國立交通大學

資訊科學與工程研究所

碩 士 論 文

在 IEEE 802.11 無線網路中以滿意度爲基準的
媒介使用機制

Satisfaction-based Differentiated Media Access Control in
IEEE 802.11 WLANs

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Abstract

In this thesis, we probe the Quality of Service (QoS) demand over IEEE 802.11 WLANs, and we found that current IEEE 802.11 media access control (MAC) mechanism is no longer fit for the fast growing application of real-time multimedia. Hence, we survey some QoS extension to IEEE 802.11 MAC scheme, including official solution IEEE 802.11e, and analyze their advantages and drawbacks. We proposed a neo IEEE 802.11e extension MAC scheme, named Satisfaction Enhanced DCF (SEDCF), based on satisfaction degree of QoS traffics and aim to (i) providing service differentiation according to required transmission rate; (ii) achieving better residual bandwidth fairness between all the flows; and (iii) maintaining high network throughput. Through the simulation with ns-2 simulator, we evaluated the performance of SEDCF and surveyed IEEE 802.11 QoS enhancements. At last, we proposed the conclusion and related future work.

摘要

在這篇論文中，我們首先探討因爲無線多媒體應用的快速發展，而產生目前在無線網路上的品質服務保證問題，進而討論目前最廣爲使用的無線網路媒介使用機制—IEEE 802.11 在品質服務保證問題上的不足之處。我們進一步探討一些目前在已經提出來 IEEE 802.11 無線網路上的品質服務保證加強機制，包括官方解決方案 IEEE 802.11e，並且分析並瞭解這些方法的優缺點。接下來我們在 IEEE 802.11e 架構上提出一個新的以滿意度爲基準的媒介使用機制，叫做 SEDCF，這個新的機制的目標是(1)提供以傳輸速率爲基準的服務區別，(2)達到對於所有連線之間的剩餘頻寬使用的公平性，以及(3)同時保持較高的網路傳輸量。透過 ns-2 網路模擬器的模擬，我們評估了 SEDCF 和其他品質服務保證加強機制的效能。最後，我們將提出結論和未來可供努力的方向。

致謝

要完成一篇論文，本來就已經有心理準備這會是一個艱鉅的任務了，只是這困難程度還遠遠超過我想像。

首先我要感謝我的指導教授—趙禧綠老師，沒有趙禧綠老師的帶領與指導，我絕對不可能完成這篇論文。從一開始剛進研究所唸 paper 開始，本來重點都不容易抓到的我，慢慢地在老師指導之下，重點、筆記整理、想法歸納…等等，漸漸步上了軌道。我從老師這邊也學到了不少做事的觀念，如何確定自己的目標、分析目前遇到的困難並且對症下藥，也改正了我做事時不少模糊焦點或是本末倒置的毛病，這些觀念不只對論文完成有絕對幫助，更對我做人處事有深遠影響，真的非常感謝老師的帶領與指導，尤其是對我們的信心與容忍，讓我們把我們心中的不可能加以實現。

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最重要的，我要感謝的是我的家人，我的父母、弟弟、外婆，沒有你們的支持，不用說論文完成，我現在是不是能唸到研究所都很難說，你們的支持是一個極其重要卻看不見的力量，得以讓我幸運的成長，並且繼續走下去。

謝謝交通大學，一個容忍我在這裡撒野了六年的地方；謝謝房東楊奶奶讓我有舒適的住宿環境；謝謝世界上所有的藝文工作者，不論是音樂電影還是劇場，你們繽紛了一切的夢想。

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Chapter 1
Introduction

1.1 Current IEEE 802.11 Media Access Control Scheme

The IEEE 802.11 media access control (MAC) for wireless local area networks (WLANs) is currently the most widely deployed wireless techniques in the world [1]. It allows people to implement a wireless network in one of two possible configurations: the infrastructure mode or the ad hoc mode. Under the infrastructure mode, all nodes reside in a particular region where all communication must go through the access point. If the connection between a node and the access point is lost, the node cannot transmit any packets. Under the ad hoc mode, all nodes can form an ad hoc network spontaneously without any centralized control. Even if a node loses direct connections with some nodes, it is still possible for the node to communicate with others through multi-hop connections. This feature allows ad hoc networks to be flexibly deployed in scenarios such as battlefields, emergencies etc., where no pre-established infrastructure exists.

The IEEE 802.11 MAC employs a compulsory contention-based media access function—Distributed Coordination Function (DCF), and an optional controlled channel access function—Point Coordination Function (PCF). The DCF and PCF represent two media access concepts: distributed management and centralized management, respectively. The DCF is based on a Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) mechanism. In DCF, when a node wants to transmit a packet, it either directly transmits the packet while sensing the channel is idle over a DCF InterFrame Space (*DIFS*) or starts decreasing a backoff timer randomly chosen from contention window (*CW*) while sensing the channel is busy. In the second scenario (deferring mode), the node can transmit the packet after the backoff is reaching zero and the channel is still idle over a DIFS. On the other hand, PCF is designed to support time-bounded multimedia applications and eliminate the contention process. In PCF, all the transmission is controlled by the coordinator with the polling mechanism. However, PCF is rarely implemented in current IEEE 802.11 equipments because of many reasons such as the coordinator may not get all the nodes' information on time and precisely while new nodes coming, or it cannot serve all the nodes in a contention free period which may causes an extra delay penalty. Hence, in the rest of this thesis, we will only focus on DCF and its enhancement.

1.2 Motivation

With the tremendous growth of wireless network, more and more applications of real-time multimedia will take wireless access as one of their communication path. And many kinds of real-time multimedia application have quality-of-service (QoS) requirements such as delay or throughput or transmission rate. However, DCF of current IEEE 802.11 provides a media access mechanism with equal probabilities to all nodes and all kinds of applications contending for the same wireless media, and does not support service differentiation, which means real-time flows will get the same treatment as best-effort flows'. Therefore, DCF is no longer suitable for real-time multimedia applications.

To deal with these problems, the IEEE 802.11 working group designed an extension to the IEEE 802.11 standard, called IEEE 802.11e [2] [3]. It mainly enhanced the QoS support with service differentiation by some MAC parameters, which we will describe in detail in chapter 2. Besides, some other QoS enhancement based on IEEE 802.11 or IEEE 802.11e are also proposed [4] [5] [6] [7] [8] [9], however, most of them aim to improve the throughput, fairness between the same priority flows and service differentiation only. It motivated me to design a neo mechanism which consider about the overall fairness and QoS satisfaction based on transmission rate. Hence, the aim of this thesis is to investigate a novel MAC mechanism to achieve three objectives: (i) improving network throughput; (ii) achieving fair share of residual bandwidth to have better global fairness; and (iii) guaranteeing QoS applications' demands to have better satisfaction.

1.3 Organization

The remaining of this thesis is organized as follows. Chapter 2 describes the related work including the EDCF architecture proposed in the802.11e and its improvement. In chapter 3, the proposed media access mechanism is described in detail. And chapter 4 is the performance evaluation via simulations. The conclusion and future works is in chapter 5.

Chapter 2
Related Work

This chapter will introduce some QoS support extensions to IEEE 802.11 DCF mechanism, including official solution, and other solutions based on different adaptation aspect.

2.1 Enhanced Distributed Channel Access of IEEE 802.11e (EDCA)

IEEE 802.11e is the QoS support extension to the original IEEE 802.11 standard, and the new media access control scheme is called Hybrid Coordination Function (HCF). Because HCF is developed based on the original IEEE 802.11 standard, it also has a distributed contention-based media access function—Enhanced Distributed Channel Access (EDCA) (named Enhance DCF (EDCF) in early version of IEEE 802.11e draft), and a centralized contention-free media access function—HCF Controlled Channel Access (HCCA), which are extended from DCF and PCF, respectively. We will only focus on EDCA in this thesis because of the same reason mentioned before.

The way IEEE 802.11e provided QoS support is via service differentiation. Unlike all the traffics are treated the same in DCF, in EDCA, all the flows are classified into four Access Category (*AC*), which represent the different priority and with different MAC parameters. In EDCA, all the flows use different Arbitration Inter Frame Space (*AIFS*[*AC*]), minimum contention window value (*CWmin*[*AC*]), maximum contention window value (*CWmax*[*AC*]) and Persistent Factor (*PF*[*AC*]) for

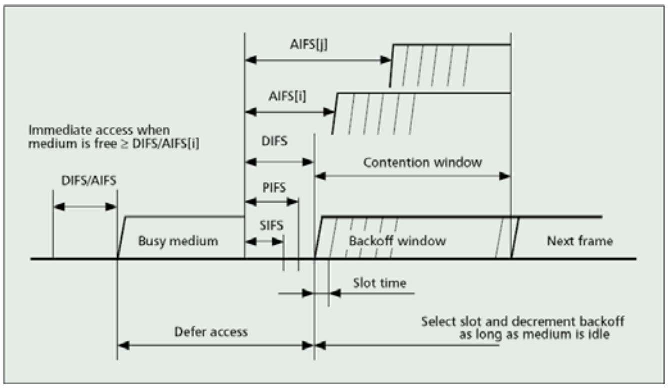

Figure 2.2-1. The timing diagram of media access in EDCF and DCF

the contention process to transmit packets belonging to the different *AC*s instead of original *DIFS*, *CWmin*, *CWmax*, *PF* in DCF. And the backoff timer is randomly chosen from [1, 1+*CW*[*AC*]], instead of [0, *CW*] as in DCF.

The timing diagram of media access in EDCA and DCF are shown in Fig. 2.2-1. As to the definitions of these parameters belonging different *AC*s, in concept, the higher priority flows get the smaller values of these parameters. Because the smaller parameter values mean the higher probability to access the media, the less latency, and the more capacity share of this priority.

In EDCA, contention to the media is becoming between ACs, and each AC with a different transmission queue. Fig. 2.2-2 shows in EDCF, there are four queues in a station, where each queue behaves as a single EDCF contending entity with different parameter sets. While more than one AC within in one station attempt to transmit

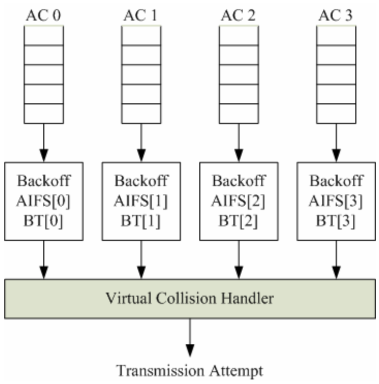

Figure 2.2-2. Four access categories (ACs) in one station in EDCF

packet concurrently, the collision is handled in a virtual manner and the packet with highest priority is chosen to transmitted, which left other queues performing the contention window updating and backoff procedure. The contention window updating procedure is basically the same as in DCF, which described in follows.

