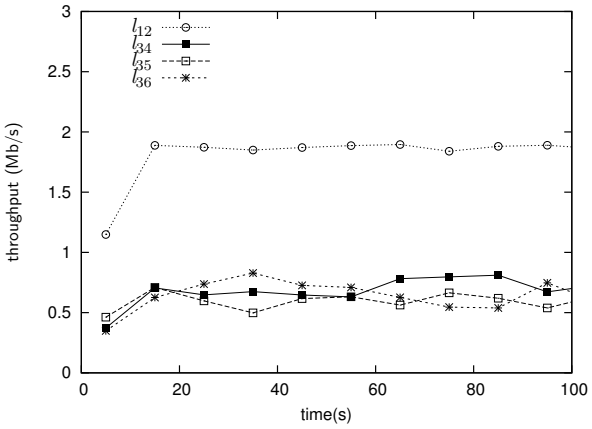
3.1.1 Network model and fairness

The IEEE 802.11 specification defines an independent basic service set (IBSS) as a network composed solely of stations within mutual communication range of each other via the wireless medium [1]. Since stations are able to communicate with each other directly, all packets are delivered through a single hop. A single hop ad hoc network can be represented by a set of stations, $S = \{s_i, i = 0, 1, ..., m\}$ and a set of active links, $L = \{l_{ij}\}$, where l_{ij} represents the active link between the station s_i and station s_j. On the active link l_{ij}, there exist a flow directed from s_i to s_j or from s_j to s_i. We assume that there is only one flow for a given source-destination pair throughout the paper.

Figure 3.1(a) shows a simple example of single hop ad hoc networks consisting of six stations, $S = \{s_1, s_2, s_3, s_4, s_5, s_6\}$ and four active links, $L = \{l_{12}, l_{34}, l_{35}, l_{36}\}$. Among the six stations, two stations (s_1, s_3) send flows to four stations (s_2, s_4, s_5, $and\, s_6$). Since the two senders s_1 and s_3 have an equal chance to send a packet under the IEEE 802.11 DCF, they send almost the same amount of packets if they always have packets to send. Therefore, the station s_2 will receive data about three times more than the other receiver stations. Figure 3.1(b) shows the simulation result for the average throughput for each link when there exists a TCP flow in each link of Fig. 3.1(a). As expected, the average throughput of TCP flow from s_1 to s_2 is larger than those of the other flows by about three times. Since the IEEE 802.11 DCF only provides equal chance for sender stations to access channel, it does not guarantee fair service for each flow. Thus, to assure fair service to each flow, each link, over which each flow is transmitted, should use the same amount of channel capacity.



(a) An example of single hop ad hoc networks



(b) Average TCP throughput for the links of Fig. 3.1(a) (transmission rate: 11Mb/s)

Figure 3.1: A single hop ad hoc network

Moreover, when links have different data rates, the channel capacity may not be shared fairly under the IEEE 802.11 DCF, leading throughput degradation. In the IEEE 802.11b, the data rate can vary from 1 to 11 Mb/s depending on the channel condition. When data is transmitted over a 2Mb/s link, it takes 5.5 times longer time than that transmitted over a 11Mb/s link. Each station has equal chance to transmit a packet under the DCF regardless of the transmission rate,

and so the transmission over a link with a lower data rate will use more channel capacity, degrading total throughput and fairness with respect to channel access time for each flow [52–54]. In order to prevent such a situation from occurring, it is desirable to allocate the channel access time, not the chance to access the channel, equally to each flow regardless of the transmission rate. Based on the above reasons, it is desirable to allocate the same amount of channel access time during a given period to each flow. For this, we propose an access time control (ATC) scheme as described in the next subsection.

3.1.2 Access time control for per-flow fairness

When a packet is transmitted, both sender and receiver stations can know which link is used, and how long the transmission will take. When station s_i receives/sends packets from/to station s_j, both s_i and s_j stations can measure the time $t_{ij}(kT)$ taken for transmitting the packets over the link l_{ij} during the k-th period $[(k-1)T, kT]$, including the overhead time for sending RTS/CTS and ACK packets. In addition, by overhearing the packets transmitted by its neighbor stations and examining the address fields and duration field in the MAC header, a station can also know which link is activated and how long the transmission takes. Since RTS and CTS frames are transmitted at a basic rate, stations can obtain these information even when each station uses a different data rate.

For example, the station s_1 in Fig. 3.1(a) can measure the time $t_{12}(kT)$ taken over the link l_{12} during the k-th period. In addition, since all stations are within the transmission range of each other in a single hop network, the station s_1 can measure the time $t_{34}(kT)$, $t_{35}(kT)$, and $t_{36}(kT)$ by overhearing the packets transmitted by the stations s_3, s_4, s_5, and s_6.

Based on the channel access time over the links and the number of the active links counted during a period $[(k-1)T, kT]$, each station determines its target access time for the next period. At each period kT, station s_i calculates its target access time $t^i_{t,ij}$ for its flow over the link l_{ij} as follows:

$$t^i_{t,ij}(kT) = \frac{1}{n(\mathcal{L})}\left(t_{idle}(kT) + \sum_{l_{mn}\in\mathcal{L}, m<n} t_{mn}(kT)\right), \tag{3.1}$$

where $\mathcal{L}$ is the set of all active links, and $t_{idle}(kT)$ is the idle time during $[(k-1)T, kT]$ which will be zero when the network is saturated.

In order to keep the access time over each link to the target value, each station controls the rate of its TCP flows. TCP flow is designed to adjust the flow rate in order to fully utilize the network bandwidth. When a packet is lost or a TCP sender receives an ACK packet marked with explicit congestion notification (ECN), it reduces its sending rate by halving its congestion window [44]. By using this congestion control mechanism, we can control the rate of a TCP flow by either dropping or marking packets with a certain probability.

In single hop ad hoc networks, there is no intermediate node that plays the role of a router in wired networks. Therefore, in order to adjust the rate of a TCP flow, both sender and receiver stations have to mark or drop packets according to their targets. We use the ECN marking instead of dropping packets in this paper. If we adopt packet dropping, only the network throughput will be reduced slightly.

