Published in IET Communications
Received on 30th March 2007
Revised on 20th December 2007
doi: 10.1049/iet-com:20070488

ISSN 1751-8628

Effective transmission opportunity allocation scheme for real-time variable bit rate traffic flows with different delay bounds

T.-H. Lee Y.-W. Huang

Department of Communication Engineering, National Chiao Tung University, HsinChu, Taiwan, Republic of China
E-mail: tlee@banyan.cm.nctu.edu.tw

Abstract: The medium access control of IEEE 802.11e defines a novel coordination function, namely, hybrid coordination function (HCF), which allocates transmission opportunity (TXOP) to stations taking their quality of service (QoS) requirements into account. However, the reference TXOP allocation scheme of HCF controlled channel access, a contention-free channel access function of HCF, is only suitable for constant bit rate traffic. For variable bit rate traffic, packet loss may occur seriously. The authors propose a TXOP allocation scheme to efficiently allocate bandwidth and meet the QoS requirements in terms of both delay bound and packet loss probability. To achieve high bandwidth efficiency, the authors take advantage of not only intra-flow multiplexing gain of traffic flows with large delay bounds, but also inter-flow multiplexing gain of multiple traffic flows with different delay bounds. According to numerical results obtained by computer simulations, the proposed TXOP allocation scheme results in much higher bandwidth efficiency than previous algorithms under the same constraints of delay bounds and packet loss probability.

1 Introduction

Nowadays, wireless network such as IEEE 802.11 WLANs [1] is deployed widely with rapidly, increasing users all over the world. As real-time applications, such as VoIP and streaming video are getting more common in daily life, quality of service (QoS) guarantee over wireless networks is becoming an important issue. Generally speaking, QoS includes guarantee of maximum packet delay, delay jitter and packet loss probability. To cope with this problem, a new enhancement of WLANs, called IEEE 802.11e [2], is introduced to support the QoS requirements of real-time traffic.

IEEE 802.11e proposes a QoS-aware coordination function, which is called hybrid coordination function (HCF). This function consists of two-channel access mechanisms. One is contention-based enhanced distributed channel access (EDCA) and the other is contention-free HCF-controlled channel access (HCCA). Because of the contention-free nature, HCCA can provide much better QoS guarantee than EDCA [3].

HCCA requires a centralised QoS-aware coordinator, called hybrid coordinator (HC), which has a higher priority than normal QoS-aware stations (QSTAs) in gaining channel control. HC can gain control of the channel after sensing the medium idle for a point coordination function interframe space (PIFS) interval that is shorter than distributed coordination function interframe space (DIFS) adopted by QSTAs. After gaining control of the transmission medium, HC will poll QSTAs according to its polling list. In order to be included in HC's polling list, each QSTA needs to make a separate QoS service reservation, which is accomplished by sending the add traffic stream (ADDTS) frame to HC. In this frame, QSTAs can give their traffic characteristics a detailed description in the traffic specification (TSPEC) field. Based on the traffic characteristics specified in TSPEC and the QoS requirements, HC calculates the scheduled service interval (SI) and transmission opportunity (TXOP) duration for each admitted flow.

Upon receiving a poll, the polled QSTA either responds with QoS-data if it has packets to send or a QoS-null frame otherwise. When the TXOP duration of some QSTA ends, HC gains the control of channel again and either sends a QoS poll to the next station on its polling list or releases the medium if there is no more QSTA to be polled.

The TXOP calculation of the reference scheduler provided in IEEE 802.11e standard document is based on mean data rate and nominal medium access control service data unit (MSDU) size. It performs well for constant bit rate (CBR) traffic. For variable bit rate (VBR) traffic, packet loss may occur seriously. A simple solution to cope with this problem is to use the maximum data rate of a VBR flow to allocate TXOP. However, such a solution tends to significantly reduce bandwidth utilisation. In [4], a new HCF scheduling scheme, called fair HCF (FHCF) scheduling algorithm, was proposed for IEEE 802.11e WLANs, which aims to be fair for both CBR and VBR traffic. Unfortunately, it does not provide guaranteed delay bound for all admitted flows. The scheduling algorithms proposed in [5] and [6] provide bounded average packet delay without considering packet loss. Since many real-time VBR applications can tolerate packet loss as long as the loss probability is under a pre-defined level [7], one can manage medium bandwidth more efficiently if such a feature is utilised.

Fan *et al.* [8] and our previous work [9] modified the TXOP calculation and admission control unit of the reference design so that under a bounded delay, packet loss probability can be controlled under a pre-defined level. To reduce computational complexity, it is assumed that all real-time traffic flows have the same delay bound and a packet is dropped if it cannot be served in the next SI right after the one it arrived in. However, different real-time traffic flows may have distinct delay bounds [7]. In other words, for a given flow with delay bound greater than one SI, the allocated TXOP in a specific SI can be shared by packets that arrived in previous SIs as long as these packets do not violate their delay bound requirement. Such a mechanism is called intra-flow multiplexing. When there are multiple VBR flows in one QSTA, the total allocated TXOP can be smaller than the sum of the TXOPs allocated to each individual flow if the service requirements of these flows are considered together. Such a mechanism is called inter-flow multiplexing. Without taking into account the earlier two multiplexing mechanisms, a TXOP-allocation scheme tends to be too conservative and results in inefficient bandwidth allocation.

