KSII TRANSACTIONS ON INTERNET AND INFORMATION SYSTEMS VOL.5, NO. 9, September 2011

Adaptive Packet Scheduling Scheme to Support Real-time Traffic in WLAN Mesh Networks

Rongbo Zhu[1], Yingying Qin[1] and Chin-Feng Lai [2]
[1] College of Computer Science, South-Central University for Nationalities
708 Minyuan Road, Wuhan 430074, China
[e-mail: rongbozhu@gmail.com, foreverqyy@yahoo.cn]
[2] Institute of Computer Science and Information Engineering, National Ilan University
Shen-lung Road, I-Lan, 260, Taiwan
[e-mail : cinfon@ieee.org]
*Corresponding author: Chin-Feng Lai

Received July 5, 2011; revised August 16, 2011; accepted September 5, 2011;
published September 29, 2011

Abstract

Due to multiple hops, mobility and time-varying channel, supporting delay sensitive real-time traffic in wireless local area network-based (WLAN) mesh networks is a challenging task. In particular for real-time traffic subject to medium access control (MAC) layer control overhead, such as preamble, carrier sense waiting time and the random backoff period, the performance of real-time flows will be degraded greatly. In order to support real-time traffic, an efficient adaptive packet scheduling (APS) scheme is proposed, which aims to improve the system performance by guaranteeing inter-class, intra-class service differentiation and adaptively adjusting the packet length. APS classifies incoming packets by the IEEE 802.11e access class and then queued into a suitable buffer queue. APS employs strict priority service discipline for resource allocation among different service classes to achieve inter-class fairness. By estimating the received signal to interference plus noise ratio (SINR) per bit and current link condition, APS is able to calculate the optimized packet length with bi-dimensional markov MAC model to improve system performance. To achieve the fairness of intra-class, APS also takes maximum tolerable packet delay, transmission requests, and average allocation transmission into consideration to allocate transmission opportunity to the corresponding traffic. Detailed simulation results and comparison with IEEE 802.11e enhanced distributed channel access (EDCA) scheme show that the proposed APS scheme is able to effectively provide inter-class and intra-class differentiate services and improve QoS for real-time traffic in terms of throughput, end-to-end delay, packet loss rate and fairness.

Keywords: Packet scheduling, real-time traffic, wireless mesh networks, wireless local area networks

---

This work was supported by the National Natural Science Foundation of China (No. 60902053), the Natural Science Foundation of Hubei Province of China (No. 2008CDB339), and the Science and Technology Research Planning of Educational Commission of Hubei Province of China (No. B20110803).

DOI: 10.3837/tiis.2011.09.001

1. Introduction

Supporting delay sensitive real-time traffic service is a challenging task in wireless local area network-based (WLAN) mesh networking [1][2], due to multiple hops, mobility and time-varying channel. WLAN mesh network uses wireless multi-hop networking to provide a cost-efficient way for community or enterprise users to have broadband Internet access and share network resources. In a typical WLAN extended service set (ESS) mesh network [3], mesh access points (MAPs) are connected to each other via one or multiple hops, and each MAP is responsible for forwarding packets for the end users associated to it as well as exchanging packets with neighboring MAPs based on the destinations of the packets [4][5][6]. And IEEE 802.11 stations (STAs) that do not have mesh capabilities can connect to MAPs in order to send data over the mesh network. MAPs have limited mobility and some of them have wired Internet connections, which become Internet gateways.

The distributed nature of WLAN mesh networks presents many challenges when facing the increasing demand for multimedia applications [7][8][9][10]. Furthermore, recent research results [11][12][13] show that WLAN mesh networks perform poorly on heavy traffic and mobile case, in particular for real-time flows subject to multiple hops and medium access control (MAC) layer control overhead, such as preamble, carrier sense waiting time and the random backoff period. WLAN mesh networks scalability problems pose additional constraints so that ensuring the required quality of service (QoS) parameters appears a challenging task even for a small number of hops [14]. Efficient allocation and management of resources [15], such as packet scheduling and throughput maximization, is a crucial element and an attractive and efficient way for interconnecting MAPs to form an efficient multihop WLAN mesh network supporting QoS.

In order to improve performance of WLAN mesh networks, current research work on traffic and packet scheduling mainly focuses on fair scheduling or routing optimization when supporting non-real time traffic [16][17]. In order to improve the support of multiple simultaneous flows, a novel opportunistic routing protocol was proposed in [17], which combines randomized opportunistic forwarding with opportunistic packet scheduling. Zong et al. [18] analyzed the impact of selfish and non-cooperative behaviors on the performance of packet scheduling algorithms using a mixed strategy game theoretic model. However, such schemes only took fair scheduling or routing optimization into consideration, and did not consider the distributed nature and medium access control (MAC) layer control overhead, such as preamble, carrier sense waiting time and the random backoff period, which make the current WLAN mesh network systems are likely to show low bandwidth utilization and poor performance for real-time traffic in multihop, noise interference and mobile environment. Another problem is that most existed schemes focus on the centralized scheduling algorithms in IEEE 802.16 mesh networks [19]. To the best of our knowledge, there has been very limited work on real-time scheduling in WLAN mesh networks.

Although a number of packet scheduling schemes have been proposed to improve the performance of WMNs, supporting real-time traffic is still a very challenging task due to its complexity and ill-posed nature and all of these works do not comprehensively consider the inter-class and intra-class service differentiation. And most existing research work has focused on fair scheduling or routing optimization. The overhead of packet transmission is generally in the WLAN mesh networks. However, in many cases the overhead of packet transmission will greatly degrade the system throughput. Another problem is that existed work always assumes that the packet length is fixed, or all traffic has the same packet length. However, in actual complex environment, due to the characteristics of traffic, different traffic always has different packet length, which greatly influences the performance of real-time traffic. Hence how to select appropriate packet length to satisfy the QoS requirements of real-time traffic through adaptive packet scheduling is an important task.

Motivated by the above reasons, in this paper, we concentrate on packet scheduling scheme to support real-time traffic in WLAN mesh networks. To provide QoS for real-time flows, an efficient adaptive packet scheduling (APS) scheme is proposed in WLAN mesh networks. Unlike existed schemes, the proposed APS in this paper aims to improve the system performance by guaranteeing service differentiation and adaptively adjusting the packet length to the current channel status, which takes both inter-class and intra-class into consideration. Moreover, APS does not deteriorate the network throughput with unnecessary control message exchanges which can cause severe throughput degradation over the wireless medium. The proposed APS classifies incoming packets by the IEEE 802.11e access class and then queued into a suitable buffer queue. APS employs strict priority service discipline for resource allocation among different service classes to achieve inter-class fairness. By estimating the received signal to interference plus noise ratio (SINR) per bit and current link condition, APS is able to calculate the optimized packet length with bi-dimensional markov MAC model to improve system performance. To achieve the fairness of intra-class, APS also takes maximum tolerable packet delay, transmission requests, and average allocation transmission into consideration to allocate transmission opportunity to the corresponding traffic.

The remainder of this paper is organized as follows. In Section 2, we discuss some related work. Section 3 describes the details of the proposed adaptive packet scheduling (APS) scheme. Section 4 presents and assesses the simulation results. Finally, Section 5 concludes the paper.

2. Related Work

The performance of WLAN mesh networks is degraded by the mobility, multi-hop, and interference and time varying property of the wireless channel. In fact, the IEEE 802.11n WLAN physical layers (PHYs) can provide multiple transmission rates up to 540 Mbps [20] by employing different modulation and channel coding schemes. However the MAC layer control overhead, such as the preamble, frame headers, carrier sense waiting time and the random backoff period, has great influence on the system throughput and QoS of real-time traffic. To address packet scheduling problem, there are two threads of research to improve the performance: MAC protocal analyzing and enhancement, packet scheduling and throughput maximization in multi-hop WLAN mesh networks.