A schematic diagram of the ATC scheme is shown in Fig. 3.2. A station that has a packet to send, marks the packet with a certain probability to control the access time over the link through which the packet will be delivered. Similarly, when a station receives a packet, it marks the packet with a certain probability to control the access time over the link through which the packet was delivered. For example, the stations s_1 and s_2 in Fig. 3.1(a) can mark the packet to control

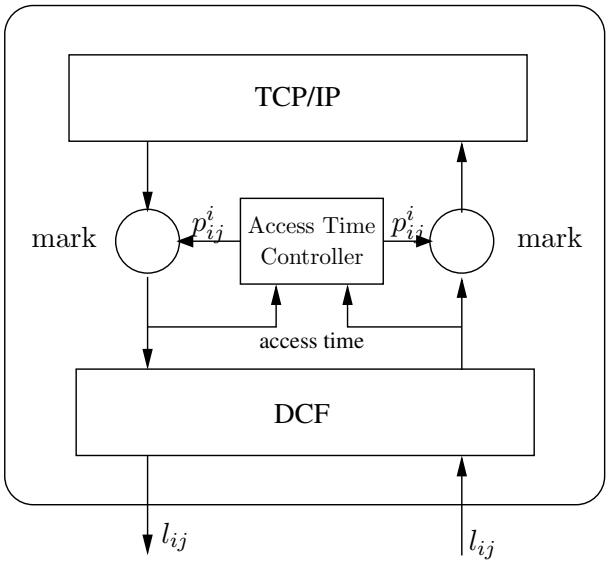


Figure 3.2: ATC scheme implemented in station s_i in ad hoc networks

the access time of the link l_{12}, when the station s_1 (s_2) sends (receives) a packet to (from) the station s_2 (s_1).

Now, let us consider how the station s_i calculates the marking probability $p_{ij}^i(kT)$ at the time kT, with which the station s_i adjust its access time over the link l_{ij} by marking the packets during $[kT,\ (k+1)T]$. Similar to the ATC and DATC schemes in an infrastructure WLAN, the station s_i calculates the marking probability $p_{ij}^i(kT)$ by applying the VRC algorithm as follows:

$$\begin{aligned} t_{v,ij}^i(kT) &= t_{t,ij}^i(kT) - \Delta t_{v,ij}^i(kT), \\ \Delta t_{v,ij}^i(kT) &= \Delta t_{v,ij}^i((k-1)T) + \beta T(t_{ij}(kT) - t_{t,ij}^i(kT)), \\ p_{ij}^i(kT) &= [\alpha(t_{ij}(kT) - t_{v,ij}^i(kT))]^+ \\ &\text{where } \Delta t_{v,ij}^i(0) = 0,\ \beta > 0. \end{aligned} \tag{3.2}$$

Assuming $p^i_{ij}(kT)$ is not saturated, i.e., $p^i_{ij}(kT) = \alpha(t_{ij}(kT) - t^i_{v,ij}(kT))$, $p^i_{ij}(kT)$ can be written from (3.2), as

$$p^i_{ij}(kT) = K_P(t_{ij}(kT) - t^i_{t,ij}(kT)) + K_I \sum_{k=1}^{n}(t_{ij}(kT) - t^i_{t,ij}(kT)), \quad \text{where } K_P = \alpha,\ K_I = \alpha\beta T \tag{3.3}$$

which is a typical PI control. When the target rate $t^i_{t,ij}(kT)$ is a constant, it can be proven that the $t_{ij}(kT)$ converges to the target access time $t^i_{t,ij}$ by marking packets with $p^i_{ij}(kT)$ in (3.2).

Figure 3.3 summarizes the behavior of access time controller in a station. During a sampling period T, each station monitors which link is activated by checking transmitter and receiver addresses of a transmitted packet. In addition, using the NAV field in the MAC header, each station measures access times over all the active links in the network. Some of WLAN cards provide the monitor mode, in which a station can examine these information easily [47]. Using these information, each station calculates its target access time $t_{t,ij}$ and updates the marking probabilities of its links. When a packet arrives from the TCP layer, the access time controller examines the receiver address and finds out which link will be used for sending the packet. With the calculated marking probability for the link, the access time controller marks the ECN bit in the IP header of the packet and sends it to the MAC layer. Similarly, when a packet is received from the MAC layer, the access time controller marks the packet with the marking probability of the corresponding link. Since the ATC controller works between the TCP and MAC layers and it does not require any modification in the MAC or TCP layers, it can be easily implemented in the logical link control (LLC) layer.

At every period T in each station s_i

count the number of active links

for all l_{ij} that belongs to s_i **do**

$t_{t,ij} \leftarrow \frac{1}{n(\mathcal{L})}(t_{idle} + \sum_{l_{mn} \in \mathcal{L},\, m<n} t_{mn})$

$error \leftarrow t_{ij} - t_{t,ij}$

$error\_sum_{ij} \leftarrow error\_sum_{ij} + error$

$p_{ij} \leftarrow K_P \cdot error + K_I \cdot error\_sum_{ij}$

end for

for all t_{ij} **do**

$t_{ij} \leftarrow 0$

end for

When receiving a packet over a link l_{ij}

calculate the time t_{pkt} taken in receiving the packet

$t_{ij} \leftarrow t_{ij} + t_{pkt}$

if the packet belongs to itself **then**

mark the packet with the probability p_{ij} and send the packet to the TCP layer

end if

When sending a packet over a link l_{ij}

calculate t_{pkt} taken in transmitting the packet

$t_{ij} \leftarrow t_{ij} + t_{pkt}$

mark the packet with the probability p_{ij} and send the packet to the MAC layer

Figure 3.3: The operation of access time controller

3.2 ATC Scheme in Multihop Ad Hoc Networks

A well-known problem of multihop ad hoc wireless networks is the hidden node problem. Nodes beyond the carrier sensing range of each other may try to send a frame at the same time, causing collisions at the receiver. TCP flows are much affected by the hidden node problem. TCP throughput often decreases dramatically with the number of hops traversed by a flow. This problem is particularly significant when the TCP window size is large, and many packets are in flight at a time [9–15, 42]. Throughput degradation is more severe when a dynamic routing protocol such as Dynamic Source Routing (DSR) [43] is used [16].
In this section, the ATC scheme is extended in order to enhance fairness and reduce the effect of interference from hidden nodes in multihop ad hoc networks.

