The Cross-Layer per-Flow Admission Control with Adaptive Reservation of Bandwidth and Multiple QoS Guarantee in EDCA of IEEE 802.11e for Internet of Things

Juling Zeng,

Computer and Information Institute of Three Gorges University ,Yichang,China,443002
julingzeng@gmail.com

Lisi Jiang

School of Information and Communication Engineering of Beijing University of Posts and Telecommunication,
Beijing,China,100876,
jianglisijianglisi@qq.com

Abstract—The static bandwidth's reservation and resource allocation in 802.11e results in bad supporting to the Internet of things. The cross-layer per-flow distributed admission control with dynamic reservation of bandwidth and multiple OoS guarantee of EDCA in IEEE802.11e is presented in this paper. The adaptive bits allocation of OFDM in the station is presented to obtain the maximum Shannon's capacity subjected to the maximum power at first and this information is crossed layer to the MAC. Then the dynamic bandwidth reservation is suggested based on it and the distributed measurement, which adapts to the requirement of the service and maximizes the efficiency of spectrum simultaneously. Meanwhile, the surplus factor's central estimation based on model is presented to overcome the measurement's inaccuracy. On the other hand, the relationship between the QoS and the probability of the collision is analyzed and the double criterion with collision probability and bandwidth is used to ensure multiple QoS such as bandwidth and delay. The simulation under MATLAB show that the mechanism presented in this paper outperforms the formers in the utilization of resource and the ability to track the network and service as well as admission of the service.

Index Terms—Distributed Admission Control, Bits allocation of OFDM, dynamic reserve of the bandwidth, computation of the collision probability and surplus factor, Multiple QoS guarantee

I. INTRODUCTION

The internet of things need a wideband wireless access technology for any thing at any time and any where so as to pervasively connect smart things, the enhanced distributed channel access(EDCA) of IEEE802.11e is one of the most competitive candidate[18][22]. However, the priority accessing by category in that EDCA cannot ensure the quantitative quality of service. The distributed admission control(DAC) in EDCA for this problem is presented in the draft of the advanced IEEE802.11e in 2003 and is improved in [2],where the time occupancy in a beacon frame of each service measured by AP(access point) and the budget of the bandwidth of each category of service in the next frame simultaneously considering the reserved bandwidth is computed and broadcast to all stations and the local station decide if a new service flow should be admitted or an old one should be continue to exist at last. But this DAC has some defects in internet of things(IOT). Most wireless sensors of IOT are energy harvesting active networked, which demands an energy adaptive mechanism in admission control to alter communications networking to satisfy energy and harvesting constraints and an ultra-low-power operation transmission for spending a few nano-Joules or less on every communicated bit[19]. On the other hand, the volatility of wireless resource due to the mobility of sensors and the time-varying of the wireless channel and the scarcity of wireless resource due to more and more services contrary to the non-renewable spectrum as well as the increase of video and voice service with requirement of quality need that the admission control should adapt to the wireless media to obtain the most effective utilization of the resource but also guarantee the quality of the service at the same time[20][21][22][23].

Some facts have shown that the presented admission control of EDCA cannot meet above mentioned demands. One fact is that no adaptive energy mechanism have been adopted in admission control of EDCA due to no physical information is crossed to it although some adaptive or optimized methods has been adopted in physical layer such as bits allocation in OFDM, which will result in the lower utilization of the wireless resource. The other is that the static reserved bandwidth decided by the ordering quantity

of the service and their weights makes no relation between the variation of the service and the dynamic of the network ,which can not guarantee the quality of service or high efficiency of the wireless resources. Moreover, the surplus factor used for estimation of transmission loss measured by a certain station is fed to the AP and then is broadcast to all stations as a category parameter for admission control, neglecting different data rates for different stations due to adaptive bits allocation in subcarrier of OFDM and different collision probability for even completely identical service of different stations[10], which will result in different surplus factor and transmission budget for different stations. This method directly measuring the resource of the network and using that as the admission criteria will led to the bandwidth stolen[6].Besides, it cannot guarantee the quality of service that the bandwidth is adopted as a single admission criterion.

The above defects have been improved in some papers. A dynamic updating of the reservation bandwidth according to the priority of the service and the loads of the network as well as the utilization of the resources is presented in [3],where the utilization of resources is some increased but the difference of the loads of the same category of the services lying in different stations or channels due to adaptive modulation and coding is adopted. In facts, for the adaptive modulation and coding system, only the distributed measurement and the global average can provide high utilization of the resources and real time updating of the bandwidth. A algorithm called HARMANIC is presented in [4], which changes the parameters of the competition window and the bandwidth of the service according to the indicator of the link quality (LQI) but there are still some defects. One important fact is that only searching under multiple services cannot reach global optimal in this algorithm due to no quantity relationship between the quality of the link and the resource allocation. An admission control based on the analytical model is presented in [5], where some parameters used for predicting the available bandwidth such as collision probability, free probability, success transmission probability, average loads and so on are measured by AP but not some stations and the final results for admission is the throughput per flow computed by bi-MARKOV model of EDCA. A similar one that give out the delay expression of the error or perfect channel based on the analytical model presented an admission control guaranteeing the throughput and the delay is in [6].An admission control based on the competition window is suggested in [7].A medium access control based on the virtual source and an admission control based on the threshold are presented respectively in [8] and [9].Although all these methods have some advantages by model-based strategy such as global optimization and so on, the complexity and the non-real time due to no consideration of the adaptive technology of the physical layer will result in inadaptability in on-line control. For