On one hand, in order to enhance the QoS support in 802.11 WLANs, IEEE 802.11e [21] developed a new enhancement contention-based channel access mechanism called enhanced distributed channel access (EDCA) for service differentiation. And many literatures focus on modeling and analyzing the IEEE 802.11 carrier sense multiple access with collision avoidance (CSMA/CA) and EDCA protocol [22][23], which analyzed the throughput performance of IEEE 802.11 MAC protocol. Kong et al. [24] presented a three dimensional discrete time Markov chain model to describe IEEE 802.11e EDCA, which took into account the backoff timer, freeze and virtual collision policy. In our previous work [25], an efficient bi-dimensional markov model was proposed to reveal the essence of EDCA scheme for the station is under unsaturation at most case, and a model-based admission control (MBAC) scheme that performs real-timely at medium access control (MAC) layer was also proposed for the decision of accepting or rejecting requests for adding traffic streams to improve performance of WLANs. Focusing on the problem of improving QoS, an intelligent MAC model [26] for traffic scheduling located at the QoS-enhanced access point (QAP) is proposed. However, such schemes just concentrate on single hop WLAN, which can enhance the very limited performance of multihop WLAN mesh networks.

On the other hand, considering that stations in wireless mesh networks are in multi-hop and mobile environment and the characteristic of each link will change timely, it is effective to schedule traffic and packet appropriately which achieves the best transmission performance at each link. In [16], a novel traffic class differentiation scheme and a QoS-aware fair packet scheduling (QFPS) algorithm were proposed to fulfill the QoS provisioning in IEEE 802.16 wireless mesh networks. In order to maximize the system throughput in IEEE 802.16 broadband access networks with mesh topology, a linear chain network was considered and an optimal scheduling algorithm was proposed to stablish an analytical result on the length of the schedule for linear chain networks [27]. Focus on routing and packet scheduling problem in IEEE 802.16 mesh network, an efficient and interference-aware centralized routing tree algorithm was proposed in [28], which takes the number of interferences into consideration. A scheduling framework and a threshold mechanism for fair scheduling of elastic data flows were proposed in time division multiple access (TDMA) wireless mesh networks [29]. However, the schemes mentioned above are not suitable for CSMA/CA WLAN mesh networks. Zhang et al. [30] developed a fine-granularity transmission distortion model for the encoder to predict the quality degradation of decoded videos caused by lost video packets, and proposed a content-and-deadline-aware scheduling (CDAS) scheme for multi-session video streaming over multi-hop mesh networks. A cross layer weighted fair scheduling based on adaptive rate control (WFS-ARC) framework is proposed in [31]. And an opportunistic packet scheduling method and MAC scheme for controlling the throughput is proposed in [32]. However, the schemes in [31][32] just considered the case in one-hop WLANs. In recent work, a novel method of voice frame aggregation for wireless mesh networks was presented in [33], which automatically regulated the degree of aggregation by the congestion level on the wireless link. In [34], a cross-layer scheme that improves the scalability of WMNs using aggregation of MAC layer frames, which coupled the routing metrics to the channel state to improve the network performance. In multihop WLAN mesh networks, shorter end-to-end delay is expected to be offered for the traffic flows with time-urgent requirements. However, the schemes [29][30] just considers the voice traffic or saturation throughput case. Considering that, in multihop WLAN mesh networks, high-rate links heavily suffer from performance degradation due to the presence of overhead.

Based on the above observations, for the approaches of MAC protocol analyzing and enhancement, such schemes only can improve performance of single hop WLAN mesh networks. For the approaches of packet scheduling and throughput maximization, most schemes require extra implementation effort and modifications to the current 802.11 standard, or suffer from performance degradation due to the presence of overhead. Furthermore, most schemes are simple heuristic methods and lack theoretical analysis, which do not consider how to adaptive optimize the packet scheduling corresponding to the quantity and the distribution of frame arrivals, the communication types and transmission categories.

3. Proposed Scheme

3.1 System Description

The WLAN mesh network is assumed herein to be composed of many STAs and MAPs shown as in **Fig. 1**, where MAPs are interconnected through one or multiple hops. All of STAs and MAPs are equipped with backhaul interfaces which are used for backbone communication. Each MAP is equipped with client interfaces which are used for providing network access for STAs within their coverage area. Some MAPs are equipped with gateway functionality to enable connectivity to the wired Internet via wired link. All resources that the STAs access on the Internet can be accessed via any MAP with gateway functionality. Thus, the MAPs mainly relay mobile STAs traffic to and from the Internet through the gateway routers. Usually, the traffic in transit over the network demands differentiated QoS.

We assume that the network topology is relatively fixed, i.e. MAPs are relatively static and not removed from the network. STAs can move randomly in the network. Moreover, even though the quality perceived by STAs on different links may change over time, links are considered stable enough not to be torn down. Furthermore, we assume that there exist pre-computed routes for each STAs pair in the network, which can be achieved during the phase of network deployment by employing the node entry process prescribed in IEEE 802.11s mesh standard.

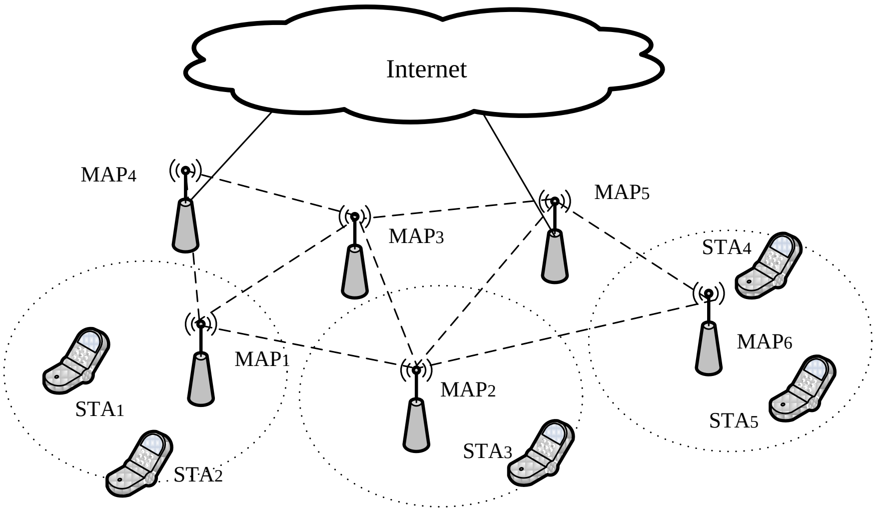

Fig. 1. An example of WLAN mesh network.

3.2 Packet Length Optimization

As mentioned above, the traffic in transit over the network demands differentiated QoS, and IEEE 802.11e introduces a new coordination function called the Hybrid Coordination Function (HCF). Within the HCF there are two access mechanisms: a non-contended polling channel access scheme HCF Controlled Channel Access (HCCA) and a prioritized contention scheme EDCA. Both EDCA and HCCA define Traffic Classes (TC). However, since HCCA support is not mandatory currently available APs do not support it. Hence, in this paper, we just consider the EDCA scheme in MAC layer. With EDCA, high priority traffic has a higher chance of being sent than low priority traffic: a STA with high priority traffic waits a little less before it sends its packet, on average, than a station with low priority traffic.

The architecture of the proposed adaptive packet scheduling scheme is sketched in **Fig. 2**. Incoming packets of network layer are classified by the EDCA classifier and then queued into a suitable Buffer queue, and the EDCA will provide traffic differentiation which pulls packets from buffers, according to the traffic class.

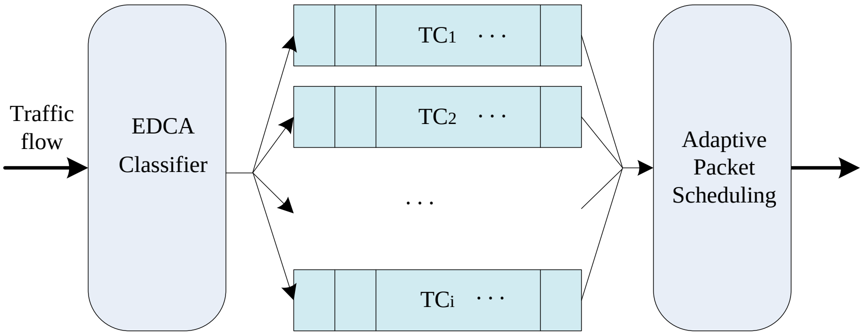

Fig. 2. The architecture of the proposed adaptive packet scheduling scheme.