3.2.1 Providing per-flow fairness in multihop ad hoc networks

In multihop ad hoc networks, the limited radio transmission range of each station often requires packets to traverse multiple intermediate nodes from a source to a destination. In addition to play the roles of a sender and a receiver, each station needs to function as a store-and-forward router for packets originating from other sources. Therefore, unlike a single hop ad hoc network, the number of flows over the links should be also considered in multihop ad hoc network. While only one link is used for each flow in a single hop network, a link in multihop ad hoc networks can be used by many flows. Therefore, allocating the same channel capacity to each link is not enough to assure fairness among flows in multihop ad hoc networks. For example, let us assume the link l_{45} in Fig. 3.4 is used for delivering packets for two flows f_1 and f_2 which belong to different source-destination pairs. If the access time over the links in the same transmission range

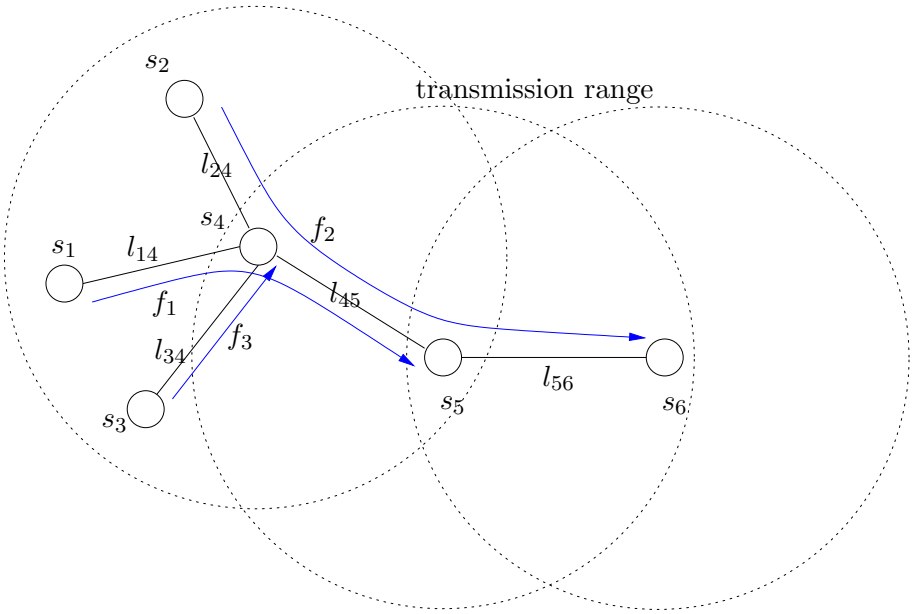


Figure 3.4: An example of multihop ad hoc networks

is allocated equally, then it may not be good for per-flow fairness. In such a case, each of the bandwidth allocated to the two flows passing the link l_{45} in Fig. 3.4 will be lower than that of f_3. Therefore, in order to provide fair service to each flow, it is necessary to consider the number of flows over each link. Therefore, in order to provide fairness among the three flows in Fig. 3.4, the access time should be allocated to the links l_{14}, l_{24}, l_{34}, l_{45}, and l_{56} in the ratio of 1:1:1:2:1, that is proportional to the number of flows over the links.

The IEEE 802.11 MAC header includes four address fields (destination, source, receiver, and transmitter addresses). By checking the transmitter and receiver address fields in the MAC header, a station can find out which link is activated. In addition, the source and destination address fields in the MAC header of data frame can be used to identify a flow. Here, we assume the data rate of each station is the same in multihop ad hoc networks so that each station can

overhear data frames transmitted by senders in its transmission range and identify flows by examining the source and address fields in the MAC header of data frames. During a sampling period T, each station measures the access time over its neighbor links as described in Section 3.1 and counts the number of flows over the links. For example, in Fig. 3.4, the station s_4 can measure the access time $t_{14}(kT)$, $t_{24}(kT)$, $t_{34}(kT)$, and $t_{45}(kT)$ during the k-th period kT. In addition, since the station s_4 is within the transmission range of the station s_5, the station s_4 can measure the time $t_{56}(kT)$ by overhearing the RTS and data frames sent by the station s_5. In addition, by examining the address fields in the MAC header, the station s_4 can count the number of flows on its neighbor links, i.e., $\mathcal{N}_{14} = \mathcal{N}_{24} = \mathcal{N}_{34} = \mathcal{N}_{56} = 1$ and $\mathcal{N}_{45} = 2$ in Fig. 3.4, where $\mathcal{N}_{mn}$ is the number of flows over the link l_{mn}. Using these numbers, each station s_i calculates the target access time for its links l_{ij} as follows:

$$t^i_{t,ij}(kT) = \frac{\mathcal{N}_{ij}(kT)}{\sum_{l_{mn} \in \mathcal{L}(s_i),\, m<n} \mathcal{N}_{mn}(kT)} \left(t_{idle} + \sum_{l_{mn} \in \mathcal{L}(s_i),\, m<n} t_{mn}(kT) \right). \quad (3.4)$$

where $\mathcal{N}_{mn}(kT)$ is the number of flows over the link l_{mn}, and $\mathcal{L}(s_i)$ is the set of links over which the station s_i can overhear transmissions and measure access time. By (3.4), as a link serves more flows, more access time is allocated to the link.

However, in multihop ad hoc networks, each station cannot measure the access time for all the links in a network. For example, let us consider the stations s_5 and s_6 in Fig. 3.4. When there are three TCP flows (f_1, f_2, and f_3), the station s_5 can measure the access time of the links l_{14}, l_{24}, l_{34}, l_{45} and l_{56}, and can count the number of flows over the links as $\mathcal{N}_{14} = 1$, $\mathcal{N}_{24} = 1$, $\mathcal{N}_{34} = 1$, $\mathcal{N}_{45} = 2$, and $\mathcal{N}_{56} = 1$, respectively. On the other hand, the station s_6 can measure the access time of l_{45} and l_{56}, and also count the number of flows on the links as $\mathcal{N}_{45} = 2$ and

$\mathcal{N}_{56} = 1$. We can see that the target access time over the link l_{56} calculated by the station s_5 is different from that by the station s_6. The station s_5 will allocate $\frac{2}{6}$ of its capacity to the link l_{56}, while the station s_6 will allocate $\frac{2}{3}$ of its capacity. The station s_5 will mark the packets of the flow f_2 in order to keep t_{56} to the target access time $t^5_{t,56}$. Since the station s_5 keeps the access time of the link l_{56} to $t^5_{t,56}$ that is smaller than target access time $t^6_{t,56}$ calculated by s_6, the station s_6 will mark the received packets with the probability $p^6_{56}(kT)$ which is less than $p^5_{56}(kT)$. Therefore, the access time over the link l_{56} is controlled almost entirely by station s_5, which allocates smaller access time than the station s_6.

3.2.2 Reducing the effect of interference in multihop ad hoc networks

In a multihop ad hoc network, sender stations can experience transmission failure due to hidden stations leading throughput degradation and unfairness. This problem is particularly significant when the TCP window size is large and many packets are in flight at the same time. Moreover, throughput degradation is more severe when a dynamic routing protocol such as Dynamic Source Routing (DSR) [43] is used. When a packet is dropped due to hidden nodes, a routing failure message is delivered from the MAC layer to a routing agent in the station. In DSR, the station floods the network with a route request packet to find a new routing path, which makes the network more loaded, hence inducing more packet drops [16]. This vicious circle degrades the throughput of TCP flows in multihop ad hoc networks. In addition to this, interference from hidden nodes can induce unfairness between flows due to capture effect and the exponential backoff scheme of the 802.11 DCF [12, 19].