1) Adjusting CW after each successful transmission

After each successful transmission, the value of contention window for class *i* is reset to its predefined minimum contention window value in EDCA, which described below,

$$CW[i] = CW_{\min}[i]\,. \tag{1}$$

2) Adjusting CW after each unsuccessful transmission

After each unsuccessful transmission, this class *i*'s contention window value become *PF*[*i*] times of previous value, of course, the new *CW*[*i*] must be bounded in the predefined value (smaller than or equal to *CWmax*[*i*]). That is,

$$CW[i] = \min\left(CW[i] * PF[i], CW_{\max}[i]\right). \tag{2}$$

where in lately versions of IEEE 802.11e draft, *PF*[*AC*]s are all set to 2, which is the same as in DCF. Except for the parameters and mechanisms mentioned above, the rest part of EDCA is basically the same as DCF.

After the QoS concept is taking seriously by IEEE 802.11e, there are further QoS enhancements about improving better service differentiation, throughput or fairness. All the further QoS enhancements can be separated into three main categories according to the aspect of adaptation: contention window based enhancement, backoff based enhancement and inter frame space (IFS) based enhancement. In the following, I am going to introduce some further QoS enhancements of different categories.

2.2 Adaptive Enhanced DCF (AEDCF) – CW-based QoS Enhancement

In the further observation of EDCA, the EDCA performance obtained are not optimal since all the MAC parameters are predefined as static values, which cannot be adapted to the network condition. Especially when the media is highly loaded, EDCA performs poorly in throughput, latency, and collision rate. This is mainly because of the immediately reducing current *CW*[*i*] to *CWmin*[*i*] after successful transmission, and leading the over high collision rate. Hence, Adaptive Enhance DCF (AEDCF) [4] is proposed to improve EDCA by taking network condition into account in MAC scheme.

The major difference between EDCA and AEDCF is that AEDCF use the network condition to adapt *CW*[*i*], instead of setting it to *CWmin*[*i*]. The whole contention window updating procedure is shown below.

1) Adjusting CW after each successful transmission

With the motivation by the fact that when a collision occurs, a new collision is likely to occur in the near future, AEDCF adopts an approach called Slow Decrease (SD) to reduce *CW* by a dynamic factor. And the SD factor used here to reflect network condition is the average collision rate, which updated periodically. First, the current measured collision rate $f_{current}^{j}$ is calculated by the following equation:

$$f_{current}^{j} = \frac{N\left(collision_j[p]\right)}{N\left(data\_sent_j[p]\right)} \qquad (3)$$

where $N(collision_j[p])$ is the number of collisions of node *p* which occurred between the $(j\text{-}1)^{th}$ and j^{th} updates, and $N(data\_sent_j[p])$ is the total number of packets that have been sent by node *p* in the same period. It's obvious that $f_{current}^{j}$ is always in the range of [0, 1].

Next, AEDCF uses an estimator of Exponentially Weighted Moving Average (EWMA) to smoothen the currently measured values to minimize the impact of transient collisions and get the average collision rate. The average collision rate $f_{average}^{j}$ is calculated by the following equation.

$$f_{average}^{j} = (1-\gamma) * f_{current}^{j} + \gamma * f_{average}^{j-1} \quad (4)$$

where $f_{current}^{j}$ is calculated from (3), and $f_{average}^{j-1}$ is the measured average collision rate of the *(j-1)*th update, and γ is the weight (or smoothing factor). The average collision rate is computed every period *Tupdate* express in time-slots, which should too long to infect the estimation preciseness and should not be too short in order to limit the complexity.

In order to ensure that the priority sequence between different *AC*s is still intact when a class updates its *CW*, each class should be assigned a different factor according to its priority level, and this factor is called Multiplication Factor (*MF*). In AEDCF, the maximum *MF* value is set to 0.8 based on set of simulations with several scenarios by the authors. And of course, the factor should not lead the calculated *CW* excess the previous *CW* because in concept, the flows transmitted successfully should not be punished more. And the *MF* value is determined based on (5).

$$MF[i] = \min\left(1 + (i \times 2) * f_{average}^{j},\ \ 0.8\right) \quad (5)$$

Obviously, based on (5), the highest priority *AC* will reset its *CW* parameter with the smallest *MF* value. Finally, we still need to guarantee that all the computed *CW* after each successful transmission of packet of class *i* are greater than or equal to *CWmin*[*i*], so *CW*[*i*] is then updated as (7).

$$CW[i] = \max\left(CW_{\min}[i],\ \ CW[i] * MF[i]\right) \quad (7)$$

2) Adjusting CW after each unsuccessful transmission

After each unsuccessful transmission AEDCF did not make any change as in EDCA scheme but reset *PF*[*i*] with different value according to priority levels, which in order to re duce the probability of a new collision and consequently decrease delay. So the *CW* updating equation is following the equation (2) described before. However, the *PF*[*i*] values are not set to 2 anymore, and the higher flows have the lower *PF*[*i*] values.

2.3 Adaptive Fair Enhanced DCF (AFEDCF) – Backoff-based QoS Enhancement

While AEDCF improves the total throughput of EDCA, the performance of low-priority flows degrades sharply at high load because of the differentiation between MAC parameters of different *AC*s. And the fairness between the same *AC* and the channel utilization also degrades when the channel is congested. Hence, Adaptive Fair Enhanced DCF (AFEDCF) [5] is proposed to extend EDCA which combined the advantages of service differentiation, fast backoff decrease, and an adaptive access scheme and aim to improve (i) the performance of multimedia applications whatever is the channel load, (ii) the total throughput obtained, and (iii) the fairness between the same priority applications.

Unlike AEDCF, the contention window adjustment procedure is not the mainly part of AFEDCF and AFEDCF just follow the original mechanism of EDCA at this part. That is, after each successful transmission, the *CW* is updating by the equation (1) mention before; after each unsuccessful transmission, the *CW* is updating by the equation (2) described above. But when a queue is in deferring mode, in AFEDCF, whenever it detects the start of a new busy period (maybe caused by a collision or a

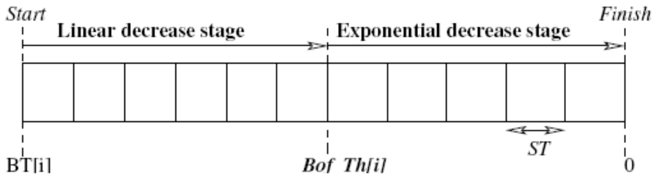

Figure 2.3-1. Backoff Timer decreasing stages in FCR mechanism

packet transmission in the media by other flows), it will react as it got through a unsuccessful transmission itself and increase the *CW* as above. The reason is to penalize the low priority flows and to improve the fairness between the same priority flows by having almost the same value of *CW* equal to *CWmax*[*i*] after the finish of a busy period, and consequently the same transmission opportunity.

The major innovation of AFEDCF is in the backoff decreasing procedure. In order to obtain better channel utilization, AFEDCF adopt a mechanism called Fast Collision Resolution (FCR) [10], and the FCR mechanism consists in using a backoff threshold value that separates two backoff states. The first backoff stage corresponds to linear decrease as in the standard. When the remaining backoff time reaches the threshold value, the queue starts the second stage by reducing the *BT* exponentially. (shown in Fig. 2.3-1)

In the linear decrease stage, Backoff Timer (*BT*) is decreasing as following:

$$if\ \ BT[i] > Bof\_th[i],\ BT[i] = BT[i] - SlotTime\,, \tag{8}$$

where $Bof\_th[i]$ is the backoff threshold of this flow i , *SlotTime* is predefined system slot time. And when the remaining backoff time reaches the threshold value, the *BT* is decreasing exponentially until *BT* is zero or less than a slot time, as following:

$$if\ \ BT[i] \le Bof\_th[i],\ BT[i] = BT[i]/2\,, \tag{9}$$

$$if\ \ BT[i] < ST,\ then\ BT[i] = 0\,. \tag{10}$$

AFEDCF also adapted FCR by dynamically adjust the backoff threshold. In concept, when media load decreases and the queue decrements its *CW*[*i*] value, the

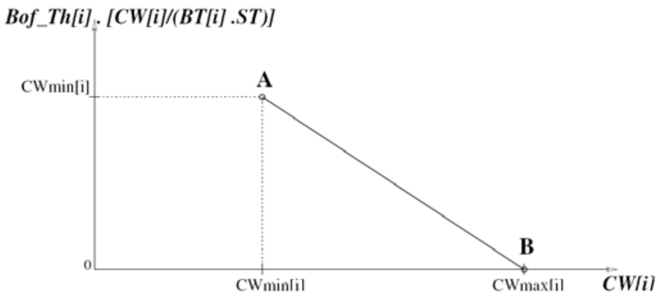

Figure 2.3-2. Backoff Threshold Adaptation Function

exponential decrease stage must be extended by increasing its *Bof_Th*[*i*] parameter, in order to reduce the idle time; when media load increases and the queue increments its *CW*[*i*] value, the exponential decrease stage must be reduced by decreasing its *Bof_Th*[*i*] parameter, in order to avoid a new collision. In fact, the backoff threshold function is derived by drawing a linear function (shown in Fig. 2.3-2 above) which joins the two points $A(CW[i]=CW_{min}[i], Bof\_Th[i]=CW_{min}[i])$ and $B(CW[i]=CW_{max}[i], Bof\_Th[i]=0)$.

Hence, the backoff threshold adaptation function is derived as below:

$$Bof\_Th[i] = \frac{CW_{\max}[i] - CW[i]}{CW_{\max}[i] - CW_{\min}[i]} * \frac{BT[i]}{CW[i]} * CW_{\min}[i] * SlotTime \qquad (11)$$

2.4 IFS-based Distributed Fair Queuing (IDFQ) – IFS-based QoS Enhancement

Except for QoS enhancement which adapting contention window computing mechanism and backoff decreasing mechanism, there is method using inter frame space to provide better QoS support. Unlike other method working for collision resolution, IFS-based Distributed Fair Queuing (IDFQ) [6] [7] just chooses appropriate IFS values for flows with different weight, and applies some randomization to avoid collisions, all based on the concept of weighted fair queuing (WFQ). There is no backoff mechanism in IDFQ and the reason is to improve aggregate throughput.