Then the differentiated traffic will enter in the proposed adaptive packet scheduler for transmission scheduling. In order to decrease the back-off contention time and improve the utility of channel, the incoming MAC frames are first classified according to their destination address. And the proposed adaptive packet scheduling scheme will aggregate several MAC service data units (MSDUs) to form the data payload of a large MAC protocol data unit (MPDU) with the same destination, which will combine the PHY header and the MAC header together and decrease physical layer overhead. Then several MSDUS will form an aggregated MSDU (A-MSDU).

In order to aggregate multiple frames into a single burst at the MAC level efficiently, the proposed adaptive packet scheduling scheme can reduce encapsulation, back-off overhead and increase the system throughput. Meanwhile, the optimized length of aggregated frame should provide to ensure the fairness of different traffic. Following our previous work [25][26], the bi-dimensional markov model is employed to depict the EDCA differentiated service and calculate the optimized frame length accurately.

Assumption that there are N priorities class in the system, and let n_i denote the number of active queues in the priority i class. In steady state, each remaining station in the priority i class transmits a packet with probability τ_i, For transmission occurs either when the backoff time counter is zero, independent of the backoff state, or if the station, after some idle period in idle state, conducts a first transmission from first transmission state. So we have:

$$\tau_i = \sum_{j=0}^{m_i+r} b_{i,j,0} + P_{Ft} = \frac{1-[(1-P_{RTS}P_{CTS}P_{MPDU}P_{ACK})p_i]^{m_i+r+1}}{1-p_i(1-P_{RTS}P_{CTS}P_{MPDU}P_{ACK})} b_{i,0,0} + P_{Ft} \quad, i \in [0, N-1] \tag{1}$$

where P_{RTS}, P_{CTS}, P_{ACK} and P_{MPDU} denote the probability that the request to send (RTS), clear to send (CTS), acknowledge (ACK) control frames and a MPDU frame be transmitted with success. The probabilities τ_i are functions of p_i, q, P_{Ft}, $P\{B|I\}$, $P\{I|-1\}$, P_I, m_i, P_{rts}, P_{cts}, P_{ack} and P_{mpdu}, where the probability p_i ($i \in [0, N-1]$), a transmission from a station in the priority i class collides in any time slot, q is the probability that the station enters in backoff with exactly one packet to transmit and no new packets arrive until current packet is transmitted successfully, $P\{F|I\}$ is the probability that the channel is sensed idle after receiving a packet from the 'Idle' state, $P\{B|I\}$ is the probability that the channel is sensed busy after receiving a packet from the idle state and $P\{I|-1\}$ is the probability that station enters in 'Idle' state from the post-backoff state.

A collision occurs when at least one other station, which can be from any priority class, transmits. Hence, we have the probability p_i ($i \in [0, N-1]$), a transmission from a station in the priority i class collides in any time slot:

$$p_i = 1-(1-\tau_i)^{n_i-1}\prod_{j=0,j\neq i}^{N-1}(1-\tau_j)^{n_j}\ ,\ i\in[0,N-1] \tag{2}$$

and the probability p_b that the channel is busy:

$$p_b = 1-\prod_{j=0}^{N-1}(1-\tau_j)^{n_j}\ ,\quad i\in[0,N-1] \tag{3}$$

The set of equations (1), (2) and (3) represent a nonlinear system of equations with $2N$ unknowns τ_i and p_i. Assuming knowledge of P_{Ft}, $P\{B|I\}$, $P\{I|-1\}$ and P_I, it can be proved that they have a unique solution and can be solved by using numerical techniques.

The probability that station enters in 'Idle' state from post-backoff state:

$$P\{I|-1\} = e^{-\lambda(W_{i,0}+1)\bar{\varepsilon}/2} \tag{4}$$

where $\bar{\varepsilon}$ is the average time between successive counter decrements:

$$\bar{\varepsilon} = (1-p_b)\sigma + p_b p_i(T_{s,i}+\sigma) + p_b[1-p_i(T_{c,i}+\sigma)] \tag{5}$$

where $T_{s,i}$ is the average length of successful slots, $T_{c,i}$ is the average length of failed slots and σ is one system slot time.

The probability q, as an approximation, we assume that the probability of entering in backoff with one packet is one. Then we can get:

$$q = e^{-\lambda E(bk)} \tag{6}$$

where $E(bk)$ is the expected value of the time that the packet spends in backoff before successfully transmitting the packet.

$$E(bk) = \frac{\bar{\varepsilon}\left(\sum_{j=0}^{m_i+r} W_{i,j}[p_i(1-P_{RTS}P_{CTS}P_{MPDU}P_{ACK})]^j\right)}{2} + \frac{2T_{C,i}p_i-\bar{\varepsilon}}{2(1-p_i)}(1-p_i^{m_i+r+1}) \tag{7}$$

The probabilities $P\{B|I\}$, $P\{F|I\}$, P_I and P_{Ft} can be calculated as follows:

$$P\{B|I\} = p_b p_i(1-e^{-\lambda T_{s,i}}) + p_b(1-p_i)(1-e^{-\lambda T_{c,i}}) \tag{8}$$

$$P\{F|I\} = (1-p_b)(1-e^{-\lambda\sigma}) \tag{9}$$

$$P_I = \frac{qP\{I\,|\,-1\}[(1-p_i)P_{RTS}P_{CTS}P_{MPDU}P_{ACK}P_{Ft} + (1-p_i)(1-P_{RTS}P_{CTS}P_{MPDU}P_{ACK})\sum_{j=0}^{m_i-1+r} b_{i,j,0} + b_{m_i+r}]}{P\{F\,|\,I\} + P\{B\,|\,I\}} \tag{10}$$

$$P_{Ft} = P_I \cdot P\{F\,|\,I\} \tag{11}$$

The RTS and CTS control frames must be transmitted at one of the rates of the basic service set (BSS) {6 Mbps, 12 Mbps, 24 Mbps} so that they can be decoded by all the stations in the same network. The positive acknowledgment (ACK) control frame must be transmitted using the basic rate that is less than or equal to the rate of the data frame it is acknowledging. In oder to calculate the propbalities P_{RTS} , P_{CTS} , P_{ACK} and P_{MPDU} that is related to the phsycal layer (PHY) modulation, in the following analyze, we just consider the IEEE 802.11a PHY for convenience. Actually, the similar way can be used to analyze other PHY without much difficulty. In IEEE 802.11a PHY, eight different data transmission rates are supported by using M-ary quadrature amplitude modulation (QAM). In order to reduce the error probability, convolutional coding and data interleaving techniques are used for forward error correction (FEC). There are three different code rates used in the standard.

The symbol error probability $P(\gamma)$ with signal to noise ratio (SNR) γ for M-ary QAM is (M =4, 16, 64):

$$P(\gamma) = 1 - \{1 - [2 \cdot (1 - \frac{1}{\sqrt{M}}) \cdot Q(\sqrt{\frac{3}{M-1} \cdot \gamma})]\}^2 \tag{12}$$

With Gary coding, we can get the bit error ratio (BER) P_e for M-ary QAM:

$$P_e = P(\gamma) \cdot \frac{1}{\log_2 M} \tag{13}$$

Quadrature phase shift keying (QPSK) and 4-ary QAM are identical, and the BER for binary phase shift keying (BPSK) modulation is:

$$P_e = Q(\sqrt{2 \cdot \gamma}) \tag{14}$$

where the Q-function is the tail of the Gaussian distribution.