improvement of admission criterion, it is presented in [10] that the collision probability be used as admission criterion which will protect the presented flows . And a guarantee rate with delay threshold is presented as an admission criteria in [11] based on a pair of token buckets and multiple parameters of QoS, which effectively protect the rate-variable flow but cannot provide multiple QoS due to the guarantee rate only relating the bandwidth with the variable arrival rate.

Furthermore, it is a good foundation for the cross-layer per-flow admission control that physical layer's protocol of IEEE802.11a is supported by IEEE802.11e.The parallel data transmission based adaptive bits allocation adopted by the OFDM(orthogonal frequency division multiplex) of that protocol make it easy to antagon frequency selection fading resulted by the multipath spreading and to improve the spectrum efficiency while it turn the cross-layer admission into realize because that the information of the bits allocation is allowed to the MAC[12].The bits allocation is studied in some papers. The rate-limited and minimum-power algorithm is presented in [13] but it cannot maximize the spectrum efficiency. An allocation with minimum-guarantee of the resource and the maximize capacity is suggested in [14].The power-limited and the error-bits-constrained suboptimal algorithm with the maximize bit-rate is presented in [15] and [16],which is not only very convenient to realize but also guarantee high efficiency of the resource closing to the optimal.

In this paper, we present the cross-layer per-flow distributed admission control with dynamic reservation of bandwidth and multiple OoS guarantee of EDCA(enhanced distributed channel access) in IEEE802.11e . In this mechanism, the bits of the subcarrier of OFDM in the station is allocated to obtain the maximum capacity of the channel subjected to the maximum power at first and the information of the bits allocation is crossed layer to the medium access control layer for adaptive admission control. And then the dynamic reservation of the bandwidth is suggested based on the bits allocation and distributed measurement, which adapt to the requirement of the service and maximize the efficiency of the spectrum simultaneously. Meanwhile, the central control estimation of the surplus factor based on the analytical model is presented to overcome the inaccuracy by direct measurement and ensure the enough bandwidth for every flow to avoid collision. In the other hand, the relationship between the requirement and the probability of the collision is analyzed and the double criterion of the collision probability and the bandwidth are adopted to ensure multiple parameters of the quality of the services such as bandwidth and delay as well as error ratio of frame. The simulation under MATLAB show that the admission control presented in this paper outperforms the former in the utilization of the resource and the ability to track the network and service as well as the admission of the service.

I. BITS ALLOCATION IN SUBCARRIER OF OFDM

A. Optimal Model

Assuming that the channel is quasi-static fading, the received symbol in the ith subcarrier of OFDM of the kth station is as :

$$r_{i,k} = H_{i,k}s_{i,k} + N_{i,k} \tag{1}$$

where，$s_{i,k}$ is the modulating data in the ith subcarrier of the kth station and $H_{i,k}$ is the gain of the complex gain of this frequency domain subcarrier featured by rayleigh fading and $N_{i,k}$ is complex Gaussian noise with zero mean. Let $p_{i,j} = E[|s_{i,j}|^2]$,then the data rate of the subcarrier modulated by M order QAM （quadrate amplitude modulation）with Gray mapping is as :

$$R_{i,k} = \log_2(1 + \frac{p_{i,k}|H_{i,k}|^2}{\Gamma \sigma_z^2}) \tag{2}$$

Where,，$\sigma_z^2 = N_0 \frac{B}{N}$ ，$\Gamma = -\ln(5BER)/1.6$ and BER is the bits error ratio of the M order QAM with Gray mapping.

The optimal model with maximize data rate and specific all power as well as the appointed error bits ratio for any one of the system with K users having N subcarriers.

$$\max_p(\frac{B}{N}\sum_{i=1}^{N}\log_2(1 + \frac{p_i|H_i|^2}{\Gamma \sigma_z^2})) \tag{3}$$

subject to: (c1) $p_i \geq 0$

(c2) $E[\sum_{i=1}^{N} p_{i,k}] \leq p_k$

B. Solving the Optimal Model

The above model's being a N-orders non-line integer optimal problem, the strict solving to it is too complex to on-line control. So it is always converted to the problem of the multi-user water-filling theory. Let $H_{i,k}$ ascending-power permutation and the $H_{1,k}$ minimum value, the following results can be obtained by Lagrange Multipler according to water-filling theory in frequency domain.

$$p_{1,k} = \frac{p_k - V_k}{N} \quad (4) \qquad p_{i,k} = p_{1,k} + \frac{H_{i,k} - H_{1,k}}{H_{i,k}H_{1,k}} \tag{5}$$