In order to prevent this unfairness and throughput degradation caused by interference from hidden nodes, a station under heavy interference must reduce its sending rate of TCP packets by shrinking its TCP window size [15, 16, 42]. To do this, we introduce the interference level I to adjust the target access time of the link. The interference level I_{ij}^i of the link l_{ij} is calculated as the ratio of the number of failures to the number of total transmissions of s_i over the link l_{ij}.

$$I_{ij}^i(kT) = \begin{cases} \lambda I_{ij}^i((k-1)T) + (1-\lambda)\frac{n_{ij}^f(kT)}{n_{ij}^t(kT)}, & n_{ij}^t(kT) > 0, \\ \lambda I_{ij}^i((k-1)T) & n_{ij}^t(kT) = 0, \end{cases} \tag{3.5}$$
$$0 < \lambda < 1,$$

where, $n_{ij}^t(kT)$ and $n_{ij}^f(kt)$ $(< n_{ij}^t(kT))$ are the number of total transmissions and failed transmissions over the link l_{ij} during $[(k-1)T, kT]$, and λ is a coefficient for exponential smoothing. It is noted that $0 \leq I_{ij}^i(kT) \leq 1$ and the interference level I_{ij} increases as there are more interference leading to transmission failure. With this interference level, each station s_i calculates the target access time $t_{t,ij}$ for its active links l_{ij} as follows:

$$t_{t,ij}^i(kT) = \frac{\mathcal{N}_{ij}(kT)(1 - I_{ij}^i(kT))}{\sum_{l_{mn} \in \mathcal{L}(s_i),\, m<n} \mathcal{N}_{mn}(kT)} \left(t_{idle} + \sum_{l_{mn} \in \mathcal{L}(s_i),\, m<n} t_{mn}(kT) \right), \tag{3.6}$$

Since the target access time is reduced as transmission failure increases, TCP flows passing a link under heavy interference keep the window size small, reducing the extent of the degradation in the TCP performance in multihop ad hoc networks.

We have extended the ATC scheme of single hop networks to multihop ad hoc networks by redefining the target access time as (3.6). Note that (3.6) becomes (3.1) by setting $\mathcal{N}_{mn} = 1$ and $I_{mn} = 0$ for all links l_{mn}.

3.3 Performance Evaluation

In this section, we investigate the performance of the ATC scheme via the *ns*-2 network simulator version 2.28 [49] in terms of throughput and fairness. We compare the performance of the ATC scheme with those of the 802.11 DCF and LRED [15]. We used the TCP-NewReno and set the TCP packet size to 1460 bytes and the advertised window size to 42 packets. The parameters of ATC scheme, K_P and K_I were set to 1.0 and 0.4 respectively, and the sampling period T was set to 300ms. When there was a need for a routing scheme, we used the Dynamic Source Routing protocol (DSR) [43] unless stated otherwise. The other parameters of IEEE 802.11 and DSR were set to the values specified in the *ns*-2 simulator.

3.3.1 Single hop ad hoc network

- **Fairness among flows**

For the ad hoc network of Fig. 3.1(a), the performance of the ATC scheme is investigated. Figure 3.5 shows the throughput of each link in the ATC scheme. In this simulation, the transmission rate of each station was set to 11Mb/s. While the 802.11 DCF displays unfairness as shown in Fig. 3.1(b), the ATC scheme provides almost the same throughput for each link because the stations s_1 and s_2 adjust the TCP flow rate based on the same target access time as the other links. Although it is not shown in this dissertation, the LRED scheme showed unfairness similarly to that of the 802.11 DCF as in Fig. 3.1 because it does not consider fair capacity allocation among links.

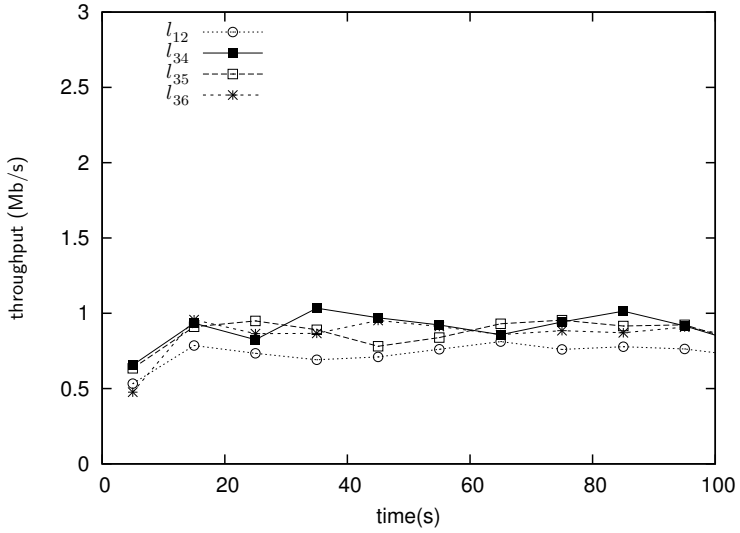


Figure 3.5: TCP throughput of each link in Fig. 3.1(a)

- **Effect of multi-rate stations**

When the data rate of the link l_{12} in Fig. 3.1(a) was set to 2 Mb/s and the data rate of the other links remained at 11 Mb/s, Fig. 3.6 shows the total access time of each link during 100 sec simulation time. In the ATC scheme, the access time was kept at nearly the same level, the channel capacity was shared fairly, and the aggregated throughput is 3.18 Mb/s. On the other hand, under the DCF and LRED, the lower data rate link l_{12} occupied the channel much of the time degrading the aggregated throughput to 2.07 Mb/s and 1.97 Mb/s, respectively.