Let P_R be the BER for IEEE 802.11a with transmission rate R , With FEC, under the assumption of binary convolutional coding and hard-decision Viterbi decoding with independent errors at the channel input the union bound for P_R is:

$$P_R < \sum_{k=d_{free}}^{\infty} c_k P_k(\gamma) \tag{15}$$

where d_{free} is the free distance of the convolutional code, c_k is the total number of error events with k bit errors, and P_k is the probability that an incorrect path at distance k from the correct path being chosen by the Viterbi decoder.

$$P_k(\gamma)=\begin{cases}\frac{1}{2}\binom{k}{i}(P_e)^{k/2}(1-P_e)^{k/2}+\sum_{i=1+k/2}^{k}\binom{k}{i}(P_e)^{i}(1-P_e)^{k-i}, k \quad is \quad even\\ \sum_{i=(1+k)/2}^{k}\binom{k}{i}(P_e)^{i}(1-P_e)^{k-i}, \qquad k \quad is \quad old\end{cases} \tag{16}$$

The RTS and CTS control frames must be transmitted at one of the rates of the basic service set (BSS) {6 Mbps, 12 Mbps, 24 Mbps} so that they can be decoded by all the stations in the same network. The positive acknowledgment (ACK) control frame must be transmitted using the BSS basic rate that is less than or equal to the rate of the data frame it is acknowledging.

Assuming that the errors inside of the hard decision Viterbi decoder are interdependent, then Pursley and Taipale have shown that the upper bound for a successful transmission of a frame with l octets is given by:

$$P(l,\gamma)<[1-P_e(\gamma)]^{8l} \tag{17}$$

For a block-fading channel, we can get:

$$P(l,\gamma)=\int_{\gamma_{\inf}}^{\infty}[1-P_e(\gamma)]^{8l}p(\gamma)d\gamma \tag{18}$$

where $P_e(\gamma)$ is a function that depends on γ and the PHY mode. The lower limit of the definite integral is chosen so that

$$[1-P_e(\gamma)]^{8l}\leq 1, \qquad \gamma\geq\gamma_{\inf} \tag{19}$$

Considering a Rayleigh fading channel, the probability distribution function (pdf) of the signal to interference plus noise ratio (SINR) per bit at the Viterbi decoder input is of gamma kind. Then we can get:

$$p(\gamma)=\frac{1}{\Gamma(L)}\left(\frac{1}{\bar{\gamma}}\right)^{L}(\gamma)^{L-1}\exp\left(-\frac{\gamma}{\bar{\gamma}}\right), \qquad \gamma>0 \tag{20}$$

where γ is the average SINR per bit at the Viterbi decoder input and L is the number of receiver diversity branches.

Then we can get $P_{RTS}(m)$, $P_{CTS}(m)$, $P_{MPDU}(m)$ and $P_{ACK}(m)$ at different PHY mode. In the following, we just give the equations with 6Mbps binary phase shift keying (BPSK). Actually, the similar way can be used to calculate the probabilities at other PHY mode easily.

$$P_{RTS}=P(L_{RTS},\gamma)=P(25.75,\gamma) \tag{21}$$

$$P_{CTS}=\frac{P(L_{RTS}+L_{CTS},\gamma)}{P_{RTS}}=\frac{P(45.5,\gamma)}{P(25.75,\gamma)} \tag{22}$$

$$P_{MPDU} = \frac{P(L_{RTS} + L_{CTS} + L_{MPDU}, \gamma)}{P_{CTS} P_{RTS}} = \frac{P(45.5 + L_{MPDU}, \gamma)}{P(45.5, \gamma) P(25.75, \gamma)} \tag{23}$$

$$P_{ACK} = \frac{P(L_{RTS} + L_{CTS} + L_{MPDU} + L_{ACK}, \gamma)}{P_{CTS} P_{RTS} P_{MPDU}} = \frac{P(45.5 + L_{MPDU} + L_{ACK}, \gamma)}{P(45.5, \gamma) P(25.75, \gamma) P(45.5 + L_{MPDU}, \gamma)} \tag{24}$$

Then we can get the normalized throughput S_i ($i \in [0, N-1]$) for the priority i class. Let $P_{tr}(i)$ ($i \in [0, N-1]$) be the probability that there is exactly one transmission from the tagged station in the service i class in the considered slot time. Let $p_{s,i}$ ($i \in [0, N-1]$) denote the probability that a successful transmission occurs in a slot time for the priority i class. We have:

$$P_{tr}(i) = \tau_i (1-\tau_i)^{n_i - 1} \prod_{j=0, j \neq i}^{N-1} (1-\tau_j)^{n_j} \tag{25}$$

$$p_{s,i} = n_i P_{tr}(i) \tag{26}$$

$$S_i = \frac{p_{s,i} E(L_i)}{(1-p_b)\sigma + p_{s,i} T_{s,i} + (p_b - p_{s,i}) T_{c,i}} \tag{27}$$

where L_i denotes payload size of priority i class.

For the system, we can get the total normalized throughput S:

$$S = \sum_{i=0}^{N-1} S_i = f(\gamma, L_i, N) \tag{28}$$

the total throughput S depends on the channel status and transmission rate, and S can be expressed as a function of the SINR γ per bit, MPDU packet length L_i and active flows number of each priority class.

It's clear that the objective is to achieve maximum throughput with the constraints mentioned above, as:

$$\max S = \max_{L_i^*} \sum_{i=0}^{N-1} S_i \,. \tag{29}$$

If given active flows number of each priority class and SINR γ per bit, we can get the optimized MPDU packet length L_i to the maximum total throughput at each PHY mode. In fact, the knowledge of each priority class and SINR γ per bit is known to each STA and active flows number of each priority class is known to MAP. Hence, we can improve the system performance by guaranteeing inter-class, intra-class service differentiation and adaptively adjusting the packet length.

Considering that in IEEE 802.11e the data frame has a 2 octets new field named QoS control field as shown in **Fig. 3**, which is used in uplink to indicate station's traffic category (0-7 bit) or traffic streams (8-15 bit), the knowledge of active flows number of each priority class can be transmitted by MAP in this field to every STA in downlink.

Octets: 2	2	6	6	6	2	6	2	0-2312	4
Frame Control	Duration/ ID	Address 1	Address 2	Address 3	Sequence Control	Address 4	QoS Control	Frame Body	FCS

Fig. 3. IEEE 802.11e MAC frame format.

In the proposed adaptive packet scheduling scheme is very simple, and the transmitter estimates the received SINR per bit γ by using the CTS frame. After receiving the active flows number n_i of each priority class i, the system will calculate the optimized packet length L_i^* with expression (29) to improve throughput. Then multiple frames will be aggregated into a single burst frame to be transmitted.

3.3 Adaptive Packet Scheduling

Packet scheduling is of big significance for the schedules in the sense that it should balance the load of multiple traffic flows in order to fulfill the objective of fairness. The proposed adaptive packet scheme guarantees a uniform QoS level in terms of maximum tolerable packet delay $D_{\max,i}$, inter-class and intra-class fairness scheduling for different priority real-time traffic to exercise a discipline to satisfy the diverse QoS requirements.