Since IDFQ is based on the concept of WFQ, each transmitted frame should be stamped with a finish tag which related to the weight predefined. And the frames with larger weight leads smaller finish tag, which should be transmitted before those frames with smaller weight, i.e. larger finish tag, basically. Each station *i* maintains a local virtual clock $v_i(t)$ as a function of real time *t*. As in [6], in order to compatible with IEEE 802.11b MAC parameters (*PIFS*=30μs and *DIFS*=50μs), the IFS value of station *i* is expressed as

$$IFS[i] = \begin{cases} 40(\mu s) + \dfrac{random(0,1)}{\alpha} * \beta, & F[i] - v_i(t) = 0 \\ 40(\mu s) + \dfrac{F[i] - v_i(t)}{\alpha} * \beta, & otherwise \end{cases} , \quad (12)$$

where $F[i]$ is the finish tag of the head of line frame in station *i*; $\alpha = L_{\max} / \phi_{\min}$, which $L_{\max}$ and $\phi_{\min}$ are the maxima frame size and minima weight in the system, respectively, and α also represent the maxima value of $F[i] - v_i(t)$; β is a positive uniformly random number in interval [0, 10] to avoid collision, since there is no backoff mechanism. The generated *IFS* value will always located in interval [*PIFS*, *DIFS*], which is also proved in [6].

2.5 Discussion

Through the necessary simulations, as these mechanisms described above are designed in purpose, most of them reach the goal they were designed for. EDCA provides service differentiation which not provided by original DCF; AEDCF lower the collision rate and increase total throughput especially when the channel is highly load, compare to EDCA; AFEDCF performs better fairness between the same priority flows while maintaining high throughput and service differentiation; IDFQ provides higher total throughput of all flows than EDCA and service differentiation for different flows in proportion to their weights, while achieving weighted fairness between different priority flows, especially.

As to comparison under different consideration angle and different protocols, in total throughput, AEDCF, AFEDCF and IDFQ prefer better than EDCA, and AFEDCF prefers better than AEDCF. In the view of fairness, AFEDCF and IDFQ should prefer better than EDCA and AEDCF, but AFEDCF and IDFQ just achieve different kind of fairness because the method they adopted in protocols.

However, all the four mechanisms do not mention about absolute fair share of residual bandwidth among all applications, including flows of different priorities. Even IDFQ provides only weighted fairness, i.e. relative fairness, regardless of usage or residual bandwidth. In other words, these mechanisms cannot provide global fairness. In fact, EDCA even perform better than AEDCF in this aspect.

Besides, no mechanism consider QoS demand in the aspect of transmission rate, which describes the real applications' demand more precisely than to just define the priorities relationship. Guaranteeing that the high priority flows will get higher probability than lower priority flow may be not enough if the high priority flows demand is too high compare to the priority relation predefined, on the contrary, these

Table 2.5-1. Characteristic summation of QoS enhancement mechanisms

	EDCA	AEDCF	AFEDCF	IDFQ
Service differentiation based on priority	●	●	●	●
Service differentiation based on QoS satisfaction	○	○	○	○
Total throughput improvement	○	●	●	●
Fairness between the same priority flows	○	○	●	○
Weighted fairness between different priority flows	○	○	○	●
Absolute fairness between all flows	○	○	○	○

●: support, ○: not support

mechanism may be too unfair for the low priority flows, especially if the high priority flows' demand are not far more the lower ones'. The characteristic analysis of mechanisms above is organized in Table 2.5-1. below.

According to the observations above, in the next chapter, I will introduce a new mechanism based on the satisfaction of applications' transmission rate demand, and it will also achieve better global fairness among all the applications, while maintaining high total throughput.

Chapter 3
Satisfaction-based Media Access Control Scheme

In this chapter, the proposed media access control scheme is described, named Satisfaction-based Enhanced DCF (SEDCF). In the following, the description of SEDCF is separated to characteristic and assumption, parameters, and algorithm.

3.1 Characteristic and Assumption

SEDCF is capable of providing QoS guarantee for multimedia flows in the view of transmission rate satisfaction, and it ensures the global fairness among all flows while maintaining high total throughput.

In SEDCF concept, all the flows must provide their QoS demand by specifying their requirement transmission rate, not just specifying *AC* and get the information about priority relationships, and if the average transmission rate is higher than the required transmission rate previous defined, the flow is said to be satisfied. After a flow is satisfied, any other transmission of this flow is extra gift, regardless what *AC* this flow is. The concept of global fairness SEDCF provide is once the QoS flows are satisfied, the total residual bandwidth is shared fairly among all the flows, including QoS flows and best effort flows.

There are some assumptions and definitions below:

A) A node cannot transmit and receive frames simultaneously.

B) Mobility is not under consideration in SEDCF.

C) Every QoS flows must provide their QoS demand by specifying their requirement transmission rate, not just specifying *AC*.

3.2 Parameters

Here are some basic parameters in SEDCF need to specify:

A) Usage

Usage means the bandwidth which s already used by a QoS flow, measure in transmission rate.

B) Minima Required transmission rate (*MR*)

Every QoS flow must specify *MR*, which represent the QoS level of this flow more precisely than just specifying *AC*. As to best effort flows, *MR* is set to be zero, that is, best effort flows are always considered to be satisfied.

C) Measuring Time Interval (*Tupdate*)

In every system defined *Tupdate*, the situations of bandwidth allocation of all flows are measured in share degree, defined below.

D) Smoothing Factor (α)

The smoothing factor is to adjust the portion of importance degree of latest estimated share degree, defined below.

E) Share Degree (*SD*)

In every *Tupdate*, Share Degree (*SD*) of every flow is computed. *SD* means how well this flow has been treated except the minima request, which also represent how much residual bandwidth this flow has used. The *SD* of flow *i* at measuring time interval *j* $SD^j[i]$ is computed by the following equation:

$$SD^j[i] = \frac{Usage^j[i] - MR^j[i]}{BW^j}. \qquad (13)$$

where $Usage^j[i]$ and $MR^j[i]$ is the real bandwidth flow *i* used and the *MR* of flow *i* at measuring time interval *j*, respectively, and BW^j means the total network available bandwidth at measuring time interval *j*. The $SD^j[i]$ value is definitely between (-1, 1), and positive $SD^j[i]$ means the flow is satisfied at time interval *j*, while the negative $SD^j[i]$ value means the flow is not satisfied at j, which must be compensated later to ensure fairness.

Like measuring the network collision rate in AEDCF, in order to alleviate the impact of transient collisions, SEDCF also adopt EWMA mechanism to smoothen the estimated values. That is,

$$SD^j_{average}[i] = \left(1-\alpha\right) * SD^{j-1}_{average}[i] + \alpha * SD^j[i], \tag{14}$$

where $SD^j_{average}[i]$ and $SD^{j-1}_{average}[i]$ is the average *SD* value at measuring time interval *j* and *j+1*, respectively, and α is the smoothing factor here. The $SD^j_{average}[i]$ will be used in contention window adjustment and backoff timer decreasing procedure later.

3.3 Algorithm

The SEDCF scheme is separate to two phases below: contention window adjustment and backoff timer decreasing procedure, while the detail algorithm is described in these sub-sections below.

3.3.1 Phase 1 – Contention Window Adjustment Procedure

As in EDCA, contention window needs to be adjusted only after a successful transmission or an unsuccessful transmission. Hence, the whole contention window adjusting procedure is shown as follows.

1) *Adjusting CW after each successful transmission*

After each successful transmission, say flow *i*, in the original EDCA concept, the value of contention window must be reset to *CWmin*[*i*], but in SEDCF, only the flows which are not satisfied yet ($SD^j_{average}[i]$ is less than zero) have this right to do so and get more opportunity to transmit packet more, hoping for getting compensated. As to those flows which have already satisfied ($SD^j_{average}[i]$ is larger than or equal to zero), basically, their contention window should be decrease slower than unsatisfied flows' to release the transmission opportunity to other flow. Of course the decreasing potion of these satisfied flows' *CW* should refer to their $SD^j_{average}[i]$, *CWmin*[*i*] and *CWmax*[*i*]. Finally, the computed *CW* value should still be bounded between (*CWmin*[*i*], *CWmax*[*i*]), hence, the whale *CW* adjusting formula is derived below:

$$
\begin{aligned}
&if\left(SD_{average}^{j}[i]<0\right), \quad CW[i]=CW_{\min}[i] \\
&if\left(SD_{average}^{j}[i]\geq 0\right), \quad \begin{cases} CW[i]=CW[i]-\left(1-SD_{average}^{j}[i]\right)*\left(CW_{\max}[i]-CW_{\min}[i]\right) \\ CW[i]=\max\left(CW_{\min}[i],CW[i]\right) \end{cases}
\end{aligned} \tag{15}
$$

2) Adjusting CW after each unsuccessful transmission

After each unsuccessful transmission, say flow *i*, on the contrary to the situation after successful transmission, as long as this flow is satisfied ($SD_{average}^{j}[i]$ is larger than or equal to zero) now, its *CW* should be set to *CWmax*[*i*] to release the transmission opportunity to other flows. As to the unsatisfied flows ($SD_{average}^{j}[i]$ is less than zero), although it should get more transmission opportunity, its *CW* still should increase to avoid further collision base on the basic concept of IEEE 802.11 MAC scheme. Hence, the *CW* of unsatisfied flows should increase slowly, and the increasing potion is computed according to their $SD_{average}^{j}[i]$, *CWmin*[*i*] and *CWmax*[*i*]. Finally, the bounded procedure of *CW* is still necessary to keep *CW* would not be larger than *CWmax*[*i*]. The whale adjusting formula is in (16).

$$
\begin{aligned}
&if\left(SD_{average}^{j}[i]\geq 0\right), \quad CW[i]=CW_{\max}[i] \\
&if\left(SD_{average}^{j}[i]<0\right), \quad \begin{cases} CW[i]=CW[i]+\left(1+SD_{average}^{j}[i]\right)*\left(CW_{\max}[i]-CW_{\min}[i]\right) \\ CW[i]=\min\left(CW_{\max}[i],CW[i]\right) \end{cases}
\end{aligned} \tag{16}
$$

3.3.2 Phase 2 – Backoff Timer Decreasing Procedure

After the contention window is computed, if the flow *i* is in collision state or deferring state, the backoff timer should be randomly chosen from [1, 1+*CW*[*i*]] and start the decreasing procedure while sensing the channel is idle longer than AIFS[i], and the flow cannot attempt to transmit packet only after the backoff timer is decreasing to zero. Unlike AEDCF, in order to maintain the global fairness between all flows, even in backoff timer decreasing procedure, those unsatisfied flows ($SD^{j}_{average}[i]$ is less than zero) should decrease their *BT*[*i*] faster to zero, which can make more opportunity to transmit next time. In SEDCF, the FCR mechanism is used on unsatisfied flows, which decreasing *BT*[*i*] exponentially, that is

$$if\left(SD^{j}_{average}[i]<0\right),\ BT[i]=BT[i]/2\,. \qquad (17)$$

As to satisfied flows ($SD^{j}_{average}[i]$ is larger than or equal to zero), their *BT*[*i*] decrease slowly than unsatisfied flows do, which the reason is that they have already get their minima request and should release the media to other flows. Hence their *BT*[*i*] still decrease linearly as the EDCA mechanism, which the formula is

$$if\left(SD^{j}_{average}[i]\geq 0\right),\ BT[i]=BT[i]-SlotTime\,. \qquad (18)$$

Chapter 4
Performance Evaluation

In this chapter, the performance evaluations of SEDCF, AEDCF and AFEDCF will be proposed by using ns-2 simulator [11].