3.3.2 Multihop ad hoc network

In order to evaluate the proposed scheme in multihop ad hoc networks, the simulations were conducted with several scenarios including chain, cross, and grid topologies as Fig. 3.7. In the simulations of multihop ad hoc networks, the data

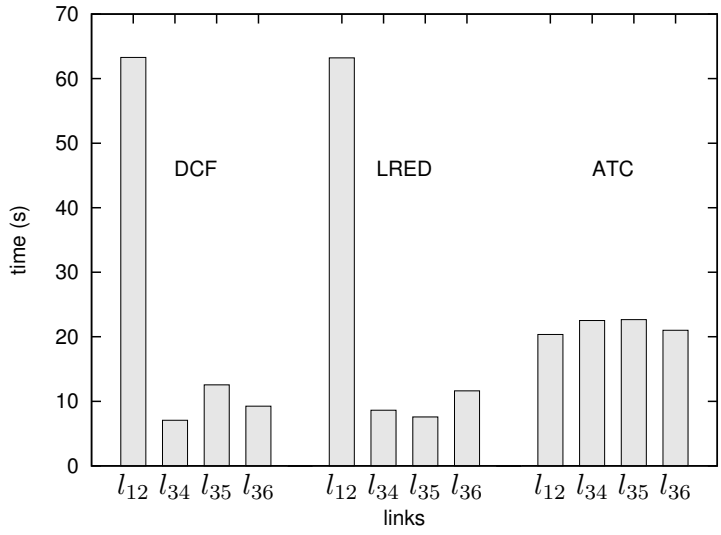
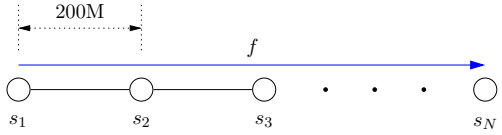


Figure 3.6: Access time of each link in Fig. 3.1(a)

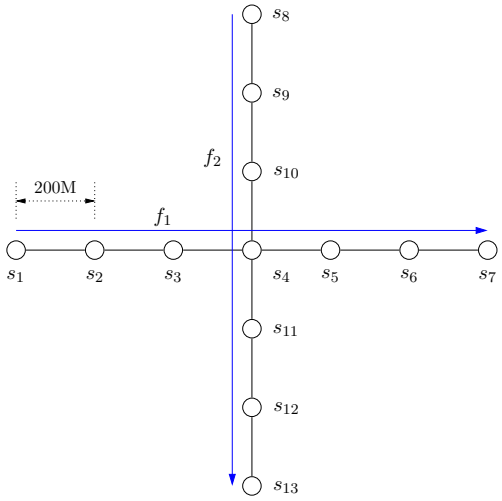
rate was always maintained 2 Mb/s and the radio propagation model was the two-ray ground model [49] with transmission range of 250m, carrier sensing range of 550m, and interference range of 550m.

- **Chain topology**

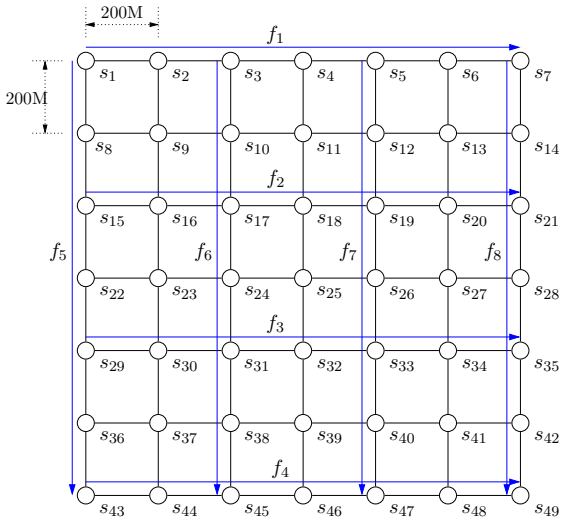
Figure 3.8 shows the throughput of each scheme in the chain topology of various lengths. When a manual static routing protocol was used, the throughput of all the schemes were nearly the same. However, when the DSR was used for routing, it can be seen that the 802.11 DCF shows severe throughput degradation while the ATC and LRED schemes show throughput improvement ranging from 80 $\sim$ 200 Kb/s over the 802.11 DCF. When the DSR was used, the packet losses activated the route maintenance process of the DSR protocol, which increased network load, which in turn degraded the throughput. Since the rate of TCP flows are controlled in the ATC and LRED schemes in order to avoid network



(a) (N-1) hop chain topology



(b) Cross topology with 13 stations



(c) 7x7 grid topology

Figure 3.7: Multihop topologies

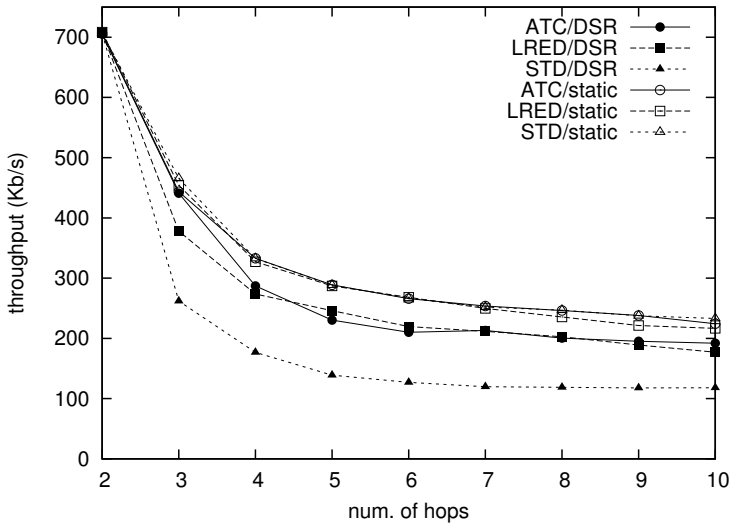


Figure 3.8: Throughput of a TCP flow in a chain topology with various lengths

overload, packet drops and routing failures are normally reduced so that the overall throughput can be increased.

Figure 3.9 shows the distribution of TCP window size for a TCP flow in the 7-hop chain topology. In the ATC and LRED schemes, the TCP flow keep the window size around 2 packets for the most of time, while the TCP window size in the 802.11 DCF is distributed over a large values.

- **Cross topology**

A cross topology was configured with 13 stations and 2 TCP flows as Fig. 3.7(b). Table 3.1 shows the throughput and the Jain's fairness index [51]. The LRED and ATC schemes work nearly the same in throughput and fairness. While the fairness index in Table 3.1 is based on the throughput during the total simulation time of 100 seconds, Fig. 3.10 shows the fairness index calculated with the throughput of every two seconds, in which several tens packets are delivered.

