Dynamic bandwidth Partition with Finer-Tune (DP-FT) scheme for Multimedia IEEE 802.11e WLANs

Yang Xiao, Frank Haizhon Li, Ming Li, Jingyuan Zhang, Bo Li, and Fei Hu

Abstract— In mobile cellular networks, bandwidth is deterministic in terms of number of channels by frequency division, time division, or code division. On the other hand, bandwidth partition schemes in the contention-based medium access control (MAC) in distributed wireless LANs are extremely challenging due to the contention-based nature. In this paper, we propose and study a Dynamic bandwidth Partition with Finer-Tune (DP-FT) scheme for integrated voice/video/data traffic in the IEEE 802.11e wireless LANs.

Keywords—Resource Allocation, IEEE 802.11e, WIFI

I. INTRODUCTION

The Medium Access Control (MAC) layer is very fundamental for wired/wireless networks, such as Ethernet and IEEE 802.11 WLAN [1]. Ethernet and the IEEE 802.11 distributed WLAN have become widely deployed since these contention-based MAC protocols are simple, robust, and they allow fast installation with minimal management and maintenance costs. The MAC of IEEE 802.11 WLAN employs a mandatory contention-based channel access function called Distributed Coordination Function (DCF), and an optional, centrally controlled, channel access function called Point Coordination Function (PCF) [1]. The DCF adopts a carrier sense multiple access with collision avoidance (CSMA/CA) with binary exponential backoff. Both the IEEE 802.11 DCF and IEEE 802.3 are very robust protocols for the best-effort service.

On the other hand, centrally controlled MAC protocols and reservation-based protocols manage QoS more easily, but they are rarely implemented in today’s products for a myriad of reasons. Central controlled MAC protocols have higher complexity and are inefficient for normal data transmission; reservation-based protocols have higher complexity, lack robustness, and make strong assumptions such as global synchronizations; and finally, end users like contention-based protocols because they plug and play. It is likely that the contention-based wireless MAC protocols will be still widely adopted in the future. In this proposed work, we choose to work the most challenging choice, i.e., QoS bandwidth allocations at the contention-based wireless MAC layer. Without QoS support at the MAC layer, QoS support from higher layers is not possible.

Although contention-based MAC protocols are very successful commercially and are robust for the best-effort traffic, they are unsuitable for multimedia applications with QoS requirements [2-5]. However, QoS is important and necessary for real-time traffic such as voice and video. A station might have to wait an arbitrarily long time to transmit a frame so that real-time applications may suffer [5]. One possible solution is to provide a good priority scheme for the DCF. Simple DCF priority schemes can be easily designed with minor changes in the DCF, and they are quite effective [3-4]. To support the MAC-level QoS, the IEEE 802.11 Working Group has recently developed IEEE 802.11e [6]. The emerging IEEE 802.11e standard provides QoS features and multimedia support to the existing 802.11b/g and 802.11a WLANs, while maintaining a full backward compatibility with these legacy standards. The IEEE 802.11e MAC employs a channel access function, called Hybrid Coordination Function (HCF), which includes both contention-based channel access and centrally-controlled channel access mechanisms. The contention-based channel access mechanism is also referred to as Enhanced Distributed Channel Access (EDCA). The EDCA provides a priority scheme by differentiating the inter-frame space as well as the initial and the maximum contention window sizes for backoff procedures.

In the previous work in [2]-[5], [7]-[12], the main focus was on studying the EDCA mechanisms and differentiated services. However, without a good admission control mechanism and a good protection mechanism, the existing multimedia traffic cannot be protected and QoS requirements cannot be met. These schemes cannot provide guaranteed QoS, and the multimedia traffic cannot be protected. In the previous work [13-14], we proposed realistic two-level QoS protection and guarantee schemes so that the existing voice and video flows are protected from the new and other existing voice and video flows. However, in [13-14], bandwidth allocation problem has never been touched. QoS guarantee and bandwidth allocation schemes [15-21] have been well studied for mobile cellular networks, where bandwidth is deterministically allocated in terms of a number of channels by frequency division, time division, or code division. On the contrary, bandwidth allocation in contention-based distributed wireless LANs is extremely challenging due to the contention constraint, the packet-based network, and, most importantly, intense competition for the only one channel available. As a consequence, both guaranteeing bandwidth and allocating...