To reduce computational complexity, we assume in this paper that the arrival process of each real-time VBR traffic flow is Gaussian, which has been widely adopted as traffic arrival model in previous works [9–16]. It is often an acceptable assumption especially when there are multiple independent traffic flows multiplexed together. Based on this assumption, we propose an effective and efficient TXOP-allocation scheme and the associated admission control unit so that delay and packet loss probability can both be bounded. To achieve high bandwidth efficiency, we take advantage of not only intra-flow multiplexing gain of traffic flows with large delay bounds, but also inter-flow multiplexing gain of traffic flows with different delay bounds. Numerical results obtained by computer simulations show that our proposed TXOP allocation scheme results in much better performance than previous works. Although we assume the arrival process is Gaussian, our idea can be applied to other arrival processes.

The remainder of this paper is organised as follows. In Section 2, we describe the system model. In Section 3, we review related previous works, including the reference scheduler, admission control unit presented in the consensus proposal and other TXOP allocation algorithms. Section 4 contains our proposed TXOP allocation scheme for traffic flows with different delay bounds. Simulation results are provided and discussed in Section 5. Finally, we draw conclusions in Section 6.

1.1 Acronyms

The acronyms that will be frequently used in the paper are listed as follows:

ADDTS	add traffic stream
CBR	constant bit rate
DCF	distributed coordination function
DIFS	DCF interframe space
EDCA	enhanced distributed channel access
EDF	earliest deadline first
FCFS	first come first serve
HC	hybrid coordinator
HCCA	HCF-controlled channel access
HCF	hybrid coordination function
HOL	head of line
MAC	medium access control
MSDU	MAC service data unit
PCF	point coordination function
PIFS	PCF interframe space
QoS	quality of service
QSTA	QoS-aware station
SI	service interval
SIFS	short interframe space
TSPEC	Traffic specification
TXOP	transmission opportunity
VBR	variable bit rate

2 System model

We assume that transmission over the wireless medium is divided into SIs and the duration of each SI, denoted by SI, is a sub-multiple of the length of a beacon interval T_b.

doi: 10.1049/iet-com:20070488

Moreover, an SI is further divided into a contention period and a contention-free period.

When the connection request of a new real-time flow arrives, a QSTA has to negotiate with the HC for admission. The QSTA needs to describe the traffic characteristics of the new flow in the TSPEC field of the ADDTS frame. As an example, the mandatory traffic characteristics defined in the consensus proposal include mean data rate (ρ), nominal MSDU size (L) and maximum service interval ($\mathrm{SI}_{\max}$). The HC uses the specified traffic characteristics and the requested QoS requirements to calculate the TXOP allocation for the new flow and accepts it if the requested QoS can be guaranteed without violating the QoS requirements of existing connections.

In this paper, we assume that the specified maximum service interval is the delay bound of a flow and, in addition, a pre-defined packet loss probability has to be guaranteed for all real-time flows.

3 Previous works

In this section, we describe previous works, including the reference scheduler proposed in the IEEE 802.11e standard document and other designs based on exactly known arrival process or Gaussian approximation. For the ease of description, we assume that there are K QSTAs with a total of n existing flows, and QSTA_k is negotiating with HC for a new flow, i.e. flow $n+1$.

3.1 The reference scheduler

Let ρ_i, L_i and $\mathrm{SI}_{\max,i}$ denote respectively the mean data rate, nominal MSDU size and maximum service interval of flow i. In the reference scheduler, the HC determines a possible new SI according to $\mathrm{SI} = \min\{\mathrm{SI}, \mathrm{SI}_{\max,n+1}\}$. Note that SI has to be a sub-multiple of the beacon interval and is initially set to T_b, the duration of a beacon interval. The HC then calculates the TXOP duration for flow i, $1 \leq i \leq n+1$, as follows.

Firstly, the HC decides the average number of packets N_i that arrive at the mean data rate during one SI

$$N_i = \left\lceil \frac{\rho_i \cdot \mathrm{SI}}{L_i} \right\rceil \tag{1}$$

Secondly, the TXOP duration is obtained by

$$\mathrm{TD}_i = \max\left\{ N_i \times \left(\frac{L_i}{R_i} + O \right), \frac{M}{R_i} + O \right\} \tag{2}$$

where R_i is the minimum physical transmission rate, M and O denote, respectively, the maximum allowable size of MSDU and per-packet overhead in time units. The overhead O includes the transmission time for an ACK frame, inter-frame space, medium access control (MAC) header, CRC field and PHY PLCP preamble and header.

Finally, the total TXOP duration of station k with m traffic flows is obtained as

$$\mathrm{TXOP}_k = \left(\sum_{i=1}^{m} \mathrm{TD}_i \right) + \mathrm{SIFS} + t_{\mathrm{POLL}} \tag{3}$$

where SIFS and t_{POLL} are, respectively, the short inter-frame space and the transmission time of a CF-Poll frame.