Where, $V_k = \sum_{i=2}^{N} \frac{H_{i,k} - H_{1,k}}{H_{i,k}H_{1,k}}$. The equation (5) show that better the channel is higher the power is . Accordingly,

$$R_{i,k} = \left\lfloor \log_2(1 + \frac{p_{i,k}|H_{i,k}|^2}{\Gamma \sigma_z^2}) \right\rfloor \tag{6}$$

Where, $R_{i,k}$ is the bits rate of the ith subcarrier of the kth station. So

$$R_k = r_s \sum_{i=1}^{N} R_{i,k} \tag{7}$$

Where, r_s is the OFDM symbol rate and R_k is the bits rate of the kth station

II. UPDATING OF THE RESERVED BANDWIDTH OF THE SERVICE BASED ON BITS ALLOCATION AND DISTRIBUTED MEASUREMENT

After bits allocation in subcarrier of OFDM is adopted, the different station has the different physical channel and bits rate, which results in that the same service need different bandwidth. So the reserved bandwidth should be independently computed according to the channel and service's load of each station so that the physical resource can be used fully and the quality of the service can be ensured at the same time. Here work is based on reference[4], where an utilization weights, load weights and effectiveness weights are designed to reserve the bandwidth.

A. New Weights of the Utilization Ratio

Assuming $ATL(i)$ (available TXOP limit) is the reserved bandwidth by AP to the $AC(i)$ (access category), the weights of the utilization ratio is defined as follows:

$$uw(i) = \frac{TX\_TIME(i)}{Total\_TXOP\_Used} \tag{8}$$

where $Total\_TXOP\_Used = \sum_i TX\_TIME(i)$ (TX_TIME(i) is the TXOP utilized by each access category).When the adaptive modulation and coding is adopted, $TX\_TIME(i)$ belong to different stations change with channel's state respectively. So it is suitable that the station monitor the channel by itself and then record the $TX\_TIME(i)$ varying with AMC and feed to AP. While, $uw(i)$ is corrected as follows

$$TX\_TIME(i) = \sum_j TX\_TIME_j(i) \tag{9}$$

$$Total\_TXOP\_Used = \sum_j \sum_i TX\_TIME_j(i) \tag{10}$$

Where, j is the amounts of the station and the weights of the utilization ratio represents the possession of the network's resources by the service. In order to ensure the bandwidth of the service, the resource of the service should be proportional to that weights..

B. New Weighting Factor

Let $pw(i)$ be the weighting factor decided by the service flow. For example, there are three services that is $i = 1,2,3$ for $AC(i)$. Assuming the proportion is $1:2:7$,then $pw(i) = 0.1 \backslash 0.2 \backslash 0.7, i = 1,2,3$ .In the initial stages setting up the network, the reserved bandwidth can be decided by the $pw(i)$ dew to the light loads. .But the time-varying network and the service required time-varying $pw(i)$. So the $pw(i)$ should be replaced by the above $uw(i)$ .Because the $uw(i)$ is the instantaneous function which is time-limited and the association method of the uplink in IEEE802.11e is associating with the next

frame, the $uw(i)$ is estimated by the liner minimum mean square error as follows:

$$uw_i(n) = \sum_{k=1}^{M} a_k uw_i(n-k)$$

(11) Where, the coefficient a_k can be computed by the steepest descent method.

C. New Loading Weights

$$lw(i) = \frac{TX\_Load(i)}{Total\_TXOP\_Need} \times \alpha \qquad (12)$$ where，α is the ratio of the useable residual bandwidth to the totals.

The distributed measurement of $Total\_TXOP\_Need = \sum_i TX\_Load(i)$ due to AMC is as follows:

$$\tau_j(i) = \frac{MSDU(i)}{R_j} + t_{ACK} + SIFS + AIFS(i) \qquad (13)$$

$$TX\_Load(i) = \sum_j queenlength_j(i) \times \tau_j(i) \qquad (14)$$

$$Total\_TXOP\_Need = \sum_i TX\_Load(i) \qquad (15)$$

Where, $\tau_j(i)$ is the time for the station j transmits a MSDU(MAC service data unit) of $AC(i)$ .In order to guarantee the QoS, the distributed resource of the service should be proportional to the loading factor which reflects the possession by this service to the network.

D. Effectiveness Weights Defined by AP

New effectiveness weights defined by AP is as follows:
$$ew(i) = pw(i) \times (1 + lw(i))$$

(16)

E. New Reserved Bandwidth

The reserved bandwidth of the service AC(i) decided by AP is as:
$$ALT(i) = (TIME\_in\_CP - Total\_TXOP\_Used) \times ew(i) \quad (17)$$