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Although contention-based MAC protocols are very successful commercially and are robust for the best-effort traffic, they are unsuitable for multimedia applications with QoS requirements [2-5]. However, QoS is important and necessary for real-time traffic such as voice and video. A station might have to wait an arbitrarily long time to transmit a frame so that real-time applications may suffer [5]. One possible solution is to provide a good priority scheme for the DCF. Simple DCF priority schemes can be easily designed with minor changes in the DCF, and they are quite effective [3-4]. To support the MAC-level QoS, the IEEE 802.11 Working Group has recently developed IEEE 802.11e [6]. The emerging IEEE 802.11e standard provides QoS features and multimedia support to the existing 802.11b/g and 802.11a WLANs, while maintaining a full backward compatibility with these legacy standards. The IEEE 802.11e MAC employs a channel access function, called Hybrid Coordination Function (HCF), which includes both contention-based channel access and centrally-controlled channel access mechanisms. The contention-based channel access mechanism is also referred to as Enhanced Distributed Channel Access (EDCA). The EDCA provides a priority scheme by differentiating the inter-frame space as well as the initial and the maximum contention window sizes for backoff procedures.

In the previous work in [2]-[5], [7]-[12], the main focus was on studying the EDCA mechanisms and differentiated services. However, without a good admission control mechanism and a good protection mechanism, the existing multimedia traffic cannot be protected and QoS requirements cannot be met. These schemes cannot provide guaranteed QoS, and the multimedia traffic cannot be protected. In the previous work [13-14], we proposed realistic two-level QoS protection and guarantee schemes so that the existing voice and video flows are protected from the new and other existing voice and video flows. However, in [13-14], bandwidth allocation problem has never been touched. QoS guarantee and bandwidth allocation schemes [15-21] have been well studied for mobile cellular networks, where bandwidth is deterministically allocated in terms of a number of channels by frequency division, time division, or code division. On the contrary, bandwidth allocation in contention-based distributed wireless LANs is extremely challenging due to the contention constraint, the packet-based network, and, most importantly, intense competition for the only one channel available. As a consequence, both guaranteeing bandwidth and allocating...
bandwidth are challenging issues. One of the key challenges is to guarantee the different QoS requirements for different traffic classes, while simultaneously ensuring that the scarce bandwidth is utilized efficiently.

In our previous work [24], we studied various sharing schemes. In this paper, we propose and study a Dynamic bandwidth Partition with Finer-Tune (DP-FT) scheme for integrated voice/video/data traffic in the IEEE 802.11e wireless LANs. The scheme contains both admission control and data control. In admission control mechanisms, video and voice flows are accepted or rejected based on the available budget (s), and a guard period is proposed to prevent bandwidth allocation from over provisioning. In the data control mechanism, best-effort data parameters are dynamically controlled based on traffic load condition. In a dynamic bandwidth partition scheme, bandwidth is dynamically partitioned among voice and video based on current voice/video/data traffic load condition. In the DP-FT scheme, bandwidth of voice and video is partitioned proportionally to voice traffic load and video traffic load in the current voice/video/data traffic load condition. In the DP-FT scheme, bandwidth of voice and video is partitioned proportionally to voice traffic load and video traffic load in the previous measurement interval, and a Finer-Tune (FT) method is adopted to handle some extreme cases to avoid starvation and over-provisioning for another real-time traffic. The FT scheme is defined as borrowing some bandwidth/budget from another Access Category (AC) if available before rejecting a flow.