After calculating TXOP_k, $1 \leq k \leq K$, flow $n+1$ is admitted if the following inequality is satisfied:

$$\sum_{k=1}^{K} \frac{\mathrm{TXOP}_k}{\mathrm{SI}} \leq \frac{T_\mathrm{b} - T_{cp}}{T_\mathrm{b}} \tag{4}$$

where T_{cp} is the time used for EDCA traffic. In case the new flow is admitted, the value of SI is updated, if necessary. Note that the reference scheduler does not consider packet loss probability in calculating TXOP. Moreover, it is only suitable for CBR traffic.

3.2 Schemes with exactly known arrival process

There are schemes which assume that the arrival process is exactly known and is used to compute TXOP with guaranteed packet loss probability [8, 17]. The basic idea is to set $\mathrm{SI} = \min_{1 \leq k \leq K} \mathrm{SI}_{\max,k}$ and assume that all traffic flows have the same delay-bound SI. As a result, the system can be modelled as a buffer-less queue such that packets arrived in the previous SI are lost if they cannot be served in the current SI. Under the buffer-less queueing model, the packet loss probability (P_L) of a single flow can be derived in terms of allocated TXOP duration (TD) and the probability density function of packet arrivals in one SI (see, e.g. (8) of [8]). The effective TXOP with guaranteed packet loss probability can be determined with some numerical algorithm, such as the bisection method for the derived equation. Obviously, one difficulty of these schemes is to exactly know the arrival process. Even if the arrival process is exactly known, the computational complexity of the numerical algorithm could make the scheme infeasible in practice. In fact, the complexity increases dramatically when there are multiple VBR flows and the inter-flow multiplexing gain is to be taken into account. Because of the buffer-less assumption, the intra-flow multiplexing gain of flows with delay bounds greater than one SI is neglected, meaning that the TXOP durations allocated to such flows tend to be too conservative.

3.3 Schemes with Gaussian approximated arrival process

Since the arrival process is difficult to be exactly known, the authors of [9] adopted the Gaussian approximation assuming that the traffic arrival of a VBR flow in an SI can be modelled as a Gaussian random variable with

IET Commun., 2008, Vol. 2, No. 4, pp. 598–608
doi: 10.1049/iet-com:20070488

probability density function $N(\mu,\sigma^2)$. It is still assumed that all traffic flows have the same delay-bound SI. Consequently, the packet loss probability can be derived as [18]

$$\begin{aligned} P_L &= \frac{E((\lambda_i - c)^+)}{E(\lambda_i)} \\ &= \frac{\int_c^\infty (y-c)\cdot f_{\lambda_i}(y)\mathrm{d}y}{\mu} \\ &= \frac{\int_c^\infty (y-c)\cdot (1/\sqrt{2\pi}\sigma)\mathrm{e}^{-((y-\mu)^2/2\sigma^2)}\mathrm{d}y}{\mu} \\ &= Q\left(\frac{c-\mu}{\sigma}\right) + \left[\frac{\sigma}{\mu\sqrt{2\pi}}\mathrm{e}^{-((c-\mu)^2/2\sigma^2)} - \frac{c}{\mu}Q\left(\frac{c-\mu}{\sigma}\right)\right] \end{aligned} \tag{5}$$

where

$$Q(\alpha) = \int_\alpha^\infty (1/\sqrt{2\pi})\mathrm{e}^{-(x^2/2)}\mathrm{d}x$$

where λ_i and c denote respectively, the amount of traffic arrival in the ith SI and the maximum amount of traffic that can be served in one SI. Variable c is called the effective bandwidth of the VBR flow because, given a bounded delay (i.e. SI), the packet loss probability is guaranteed to be no greater than the pre-defined threshold P_L. The effective bandwidth c can be represented as

$$c = \mu + \alpha\sigma \tag{6}$$

where α is called the QoS parameter of the flow and is dependent on the packet loss probability. Substituting (6) into equation (5), we get

$$\begin{aligned} P_\mathrm{L} &= Q(\alpha) + \left[\frac{\sigma}{\mu\sqrt{2\pi}}\mathrm{e}^{-(\alpha^2/2)} - \left(1+\frac{\alpha\sigma}{\mu}\right)Q(\alpha)\right] \\ &= Q(\alpha) + R(\alpha, \mu, \sigma) \end{aligned} \tag{7}$$

where $R(\alpha,\mu,\sigma)$ represents the bracket term in (7).

Since [19]

$$Q(\alpha) > \frac{1}{\sqrt{2\pi}}\,\frac{\alpha}{1+\alpha^2}\mathrm{e}^{(-\alpha^2/2)} \tag{8}$$

we conclude that $R(\alpha,\mu,\sigma)$ has a lower bound shown as follows

$$R(\alpha, \mu, \sigma) < \frac{1}{\sqrt{2\pi}}\,\frac{1}{1+\alpha^2}\,\mathrm{e}^{-(\alpha^2/2)}\left(\frac{\sigma}{\mu} - \alpha\right) \tag{9}$$

As a result, we can use $P_\mathrm{L} = Q(\alpha)$ to approximate the packet loss probability with a negative deviation if $\frac{\sigma}{\mu} < \alpha$. From the numerical results obtained with computer simulations for some typical applications, it was found that the packet loss probability can still be guaranteed and the deviation from the desired packet loss probability is small if the term $R(\alpha,\mu,\sigma)$ is neglected [9]. Therefore we will use $P_\mathrm{L} = Q(\alpha)$ to determine the QoS parameter in this paper.