In the proposed adaptive packet scheme, when there is a flow with packet delay close to $D_{\max,i}$, the flow is considered to have the higher priority to send packet. Each STA has a virtual time parameter that is equal to the start time tag of the serving packet. The virtual time tag that used by the scheduler to make packet selection decision. Each packet is assigned a start tag S_i^k as the finish tag F_i^{k-1} of last incoming packet if the corresponding flow has queued packets. The value is equal to the summation of start tag and the result of packet length divided by the flow weight. Then we can get start tag S_i^k and finish tag F_i^k as:

$$S_i^k = F_i^{k-1} \tag{30}$$

$$F_i^k = S_i^k + L_i^k / w_i \tag{31}$$

where w_i is the weight of each flow i. We use some MAC management messages specified in IEEE 802.11e to transmit the required information. Each STA first transmits a request message to the MAP before transmitting a traffic class, which contains the traffic class information of the stream and the MAC address of this STA. The MAP collects information of load conditions from each STA to estimate the radio performance. Consider the case where a WLAN mesh network is operated in infrastructure mode and each subscriber makes a contract with a wireless internet service provider to establish a service profile. The weight w_i represents the relative fair share of each traffic flow. For example, a station which has the weight of two has the right to occupy a channel twice as long as the station which has the weight of one for a given time interval. Let us define the weight and fair index of the i th flow as w_i and f_i, respectively. In order to provide weighted fair sharing among traffic flow, f_i should be in proportion to w_i:

$$f_i = \frac{w_i}{\sum_j w_j} \tag{32}$$

to minimize the bias of fair index, a smooth factor α are introduced to smoothen the values. At the nth transmission interval, the fair index is updated as:

$$f_i'(n) = (1-\alpha) f_i(n) + \alpha f_i(n-1)\,. \tag{33}$$

In most scenarios, some traffic flows demand pre-emption on current wireless resources against others. Therefore, a priority discipline that specifies the order in which the services will be granted to each traffic class. Meanwhile, the QoS provided for the currently existing real-time flows are not affected. We employ strict priority service discipline for resource allocation among different service classes to achieve inter-class fairness. The scheduler serves the traffic from the queue at priority level K only if there are no packets in the queues of higher priority levels. Under this policy, the sessions of higher priority class are guaranteed to have lower queuing delays at the current STA. In particular, our scheme consists of a priority-based scheduler for inter-class scheduling based on their priority levels and a session fairness-based scheduler for each incoming flow. Packet arrivals are sent to different queues according to their priority and destination node. The queues are grouped based on their priority levels. We employ strict priority service discipline for allocating bandwidth among service class flows, which means for that four priority classes, the scheduler serves the messages from the queue at priority level only if there are no packets in the queues of any other priority levels. Note that such a discipline is not influenced by the unfinished round result from the last schedule, i.e. the flows with higher priority level are pre-emptive. Therefore, the higher priority traffic flows always get scheduled first even if some lower priority traffic flows are still left unassigned by the last scheduling period. The flows of higher priority are therefore guaranteed to have lower queuing delays at the current node. An end-to-end QoS guarantee, consequently, can be provisioned since every mesh node assumes the same strategy distributive yet collaboratively.

Let θ_i be the proportion of the transmission request $R_{n,i}$ from traffic flow i to that from all traffic flows, namely:

$$\theta_i = R_{n,i} / \sum_j R_{n,j} \tag{34}$$

within the traffic flow group of the same class, the basic idea of the proposed scheduler is that each node assigns transmission opportunities to traffic flows is proportional to the amount that each flow requests with the consideration of the constraint maximum tolerable packet delay $D_{\max,i}$. We allocate transmission fairly according to the fairness model as:

$$f_{s1}'(n)\theta_{s1} \frac{\sum_{d \neq s1} Q_{s1d}^k}{R_{n,s1}} = f_{s2}'(n)\theta_{s2} \frac{\sum_{d \neq s2} Q_{s2d}^k}{R_{n,s2}} \tag{35}$$

where $R_{n,s1}$ and $R_{n,s2}$ are transmission requests of traffic flows s_1 and s_2, Q_{s1d}^{k} and Q_{s2d}^{k} are the average allocation transmission opportunities to satisfy the requests in the packet queue.

The proposed adaptive packet scheduling scheme works as follows:

1) When new application flows arrive in MAC layer, they will be categorized into four traffic classes with eight priorities immediately and subsumed in four scheduling services accordingly by the EDCA classifier. These packets are fed into queues according to its priority, waiting to be scheduled. The traffic flow of priority i transmits a REQUEST message by QoS control field to the MAP, which contains requirement $D_{\max,i}$.
2) The scheduler estimates the received SINR per bit γ by using the CTS frame, and after receiving the active flows number n_i of each priority class i, it will calculate the throughput S_i and the optimized packet length L_i^* with expression (29) to improve throughput. The scheduler will calculate θ_i by (34), fair index $f_i'(n)$ with (32) and (33) of the i th traffic flow in the n th transmission interval.
3) The scheduler calculates fairness index with expression (35), and allocates transmission opportunity to the corresponding traffic flow i satisfy the requests in the packet queue.
4) For the selected traffic flow i, the proposed adaptive packet scheduling scheme will aggregate several MSDUs to form an A-MSDU with the optimized packet length L_i^*. Then the packet scheduler selects the A-MSDU to be transmitted, tagging them in the mesh header and packing the packet as the payload. Packet scheduler subsequently attempts to find an appropriate transmission opportunity and pass the MSDUs to the physical layer.

In the WLAN mesh backbone, the medium is shared by all the wireless STAs in a neighborhood. All the traffic within the conflict domain should be carefully considered to avoid the collision. As a result, each traffic class in one MAP will not only experience intra-MAP but also inter-MAP competition. In the proposed scheduling scheme, we take advantage of the distributed mesh scheduling scheme to achieve our objective, in which coordinated distributed scheduling mechanism runs election algorithm in each node to coordinate the transmission opportunity within the two-hop or more extended hop neighborhood. Owing to the pseudo-random nature of the election algorithm, statistical fairness, along with the conflict resolution, is able to be achieved.

4. Simulation Results

4.1 Simulation Setup

The well-known simulation tool NS-2 [35] is used to validate the proposed scheme, and the proposed adaptive scheduling scheme is implemented MAPs and STAs. The simulation system is an infrastructure-based system, where they are served via an MAP. A random topology connecting STAs is used for evaluation. All the STAs are randomly placed in a region within the radio transmission range. The transmission range of MPDU is 120 m, and the carrier sense range of MPDUs is about 280 m. Each transmitter STA transmits with 16 dBm transmission power and the background noise level is set to -90 dBm. The allocated buffer size for all the queues in MAP and STA is set to 100 packets. Rayleigh fading channel used the well-known Jakes' method. The IEEE 802.11a PHY layer was selected for the simulations and the transmission rate is set to 54Mb/s. Smooth factor α is 0.1. The simulation results are the average value of 100 runs. Simulation parameters are shown in **Table 1**.

Table 1. Simulation parameters

Parameter	Value
DIFS (Data inter frame space)	60 μ s
SIFS (Short inter frame space)	20 μ s
PIFS (point inter frame space)	40 μ s
Slot time	20 μ s
Propagation delay	1 μ s
Beacon internal	100ms
Delay bound	100ms
Priority class	6(VoIP), 5(Video), 0(Data)
Mean data rate	64 kbps(VoIP), 375 kbps(Video)
Maximum MSDU size	60 bytes(VoIP), 1000 bytes(Video, Data)
MAC header	34Bytes
Sub-frame header in A-MSDU	14 Bytes
PHY Rate	54Mbps
ACK, CTS	112 bits + MAC header
RTS	160 bits + MAC header

In the simulations, data, video and voice-over-internet protocol (VoIP) traffic are used. Data traffic and video traffic are generated over a transmission control protocol (TCP) connection and a constant bit rate (CBR) 375kbps application over a user datagram protocol (UDP) connection, respectively. Also, VoIP traffic is taken into account to be generated by the on/off traffic model. During the on periods, packets are generated at a constant rate of 64 Kb/s, whereas no packets are generated during the off periods. Next, we compare the performance of our scheme with EDCA in terms of average values of throughput, packet loss rate and end-to-end delay.

4.2 Results

In the first scenario, every STA has three different priority classes: bi-directional VoIP traffic, unidirectional upstream video traffic and unidirectional upstream burst data traffic. We investigate the effect of traffic load on the performance of these schemes. The average inter-arrival times are 10 ms, 3 ms and 15 ms, respectively. The simulation time is 100s. In this scenario, five STAs begin to transmit, and then the number of STAs increases from 5 to 30.