4.1 Simulation Environment

Besides using ns-2 simulator, other simulation environment is described as follows. IEEE 802.11a is adopted as the PHY layer, and detailed parameters are listed in Table 4.1-1, including total data rate, Slot_time, which is significant to the proposed mechanism.

Because SEDCF, AEDCF and AFEDCF are proposed based on EDCA of IEEE 802.11e, all the parameters used in IEEE 802.11e MAC layer to provide service differentiation are in Table 4.1-2 for the general simulations later. We generate three classes of traffic in our simulations, i.e., phone, video, and best effort flows, respectively. These three types of flows represent the highest, the second, and the lowest priority, accordingly. All three classes of flows send data with constant data rates of 160 bytes per 20 ms, 1280 bytes per 10 ms, and 200 bytes per 12.5 ms, respectively. The simulation time is 12 seconds. We assume that all flows are backlogged during the simulation time. We set the QoS demands for phone and video flows are both 50% transmission successful rate. That is, to be satisfied, the minima transmission rate requirement for phone and video flows are 32 Kbps and 512Kbps, respectively. Furthermore, based on [4], the smoothing factor and *Tupdate* is set to 0.8 and 5000 Slot_time, accordingly.

In order to increasing the network load, the number of nodes will increase gradually to simulation. All the nodes locate in the same Basic Service Set (BSS), and the diagram of the traffic is shown in Fig. 4.1-1, which is that every node sends three distinct flows to next node, and all the traffics are one-hop.

Table 4.1-1. Parameter settings of PHY layer

SIFS	16 μs
DIFS	34 μs
ACK size	14 bytes
Data rate	36Mbits/s
Slot_time	9 μs
CCA Timer	3 μs
MAC header	28 bytes
Modulation	16-QAM
Preamble Length	20 μs
RxTxTurnaround time	1 μs
PLCP header length	4 μs

Table 4.1-2. Parameter settings of IEEE802.11e MAC layer

Parameters	Phone Traffic	Video Traffic	Best Effort Traffic
CWmin	7	15	31
CWmax	600	800	1023
AIFS (μs)	25	34	43
PF	2	2	2
Packet Size (bytes)	160	1280	200
Packet Interval (ms)	20	10	12.5
Sending Rate (Kbit/s)	64	1024	128
Require Rate (Kbit/s)	32	512	0

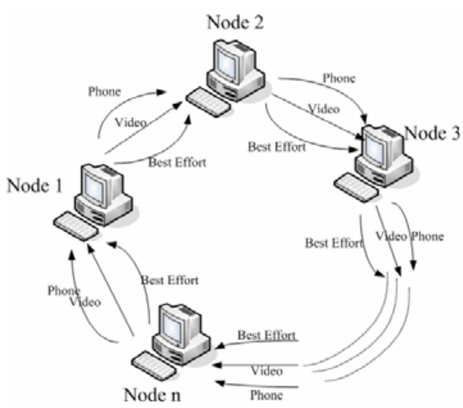

Figure 4.1-1. Simulation scenario

4.2 Performance Metrics

The performance metrics measured in the simulation include the network throughput, satisfaction index, and fairness index, which extended from [12] as defined below:

A) Network throughput (φ)

The summation of all flows' Usage, i.e.,

$$\varphi = \sum_i Usage(i), \quad \forall i \in F, \tag{19}$$

where *F* is the set of all flows, $Usage(i)$ is the usage of flow *i*.

B) Satisfaction index (η)

It only counts for QoS flows and is used to indicate the satisfaction degree. Its definition is

$$\eta = \left(\sum_{i \in F_Q} x_i \right)^2 \Big/ n \sum_i \left(x_i^{\,2} \right), \tag{20}$$

$$\text{where} \begin{cases} x_i = 1, & if \quad Uasge[i] \geq MR[i] \\ x_i = 1 + \dfrac{Usage[i] - MR[i]}{MR[i]}, & if \quad Uasge[i] < MR[i] \end{cases},$$

and, F_Q is the set of QoS flows, and *n* is the number of QoS flows, $Usage(i)$ and $MR[i]$ are the usage and the minima required transmission rate of flow *i*. The concept is, say flow *i*, once $Uasge[i] \geq MR[i]$, it is said satisfied, for satisfaction index, how much the media is over used by this flow is meaningless, so we do not have to consider $Usage(i)$ to compute x_i; while for the unsatisfied flows, how much more the usage needs for them to satisfy is very important, for satisfaction index, and the closer x_i to 1, the closer this flow is satisfied. And the final value of η is between 0 and 1 after indexing normalization. The larger the value η is, the better the overall satisfaction degree of QoS flows is.

C) Fairness index (κ)

It counts all flows and is to show how fair share about the residual bandwidth. Its definition is

$$\kappa = \left(\sum_{i \in F} y_i \right)^2 \Big/ m \sum_{i \in F} \left(y_i^{\,2} \right), \tag{21}$$

where $\begin{cases} y_i = Usage[i] - MR[i], & if \quad Usage[i] \geq MR[i] \\ y_i = 0, & if \quad Usage[i] < MR[i] \end{cases}$,

and F is the set of all flows, and m is the number of all flows, $Usage(i)$ and $MR[i]$ are the usage and the minima required transmission rate of flow i. On the contrary concept of satisfaction index, for any unsatisfied flow, the difference between its usage and its minima required transmission rate is not important, because fairness index is about residual bandwidth. As to satisfied flows, how much usage a flow over used is very significant, and the fairness we attempt is among all flows regardless of priority, so y_i is not concern about $MR[i]$ in denominator. The larger difference between all flows' y_i leads the worse fair share among all flows. After the indexing normalization, the final value of κ is also between 0 and 1. The larger the value κ is, the more fairly share of the residual bandwidth among all flows.

D) Mean Delay (δ)

The mean end-to-end delay is the time difference of a QoS packet from source to destination, i. e.,

$$\delta = Mean\_Delay(i), \ \forall i \in F \tag{22}$$

where F is the set of all flows, $Mean\_Delay(i)$ is the average value of all the end-to-end delay of flow i.

4.3 Simulation Result

In order to understand the performance of SEDCF precisely, the simulation results will be apart to phase by phase. That is, in the following sub-section, I will propose the baseline comparison of those related works, performance comparison of SEDCF phase 1 vs. AEDCF, and then SEDCF phase 1 vs. SEDCF phase 1+2, SEDCF phase 1 vs. AFEDCF, the delay comparison, finally is SEDCF phase 1 vs. AFEDCF.

4.3.1 Baseline Comparison of Related Works

First of all, we propose the baseline comparison of related works, which is include EDCA, AEDCF and AFEDCF, and IDFQ is not included because it is based on WFQ, which is totally different concept from others.

Fig. 4.3-1 shows the throughput of EDCA, AEDCF and AFEDCF. We can see that the throughput lines increase before there are 15 nodes, and decrease after that, because after there are 15 nodes, the total available bandwidth is not enough to handle

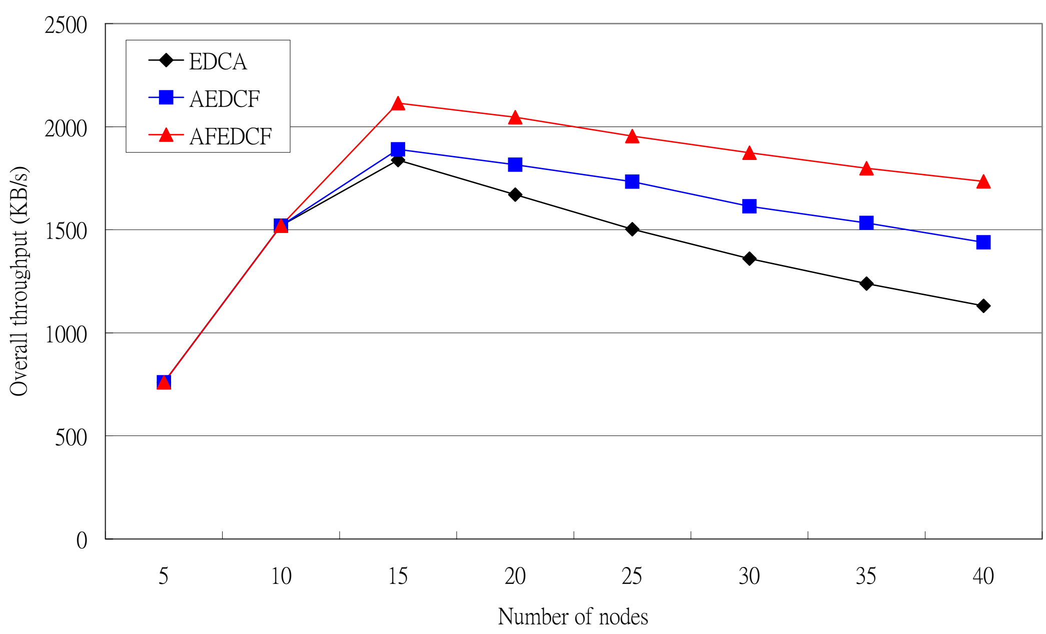

Figure 4.3-1. Overall throughput of EDCA, AEDCF and AFEDCF

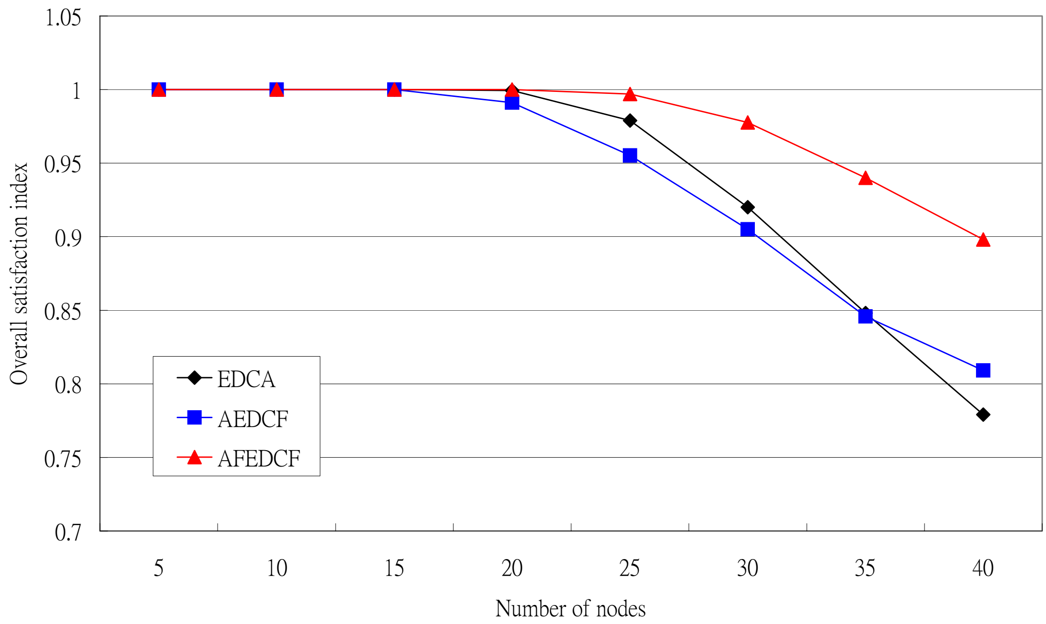

Figure 4.3-2. Overall satisfaction index of EDCA, AEDCF and AFEDCF

all the traffic. After there are 15 nodes, AFEDCF performs outstandingly in the related works.