It is clear that inter-flow multiplexing gain can be easily achieved under the Gaussian approximation because sum of independent Gaussian random variables remains Gaussian. Again, the Gaussian approximation does not take advantage of the intra-flow multiplexing gain by keeping packets for more than one SI and thus is too conservative for flows with large delay bounds.

4 Our proposed TXOP allocation scheme

In this section, we present our TXOP allocation scheme which allows packets to be kept for more than one SI as long as they do not violate the delay-bound requirement. To simplify TXOP calculation and admission control, we adopt the Gaussian approximation.

4.1 Flows with identical delay bounds

It is easy to obtain inter-flow multiplexing gain under Gaussian approximation if all flows have identical delay bounds. Therefore it suffices to study the single flow case. Let $N(\mu,\sigma^2)$ be the distribution of traffic arrival in one SI and $\mathrm{SI}_{\max}$ the maximum service interval, which satisfies

$$\left\lfloor \frac{\mathrm{SI}_{\max}}{\mathrm{SI}} \right\rfloor = \beta \tag{10}$$

where β is a positive integer ≥ 1. If $\beta = 1$, then packets cannot be kept for more than one SI and the buffer-less model applies. Assume that $\beta > 1$. In this case, packets can be kept for up to β SI. Let λ_i and Q_i denote, respectively, the amount of traffic arrival in the ith SI and buffer occupancy at the end of the ith SI. We assume that packet arrivals in the ith SI happen at the beginning of the ith SI and are eligible for service starting from the ith SI. The service discipline is assumed to be first come first served (FCFS). Let c represent the maximum amount of traffic that can be served in one SI. As a result, the system can be modelled as a finite buffer queue with buffer size βc. Note that it is not helpful to use a buffer of size greater than βc because a packet will violate its delay bound if buffer occupancy is larger than or equal to βc upon its arrival. Based on the finite buffer queue model, one can derive the traffic lost in the ith SI as

$$\begin{aligned} &\mathrm{Loss}_i = (\lambda_i + Q_{i-1} - \beta c)^+ \\ &\text{where } a^+ = \max(a, 0) \end{aligned} \tag{11}$$

Moreover, the buffer occupancy can be updated according to

$$Q_i = (\min(\lambda_i + Q_{i-1} - c, (\beta - 1)c))^+ \quad (12)$$

It is a difficult task to derive the exact packet loss probability for a finite buffer system. An approximated calculation of packet loss probability for a finite buffer system under Gaussian input arrival was presented in [10]. Its basic idea is to approximate the loss probability of the finite buffer system with the tail probability of an infinite buffer system and the loss probability of the buffer-less system, as shown in (13)

$$P_L(x) \simeq \frac{P_{\mathrm{L}}(0)}{P(Q > 0)} P(Q > x) \quad (13)$$

where $P_{\mathrm{L}}(x)$ represents the packet loss probability of a finite buffer system with buffer size x and $P(Q > x)$ denotes the tail probability above level x of an infinite buffer system. Applying the approximation to (11), one can obtain the approximate packet loss probability as follows.

$$P_{\mathrm{L}}(\beta c) \simeq \frac{\sigma}{\mu\sqrt{2\pi}} \mathrm{e}^{(-\alpha\beta c/\sigma)} - \frac{\alpha\sigma}{\mu} \mathrm{e}^{(\alpha^2/2)-(\alpha\beta c/\sigma)} Q(\alpha) \quad (14)$$

Given μ, σ, β and P_{L}, the bisection method can be adopted to derive the effective QoS parameter α, and thus effective bandwidth c can be obtained.

We conclude our proposed effective TXOP allocation scheme for a single VBR flow as follows.

Step 1. Find mean and variance of the VBR traffic flow.

Step 2. Determine the value of β.

Step 3. Compute α and use it to derive c.

Step 4. Estimate the number of packets N which can be transmitted in one SI. Since the nominal MSDU size is L, we estimate N by

$$N = \left\lceil \frac{c}{L} \right\rceil \quad (15)$$

Step 5. Calculate the allocated TXOP by

$$\mathrm{TXOP}_{\text{effective}} = \frac{c}{R_{\text{phy}}} + \text{per packet\_overhead} \times N$$

where R_{phy} is the physical transmission rate in TSPEC. (16)

The admission control can be decided by substituting the result of (16) into (4).

Note that the TXOP-allocation scheme presented earlier achieves intra-flow multiplexing gain for a VBR flow with delay bound greater than one SI. In the next sub-section, we propose a technique to achieve inter-flow multiplexing gain of flows with different delay bounds.

4.2 Flows with different delay bounds

Consider the multiplexing of two flows, called flow 1 and flow 2, such that the delay bound of flow 1 is one SI and that of flow 2 is β SI, where β is an integer greater than 1. Let $N(\mu_1,\sigma_1^2)$ and $N(\mu_2,\sigma_2^2)$ represent, respectively, the traffic arrival distributions of flow 1 and flow 2 in one SI.