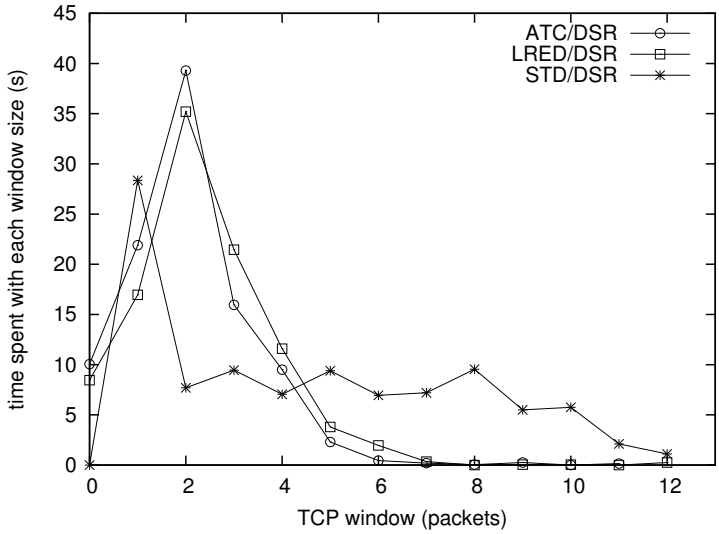


Figure 3.9: Distribution of TCP window size in 7-hop chain topology

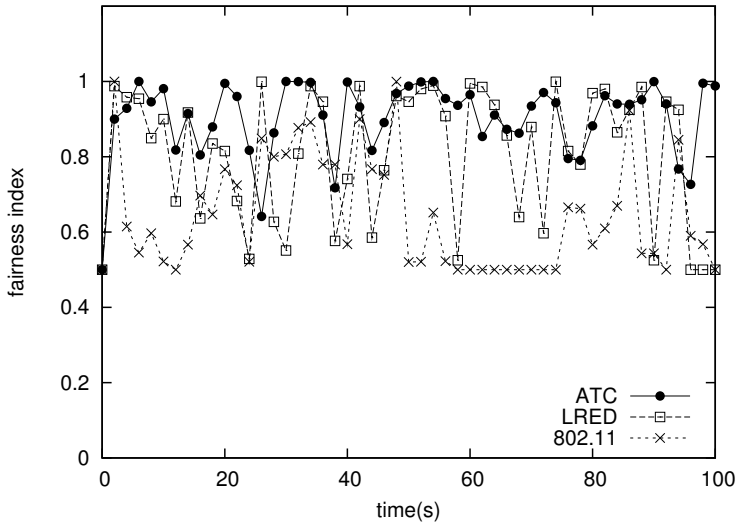


Figure 3.10: Short-term fairness in cross topology

Table 3.1: Throughput and fairness index for the cross topology

	802.11 DCF	LRED	ATC
Flow 1 (Kb/s)	81.8	125	123.2
Flow 2 (Kb/s)	52.7	118.2	117.1
Aggregate throughput (Kb/s)	134	243	240
Fairness index (100s)	0.95	0.99	0.99
Fairness index (2s)	0.64	0.81	0.91

Table 3.2: Throughput and fairness index for the grid topology

	802.11 DCF	LRED	ATC
Aggregate throughput (Kb/s)	377	387	406
Fairness index	0.87	0.88	0.97

The average fairness index for two seconds in the ATC scheme is 0.91, while that of LRED is 0.81. This shows that the ATC scheme is better in the short-term fairness.

- **Grid topology**

In a 7x7 grid topology with 8 TCP flows as Fig. 3.7(c), the throughput and fairness index were examined. Table 3.2 shows the aggregated throughput of 8 flows and the Jain's fairness index. The fairness index of the ATC scheme shows about 10% improvement over those of the 802.11 DCF and LRED scheme, while the ATC scheme gives about 5% more throughput than the 802.11 DCF and LRED.

- **Asymmetric topology**

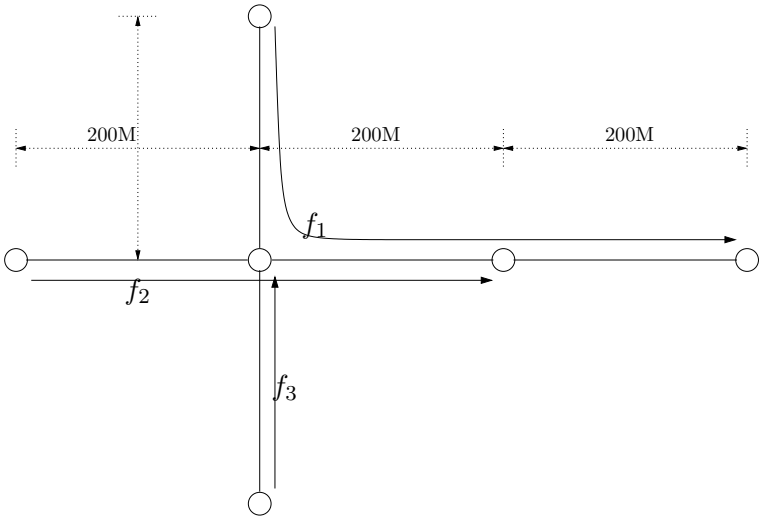
Table 3.3: Throughput and fairness index for the topology of Fig. 3.11

	802.11 DCF	LRED	ATC
f_1 (Kb/s)	49	56	209
f_2 (Kb/s)	21	52	187
f_3 (Kb/s)	1200	1100	218
Aggregate throughput (Kb/s)	1270	1208	614
Total link throughput (Kb/s)	1390	1372	1219
Fairness index	0.37	0.40	0.99

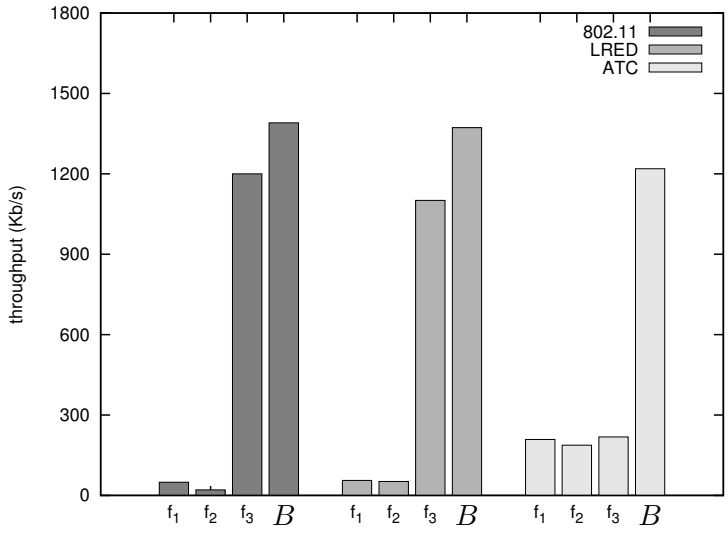
Since the topologies that were investigated were symmetric, all the schemes have a better chance to provide good fairness among flows. However, in the asymmetric topology such as Fig. 3.11(a), it is more difficult to provide good fairness among flows. Table 3.3 and Fig. 3.11(b) show the throughput and fairness index of three flows in Fig. 3.11(a). In the 802.11 DCF and LRED schemes, the single hop flow f_3 monopolizes most of the capacity, while the ATC keeps the throughput of each flow at nearly the same level. In Table 3.3 and Fig. 3.11(b), the total link throughput is a sum of aggregated throughput of TCP flows over each link. Since the ATC scheme provides higher throughput for multihop flows f_1 and f_2 than the other schemes, the network ends up experiencing more collisions and route maintenance procedures are more frequently invoked. Due to this overhead, the total link throughput in the ATC scheme is a little smaller than those in the other schemes.