There are overall satisfaction index and overall fairness index comparison shown in Fig. 4.3-2 and Fig 4.3-3. In the overall satisfaction index, it includes all the QoS flows, which mean it does not include best effort flows. All the satisfaction indexes start to degrade after there are 20 nodes, and EDCA and AEDCF have no big difference while AFEDCF is the outstanding method (over 0.9 even when there are 40 nodes) again.

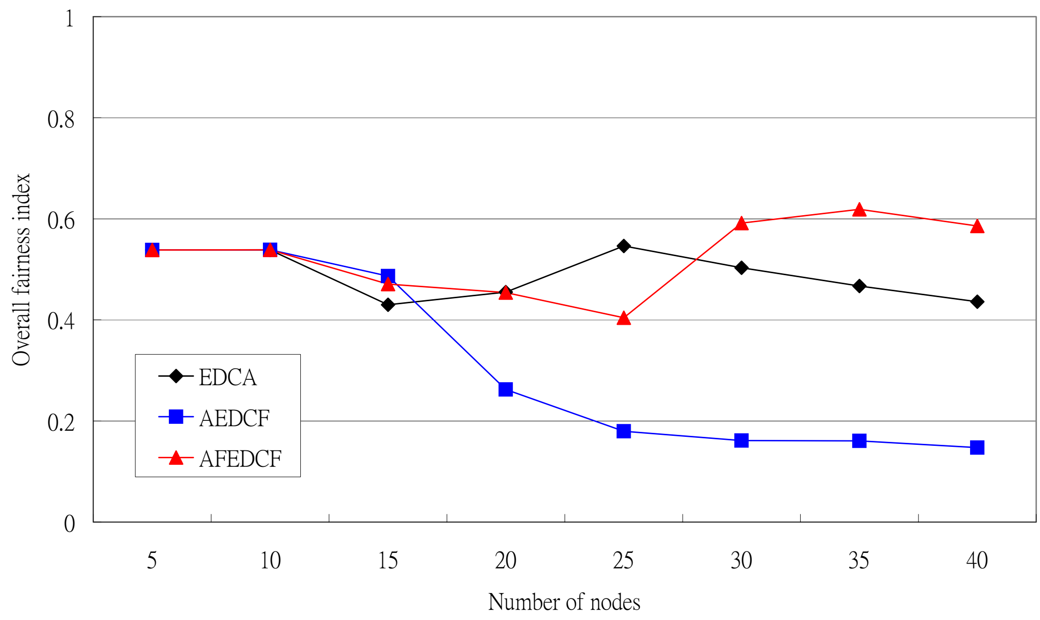

Figure 4.3-3. Overall fairness index of EDCA, AEDCF and AFEDCF

As to overall fairness index, after there are 15 nodes, AEDCF performs worst in three protocols, while EDCA and AFEDCF performs overall satisfaction index over 0.4. AFEDCF performs satisfaction index about 0.6 by indirectly achieving inter class fairness after there are 30 nodes.

4.3.2 SEDCF phase 1 vs. AEDCF

Since SEDCF phase 1 and AEDCF are similar to adjust contention window by a periodically estimated factor, and neither adapt the original backoff timer decreasing procedure, we propose their performance comparison first.

The throughputs of SEDCF phase 1 and AEDCF are shown in Fig. 4.3-4. We found that SEDCF phase 1 has better video-type flow and overall throughput than that of AEDCF. The reason is we lower the sending failure rate of QoS flows by adjusting the *CW* of satisfied flows more flexibly. And the throughout of phone–type flow is maintained the same as that of AEDCF. Furthermore, as the number of nodes

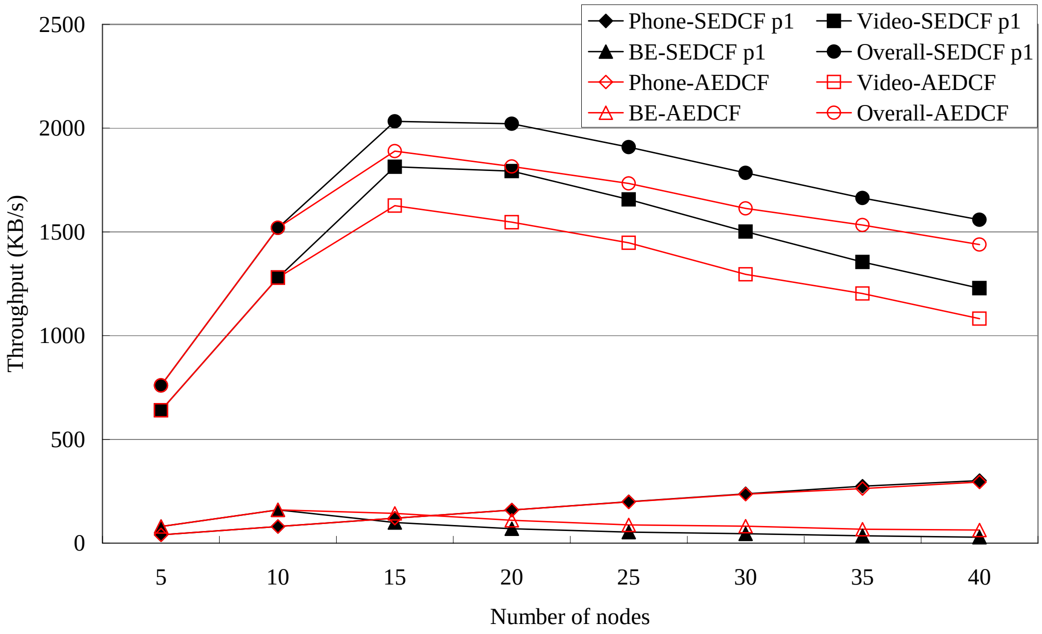

Figure 4.3-4. Throughput of SEDCF phase 1 and AEDCF

increasing, the throughput of phone-type flows keeps increasing; contrarily, the throughput of video-type flows starts to decreasing when the number of nodes is larger than 15. The reason is that in such a case that 15 nodes are backlogged to send data, the total required bandwidth to satisfy their QoS demands almost equals to the available bandwidth. Thus, more number of nodes, more number of the highest-priority flows (i.e., phone-type flows). In such situation, to guarantee phone-type flows' QoS demands, best effort-type and even video-type flows should sacrifice to release some bandwidth.

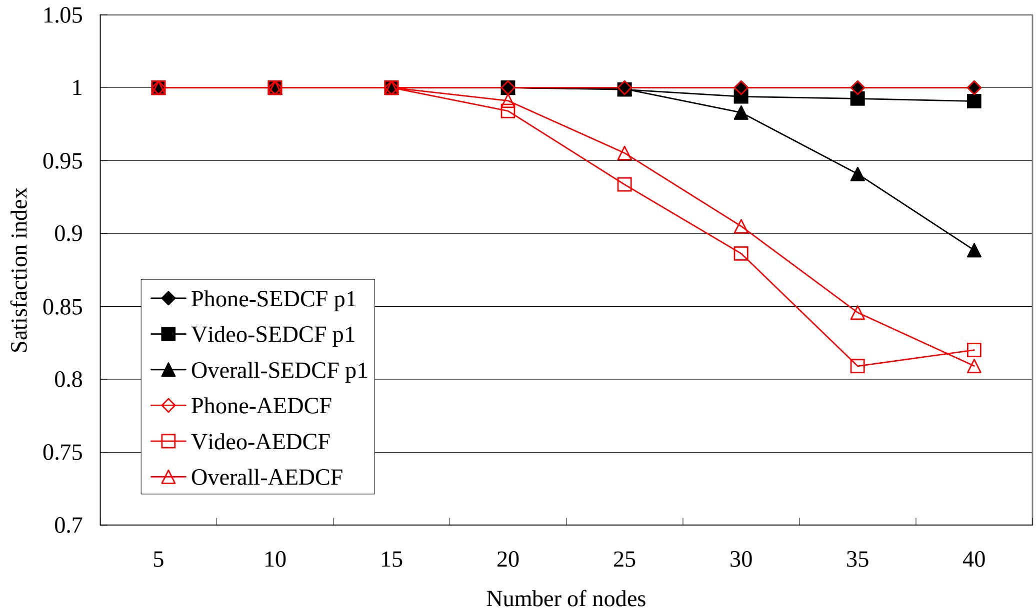

Figure 4.3-5. Satisfaction index of SEDCF phase 1 and AEDCF

Fig. 4.3-5 shows the satisfaction index of SEDCF phase 1 and AEDCF. We found both phone-type traffics of SEDCF phase 1 and AEDCF have same high value satisfaction index, however, the other satisfaction index of both SEDCF phase 1 and AEDCF are slightly decreasing while the number of nodes increase because the available bandwidth is no longer enough to satisfy the QoS demand of those Video-type flows. But because we take account of *SD* into *CW* adjustment, most of the flow satisfaction index of SEDCF phase 1 aggregate better than those of AEDCF, which leads the higher overall and video-type flow satisfaction index.