II. THE SCHEMES

In this paper, measurements are conducted during each regular time interval, which can be a beacon interval or several beacon intervals. We define bandwidth as the time interval between two measurements, and it is a constant value. If a portion of bandwidth is partitioned, each AC occupies one partition of the whole portion, although the partition may be changed dynamically based on traffic load if it is a dynamic bandwidth partition scheme. On the other hand, if a portion of bandwidth is reserved, one particular AC occupies the whole portion.

Let $T$ denote the total bandwidth, which is the time interval between two measurements. In a fixed bandwidth partition scheme as shown in Fig.1a, total bandwidth is divided into three portions: a portion $(\alpha_1 T)$ for video only, a portion $(\alpha_2 T)$ for voice only, and a guard period $(\beta T)$ to prevent bandwidth allocation from over provisioning and for best-effort data traffic, where $\beta+\alpha_1+\alpha_2=1$.

![Fig.1 DP-FT scheme](image-url)

We propose two different bandwidth partition schemes, shown in Fig.1 for integrated voice/video/data traffic in the

IEEE 802.11e wireless LANs: a Static/fixed bandwidth Partition (SP) scheme and a Dynamic bandwidth Partition with Finer-Tune (DP-FT) scheme. Fig.1a is a special case of Fig.1b when $\alpha_2=\alpha_3=0$. The DP-FT scheme is given in the next subsection.

A. Static Partition or Fixed Partition

The Static Partition (SP) scheme is shown in Fig.1a. We only discuss admission control part here, and data control is discussed in the later subsection. In this subsection, an AC stands for either voice AC or video AC.

The distributed admission control is developed to protect active QoS flows, i.e., voice and video flows. The QAP announces the transmission budget via beacon frames, and the budget is shared by both voice and video. The budget indicates the allowable transmission time in addition to how much is being utilized. QoS Stations (QSTAs) determine an internal transmission limit per AC for each beacon interval, based on the transmission count during the previous beacon period and the transmission budget announced from the QAP. The local voice/video transmission time per beacon interval shall not exceed the internal transmission limit per AC. When the transmission budget is depleted, new flows will not be able to gain transmission time, while existing flows will not be able to increase the transmission time per beacon interval, which they are already using. This mechanism protects existing flows.

1) Procedure at QAP

The QoS Parameter Set Element (QPSE) provides information needed by QSTAs for a proper operation of the QoS facility during a contention period. The QPSE includes $CW_{\text{min}}[i]$, $CW_{\text{max}}[i]$, $AIFS[i]$, for $(i=0,\ldots,3)$, and $TXOPBudget[i]$ for $(i=1,2,3)$, and $\text{SurplusFactor}[i]$ for $(i=1,2,3)$. These are global variables in the sense that they are maintained by QAP and transmitted to QSTAs via beacon frames. The first three variables/parameters were already discussed in the previous sections. $TXOPBudget[i]$ specifies the additional amounts of time available for AC $i$, respectively, during the next beacon interval, and $\text{SurplusFactor}[i]$ ($>1$) represents the ratio of over-the-air bandwidth reserved for AC $i$ to bandwidth of the transported frames required for successful transmission. Note that bandwidth more than the minimum required is typically reserved to compensate potential transmission failures, e.g., due to collisions. The QPSE is calculated by the QAP for each beacon interval and embedded into the next beacon frame.

The QAP shall measure the amount of time occupied by transmissions from each AC during the beacon period, including associated SIFS and ACK times if applicable. The QAP shall maintain a set of counters $TxTime[i]$, which shall be set to zero immediately following the transmission of a beacon. For each data frame transmission (either uplink or downlink), the QAP shall add the time, equal to the frame transmission time and all overhead involved such as SIFS and ACK, to the $TxTime$ counter corresponding to the AC of that frame. The QAP determines $TXOPBudget[i]$ by:

$TXOPBudget[3] = \max(\alpha_3 T - TxTime[3] \times \text{SurplusFactor}[3], 0)$

$TXOPBudget[2] = \max(\alpha_2 T - TxTime[2] \times \text{SurplusFactor}[2], 0)$

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where $\alpha, \tau$ and $\alpha, \tau$ are defined in Fig.1a ($\alpha + \alpha, \beta + 1$) and AC=1 is not used. How to choose SurplusFactor is well studied in our previous work [15].