Figs. 4, **5** and **6** show the aggregate throughput, packet loss rate and end-to-end delay of different scheme with varying number of STA respectively. As the number of STAs increases from 5 to 25, the aggregate throughputs of two schemes increase. When the number of STAs exceeds 25, however, the throughputs of APS and EDCA decrease due to the high probability of transmission collision. It can be noted that from **Fig. 4**, the throughput is the same for the system with the EDCA and APS scheme when the number of STAs is 5. Because all traffic flows' requirements can be met. When new STAs join in the system to access the network, we can observe that the new traffic flows can obtain transmission without any difficulty, and the total throughput increase to 13.51 Mbps. With the STAs number increasing, the aggregate throughput of APS scheme increases to 19.73 Mbps, which indicates that each video stream only occupies about 341kbps, lower than its requirement. At the same time, the total throughput of the WLAN is spoiled, with obvious oscillation. When the number of STAs increases to 30, the throughput slightly decreases to 18.92 Mbps, which is higher than EDCA about 5.31Mbps. The reason is as follows: APS can aggregate several MSDUs to form an A-MSDU with the optimized packet length and well regardless of traffic load by providing inter-class and intra-class fairness scheduling for different priority real-time traffic to exercise a discipline to satisfy the diverse QoS requirements, which makes MAPs and STAs adjust their transmissions so that packets are not transmitted excessively. Thus, APS prevents the increase in the probability of collision, even though the number of STAs and traffic flows increase.

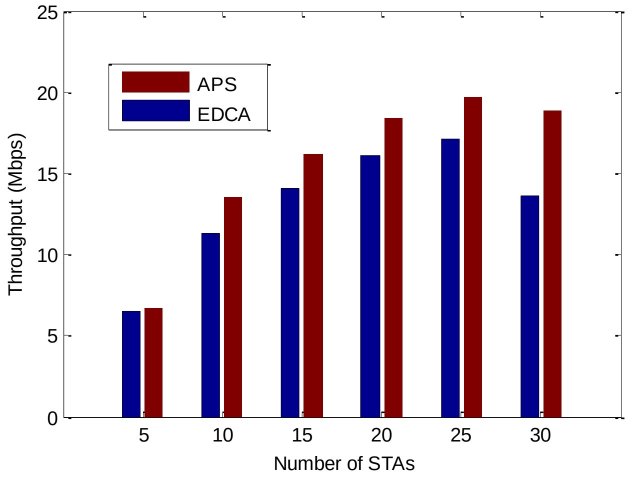

Fig. 4. Throughput of different scheme with varying number of STAs.

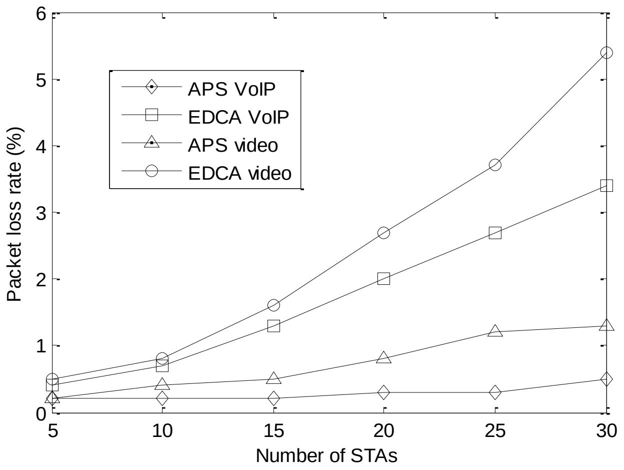

Fig. 5. Packet loss rate of different scheme with varying number of STAs.

Also, APS significantly reduces packet loss rate by aggregating scheduling scheme, as shown in **Fig. 5**. In the case EDCA, packet loss rate of VoIP and video traffic increases greatly. When the number of STAs increases from 5 to 30, the packet loss rate of VoIP increases obviously from 0.4% to 3.4%, which is higher than those of APS 2.3% on average. The packet loss rate of video traffic increases to 5.4% when the number of STAs is 30, which is higher than that of APS 4.1%. Meanwhile, the packet loss rate of APS is almost independent of the traffic load, and its value is smaller than those of DCF and EDCA by more than 4 times in most cases.

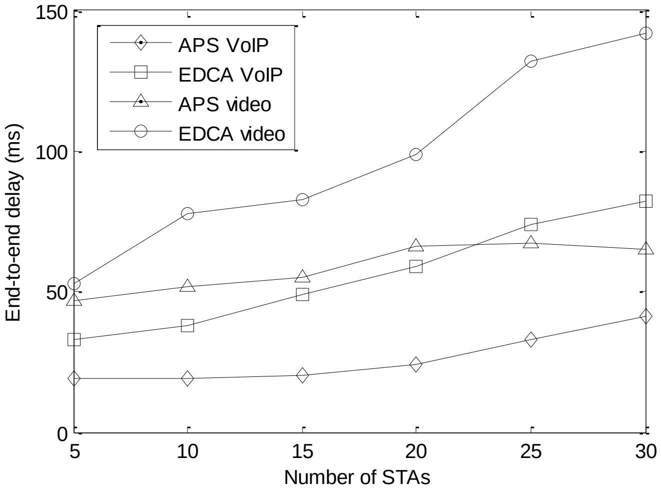

Fig. 6. End-to-end delay of different scheme with varying number of STAs.

From **Fig. 6**, we can observe that the QoS of video traffic and VoIP traffic are improved in terms of less delay with the APS scheme, for the traffic scheduled by the APS to transmit in appropriate period experience less delay and less fluctuation. When the number of STAs is less than 15, the error of delay between EDCA and APS is low, because all flows requirements can be met and the delays of all traffic are very low. As the number of STAs increasing, new real-time streams launch the system, which makes the delay of VoIP traffic oscillates obviously and increases abruptly to 82ms for EDCA. As the number of STAs increasing, the delay of video traffic also obviously increases to142ms because its actual bandwidth is just 231kbps, less than its requirement 375kbps. On the other hand we observe that with the APS scheme, the maximum delays of video traffic and VoIP traffic are less than 65ms and 41ms, respectively, whereas for EDCA the maximum delays are higher than 80ms and 140ms, respectively.

These simulation results in **Figs. 4**, **5** and **6** confirm that the APS scheme significantly enhances QoS for real-time traffic by employing maximum tolerable packet delay. And it also can guarantee intra-class fairness scheduling with strict priority service for different priority real-time traffic to exercise a discipline to satisfy the diverse QoS requirements.

In the second scenario, we evaluate the performance of APS in terms of fairness among STAs, and the Jain’s fairness index [36] is applied. We classify the STAs into three different groups depending on the traffic type the STA sends. The first group of STAs sends data traffic, the second send video traffic, and the third send/receive VoIP traffic. **Figs. 7** and **8** show the aggregate throughput of each STA and fairness index of difference scheme with varying number of STA respectively.

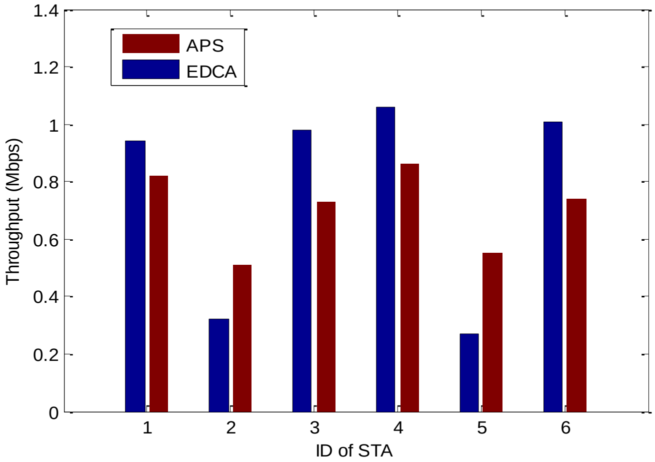

Fig. 7. Throughput of each STA with different scheme.