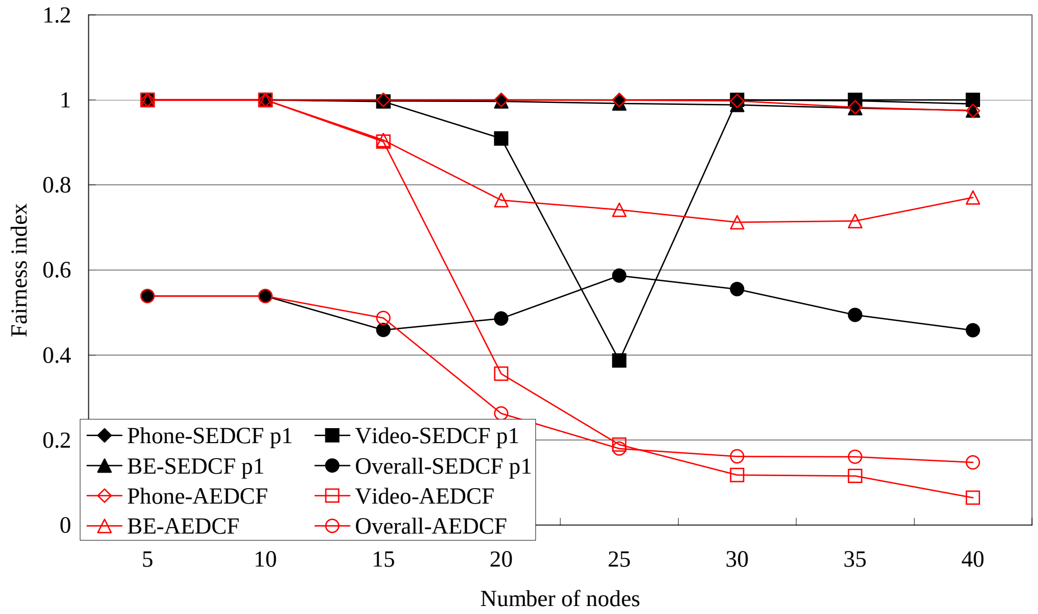

Figure 4.3-6. Fairness index of SEDCF phase 1 and AEDCF

The measured fairness index is shown in Fig. 4.3-6 Similar to satisfaction index, flows of phone-type have the best fairness index (more than 0.98) than others. As to the other flows of AEDCF, the fairness index is decreasing distinctly after the number of node is more than 15. While the other flows of SEDCF phase 1 have generally constant fairness indexes, which result from taking *SD* into account in adjusting *CW* provides well intra-class (local) and inter-class (global) fairness. But there is an exception while there are 25 nodes in topology, at this time, the available bandwidth can just no longer provide the video-type QoS demand (529.99 kbits/s per flow, which is very close to the require transmission rate 512 kbits/s per flow), which leads to the residual bandwidth of video-type flows distributed separately, and the fairness index is lower. But while the number of nodes keeps growing, the residual bandwidth of video-type flows aggregated soon although the QoS demand is no longer satisfied, so the fairness indexes afterward go back to higher value.

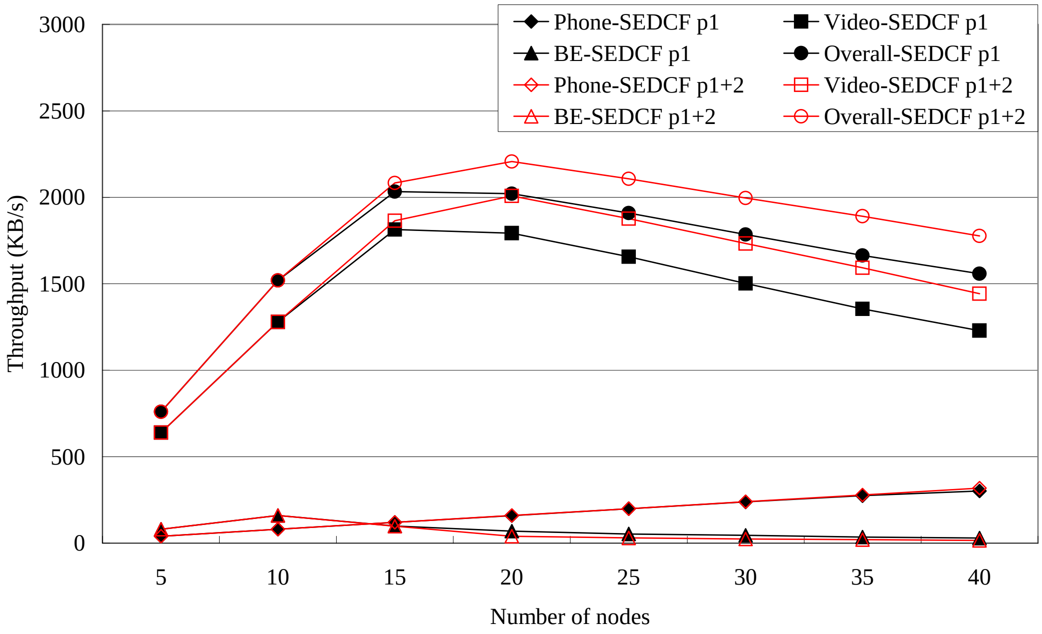

Figure 4.3-7. Throughput of SEDCF phase 1 and SEDCF phase 1+2

4.3.3 SEDCF phase 1 vs. SEDCF phase 1+2

After tuning of the contention window, the performance of adding the new backoff timer decreasing procedure should be evaluated. Fig. 4.3-7 shows the throughput comparison of SEDCF phase 1 and SEDCF phase 1+2. The phone-type flow throughput is still increasing gradually and stably while the number of node increase. As to video-type flows, after there are 20 nodes, the throughput of SEDCF phase 1+2 video-type flows start decreasing because of the total available bandwidth is running out for the total QoS demand of QoS flows, which makes the total throughput of SEDCF phase 1+2 reached high peak about 2200 KB/s, even higher than SEDCF phase 1 at all time. The reason is for the unsatisfied flows, SEDCF phase 1+2 provide even better protection by counting their backoff timer faster than satisfied flows, and since the best effort are always considered as satisfied, they can never benefited from the mechanism and start sacrifice to maintain QoS flows demand

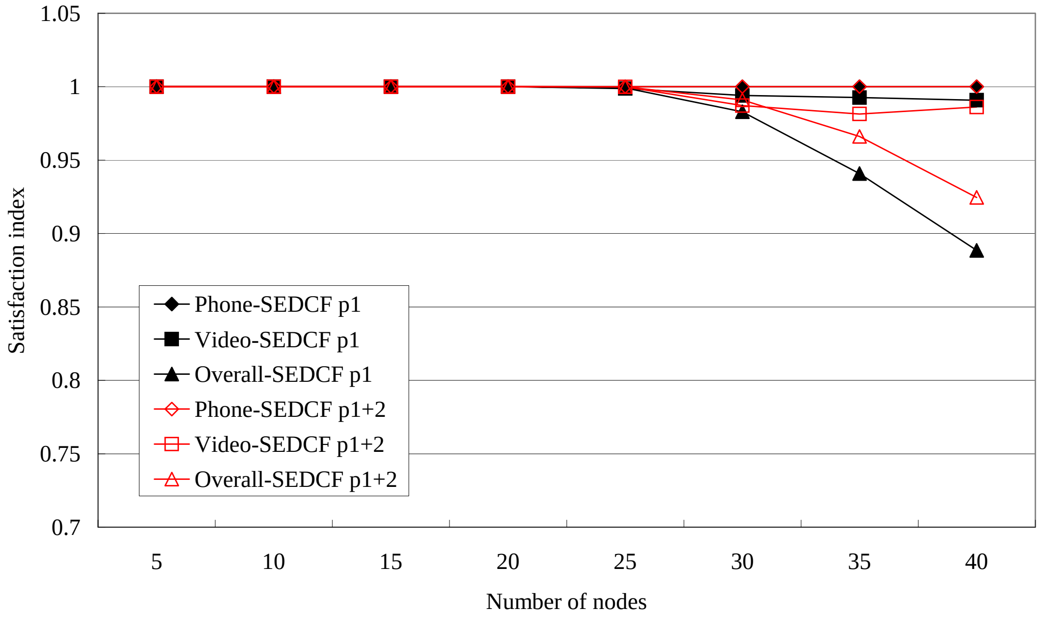

Figure 4.3-8. Satisfaction index of SEDCF phase 1 and SEDCF phase 1+2

earlier, which makes the highest throughput ever.

The satisfaction index of SEDCF phase 1 and SEDCF phase 1+2 are shown in Fig. 4.3-8. SEDCF phase 1+2 also performs well at this part. The satisfaction indexes of phone-type and video-type flows are close to 1, although there are lightly degrade as the number of node increase, they are never lower than 0.98. And the overall satisfaction index is never lower than 0.9 even when there are 40 nodes in the network. The reason is SEDCF provide almost perfect protection to QoS flows by taking *SD* into account to compute *CW* and decreasing *BT*.

Fig. 4.3-9 shows the fairness index of SEDCF phase 1 and SEDCF phase 1+2. For fairness between flows of the same priority, SEDCF phase 1 and SEDCF phase 1+2 almost performs the same, and the fairness indexes are almost 1 at all time except when there are 25 nodes in SEDCF phase 1 and there are 30 nodes in SEDCF phase 1+2. The fairness index drop reason of SEDCF phase 1+2 is the same as SEDCF phase1: at this time, the available bandwidth can just no longer provide the video-type

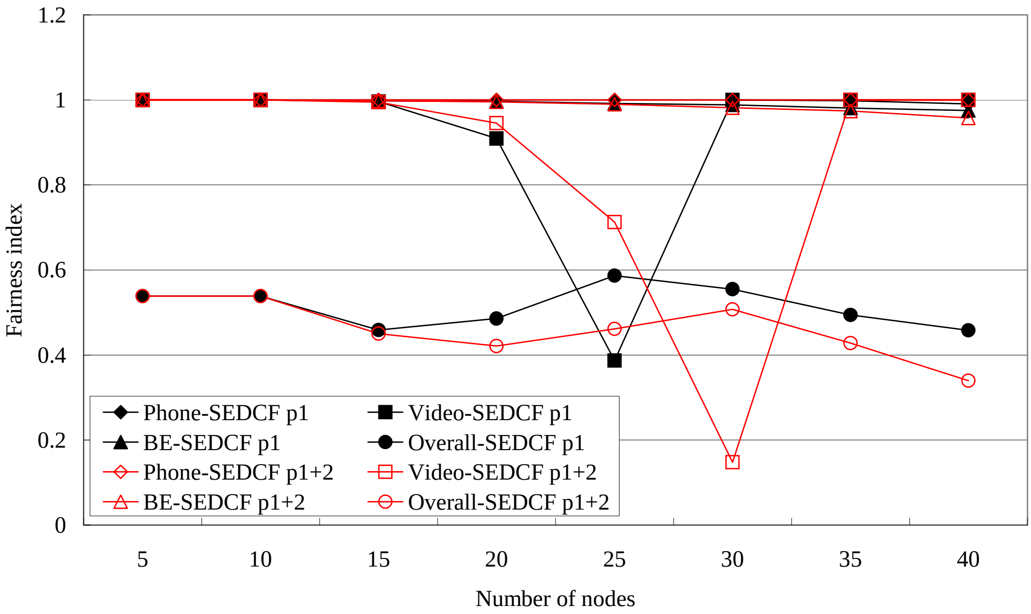

Figure 4.3-9. Fairness index of SEDCF phase 1 and SEDCF phase 1+2

QoS demand, which leads to the residual bandwidth of video-type flows distributed separately. However, the cause leads in the fairness index drop time difference is the QoS flows protection again: SEDCF phase 1+2 take *SD* into backoff timer decreasing procedure, and extend the video-type flow satisfied life to about there are 30 nodes (462.05 kbits/s per flow, which is very close to the require transmission rate 512 kbits/s per flow). For the overall fairness index, since best effort flows can not benefit from adding new backoff timer decreasing procedure and their *CWmin* and *CWmax* and other MAC parameters are week to get media access compare to QoS flows, the overall fairness index of SEDCF phase 1+2 is lower than SEDCF phase 1, even the intra-class fairness of best effort flows in SEDCF phase 1+2 is maintained higher than 0.95 at all time.