2) Procedure at Each QSTA

When the transmission budget is depleted, new QSTAs cannot gain transmission time, while existing QSTAs cannot increase the transmission time per beacon interval, which they are already utilizing. Accordingly, this mechanism protects existing flows.

Each QSTA has to maintain the following local variables for each AC: $TxUsed[i]$, $TxSuccess[i]$, $TxLimit[i]$, $TxRemainder[i]$, and $TxMemory[i]$. These are local variables in the sense that each station locally updates these variables by counting only those related to the station itself. In other words, local variables are those related to a particular station and obtained from the viewpoint of this station, whereas global variables are related to all stations in the common wireless channel, and obtained from the viewpoint of the AP. $TxUsed[i]$ counts the amount of time occupied on-air by transmissions, irrespective of success or not, from AC $i$ of this station, including associated SIFS and ACK times if applicable. $TxSuccess[i]$ counts for the transmission time for successful transmissions. A station shall not transmit a data frame if doing so would result in the value in $TxUsed[i]$ exceeding the value in $TxLimit[i]$, where to how determine this value is presented below. If the QSTA is prevented from sending a frame for this reason, it may carry over the partial frame time remainder to the next beacon interval, by storing the remainder in $TxRemainder[i]$, where $TxRemainder[i] = TxLimit[i] - TxUsed[i]$. Otherwise, $TxRemainder[i] = 0$. $TxMemory[i]$ ‘memorizes’ the amount of resource that AC $i$ of this station utilized during a beacon interval. Let $f$ denote the damping factor whose function will be explained below. Let $B[i]$ denote a predefined threshold. Note that $B[i]$ is also referred as to an inside guard period, which will be discussed in a later subsection. $B[i]$ is also referred as to non-zero-budget in [14-15] to prevent over-provisioning, and it is essential to provide a stable quality of video and voice. It is similar to the guard period, and it has been well studied in [14-15]. At each target beacon transmission time, the $TxMemory$, $TxLimit$ and $TxSuccess$ variables are updated according to the following procedure:

- If $TXOPBudget[i]<B[i]$
  - Both $TxMemory[i]$ and $TxRemainder[i]$ shall be set to zero for new QSTAs which start transmission with this AC in the next beacon interval. All other QSTAs’ $TxMemory[i]$ remains unchanged;
- Else
  - For new QSTAs, which start transmission with this AC in the next beacon interval, an initial value for $TxMemory[i]$ is assigned a number between 0 and $TXOPBudget[i]/SurplusFactor[i]$. All other QSTAs’ $TxMemory[i]$ are updated according to the following procedure:
    - $TxMemory[i] = f \times TxMemory[i] + (1-f) \times (TxSuccess[i] \times SurplusFactor[i] + TXOPBudget[i]);$
- $TxSuccess[i] = 0$;
- $TxLimit[i] = TxMemory[i] + TxRemainder[i]$;

Note that in the above procedure, only $TXOPBudget[i]$ and $SurplusFactor[i]$ are global variables, and the others are local variables. From the above procedure, when the transmission budget for an AC becomes zero,

- Its $TxLimit[i]$ will become zero for new STAs, and hence AC $i$ of any new QSTA will not be able to gain a transmission time in the next beacon interval.
- The existing QSTAs’ $TxMemory[i]$ remains unchanged, and hence the existing QSTAs’ $TxLimit[i]$ remains basically unchanged. In other words, existing stations will not be able to increase the transmission time above what they are already using. Note that this mechanism protects existing flows.