III. The Improved Computation Of Collision Probability And Surplus Factor

The surplus factor is defined as the ratio of the total bandwidth for transmitting some a service to that for successfully transmission, which is used to show the extra bandwidth for collision. In reference [10], the surplus factor is measured by some a station in the network and transmitted to the AP as a parameter of all the same kind of the service. In facts, even completely identical service has different collision probability in different station due to different bits rate for different stations adopting adaptive bits allocation in physical layer. So the different stations have different surplus factor and transmission budget and the surplus factor is measured and computed by the station and then is fed to AP to be averaged, which will increase the times of handshaking between the station and AP and result in wasting of the resource. On the other hand, the above surplus factor includes the lost packets resulted by the local queuing, which will lead to too large bandwidth to waste the resource. Contrary to that, because AP can know the channel's state of each station and the bits rate of

each station, it is more suitable that the surplus factor is measured and computed by AP. But AP can't know which packet has been corrupted, so it is difficult to directly measure the surplus factor. Another cause to no directly measuring is that the admitted flow will greatly affect the state of the network in EDCA, which will result in the bandwidth embezzlement if AP directly measure the surplus factor. Because AP can sense the channel easily, it can compute the collision probability and surplus factor of the service flow based on the analytical model of EDCA and those sensing results of each station. And also AP decide the biggest value of the surplus factors of all stations with same service as the surplus factor.

An indirect measurement is presented here .Let p_b and p_f are busy and free probability of the channel measured by AP, respectively and the parameter set of $AC(i)$ in EDCA is $CW_{j,min}(i), AFIS_j(i), CW_{j,max}(i)$ as well as R_i is the times of backoff or retransmission, the equation (18) is established for the station j according to double Markov chain when the network is in saturation state and the backoff window cannot reach the biggest one

$$wp_{j,0,0}(i) = \frac{2d_{j,i}(1 - 2p_{j,c}(i))(1 - p_{j,c}(i))}{CW_{j,min}(i)(1 - p_{j,c}(i)) + (1 - 2p_{j,c}(i))}$$
$$\times \frac{1}{(1 - p_{j,c}^{R_i+1}(i))(2d_{j,i} - 1)(1 - (2p_{j,c}(i))^{R_i+1})} \qquad (18)$$

Where, $d_{j,i} = p_f^{aifs(i)}$, $d_{j,i}$ is the state's transferring probability of $AC(i)$ of station j and $p_{j,c}(i)$ is the probability of the $AC(i)$ .Assuming $wp_{j,0,0}(i)$ is the probability of the station's competition window's beginning when it transmit AC(i),the following equation is obtained by double Markov chain.

$$wp_{j,k,0}(i) = p_{j,c}(i) * wp_{j,k-1,0}(i) = p_{j,c}^k(i) * wp_{j,0,0}(i) \qquad (19)$$

The probability of the $AC(i)$'s of the station j accessing the channel in some a time slot is

$$\tau_j(i) = \sum_{k=0}^{R_i} wp_{j,k,0}(i) = \frac{1 - p_{j,c}^{R_i+1}(i)}{1 - p_{j,c}(i)} wp_{j,0,0}(i) \qquad (20)$$

If the station adaptively allocate the bits in subcarrier while keeps the other condition invariant, it has the different bits rate from others for the same kind service having different collision probability to access the channel. Assuming r_j be the bits rate of the station j ,then

$$p_{j,c}(i) = 1 - \prod_{k=1}^{N} (\prod_{i=0}^{M-1} (1 - \tau_k(i)/(1 - \tau_j(i))) \qquad (21)$$

where，$\tau_k(i) = \frac{r_j}{r_k} \tau_j(i)$.

The internal collision probability of AC(i) of station j is as:

$$p_{j,v}(i) = 1 - \prod_{k=0,k\neq i}^{M-1} (1 - \tau_j(k))$$

$$(22) \qquad\qquad p_{j,c}(i) = p_{j,v}(i) + p_{j,r}(i)$$

(23)

The free probability of the channel is as:

$$p_{idle} = 1 - \prod_{k=1}^{N} \prod_{i=0}^{M-1} (1 - \tau_k(i))$$

where， $\tau_k(i) = \dfrac{r_j}{r_k} \tau_j(i)$ (24)

and $p_{j,c}(i), (i = 0,1, \cdots M - 1)$ is solved out through combining those equations from (18) to (24).

For surplus factor, let $p_{j,c}(i) = 1 - \dfrac{1}{SBA_J(i)}$ (25)

and $SBA(i) = \max_j (SBA_j(i))$ (26)

Because the biggest value of the surplus factors of the same service is selected, the most sufficient bandwidth is supplied to avoid the collision.