The basic idea is to find an 'equivalent' flow with delay bound equal to one SI for flow 2 and then multiplex the equivalent flow with flow 1. Let c and α denote, respectively, the effective bandwidth and QoS parameter of flow 2. Further, let $N(\mu, \sigma^2)$ be the distribution of traffic arrival in one SI for the equivalent flow of flow 2. The equivalent flow is obtained by letting its mean and effective bandwidth equal to those of flow 2, that is, $\mu = \mu_2$ and $\alpha'\sigma = \alpha\sigma_2$, where α' is the QoS parameter of the equivalent flow, which can be easily determined with the table of the cumulative distribution function of $N(0,1)$. Consequently, we have $\sigma = \alpha\sigma_2/\alpha'$. Note that, because of intra-flow multiplexing gain, the QoS parameter of flow 2 is smaller than that of the equivalent flow. In other words, we have $\sigma < \sigma_2$.

After obtaining the equivalent flow, the aggregate effective bandwidth of flow 1 and flow 2 can be calculated with (7) using $N(\mu_1 + \mu, \sigma_1^2 + \sigma^2)$ as the distribution of traffic arrival in one SI. To handle multiple flows with various delay bounds, one can follow the following steps. In the following steps, we assume that there are M admitted flows which can be classified into g different groups according to their delay bounds. Let the delay bound of flow i be β_i SIs, $1 \le i \le M$. Flow i and Flow j are in the same group if and only if $\beta_i = \beta_j$. Notation G_i denotes the ith group. Moreover, flow (G_i) represents the aggregate flow of all the flows in group G_i.

Step 1. For $1 \le i \le g$, compute the mean $\tilde{\mu}_i$ and variance $\tilde{\sigma}_i^2$ of traffic arrivals in one SI for flow(G_i) by

$$\tilde{\mu}_i = \sum_{k \in G_i} \mu_k \quad (17)$$

$$\tilde{\sigma}_i^2 = \sum_{k \in G_i} \sigma_k^2 \quad (18)$$

where μ_k and σ_k^2 represent, respectively, the mean and variance of the traffic arrival distribution of Flow k.

Step 2. For $1 \le i \le g$, determine the effective bandwidth α_i for flow (G_i). It can be obtained by plugging $\tilde{\mu}_i$, $\tilde{\sigma}_i^2$, P_{L}, and $\tilde{\beta}_i$, the delay bound of flow (G_i), into (7) or (14).

doi: 10.1049/iet-com:20070488

Step 3. For $1 \le i \le g$, find the variance of the traffic arrival distribution $N(\tilde{\mu}_i, \tilde{\sigma}^2_{e,i})$ of the equivalent flow of flow(G_i) by

$$\tilde{\sigma}^2_{e,i} = \left(\frac{\alpha_i \cdot \tilde{\sigma}_i}{Q^{-1}(P_L)}\right)^2 \tag{19}$$

Step 4. Compute the aggregate effective bandwidth c for all the M flows.

$$c = \mu + \alpha\sigma_e$$
$$\text{where } \mu = \sum_{i=1}^{g} \tilde{\mu}_i \text{ and } \sigma_e^2 = \sum_{i=1}^{g} \tilde{\sigma}^2_{e,i} \tag{20}$$

Note that α represents the QoS parameter for all the M flows and can be obtained with (7).

The remaining steps to compute the aggregate TXOP for all the M flows is similar to the steps (i.e. steps 4 and 5) shown in previous sub-section. Again, we can perform admission control by plugging the calculated TXOP into (4). Numerical results presented in the following section show that inter-flow multiplexing gain can be obtained by converting the flows with different delay bounds into the equivalent flows defined in this sub-section without violating the packet loss probability requirement.

Note that, although an aggregate TXOP is allocated to multiple flows with different delay bounds, we need to maintain a separate queue for each group of flows. Since there are multiple queues, a service discipline is required to decide which head-of-line (HOL) packet is to be served. We suggest to use the earliest deadline first (EDF) scheduling to reduce the packet loss probability. Either round robin or random selection can be used when there are multiple HOL packets with the same minimum deadline.

5 Simulation results

The PHY and MAC parameters in our simulations are shown in Table 1. Note that the sizes of QoS-ACK and QoS-Poll in the table only include the sizes of MAC header and CRC overhead. We assume the minimum physical rate is 2 Mbps and t_{PLCP} is reduced to 96 µs. All related information is presented in Table 2.

The bit rate of ordinary streaming video is chosen from 500 kbps to 1.5 Mbps [20]. In our simulations, we consider three kinds of data rate: 500 kbps, 1 Mbps and 1.5 Mbps. As for nominal MSDU size, 750, 1000 and 1250 bytes are studied for each data rate. The behaviour of packet arrival is assumed to be Poisson process. The packet length varies according to exponential distribution with mean packet size equal to nominal MSDU size. All related parameters are summarized in Table 3. In addition, we assume that the scheduled service interval is 80 ms. Simulations are performed for 100 000 SIs.

The traffic is assumed to be delivered from QSTA to AP and the contention-free period occupies the entire scheduled service interval, i.e. 80 ms. The TXOP duration of the reference scheduler is calculated by plugging the simulation parameters into (1) and (2) shown in Section 2. The TXOP duration of the buffer-less queueing system is obtained by (7), whereas the TXOP duration for our proposed scheme is calculated by (14).