From the above procedure, as long as the transmission budget is larger than zero, both $TxMemory[i]$ and $TxLimit[i]$ need be adjusted periodically. The new $TxMemory[i]$ value is a weighted average of the old $TxMemory[i]$ value and the sum of the successful transmission time and the budget. The value $TxSuccess[i] \times SurplusFactor[i]$ + $TXOPBudget[i]$ is the target to which $TxMemory$ converges. The $TxLimit$ is equal to $TxMemory$ plus a possible capped remainder, where $TxMemory$ ‘memorizes’ the amount of time, which a specific AC of the QSTA has been able to utilize per beacon interval. Once the budget is depleted (i.e., $TXOPBudget[i]$ hovers around 0), $TxMemory$ converges to $TxSuccess$, which is the lower limit. This ensures that a QSTA can continue consuming the same amount of time in subsequent beacon intervals. The damping allows for some amount of fluctuation to occur. However, $TxMemory$ cannot grow any further in the saturated state. This prevents new flows from entering a specific AC when it is saturated.

The damping factor does not affect the entrance of a newly entered flow into the system when an enough budget is available, because the decreased $TXOPBudget[i]$ is offset by an increased $TxSuccess$ instantaneously, so $TxMemory$ does not change a lot. In other words, for a newly entered flow, $TXOPBudget[i]$ is decreased due to this new entrance, and $TxSuccess$ is increased since it is changed from zero to a positive value so that the sum of these two in the algorithm above does not change a lot. The damping factor does affect $TxMemory$ when a new flow starts up in a QSTA, which does not have an existing flow of the corresponding AC. In such a case, the decreased $TXOPBudget[i]$ is not offset by an increased $TxSuccess$, and the $TxMemory$ converges to the lower target value consequently. QSTAs shall not increase their $TxLimit[i]$ if they did not transmit traffic of AC $i$ during the previous beacon interval.

For each video/voice flow, a Leaky-Bucket algorithm plus a Token-Bucket algorithm can be also implemented at the QSTA to control the flow rate.

B. Data Control

Since too many data transmissions can degrade the performance of existing voice and video flows, we propose a
retry-based data control mechanism in this paper to dynamically adjust data traffic parameters based on traffic condition. In the proposed approach, stations dynamically adjust the EDCA data parameters based on the behavior of one or more frame transmission(s). During each frame transmission, whenever the number of retries ever reaches a threshold $K$, the next frame’s initial window size is increased by $CW_{\text{min}}[0] = \alpha \times CW_{\text{min}}[0]$; whenever there are $L$ consecutive successful transmissions, the next frame’s initial window size is decreased by $CW_{\text{min}}[0] = CW_{\text{min}}[0]/\theta$. Note the above changes should be within the data EDCA parameter’s range, e.g., they are changed into integers, $CW_{\text{min}}[0] \geq CW_{\text{min}}[1]$ holds all the time, and otherwise, no change should be made.

C. Dynamic Partition Scheme

In a dynamic bandwidth partition scheme, bandwidth is dynamically partitioned among voice and video based on current voice/video/data traffic load condition. We propose and study two different bandwidth partition schemes, shown in Fig.1: the SP and DP-FT schemes. Since data control and guard period are the same as the SP scheme, the section only focuses on how to implement the dynamic partition concept. Furthermore, due to the limited space, we cannot present the whole schemes in details, but we only present major differences in each scheme, and many functions are similar to Section IIIA with some revisions (e.g.,) about $TXOPBudget$.

D. Dynamic Budget Partition

In a dynamic bandwidth partition scheme, bandwidth is dynamically partitioned among voice and video based on current voice/video/data traffic load condition. We propose and study two different bandwidth partition schemes, shown in Fig.1: the SP and DP schemes. Since data control and guard period are the same as the SP scheme, the section only focuses on how to implement the dynamic partition concept. Furthermore, due to the limited space, we cannot present the whole schemes in details, but we only present major differences in each scheme, and many functions are similar to Section II A with some revisions (e.g.,) about $TXOPBudget$.