In **Fig. 7**, we observe the throughput of selected 6 STAs in the three groups. And STA2 and STA5 belong to the first group and send data traffic, STA1 and STA4 belong to the second group and send video traffic, STA3 and STA6 are selected from the third group send/receive VoIP traffic. From **Fig. 7**, for EDCA, it is obvious that the throughputs of STA2 (0.32Mbps) and STA5 (0.27Mbps) are lower than those of STAs sending real-time traffic. The unfairness result is that the contention-based channel access mechanism of EDCA will give more channel access opportunity to higher priority traffic flows. Therefore, STA2 and STA5 have significantly lower throughput than the other STA. On the other hand, for the proposed APS, the unfairness problem can be significantly alleviated. And we can see that the throughputs of STA2 and STA5 increase to 0.51Mbps and 0.55Mbps respectively. It is because the APS scheme implemented in STA and MAP takes into account both inter-class and intra-class fairness. Moreover, APS also uses the proportion of the transmission request from each traffic flow to adjust the channel access time. Since STA2 and STA5 tend to occupy channels shorter than other STAs for their lower priorities, the service levels of packets sent by these STAs will be guaranteed by the fairness model in (35).

Fig. 8 shows the fairness index for EDCA and APS with varying number of STAs. When the number of STAs increasing, the fairness index of EDCA decrease obviously from 0.84 to 0.61. The reason is that the high traffic load and probability of transmission collision, which will decrease the access time of STAs with lower priority. Hence, the fairness index of STAs will degrade obviously. It can be noted that from **Fig. 8**, the error of fairness index between EDCA and APS is just 0.1 when the number of STAs is 5. Because all traffic flows' requirements can be met. When new STAs join in the system to access the network, we can observe that the new traffic flows can obtain transmission without any difficulty. Hence the aggregate throughput of APS scheme increases, which indicates that the STAs with real-time traffic will occupy more bandwidth. At the same time, the throughput of STAs with lower priority is spoiled, which will degrade the fairness index. However, APS maintains significantly higher value of fairness index compared to EDCA with varying number of STAs. When the number of STAs increases to 30, the value of fairness index is 0.89, which is higher than that of EDCA about 0.27. The reason is that APS can provide inter-class and intra-class fairness scheduling for different priority traffic to exercise a discipline to satisfy the diverse QoS requirements, which makes MAPs and STAs adjust their transmissions so that packets are not transmitted excessively. Moreover, APS will aggregate several MSDUs to form an A-MSDU with the optimized packet length and well regardless of traffic load. Hence, it can decrease the probability of collision, even though the number of STAs and traffic flows increase.

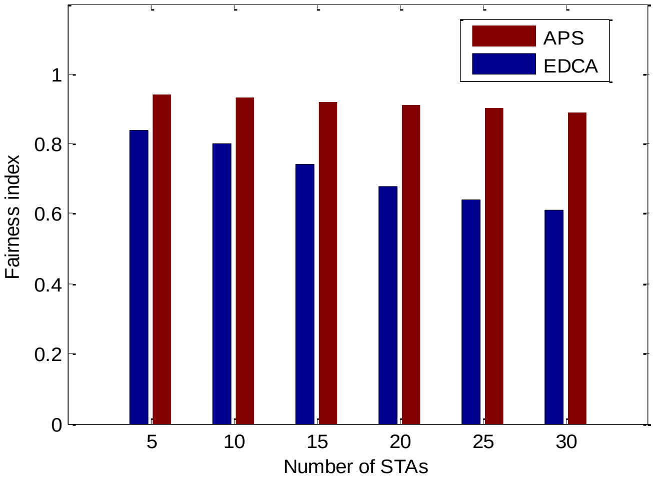

Fig. 8. Fairness index of different scheme with varying number of STAs.

5. Conclusions

In this paper, we have proposed an efficient adaptive packet scheduling (APS) scheme with inter-class and intra-class service differentiation and packet aggregation policy in WLAN mesh networks. The proposed scheme has been implemented and evaluated in NS2. Detailed simulation results and comparison with EDCA show that the proposed APS scheme is able to effectively provide inter-class and intra-class differentiate services and improve QoS for real-time traffic in terms of throughput, end-to-end delay, packet loss rate and fairness.

Future work includes taking the new features of IEEE 802.11n into account and the co-existence of IEEE 802.11s. The second one involves in extending the current scheduling framework to enhance the throughput and QoS by cross-layer design in testbed of multi-hop WLAN mesh network.

Acknowledgement

The authors gratefully acknowledge the helpful comments and suggestions of the reviewers.

References

[1] G.R. Hiertz, D. Denteneer, S. Max, R. Taori, J. Cardona, L. Berlemann, B. Walke, “IEEE 802.11s: the WLAN mesh standard,” *IEEE Wireless Communications*, vol. 17, no. 1, pp. 104-111, 2010. Article (CrossRef Link)

[2] S.M. Faccin, C. Wijting, J. Kneck, "Mesh WLAN Networks: Concept and System Design," *IEEE Wireless Communications*, vol. 13, no. 2, pp. 10-17, 2006. Article (CrossRef Link)

[3] J. Camp, E. Knightly, "The IEEE 802.11s Extended Service Set Mesh Networking Standard," *IEEE Communications Magazine*, vol. 46, no. 8, pp. 120-126, 2008. Article (CrossRef Link)

[4] S. Avallone, I.F. Akyildiz, G. Ventre, "A Channel and Rate Assignment Algorithm and a Layer-2.5 Forwarding Paradigm for Multi-radio Wireless Mesh Networks," *IEEE/ACM Transactions on Networking*, vol. 17, no. 1, pp. 267-280, 2009. Article (CrossRef Link)

[5] L. Zhou, X. Wang, W. Tu, G. Mutean, B. Geller, "Distributed Scheduling Scheme for Video Streaming over Multi-Channel Multi-Radio Multi-Hop Wireless Networks," *IEEE Journal on Selected Areas in Communications*, vol. 28, no. 3, pp. 409-419, Apr. 2010. Article (CrossRef Link)

[6] L. Zhou, H.-C. Chao, "Multimedia Traffic Security Architecture for Internet of Things", *IEEE Network*, vol. 25, no. 3, pp. 35-40, May/June 2011. Article (CrossRef Link)

[7] L. Zhou, Y. Zhang, K. Song, W. Jing, A.V. Vasilakos, "Distributed Media-Service Scheme for P2P-based Vehicular Networks," *IEEE Transactions on Vehicular Technology*, vol. 60, no. 2, pp. 692-703, February 2011. Article (CrossRef Link)

[8] L. Zhou, H. Wang, S. Lian, Y. Zhang, A.V. Vasilakos, W. Jing, "Availability-Aware Multimedia Scheduling in Heterogeneous Wireless Networks," *IEEE Transactions on Vehicular Technology*, vol. 60, no. 3, pp. 1161-1170, March 2011. Article (CrossRef Link)

[9] S. M. Elrakabawy, S. Frohn, C. Lindemann, "A Scalable Dual-radio Wireless Testbed for Emulating Mesh Networks," *Wireless Networks*, vol. 16, no. 8, pp. 2191-2207, 2010. Article (CrossRef Link)

[10] D. Wu, H. Luo, S. Ci, Song, H. Wang, A. Katsaggelos, "Quality-driven Optimization for Content-aware Real-time Video Streaming in Wireless Mesh Networks," in *Proc. of IEEE Global Telecommunications Conference*, pp. 1810-1814, 2008.

[11] E. Rozner, J. Seshadri, Y. Mehta, L. Qiu, "SOAR: Simple Opportunistic Adaptive Routing Protocol for Wireless Mesh Networks," *IEEE Transactions on Mobile Computing*, vol. 8, no. 12, pp. 1622-1635, 2009. Article (CrossRef Link)

[12] X. Wang, O.L. Azman, "IEEE 802.11s Wireless Mesh Networks: Framework and Challenges," *Ad Hoc Networks*, vol. 6, no. 6, pp. 970-984, 2008. Article (CrossRef Link)

[13] J. Yackoski, C.-H. Shen, "Managing End-to-End Delay for VoIP Calls in Multi-hop Wireless Mesh Networks," in *Proc. of IEEE INFOCOM*, 2010.