- **Random topology**

Here, the performance of each scheme in a random topology with 50 stations in 1000m by 1000m space and 16 TCP flows as in Fig. 3.12, was evaluated. Table 3.4



(a) An asymmetric topology



(b) Throughput of flows (B is the total link throughput)

Figure 3.11: Throughput in an asymmetric topology

shows the throughput of 16 flows for the three schemes. The throughput of each flow is 13.5 $\sim$ 74.6 Kb/s in the ATC scheme, while those of the DCF and LRED are 0 $\sim$ 1285 Kb/s and 0 $\sim$ 1252 Kb/s, respectively. The aggregated throughput in the 802.11 DCF and LRED schemes is about five times larger than that of the ATC scheme, because single hop flows f_1, f_2, and f_{11} use almost all the capacity, hence depriving the other multihop flows. The total link throughput in the ATC scheme is about 40 % of those of the other schemes, which is much less than expected. This is because many DSR messages are required in the ATC scheme in order to maintain routes for the multihop flows, most of which were practically deactivated in the other schemes. During simulation time of 100 seconds, there were 10508 DSR messages in the ATC scheme, while there were 3711 and 3147 DSR messages in the 802.11 DCF and LRED schemes, respectively. This shows that a more efficient routing scheme is needed to increase the throughput in the multihop topology.

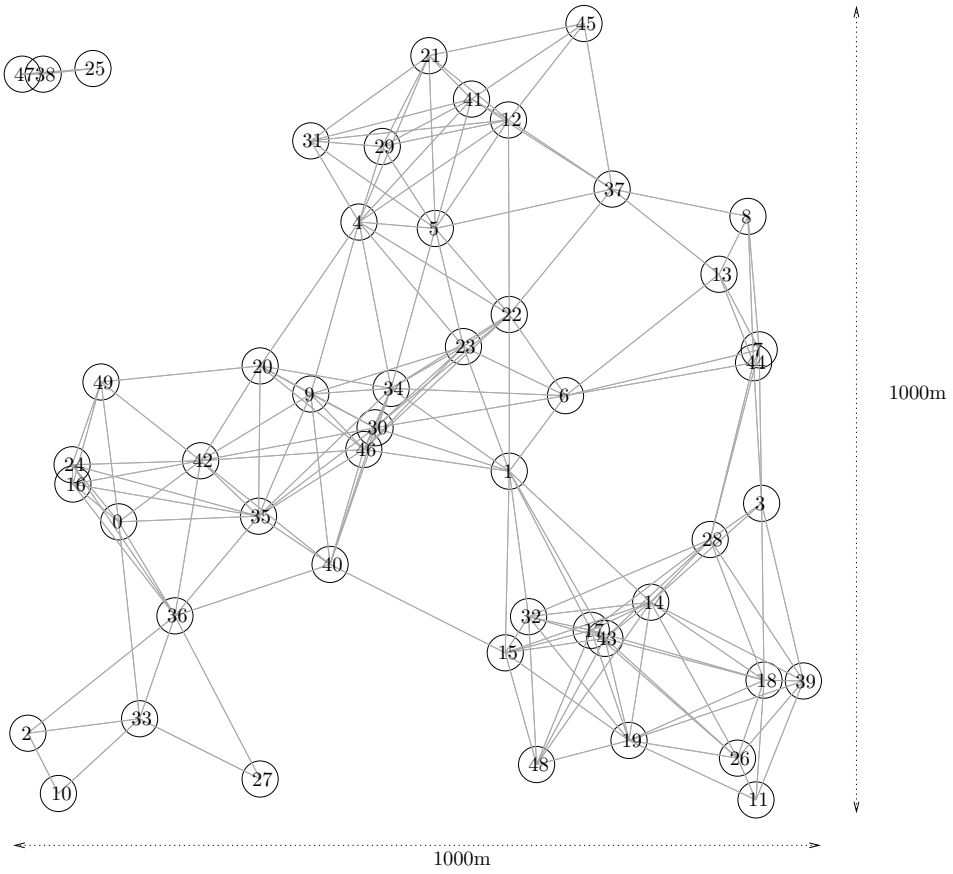


Figure 3.12: A random topology consisting of 50 stations

Table 3.4: Throughput and fairness index for the random topology

flow	src-dst	# of hops (shortest path)	802.11 DCF	LRED	ATC
f_1	11-18	1	1285	1252	74.6
f_2	15-32	1	66.1	97.1	62.0
f_3	17-5	3	0.4	0.4	35.1
f_4	0-13	3	6.1	11.8	38.1
f_5	2-43	4	0	0	13.6
f_6	2-26	5	0	0	14.7
f_7	26-22	3	0	0.4	18.2
f_8	34-0	2	81.4	51.4	26.3
f_9	35-15	2	0	0	17.5
f_{10}	40-20	2	1.6	2.8	54.6
f_{11}	41-37	1	1168	1170	51.6
f_{12}	42-26	4	0	0.4	14.0
f_{13}	43-20	3	0.2	0	16.4
f_{14}	44-30	2	0.2	0.7	22.7
f_{15}	45-0	5	8.8	9.2	14.4
f_{16}	49-18	5	0	0	13.5
total end-to-end TCP throughput Kb/s			2618	2597	487
# of TCP packets transmitted over all links			23766	23067	13450
# of DSR messages transmitted over all links			3711	3147	10508
# of collisions			6881	5615	16278
total TCP link throughput Kb/s			2833	2749	1603
fairness index			0.14	0.14	0.71

Chapter 4

Conclusions

In this dissertation, the fairness of TCP flows in the IEEE 802.11 WLANs has been studied. In the first part of the dissertation, fairness in infrastructure WLANs has been dealt. IEEE 802.11 infrastructure networks are known to induce unfairness among stations having a different number and traffic direction of TCP flows. In order to solve this problem, the ATC and DATC schemes have been proposed, which can control the rate of TCP flows based on the channel access time of each station in an infrastructure WLAN. It is showed that the rate of each station converges to the target rate by the proposed schemes so that each station has fair bandwidth regardless of the traffic direction and the number of TCP flows. The proposed schemes can be easily implemented either in an AP or mobile stations without any modification of the MAC and TCP layers. The performance of the proposed schemes was evaluated under various scenarios and the results showed that the proposed schemes work, in terms of throughput and fairness, as good as, or in certain cases, even better than the FAW scheme [4], which is impractical to implement. The proposed schemes havs the following

properties. 1) They provide per-station fairness under various scenarios without reducing throughput. 2) They guarantee QoS of differentiated services in IEEE 802.11e networks. 3) Different data rates of mobile stations do not reduce overall throughput.