In the DP scheme, shown in Fig.1b, instead of partitioning bandwidth, budget of voice and video is partitioned proportionally to voice traffic load and video traffic load for the previous measurement interval, we need to calculate the total budget ($B$)

$$B = \sum [A[i] \times T_{\text{TxTime}}[i] \times \text{SurplusFactor}[i]]$$

For the next measurement interval, we have $TXOPBudget[3] = \alpha_i \times B$ for voice $TXOPBudget[2] = \alpha_i \times B$ for video $A[i] = TXOPBudget[i] + T_{\text{TxTime}}[i] \times \text{SurplusFactor}[i]$ Note that in this paper AC=1 is not used.

E. Dynamic Partition with Finer-Tune

In the DP-FT scheme, shown in Fig.1b, bandwidth of voice and video is partitioned proportionally to voice traffic load and video traffic load in the previous measurement interval, and a Finer-Tune (FT) method is adopted to handle some extreme cases to avoid starvations and over-provisioning for another real-time traffic. The FT scheme is defined as borrowing some bandwidth/budget from another AC if available before rejecting a flow.

An intuitive approach is that bandwidth of voice and video is partitioned proportionally to voice traffic load and video traffic load in the previous measurement interval. Let’s define the following notations for the previous measurement interval:

$$U[i] = T_{\text{TxTime}}[i] \times \text{SurplusFactor}[i]$$

$$A[i] = TXOPBudget[i] + U[i]$$

We should have $\sum A[i] = T$. For the next measurement interval, we should define $A[i] = U[i]/\left( \sum_{i} T_{\text{TxTime}}[i] \right)$ for proportional (which is better than simulations, omitted) such that $\forall i, A[i] - U[i] \geq 0$. However, the above scheme may cause starvations or over-provisioning for another real-time traffic under some extreme cases. For example, 1) in the previous measurement interval, if $U[3]=0$ (voice), $U[1]=U[0]=0$, $U[2]>0$ (video), it may cause starvations for voice traffic later; 2) based on our simulations, when one AC (e.g., voice)’s left budget is no large enough for acceptance of another voice flow and there is a large portion of available video’s budget, the newly arrived voice flow may be rejected due to a large difference of the required bandwidth for a video flow and the required bandwidth for a voice flow — but a good approach should be that voice can borrow some bandwidth/budget from video so that the newly arrived voice flow is accepted. The borrowing bandwidth idea in case 2) can also avoid the problem of case 1). Therefore, we propose the FT method as a fix for the problems, where the FT scheme is defined as to try to borrow some bandwidth/budget from another AC (voice form video or video from voice) if available before rejecting a real-time flow.

III. PERFORMANCE EVALUATION

In this section, we conduct performance evaluation for the proposed partition schemes via extensive simulations. We adopted IEEE 802.11a [1] and IEEE 802.11e draft [2] in these simulations. Three traffic types are considered in our simulations: voice (AC 3), video (AC 2), and best-effort data (AC 0).

A. Performance Metrics and Simulation Setup

We adopt the following performance metrics in our simulations: (1) average throughput per voice flow, video flow, or data station, (2) total throughput, (3) $T_{\text{Txlimit}}$, (4) $T_{\text{xBudget}}$, (5) number for accepted and active flows (NAAF) in the system per AC, and (6) throughput square relative difference (SRD). Throughput SRD is proposed to characterize the normalized difference of achieved throughput and required throughput. Let $k(t)$ and $T(t)$ denote the number of flows in an AC (AC>0) and the average
throughput, respectively, at the $t$-th measurement interval. Let $T_i$ denote the required throughput for flow $i$ $(i = 1, 2, ..., k(t))$. Throughput SRD at the $t$-th measurement interval for this AC is defined

$$\text{SRD}_i(t) = \sum_{t=1}^{k(t)} (T_i(t) - T_i)^2$$

We assume that the transmitted traffic is not larger than required throughput on average; otherwise, a token bucket algorithm can be also implemented to control the traffic rate. Throughput SRD can be only applied to voice and video traffic, but not data traffic.