IV. THE RELATION BETWEEN QOS AND COLLISION PROBABILITY

A. Delay of the Flow Versus the Numbers of Retransmission

The delay of the flow is decided by the times of retransmission R_i and the minimum backoff window $CW_{i,mim}$ by EDCA model. When the times of retransmission R_i is determined, the mean delay of the packets is as follows:

$$E_i(T_{drop}) = \frac{CW_{\min,i}(2^{R_{i,\max}+1} - 1) + (R_{i,\max} + 1)}{2} E_i(slot)$$ (27)

Where $R_{i,\max}$ is the biggest times of retransmission and $E_i(slot)$ is the mean time slot.

$$E_i(slot) = \frac{\Gamma_i + \Gamma_s + \Gamma_c}{N_i + N_s + N_c} = p_{idle}T_i + p_s T_s + p_c T_c$$ (28)

Where， $\Gamma_i, \Gamma_s, \Gamma_c$ are the total idle period, the total successful period and collision period. T_i, T_s, T_c are free time, successful transmission time and collision time.

$$T_s = H + E[P] + SIFS + ACK + AIFS$$ (29)

$$T_c = H + E*[P] + SIFS + ACK\_Timeout$$ (30)

Where, H is the length of the head of the data frame and $E[P]$ is mean length of the data frame successfully transmitted and $E*[P]$ is the expected value of the longest data frame in collision and p_{idle}, p_s, p_c are free probability, successfully transmission probability and collision probability, respectively. p_{idle} is as shown in (24).We can obtain that:

$$p_s = \sum_{j=1}^{N} \tau_j(i)(1 - p_j(i)) , p_c = 1 - p_{idle} - p_s$$ (31) AP can

compute the $E[slot]$ through those equation from (27) to (31). Then let D_i is the delay of the flow i. The biggest numbers of retransmission of each flow can be computed by AP as:

$$R_{i,\max} = \left\lfloor \log_2 \left(\frac{2D_i}{CW_{i,\min} E(slot)} + 1 \right) - 1 \right\rfloor$$ (32)

B. Packet Lost Ratio versus the Upper Limitation of the Collision Probability

The main cause of the packet's lost in EDCA is collision and error in channel. When the lost packets due to full buffer is neglected, the packet lost ratio is as follows:

$$p_{i,drop} = (1 - (1 - p_{i,err})(1 - p_{i,col}))^{R_i+1}$$ (33)

Where, $p_{i,err}$ is the error frame ratio, which is known by AP easily due to the record that AP allocates the bits in subcarrier for the stations. And $p_{i,col}$ is collision probability which is computed by AP. Because $0 < p_{i,drop} < 1$,when the biggest numbers of retransmission is reached, the biggest collision probability is as follows:

$$p_{i,col,\max} = \frac{\sqrt[R_i+1]{p_{i,drop}} - p_{i,err}}{1 - p_{i,err}}$$ (34)

V. PER-FLOW ADMISSION CONTROL OF EDCA WITH MULTIPLE QOS GUARANTEE.

The admission control is accomplished by some times handshaking between AP and station. AP and station should finish following tasks, separately.

A. AP

(1) Allocate bits in subcarrier of OFDM of accessing station based on the channel state

(2)Compute ATL(i) based on service requirement and the station state

(3) Measure the free or busy probability of the channel and then compute surplus factor based on those according to the model of EDCA.

(4) Compute the biggest collision probability of each kind of service based on the packets' lost ratio and error frame ratio as well as the biggest numbers of retransmission and then transmit to each station with beacon frame.

(5)Compute the surplus ATL(i) of each service and then transmit to each station with beacon frame.

(6)After that, AP restore these parameters such as $AIFS(i) \backslash CW_{\min}(i) \backslash CW_{\max}(i) \backslash surplusfactor(i) \backslash ALT(i) \backslash Txtime(i)$ .The usable transmitting time of $AC(i)$ in the next beacon frame is as

$$TXOPBudget(i) = \max(ALT(i) - Txtime(i) \times surplusfactor(i), 0)$$ (35)

At last, AP transmits this parameter to each station.

B. STA

The parameters restored by the station are $TxUsed(i), TxCounter(i), TxLimiter(i), Tx\operatorname{Re}mainder(i), TxM emory(i)$ $TxUsed(i)$ is used for recording the on-line transmitting time of $AC(i)$ having been admitted by this station or the required bandwidth of the newly admitted by this station. $TxCounter(i)$ is used for recording successfully transmitted time of $AC(i)$. $TxMemory(i)$ is used for recording all resource used by $AC(i)$ of this station in a beacon frame. $TxLimiter(i)$ represents the largest resources used by $AC(i)$ of the station. $Tx\operatorname{Re}mainder(i)$ represents

the residual resource of $AC(i)$ of the station when a frame is forbidden. The process finished by the station is as follows:

(1)If $TXOPBudget(i) = 0$,the flow of any station accessing the network with $AC(i)$ in the next frame is rejected and that station will keep following equation. $TxM\,\mathrm{e}mory(i) = 0$, $Tx\,\mathrm{Re}mainder(i) = 0$ $TxLimiter(i) = 0$ (36) For others, $TxM\,\mathrm{e}mory(i)$, $Tx\,\mathrm{Re}mainder(i)$, $TxLimiter(i)$ will be kept invariant.

(2)If $TXOPBudget(i) > 0$, the collision probability and bandwidth are adopted as double admission criteria.