Table 1 PHY and MAC parameters

SIFS	10 µs
MAC header size	32 bytes
CRC size	4 bytes
QoS-ACK frame size	16 bytes
QoS CF-Poll frame size	36 bytes
PLCP header length	4 bytes
PLCP preamble length	20 bytes
PHY rate (R)	11 Mbps
minimum PHY rate(R_{min})	2 Mbps

Table 2 Transmission time for different header and per-packet overhead

PLCP preamble and header (t_{PLCP})	96 µs
Data MAC header (t_{HDR})	23.2727 µs
Data CRC (t_{CRC})	2.90909 µs
ACK frame (t_{ACK})	107.63636 µs
QoS-CF Poll (t_{POLL})	122.1818 µs
Per-packet overhead (O)	249.81818 µs

Table 3 TSPEC of different traffic

mean data rate (ρ)	500 k, 1 M, 1.5 M (bps)
nominal MSDU size (L)	750, 1000, 1250 (bytes)
scheduled service interval (SI)	80 (ms)
maximum service interval (SI_{max})	160, 240 (ms)
packet loss rate requirement (P_L)	0.01

doi: 10.1049/iet-com:20070488

Table 4 Simulation results for VBR traffic with $SI_{max} = 160$ ms

		Reference scheme				Buffer-less model				Our proposed scheme			
ρ, bps	L, bytes	N	TD, ms	P_L	P_W	N	TD, ms	P_L	P_W	N	TD, ms	P_L	P_W
500 K	750	7	5.567	0.0666	0.0690	15.704	12.563	0	0.5574	8.931	7.120	0.0041	0.2248
	1000	5	4.886	0.0930	0.0938	12.356	12.234	0	0.6005	6.863	6.740	0.0075	0.2815
	1250	4	4.636	0.1139	0.1153	10.580	12.366	0	0.6250	5.805	6.776	0.0098	0.3237
1 M	750	14	11.134	0.0388	0.0375	26.310	21.096	0	0.4715	15.984	12.716	0.0038	0.1270
	1000	10	9.771	0.0523	0.0523	20.404	20.085	0	0.5147	11.792	11.705	0.0044	0.1716
	1250	8	9.271	0.0641	0.0641	17.305	20.229	0	0.5432	9.952	11.545	0.0057	0.2051
1.5 M	750	20	15.905	0.0280	0.0280	34.713	27.678	0	0.4250	21.967	17.478	0.0028	0.0934
	1000	15	14.656	0.0368	0.0368	27.742	27.171	0	0.4602	16.983	16.598	0.0041	0.1203
	1250	12	13.907	0.0466	0.0466	23.396	27.265	0	0.4896	13.984	16.210	0.0055	0.1470

The simulation results for VBR traffic flows with delay bounds two or three times of SI are shown in Tables 4 and 5, respectively. In these tables, N means the number of packets that can be sent during one SI. It is clear that for the same traffic flow, packet loss probability (P_L) increases as the allocated TXOP duration decreases. On the other hand, the medium waste ratio (P_W), which is defined as the ratio of the wasted transmission time over the allocated TXOP duration, increases as the allocated TXOP increases. A good TXOP allocation mechanism should allocate TXOP durations as small as possible to reduce the medium waste ratio but simultaneously not to violate the pre-defined packet loss probability. Theoretically, the allocated TXOP based on the buffer-less model is independent of delay bound. This implies that both P_L and P_W are independent of delay bound as verified by the numerical results shown in Tables 4 and 5. In fact, the packet loss probability for the buffer-less model is always zero because the allocated TXOP is so large that all packets are served within one SI after their arrivals.

As one can see from Fig. 1, for both 160 and 240 ms delay bound requirements, the packet loss probabilities of our proposed scheme are under the pre-defined level, whereas those of the reference scheduler are not. For the buffer-less model, the packet loss probabilities can meet the pre-defined threshold. However, compared with our proposed scheme, it allocates larger TXOP durations and thus results in much higher medium waste ratio as shown in Fig. 2. The reason is that the buffer-less model does not take advantage of the intra-flow multiplexing gain. Note that since its medium waste ratio is independent of delay

Table 5 Simulation results for VBR traffic with $SI_{max} = 240$ ms

		Reference scheme				Buffer-less model				Our proposed scheme			
ρ, bps	L, bytes	N	TD, ms	P_L	P_W	N	TD, ms	P_L	P_W	N	TD, ms	P_L	P_W
500 K	750	7	5.567	0.0402	0.0419	15.704	12.563	0	0.5578	8.441	6.852	0.0008	0.1908
	1000	5	4.886	0.0585	0.0584	12.356	12.234	0	0.6007	6.409	6.410	0.0016	0.2386
	1250	4	4.636	0.0733	0.0737	10.580	12.366	0	0.6252	5.377	6.387	0.0025	0.2758
1 M	750	14	11.134	0.0229	0.0210	26.310	21.096	0	0.4714	15.439	12.419	0.0006	0.1028
	1000	10	9.771	0.0306	0.0327	20.404	20.085	0	0.5147	11.453	11.327	0.0009	0.1405
	1250	8	9.271	0.0377	0.0424	17.305	20.229	0	0.5437	9.448	11.088	0.0013	0.1682
1.5 M	750	20	15.905	0.0158	0.0159	34.713	27.678	0	0.4255	21.410	17.174	0.0006	0.0743
	1000	15	14.656	0.0222	0.0216	27.742	27.171	0	0.4610	16.438	16.202	0.0008	0.0961
	1250	12	13.907	0.0278	0.0285	23.396	27.265	0	0.4903	13.450	15.724	0.0011	0.1169