The defaults EDCA access parameters used for our simulations are listed as follows: $\text{CWmin}[3] = 16$; $\text{CWmax}[3] = 256$; $\text{AIFS}[3] = 25\mu s$; $\text{CWmin}[2] = 32$; $\text{CWmax}[2] = 2048$; $\text{AIFS}[2] = 25\mu s$; $\text{CWmin}[0] = 256$; $\text{CWmax}[0] = 51200$; $\text{AIFS}[0] = 34\mu s$; and queue size is 30 frames for each AC (voice, video, data). For other parameters, the following values are adopted unless stated otherwise: beacon interval is 100ms; damping factor is 0.9; each voice flow is 0.0832Mbps, which is generated by a constant inter-arrival time 20ms with an exponential distribution with a mean inter-arrival time $2\mu s$; Slot time is 9$\mu$s; and a symbol time is 4$\mu$s. We also have $\alpha_2 = \alpha_i = 0.4$, $\beta = 0.2$.

### B. DP vs. DP-FT

In our previous work [23], we show that a Dynamic budget Partition (DP) scheme is better than the SP scheme. Therefore, we just need to compare the DP and DP-FT schemes.

This subsection compares the DP scheme and the DP-FT scheme under the following traffic pattern: at 5s, 1 voice flow, 1 video flow, and 1 data are added; then 1 video flow is added at each 5s till the number of voice flows is 50, and 1 data station is added at each 5s till the number of data stations is 10; the simulation time is 250s.

Fig. 2 shows the NAAF and the average throughput for the DP and DP-FT schemes. The DP-FT scheme outperforms the DP scheme by accepting 49 more additional voice flows (50 voice flows for the DP-FT scheme vs. 1 voice flow for the DP scheme at 80s) while maintaining the similar average throughputs for voice and video. The reason is that the DP scheme still can cause starvation in this special traffic case due to the large difference of the required budget for a video flow and the required budget for a voice flow. In other words, access time of one voice flow is very small compared to access time of one video flow so that it causes that the ATL for voice is very small after proportional partition in certain traffic conditions.

Fig. 3 shows that 1) the throughput SRDs are almost the same (near zero, indicating good QoS); 2) the number of collisions for the DP-FT is larger than that of the DP scheme since 49 more voice flows (small frames) are accepted; 3) the total throughput of the DP-FT scheme is a little better than that of the DP scheme. In other words, the DP-FT scheme is better than the DP scheme by accepting more voice flows, having a little better total throughput, and being with the similar QoS.

### IV. CONCLUSIONS

In this paper, we study bandwidth allocation for contention-based MAC with QoS guarantee. We propose and study the DP-FT scheme. In the DP-FT scheme, bandwidth of voice and video is partitioned proportionally to voice traffic load and video traffic load in the previous measurement interval, and the FT method is adopted to handle some extreme cases to avoid starvation and over-provisioning for another real-time traffic, via borrowing some bandwidth/budget from another AC if available before rejecting a flow. Our simulations show that the DP scheme is better than the SP scheme, and the DP-FT scheme is better than the DP scheme. The DP scheme still has some starvation in some particularly designed special cases, and this can be avoided by either adopting the DP-FT scheme.

### ACKNOWLEDGEMENT

Bo Li’s research was supported in part by grants from RGC under the contracts 616406 and 616207, by a grant from NSFC/RGC under the contract N_HKUST603/07, by a grant from HKUST under the contract RPC06/07.EG27, by a grant from National Basic Research Program of China (973 Program) under the contract 2006CB303000, and by a grant from Nokia Research under the contract NOKIA001-BL/06/07.

### REFERENCES