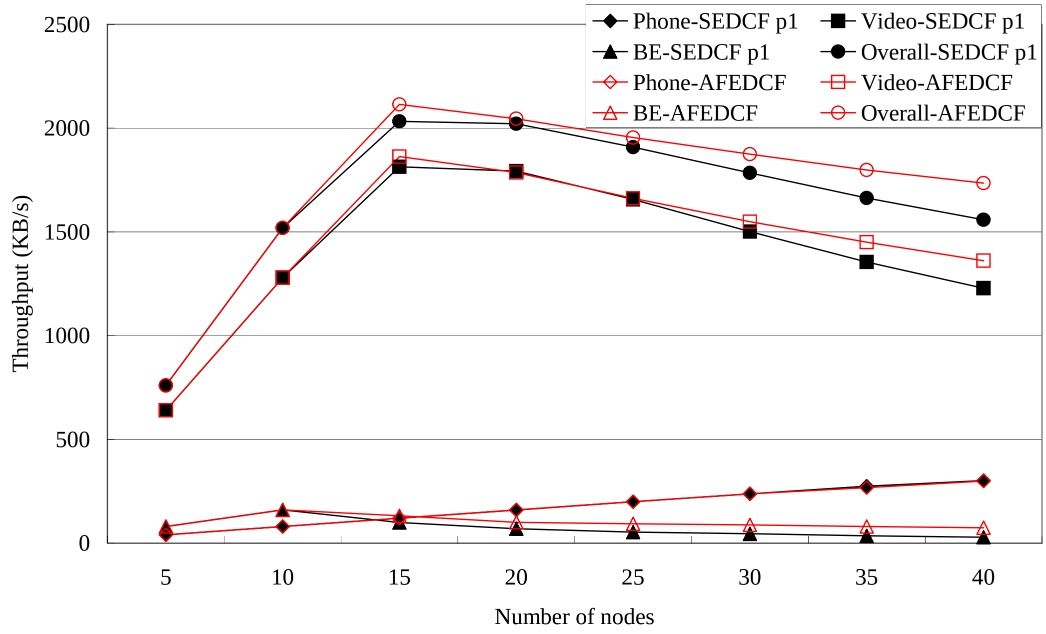

Figure 4.3-10. Throughput of SEDCF phase 1 vs. AFEDCF

4.3.4 SEDCF phase 1 vs. AFEDCF

In section 4.3.1, generally speaking, AFEDCF performs best in the view of throughput and satisfaction index, even in the view of fairness index. In section 4.3.3, SEDCF phase 1+2 achieves higher throughput but lower fairness index than SEDCF phase 1 does. Hence, in this section, we are going to exam the performance of SEDCF phase 1 and AFEDCF.

Fig. 4.3-10 shows the throughput of SEDCF phase 1 and AFEDCF. The overall trend of these results is the same as above, which also means the overall throughput is decreasing after there are 15 nodes, while the throughput of phone-type flow increases steady. The difference between the overall throughput of SEDCF phase 1 and AFEDCF is not really large, which means SEDCF phase 1+2 will achieve higher throughput than AFEDCF does. In basic, the throughput performance of SEDCF phase 1 and AFEDCF is similar.

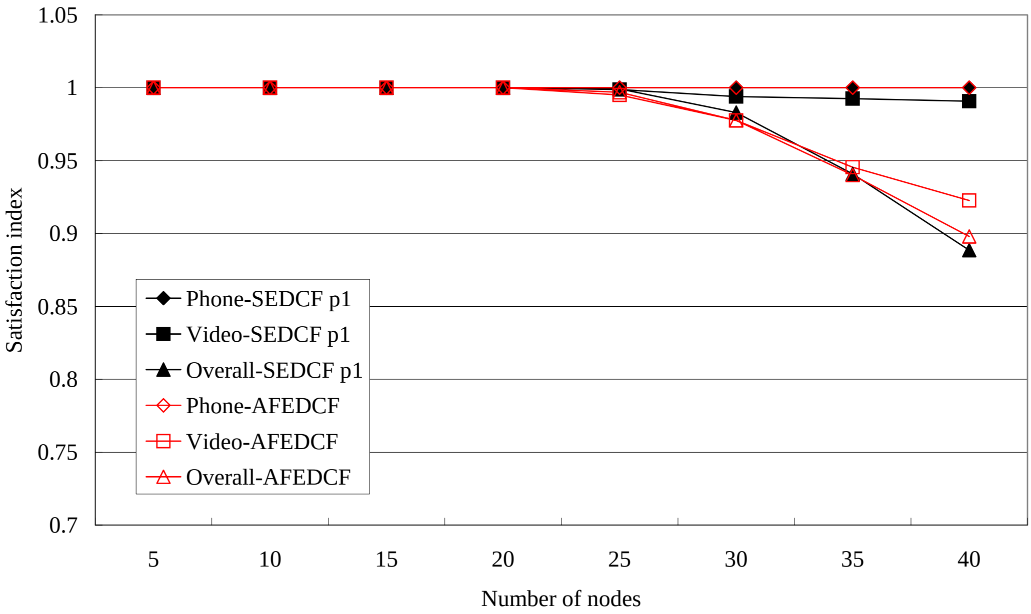

Figure 4.3-11. Satisfaction index of SEDCF phase 1 vs. AFEDCF

Similar as throughput performance, the difference between overall satisfaction indexes of SEDCF phase 1 and AFEDCF is not large. But as we can see in Fig. 4.3-11, the video-type is better protected by SEDCF phase 1, because SEDCF phase 1 take minima required transmission rate to adjust *CW*, while AFEDCF just provides priority-based QoS support to QoS flows, which may lead lower QoS flows (video-type flows) may sacrifice sooner under the consideration of required transmission rate.

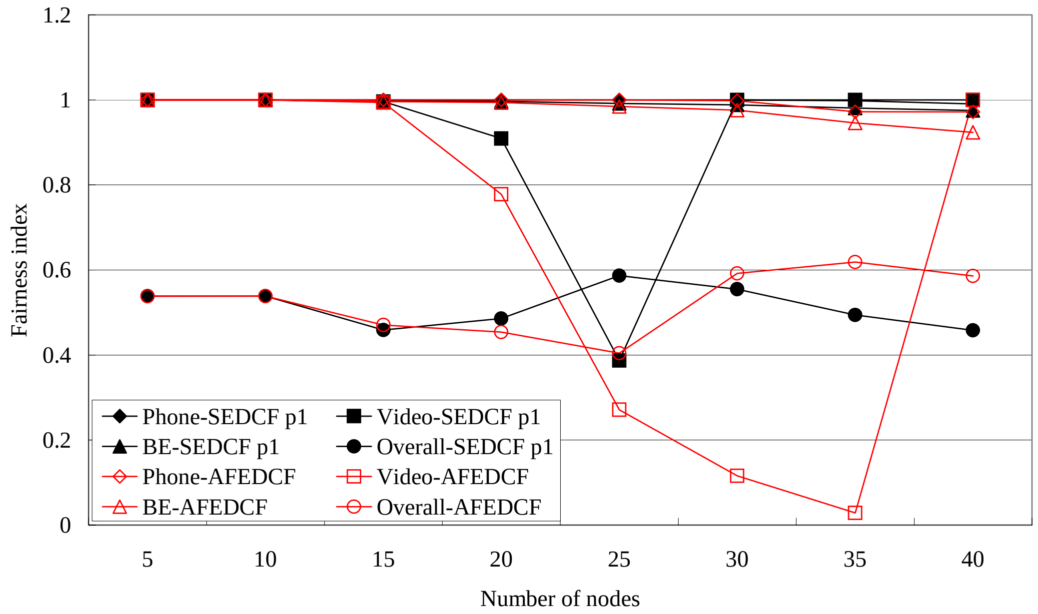

Figure 4.3-12. Fairness index of SEDCF phase 1 vs. AFEDCF

The fairness index of SEDCF phase 1 and AFEDCF is shown in Fig. 4.3-12. The performance of SEDCF phase 1 and AFEDCF are not different until there are more than 15 nodes. The fairness indexes of phone-type flows and best effort flows under two protocols are all over 0.9 no matter how many nodes are there. After there are 15 nodes, both the fairness indexes of video-type flows in SEDCF phase 1 and AFEDCF degrade sharply because the total bandwidth is running out, but fairness index of video-type flows in SEDCF phase 1 reach back to high value sooner (after there are 25 nodes), while the same situation happens in AFEDCF while there are 40 nodes. This represents SEDCF phase 1 provides better intra class fairness between video-type flows. As to overall fairness index, the performance of SEDCF phase 1 and AFEDCF are almost on a par, the two protocols both provide over certain degree of inter class fairness.