First, the collision probability is computed by equation (26) based on the surplus factor. If this collision probability is bigger than the biggest one ,the flow of the station accessing the network with $AC(i)$ in the next beacon frame is rejected. This station will keep the following equations.

$$TxM\,\mathrm{e}mory(i) = 0 \quad , \quad Tx\,\mathrm{Re}mainder(i) = 0 \quad , \quad TxLimiter(i) = 0$$

(37)

If $TXOPBudget(i) > 0$, the collision probability is smaller than the biggest one, the station accessing the network with $AC(i)$ in the next beacon frame will modify the parameters as follows:

$$TxM\,\mathrm{e}mory(i) \in [0, TXOPBudget(i) / Surplusfactor(i)] \quad (38)$$

For others,

$$TxMemory(i) = f \times Txmemory(i) + (1 - f) \times \\ ((TxCounter(i) \times Surplusfactor(i) + TXOPBudget(i))$$

(39)

After that, let $TxCounter(i) = 0$ and $TxLimiter(i) = TxMemory(i) + Tx\,\mathrm{Re}mainder(i)$ whether the old admitted flow or the newly one. The admission is as follows:

(1)For new $AC(i)$, if $TXOPBudget(i) > 0$ and $TxUsed(i) < TxLimiter(i)$, it is admitted. Otherwise, it is rejected.

(2)For existing $AC(i)$, if $TxUsed(i) > TxLimiter(i)$, it is prevented from continue transmitting. Let $Tx\,\mathrm{Re}mainder = TxLimiter(i) - TxUsed(i)$ for this $AC(i)$. Otherwise, it is continued to be transmitted.

C. The Protection for the Existing Flow in the Distributed Admission Control

When $TXOPBudget(i) = 0$,the newly arriving $AC(i)$ is rejected and $TxM\,\mathrm{e}mory(i)$ and $Tx\,\mathrm{Re}mainder(i)$ are kept invariant. So $TxLimiter(i)$ is kept invariant that is the existing flows are transmitted as before.

When $TXOPBudget(i) > 0$, $TxM\,\mathrm{e}mory(i)$ and $TxLimiter(i)$ are changed periodicity. $TxM\,\mathrm{e}mory(i)$ converge to $TxCounter(i) \times Surplusfactor(i) + TXOPBudget(i)$ when $f < 1$. When $TXOPBudget(i)$ is exhausted, $TxM\,\mathrm{e}mory(i)$ converges to $TxCounter(i) \times Surplusfactor(i)$ and $TxLimiter(i)$ converges to $TxCounter(i) \times Surplusfactor(i)$,which will ensure a constant transmitting bandwidth for real-time service.

VI. SIMULATION

We study the performance of the proposed adaptive reservation of the bandwidth and admission control under MATLAB. We first study adaptive bits allocation and the improvement of the spectrum efficiency. And then we present the performance of the proposed adaptive reservation of the bandwidth in the following two system:(1)ARB: system adopting adaptively reserving bandwidth based on bits allocation and (2)NARB: system adopting static reservation of bandwidth. At last we present the simulation performance of the proposed admission control in following three system:(1) ARB-CAC: system adopting adaptively reserving bandwidth and cross-layer admission control based on bits allocation in subcarrier and (2)NARB-CAC:system with no adaptively reserving bandwidth and but cross-layer admission control based on bits allocation in subcarrier and (3)NARB-NCAC: system with non adaptively reserving bandwidth and non cross-layer admission control based on bits allocation in subcarrier.

A. Bits Allocation in Subcarrier of OFDM

For OFDM, let 128 subcarriers be in a OFDM symbol and 28 subcarriers of that as a cyclic-prefix and the remained 100 subcarriers be used for bits allocation and let these 100 subcarriers be divided into three sub bands where each subcarrier has the same gain of channel in a same sub band. These three sub band are uniformly arranged from low number to high ones whose gain are -8dB,-4dB and -2dB respectively. For every subcarrier, let single edge power spectrum of the white gauss noise is $14 \times 10^{-12} W / Hz$ and the limiting of the uplink is 100mw.When the bits allocation is not used, the 16QAM is adopted in all subcarriers. While the bite allocation is used, 16QAM,32QAM and 64QAM are adopted in the three subcarriers, respectively, which is shown in fig.1.In this condition, assuming the symbol rate is 80K symbols per second, the total bits rate can be reached to 40Mbits/s higher than the past 32Mbits/s, which shows the frequency spectrum is greatly increased when the bits allocation is used.

B. Simulation Scenario

We set up our IOT model with 4 stations and 1 AP and each sensor of the network transmits with 802.11e . Assume identical arrival for each service and three kinds service for every station and 80B,500B and 1024B loads for these three service whose QoS is shown in table 1.The data service obey on-off exponential distribution and can be only transmitted on up direction. The initial parameters of the other service flow is as follows: $A\,\mathrm{IFS\,(voice)} = 25\mu s$, $CW_{\min}(\mathrm{voice}) = 15$, $A\,\mathrm{IFS\,(video)} = 25\mu s$, $CW_{\min}(\mathrm{video}) = 31$, $A\,\mathrm{IFS\,(data)} = 34\mu s$, $CW_{\min}(\mathrm{data}) = 63$, $\mathrm{SIFS} = 16\mu s$, $\mathrm{slot\,time} = 9\mu s$, $\mathrm{beacon\,interval} = 2s$ .The transmission of the flow begin at the speech flow which arrive every 30 seconds one by one and the arrival order is A,B and C. Video stream arrive after speech flow in 10 seconds and after that it has the

same arrival as the speech flow. The data flow begin at 20 second after speech flow and also has the same arrival as that of the speech. For data flow, when the loads of the network is light, they all can be admitted. Otherwise, some data service can be discarded so as the channel can be released. The method is to discard least a data service but not a packet each time. The minimum bandwidth for data service is 10% to the whole. The physical channel supporting the whole system is the above OFDM and the error frame ratio is assumed 0.