[14] S. Kompella, S. Mao, Y.T. Hou, H.D. Sherali, "On Path Selection and Rate Allocation for Video in Wireless Mesh Networks," *IEEE/ACM Transactions on Networking*, vol. 17, no. 1, pp. 212-224, 2009. Article (CrossRef Link)

[15] M. Kas, I. Korpeoglu, E. Karasan, "Utilization-based Dynamic Scheduling Algorithm for Wireless Mesh Networks," *Eurasip Journal on Wireless Communications and Networking*, vol. 2010, 2010. Article (CrossRef Link)

[16] Y. Li, Y. Yang, L. Zhou, A. Wei, C. Cao, "QoS-aware Fair Packet Scheduling in IEEE 802.16 Wireless Mesh Networks," *International Journal of Communication Systems*, vol. 23, no. 6-7, pp. 901-917, 2010. Article (CrossRef Link)

[17] R. Bruno, M. Conti, M. Nurchis, "Opportunistic Packet Scheduling and Routing in Wireless Mesh Networks," in *Proc. of IFIP Wireless Days*, 2010. Article (CrossRef Link)

[18] Z. Kong, Y.-K. Kwok, J. Wang, "Game Theoretic Packet Scheduling to Combat Non-cooperativeness in Wireless Mesh Networks," in *Proc. of International Conference on Distributed Computing Systems Workshops*, pp. 162-167, 2008.

[19] M.M. Alam, A. Hamid, M.A. Razzaque, C.S. Hong, "Fair Scheduling and Throughput Maximization for IEEE 802.16 Mesh Mode Broadband Wireless Access Networks," *IEICE Transactions on Communications*, vol. E93-B, no. 6, pp. 1459-1474, 2010.

[20] IEEE P802.11n/D7.0, "Amendment: Medium Access Control (MAC) and Physical Layer (PHY) Specifications, Enhancement for Higher Throughput," *IEEE P802.11n/D7.0*, Jan. 2009.

[21] IEEE Std 802.11e, "Specific Requirements Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications Amendment 8: Medium Access Control (MAC)

Quality of Service Enhancements," *IEEE Std 802.11e*, Nov. 2005.

[22] G. Bianchi, "Performance Analysis of the IEEE 802.11 Distributed Coordination Function," *IEEE Journal on Selected Areas in Communications*, vol.18, no. 3, pp. 535-547, 2000. Article (CrossRef Link)

[23] B.-G. Choi, J.Y. Lee, M.Y. Chung, "Adaptive Binary Negative-Exponential Backoff Algorithm Based on Contention Window Optimization in IEEE 802.11 WLAN," *KSII Transactions on Internet and Information Systems*, vol. 4, no. 5, pp. 896-909, 2010. Article (CrossRef Link)

[24] Z.-N. Kong, D.H.K. Tsang, B. Bensaou, D. Gao, "Performance Analysis of IEEE 802.11e Contention-Based Channle Access," *IEEE Journal on Selected Areas in Communications*, vol. 22, no. 12, pp. 2095-2106, 2004. Article (CrossRef Link)

[25] R. Zhu, Y. Yang, "Model-based Admission Control for IEEE 802.11e Enhanced Distributed Channel Access," *International Journal of Electronics and Communications*, vol. 61, no. 7, pp. 388-397, 2007. Article (CrossRef Link)

[26] R. Zhu, J. Wang, M. Ma, "Intelligent MAC Model for Traffic Scheduling in IEEE 802.11e Wireless LANs," *Applied Mathematics and Computation*, vol. 205, no. 1, pp. 109-122, 2008. Article (CrossRef Link)

[27] F. Jin, A. Arora, J. Hwang, H.-A. Choi, "Routing and Packet Scheduling in WiMAX Mesh Networks," in *Proc. of the 4th International Conference on Broadband Communications, Networks, Systems, BroadNets*, pp. 574-582, 2007.

[28] P.-R. Sheu, C.-F. Hu, C.-C. Liou, F.-C. Chuang, Y.-C. Chen, "An Efficient and Interference-aware Centralized Routing Tree Algorithm for the Routing and Packet Scheduling Problem in IEEE 802.16 Mesh Networks," in *Proc. of WRI International Conference on Communications and Mobile Computing*, pp. 496-503, 2010. Article (CrossRef Link)

[29] V.S. Naeini, N. Movahhedinia, "Fair Packet Scheduling for Integrated WLAN and TDMA Wireless Mesh Networks," *International Review on Computers and Software*, vol. 6, no. 1, pp. 126-133, 2011.

[30] Y. Zhang, S. Qin, Z. He, "Fine-Granularity Transmission Distortion Modeling for Video Packet Scheduling over Mesh Networks," *IEEE Transactions on Multimedia*, vol. 12, no. 1, pp. 1-12, 2010. Article (CrossRef Link)

[31] Q. Xia, X. Jin, H. Mounir, "Cross Layer Design for the IEEE 802.11 WLANs: Joint Rate Control and Packet Scheduling," *IEEE Transactions on Wireless Communications*, vol. 6, no. 7, pp. 2732-2740, 2007. Article (CrossRef Link)

[32] S.W. Kim, "Opportunistic Packet Scheduling over IEEE 802.11 WLAN," *Lecture Notes in Computer Science*, vol. 4159, pp. 399-408, 2006. Article (CrossRef Link)

[33] H. Kim, S. Yun, H. Lee, "Boosting VoIP Capacity of Wireless Mesh Networks Through Lazy Frame Aggregation," *IEICE Transactions on Communications*, vol. E90-B, no. 5, pp. 1283-1285, 2007.

[34] R. Riggio, F. De Pellegrini, N. Scalabrino, P. Li, Y. Fang, I. Chlamtac, "Performance of a Novel Adaptive Traffic Aggregation Scheme for Wireless Mesh Networks," in *Proc. of IEEE Military Communications Conference*, 2007. Article (CrossRef Link)

[35] NS2, Network Simulator. <http://www-mash.cs.berkeley.edu/ns>.

[36] D.-M. Chiu, R. Jain, "Analysis of the Increase and Decrease Algorithms for Congestion Avoidance in Computer Networks," *Computer Networks and ISDN Systems*, vol. 17, no. 1, pp. 1-14, June 1989. Article (CrossRef Link)

Rongbo Zhu is currently an associate professor in College of Computer Science of South-Central University for Nationalities, China. He received the Ph.D. degree in communication and information systems from Shanghai Jiao Tong University, China, in 2006. Dr. Zhu has published over 50 papers in international journals and conferences in the areas of wireless networks and mobile computing. He is the Editor-In-Chief of International Journal of Satellite Communications Policy and Management, Associate Editor of International Journal of Radio Frequency Identification Technology and Applications. He serves as a guest editor for several journals, such as, Future Generation Computer Systems, Telecommunication Systems, and as a reviewer for numerous referred journals such as IEEE System Journal etc. He has been actively involved in around 10 international conferences. Dr. Zhu is a member of the ACM and IEEE.

Yingying Qin received the B.S. degree in Computer Science from South-Central University for Nationaliteis, China, in 2009. She is currently a graduate in College of Computer Science of South-Central University for Nationalities. Her research interests include protocols design and performance optimizaiton in wireless mesh networks.

Chin-Feng Lai is an assistant professor in Institute of Computer Science and Information Engineering at National Ilan University (NIU). He has worked as a Post-Doctoral Fellow in Dept. of Engineering Science at National Cheng Kung University (NCKU) for one and a half year since Jan. 2008. He has published more than 60 technical papers. Dr. Lai serves as an editor of KSII Transactions on Internet and Information Systems. He has worked as a session chair in ICIMCS'09. He was the workshop Co-Chair of EMC-10, ACM Mobility Conference 2010, publicity co-chair of MMASN 2010. He serves as the corresponding guest editor for Journal of Telecommunications Management (JTM). He was selected an honorary member of the Phi Tau Phi Scholastic Honor Society of the Republic of China in 2009 and TC member of Multimedia Systems & Applications Technical Committee (MSATC), IEEE Circuits and Systems Society, 2009. He is also a project leader of several international, industrially-funded multimedia projects in NCKU. He is an IEEE member.