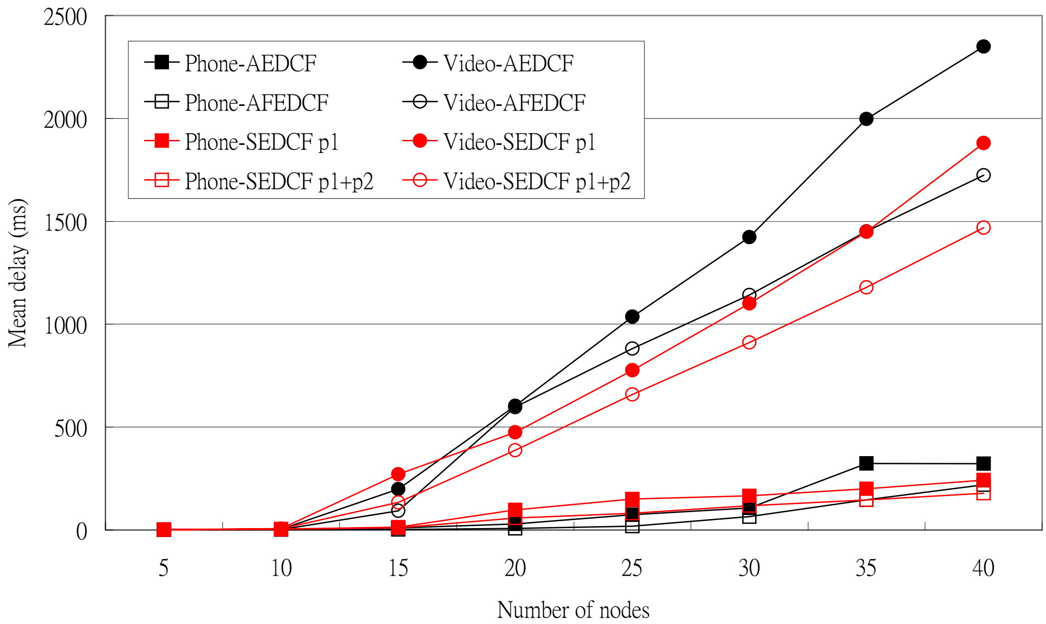

Figure 4.3-13. Mean delay of AEDCF vs. AFEDCF vs. SEDCF phase 1 vs. SEDCF phase 1+2

4.3.5 Mean delay of AEDCF vs. AFEDCF vs. SEDCF phase 1 vs. SEDCF phase 1+2

Here this section illustrates the comparison of mean end-to-end delay between AEDCF vs. AFEDCF vs. SEDCF phase 1 vs. SEDCF phase 1+2. As satisfaction index, mean delay is also calculated for QoS flows. As we can see in Fig. 4.3-13, the delay of phone-type flow is always bounded in certain area, even in the traffic load is high, which shows that the high priority flows is protected well no matter what protocol is adopted. As to video-type flows, after there are 15 nodes, the delay increase more sharply than that of flow flows because the total available bandwidth is running out, generally speaking, SEDCF performs better than AEDCF and AFEDCF and the difference is getting larger while the number of nodes is increasing, although SEDCF phase1 and SEDCF phase 1+2 are not designed for controlling delay.

Table 4.3-1. Parameter settings of IEEE802.11e MAC layer

Parameters	High Priority Traffic	Media Priority Traffic	Low Priority Traffic
CWmin	7	7	7
CWmax	1023	1023	1023
AIFS (μ s)	25	25	25
PF	2	2	2
Packet Size (bytes)	320	320	320
Packet Interval (ms)	5	5	5
Sending Rate (Kbit/s)	512	512	512
Require Rate (Kbit/s)	384	256	0

4.3.6 SEDCF phase 1 vs. SEDCF phase 1+2 vs. AFEDCF

In order to investigate SEDCF's performance of QoS guarantee more detail under admission control. This special scenario is upon the same ring topology and assumes at there are just 10 nodes in the ad hoc network to definitely be sure that each QoS flow's minimum demand can be guaranteed. The MAC parameters used in this scenario are listed in Table 4.3-1. There are still three flow priorities, and all are with the same MAC parameters, constant sending rate and same packet size to eliminate the defect of best effort flows, and the setting here is also compatible to original IEEE 802.11 MAC protocol. The major difference between priorities is the minima required transmission rate. To be satisfied, the QoS demand for high priority and media priority flows are set to be 384 Kbps and 256Kbps, i.e. 75% and 50% successful transmission rate, respectively. Furthermore, the smoothing factor and *Tupdate* is still 0.8 and 5000 Slot_time, accordingly.

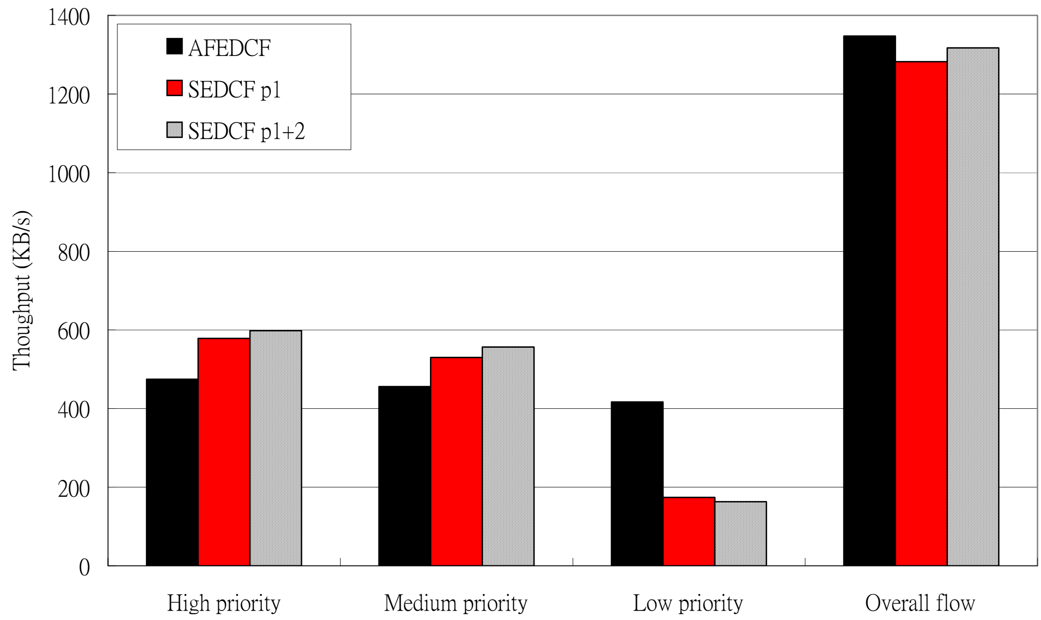

Figure 4.3-14. Throughput of SEDCF phase 1 vs. SEDCF phase 1+2 vs. AFEDCF

The network throughput and fairness index of SEDCF phase 1 and SEDCF phase 1+2 and AFEDCF are in Figs. 4.3-14 and 4.3-15. We found that the throughputs of these three mechanisms have no much difference in overall throughput, while SEDCF phase 1 and SEDCF phase 1+2 provide QoS guarantee to sacrifice best effort flows. The reason is that SEDCF no matter phase 1 or phase 1+2 integrates the concept of "satisfaction degree" and thus an unsatisfied flow has some opportunities to have a smaller *CW* to contend channel easier, even a fast backoff decreasing procedure. Besides, due to the same reason, SEDCF no matter phase 1 or phase 1+2 also has better intra-class and inter-class fairness indexes.

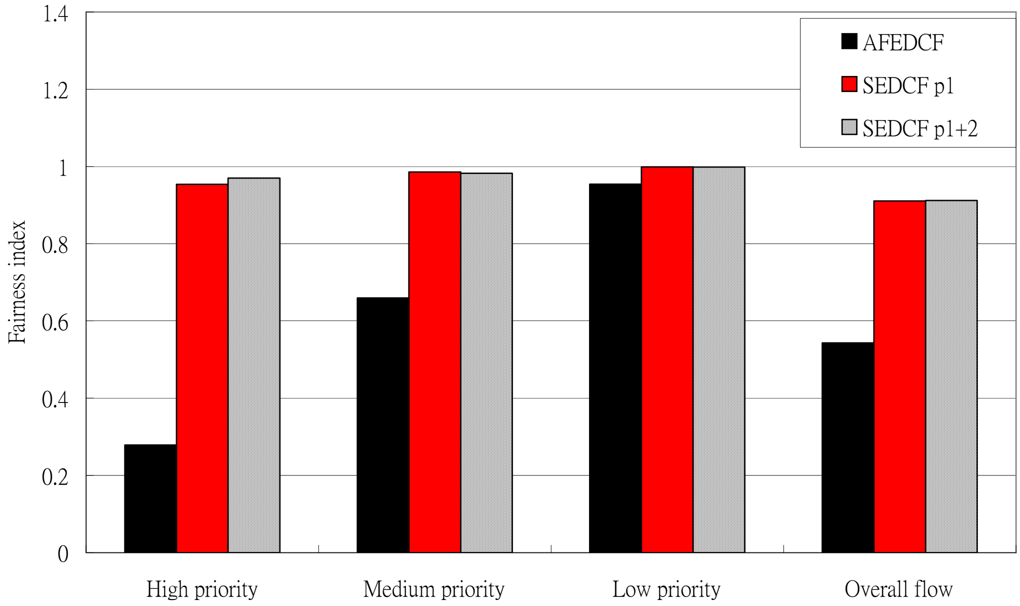

Figure 4.3-15. Fairness index of SEDCF phase 1 vs. SEDCF phase 1+2 vs. AFEDCF

As to fairness index analysis, SEDCF no matter phase 1 or phase 1+2 perform better than AFEDCF, since at last we consider *SD* in contention window adjusting procedure, the fair sharing of residual bandwidth is related to the required transmission rate. And AFEDCF do not consider about the required transmission rate, that is the reason the fairness index of AFEDCF is obvious lower than SEDCF phase 1 and SEDCF phase 1+2.

Chapter 5
Conclusion and Future Work

With the growth of wireless network and real-time multimedia application, the importance of Quality of Service (QoS) has been taken more and more seriously. However, currently the widest used wireless MAC scheme IEEE 802.11 DCF has no QoS support. In this thesis, we discussed the reason DCF cannot offer QoS demand and surveyed some QoS extension to original DCF, including official solution IEEE 802.11e EDCA, and other QoS enhancement based on EDCA, they are AEDCF, AFEDCF and IDFQ, which adapt contention window decision, backoff decreasing mechanism and inter frame space calculation, respectively. Unfortunately, except the service differentiation based on predefined priority, these mechanism provide either higher throughput or partial fairness between the same priority flows or weighted fairness.

Hence, a new media access scheme called Satisfaction Enhanced DCF (SEDCF) is proposed. SEDCF can provide not only priority relationship service differentiation but also satisfaction QoS demand, and global fairness of residual bandwidth, while maintaining high throughput. SEDCF algorithm is separated into two phases, and the performances of SEDCF and other QoS enhancement scheme is also evaluated. SEDCF phase 1 performs slightly better than SEDCF phase 1+2 in global fairness, while SEDCF phase 1+2 achieves higher total throughput than SEDCF phase 1. As to SEDCF compare to other mechanism, SEDCF achieve higher local and global fairness performance while maintaining or improving throughput.

As to the future work, the global fairness still has space to improve. And SEDCF can work with well designed admission control mechanism, because SEDCF provide absolute QoS satisfaction based on transmission rate, the network available bandwidth will definitely run out along the number of flows increasing. To operate in coordination with admission control, SEDCF can be applied in situations closer to real world network scenarios, even considering the nodes' mobility.

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