C. Analysis of the Reserved Bandwidth, Collision Probability and Surplus Factor

For the NARB system, the static reserved bandwidth is adopted, which assign 2.399Mbits/s and 26.401Mbits/s to speech flow and video flow respectively based on the rate of the service. For the ARB system, the bandwidth assigned to speech flow and video flow are 4Mbits/s and 32Mbits/s based on the rate of the service at the beginning. When the loads of the network is low, the bandwidth is reserved according this amount all the time. While the loads is increased, the utilization $uw(i)$ is recorded and it is estimated again by liner prediction with 5 stage filter. When the loads reach a certain amount, $pw(i)$ is represented with $uw(i)$ to compute $ew(i)$. The results in fig.6 show that $uw(i)$ is less than $pw(i)$ for speech flow and $uw(i)$ is bigger than $pw(i)$. Assuming the time for service's arrival is just the time for some a existing service's transmission end, there isn't retransmission due to collision before the network enter into saturation and the length of the queue is zero. When the network enters into saturation, $lw(i)$ is observed and recorded. The results should be that $lw(i)$ is zero for speech flow and is bigger than zero for video flow . $ew(i)$ in fig.5 computed by (16) shows that it is kept invariant before the network enter into saturation while it is adaptively changed to the state of the network when it is in saturation. Generally speaking, the efficient factor is less than the static allocation factor for speech flow and is bigger for video flow, which balances the traffic between the speech flow and the video flow. This results can be shown in Fig3,4 and 5,where the speech flow using the efficient factor admitted by the network is less than before while the video flow is bigger than before. At the same time, fig.2 shows that the throughout of the network is increased by efficient factor.

For surplus factor, when the error frame and the lost packets due to the full buffer are neglected and the loads of the network is light, there isn't collision and lost packets certainly. So let $SBA_i = 1$ .With the increasing of the network's loads, collision is increased, which results in more lost packets and the SBA_i increase gradually. The biggest SBA_i is computed by (26) and it is 1.72,1.47,1.67,1.47 for speech flow and video flow A,B,C respectively. The parameters of the competition window are computed by these equations from (32) to (40), the collision probability of the video flow C is shown in fig.7.From fig.7,it can be seen that the collision probability will increase with the loads at the beginning but decrease

with loads when the loads are increased to a certain amount due to some stations or flows cannot being admitted if no admission control is adopted. While, when the admission control in this paper is adopted, the collision probability can be stable in some a value when the network is in saturation dew to the flow's admission control.

D. Analysis of Throughput

It is shown in fig.2 that the network in NARB-CAC reach the saturation 50s later than that in NARB-NCAC and the throughput of the former is increased 8Mits than the latter due to the improvement of the frequency spectrum. In saturation, the throughput of the network in ARB-CAC is some increased than that in NARB-CAC but it is not enough obvious, which is because that adaptively reserving bandwidth can only assign the bandwidth based on the features of the service flow so as more fairness can be shared in the services but can not increase the limited bandwidth. It is shown in fig.3 and fig.4 that the speech flow reach the saturation earlier than before due to the adaptively reserved bandwidth being less than that of static reserved and the opposite effect is for the video flow, which make the speech flow and the video flow reach saturation almost at the same time so as to avoid that the speech flow reach saturation 170s earlier than the video flow and ensure the fairness and integrity of the multimedia service. Curves in fig.2,3 and 4 shows that the system reach the saturation almost at the same time whether in NARB-CAC or in NARB-NCAC but the data service's releasing the channel in heavy load will let the speech flow reach saturation later than the whole system and the same results is for the video flow. The speech flow reach saturation in NARB-CAC 170s later than in NARB-NCAC and the video flow 60s later in NARB-CAV than in NARB-NCAC. The comparison of these three system's data service's throughput is shown in fg.5.It can be seen that these three system all is increased at first until the top point and then is decreased gradually until stable, which is because that the data service can be admitted in time when the loads is light but it should be discarded to release the channel until the lowest limit when the loads is too high to result in the saturation of the network. It also can be seen in fig.5 that the data service of the network reach the saturation in NARB-CAC and ARB-CAC about 60s later than in NARB-NCAC. Specially, in ARB-CAC,the mechanism adaptively reserving bandwidth make the data service release the channel more quickly and give some bandwidth to video flow when the loads gets heavier, which is uniform with the fig.4.
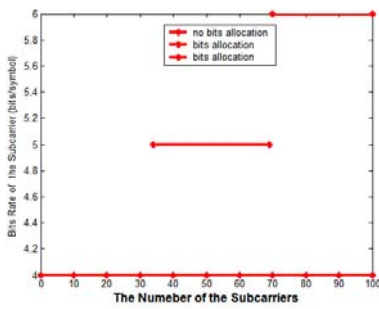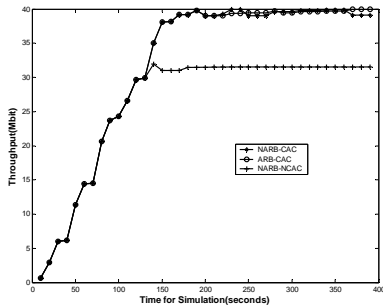
Fig.1 Spectrum efficiency comparison