In the second pard of the dissertation, improving fairness in multihop ad hoc networks has been studied. In ad hoc networks based on the IEEE 802.11 WLAN, the channel capacity may not be shared fairly among flows because each sender station has an equal chance to access the channel regardless of the number of its flows. In addition, in multihop ad hoc networks, the interference from hidden nodes may degrade throughput and fairness. In order to resolve these problems, a network capacity allocation scheme based on the access time over each link has been proposed. By using the congestion control mechanism of TCP flows, the proposed scheme controls the access time of each link in order to provide per flow fairness and reduce the interference from hidden nodes. Compared with other schemes, the proposed scheme can provide much better per-flow fairness in ad hoc networks without sacrificing throughput. Since the proposed scheme does not require any modification of the TCP and MAC protocols, it can be easily applied to existing devices. Through various simulations, we showed that the proposed scheme can improve fairness, even though there is a possibility that throughput performance may be reduced due to the control messages in routing algorithms in multihop ad hoc networks.

For future work, the idea of the ATC scheme will be extended to mesh networks, in which fairness among stations is an important issue. In mesh networks, each station is served by an AP that is also connected to an access router through several wireless hops. Therefore, the mesh network will inherit the problems of

infrastructure and ad hoc networks. By extending the proposed schemes, it is expected that fair service can be provided for users in mesh networks.

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초록

IEEE 802.11 표준에 기반한 무선랜은 인프라스트럭쳐 네트워크와 애드 혹 네트워크를 구성하는데 널리 사용되고 있다. 802.11 무선랜의 매체 접근 계층 방식은 각 기기간의 무선 채널 획득 기회를 공평히 보장하지만, TCP와 같은 상위 계층 프로토콜의 영향 및 서로 다른 네트워크 조건으로 인해 기기간 혹은 플로우간에 무선 자원이 불공평하게 할당되는 문제가 있다. 본 논문에서는 802.11 표준에 기반한 인프라스트럭쳐, 애드 혹 네트워크에서 TCP 트래픽이 사용될 때 발생되는 불공평한 자원 할당 문제를 해결하기 위해 무선 채널 사용 시간을 제어하는 방법을 제안한다.

논문의 전반부에서는 인프라스트럭쳐 무선랜 환경에서 액세스 포인트에 연결된 각 기기간 공평성 보장 문제를 다룬다. 인프라스트럭쳐 무선랜 환경에서는 802.11 매체 접근 계층 방식인 DCF를 사용할 때 상향 플로우와 하향 플로우 사이에 불공평한 자원 분배가 발생되는 문제가 있다. 또한, 각 기기들이 사용하는 플로우 수가 다를 경우에도 이에 따라 각 기기들 간에 무선 자원이 불공평하게 사용되는 문제가 있다. 이러한 불공평한 자원 할당 문제를 해결하기 위해, 본 논문에서는 각 기기의 사용 시간을 제어하는 ATC, DATC 두가지 방법을 제안한다. ATC 방법에서는 액세스 포이트가 각 기기의 무선 채널 사용 시간에 기반하여 각 기기가 사용하는 TCP 플로우의 전송률을 제어하고, DATC 방법에서는 같은 액세스포인트에 연결된 기기들이 자신의 무선 채널 사용 시간에 기반하여 자신의 TCP 플로우의 전송률을 제어한다. 이를 통해 각 기기가 사용하는 플로우의 방향, 플로우 수, 무선 데이타 전송 속도 등에 상관없이 무선 채널을 사용하는 시간을 각 기기에 공평히 할당함으로써 전체 네트워크의 효율을 떨어뜨리지 않으면서도 공평성을 보장할 수 있다. 또한 제안된 방법이 기존의 MAC, TCP 계층의 표준을 수정하지 않고 구현이 가능하기 때문에 현재 사용되고 있는 무선

랜 기기에 쉽게 구현될 수 있다는 장점이 있다. 다양한 환경에서 ns2 네트워크 시뮬레이션을 수행하여 제안된 방법의 성능을 검증한다.

논문의 후반부에서는 애드 혹 네트워크에서의 공평성 문제를 다룬다. 에드 혹 네트워크에서는 각 무선랜기기가 자신의 플로우 뿐만아니라 다른 기기의 플로우도 같이 전달할 필요가 있어, 기기마다 전송해야 하는 플로우 수가 달라지게 된다. 802.11의 매체 접근 방식에 의하여 전송을 시도하는 모든 기기들은 공평한 채널 획득 기회를 가지기 때문에, 각 기기가 전송해야 하는 플로우 수가 서로 다른 애드 혹 네트워크에서는 각 플오우들에게 불공평하게 자원이 할당되는 문제가 있다. 또한, hidden node problem으로 인한 간섭이 발생되어 멀티 홉 애드 혹 네트워크에서는 TCP 플로우의 성능 및 공평성이 현저히 떨어지는 문제가 있다. 본 논문에서는 이러한 불공평성과 성능 감소 문제를 해결하기 위해 애드 혹 네트워크에서 각 링크 상에 존재하는 플로우 수와 간섭으로 인한 전송 실패 횟수를 고려하여 플로우들이 공평한 전송률을 가질 수 있도록 각 링크를 사용하는 시간을 제어하는 방법을 제안하고, 다양한 애드 혹 네트워크 환경에서 ns2 네트워크 시뮬레이션을 수행하여 제안된 방법이 공평성 및 전송률 향상을 가져옴을 보인다.

주요어: IEEE 802.11 무선랜, TCP, 공평성, 인프라스크럭쳐 네트워크, 애드 혹 네트워크

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십년이 넘는 시간을 보내며 정든 관악을 떠나면서, 부족하나마 작은 결실을 맺을 수 있도록 도와 주신 많은 분들에게 감사의 마음을 전합니다.

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