doi: 10.1049/iet-com:20070488

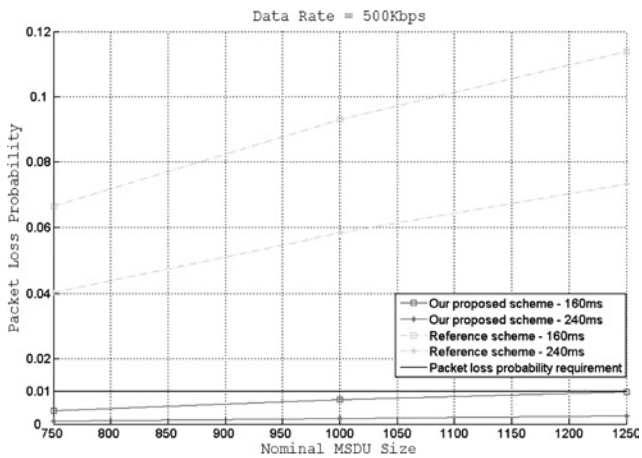

Figure 1 *Packet loss probability comparison*

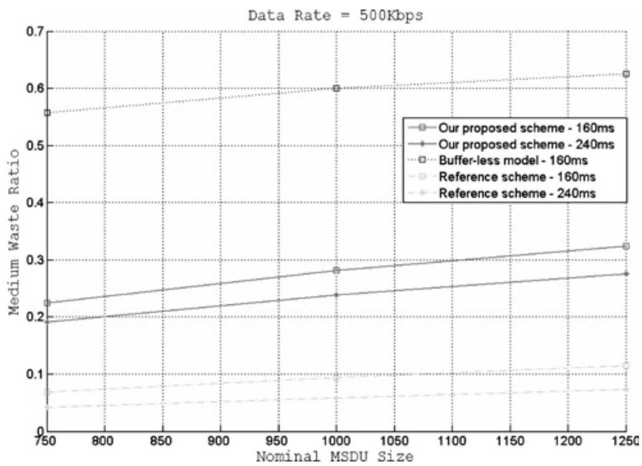

Figure 2 *Medium waste ratio comparison*

bound, only one curve for the buffer-less model appears in Fig. 2. Owing to space limitation, we only show figures for data rate equal to 500 kbps. The results for data rate equal to 1 or 1.5 Mbps are similar.

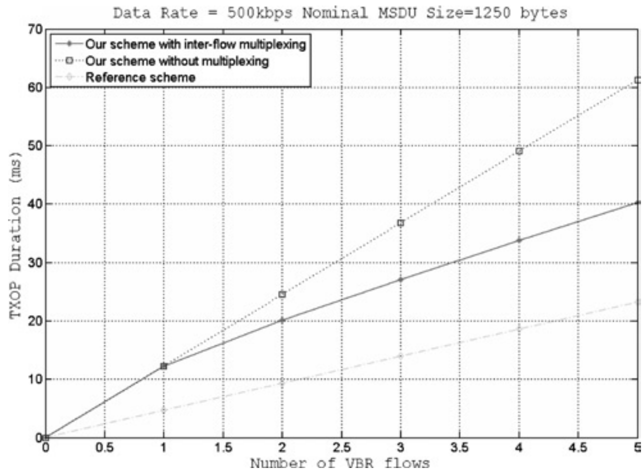

Figure 3 *TXOP durations against number of VBR flows*

All SI_{max} are equal to 80 ms

Table 6 shows the results when one QSTA requests multiple VBR flows, all of which have the same maximum service intervals as the scheduled service interval, that is, 80 ms. For $M = 2$ (i.e. two flows), the allocated TXOP is about 18% less than two times the TXOP allocated to an individual flow. The percentage of reduction increases as the number of flows increases, as illustrated in Fig. 3. Table 7 illustrates the results when one QSTA requests multiple VBR flows, one of which has the maximum service interval equal to the scheduled service interval (i.e. 80 ms), and all other M flows have two times the scheduled service interval (i.e.160 ms). For $M = 1$, the allocated TXOP with both intra- and inter-flow multiplexing gains is about 41% less than two times the TXOP allocated to an individual flow (without multiplexing gain), while the allocated TXOP with only intra-flow multiplexing gain is about 22% less than that. Comparing these two cases, we conclude that the introduction of inter-flow multiplexing of flows with different delay bounds can contribute about 19% reduction of TXOP allocation. The percentage of reduction increase as the number of concurrent flows increase, as illustrated in Fig. 4. Again, for these two cases, Figs. 5 and 6 show that our proposed scheme provides packet loss probability guarantee, whereas the reference

Table 6 Simulation results for multiple VBR traffic flows with $SI_{max} = 80$ ms