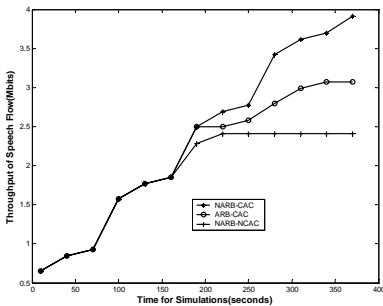Fig.2 System throughput comparison

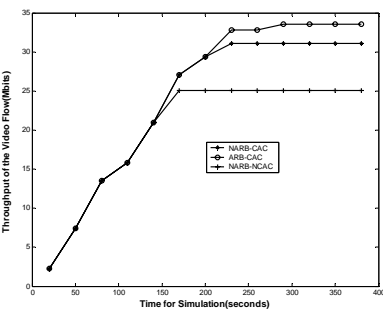Fig.3 Voice traffic throughput comparison

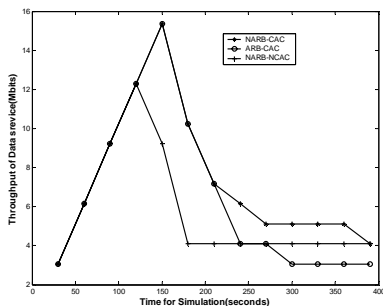Fig.4 Video traffic throughput comparison

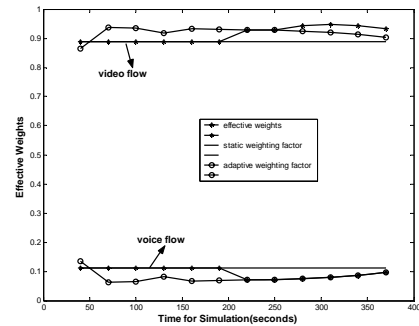Fig.5 Data service traffic throughput comparison
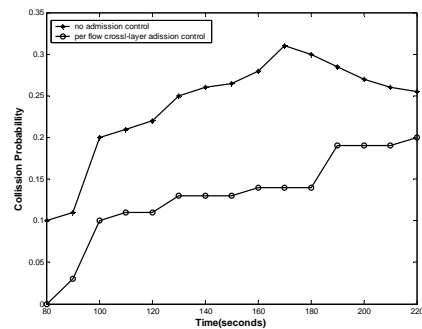

Fig.6 Effective weights comparison

Fig.7 Collision probability comparison

TABLE 1
PARAMETERS OF SERVICES

Service	Bits rate	Delay	Packets loss ratio
Voice(A)	128Kbps	<5ms	<3%
Voice(B)	64Kbps	<5ms	<3%
Voice(C)	16Kbps	<5ms	<3%
Video(A)	384Kbps	<30ms	<1%
Video(B)	768Kbps	<60ms	<1%
Video(C)	1024Kbps	<30ms	<1%

VII. CONCLUSION

The cross-layer per-flow distributed admission control with dynamic reservation of bandwidth and multiple QoS guarantee of EDCA(enhanced distributed channel access) in IEEE802.11e is presented in this paper. In this mechanism, the bits of the subcarrier of OFDM in the station is allocated to obtain the maximum capacity of the channel objected to the maximum power. And then the dynamic reservation of bandwidth is suggested based on the bits allocation and distributed measurement, which adapts to the requirement of the service and maximize the efficiency of the spectrum simultaneously. Meanwhile, the central control estimation of the surplus factor based on model by AP is presented to overcome the inaccuracy by direct measurement and ensure the enough bandwidth for every flow to avoid collision. The double criterion of the collision probability and the bandwidth are adopted to ensure multiple parameters of the quality of the services

such as bandwidth and delay as well as error ratio of frame.

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Juling Zeng, received the B.Sc. and M.Sc. degrees from Xidian University, China, and the Ph.D. degree from the Beijing University of Posts and Communications, China, all in electrical engineering. Her current research interests include quality-of service provisioning and resource allocation for bandwidth wireless networks and internet of things. She is a professor of the College of Computer and Information Technology, China Three Gorges University.

Lisi Jiang is currently a student of the school of information and communication engineering of the Beijing University of Posts and Communications, China. She has ever received the first and the second class scholarship each once, respectively. Now she is a main member of a innovation project of the university,where she engage in the research of resource allocation.