M	Reference scheme			Our proposed scheme					
				Without multiplexing			With inter-flow multiplexing		
	TD_{total}, ms	TD_{avg}, ms	P_L	TD_{total}, ms	TD_{avg}, ms	P_L	TD_{total}, ms	TD_{avg}, ms	P_L
1	4.636	4.636	0.2504	12.261	12.261	0.0070	12.261	12.261	0.0070
2	9.272	4.636	0.1762	24.522	12.261	0.0004	20.055	10.028	0.0030
3	13.908	4.636	0.1446	36.783	12.261	0	27.114	9.038	0.0023
4	18.544	4.636	0.1265	49.044	12.261	0	33.793	8.448	0.0017
5	23.180	4.636	0.1124	61.305	12.261	0	40.229	8.046	0.0014

doi: 10.1049/iet-com:20070488

Table 7 Simulation results for multiple VBR traffic flows

M	Reference scheme			Our proposed scheme								
				Without multiplexing			With only intra-flow multiplexing			With both intra- and inter-flow multiplexing		
	TD_{total}, ms	TD_{avg}, ms	P_L	TD_{total}, ms	TD_{avg}, ms	P_L	TD_{total}, ms	TD_{avg}, ms	P_L	TD_{total}, ms	TD_{avg}, ms	P_L
0	4.636	4.636	0.2508	12.261	12.261	0.0070	12.261	12.261	0.0070	12.261	12.261	0.0070
1	9.272	4.636	0.0960	24.522	12.261	0	19.037	9.519	0.0001	14.475	7.238	0.0028
2	13.908	4.636	0.0602	36.783	12.261	0	25.813	8.604	0	18.530	6.177	0.0018
3	18.544	4.636	0.0446	49.044	12.261	0	32.589	8.147	0	22.709	5.677	0.0016
4	23.180	4.636	0.0355	61.305	12.261	0	39.365	7.973	0	26.968	5.394	0.0016

One flow has $SI_{max} = 80$ ms and the other M flows have $SI_{max} = 160$ ms

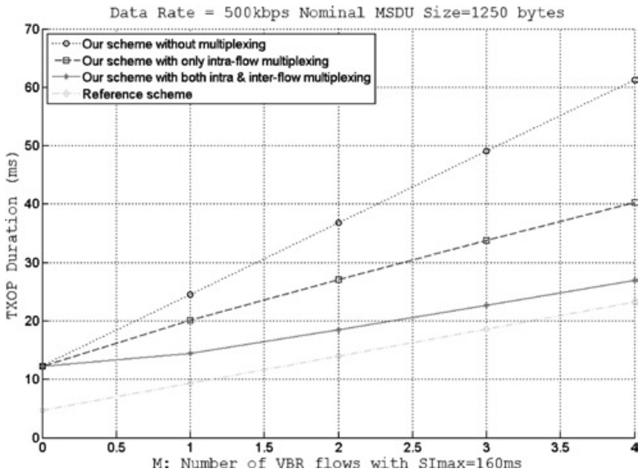

Figure 4 *TXOP durations against number of VBR flows*

One flow with $SI_{max} = 80$ ms and M flows with $SI_{max} = 160$ ms

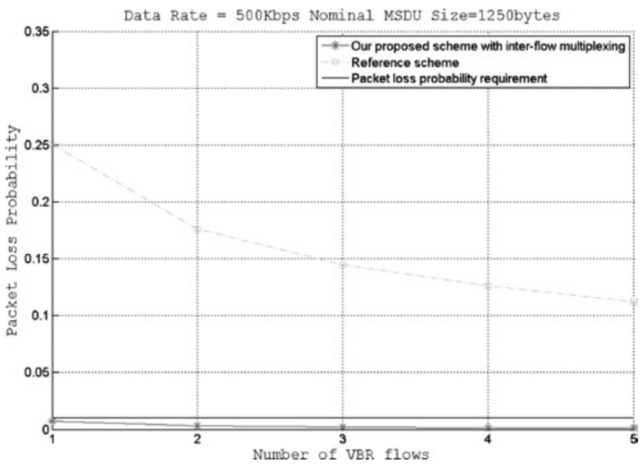

Figure 5 *Packet loss probability comparison*

All SI_{max} are equal to 80 ms

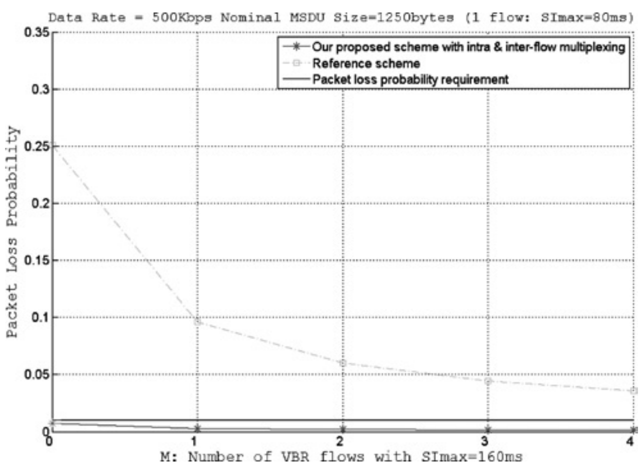

Figure 6 *Packet loss probability comparison*

One flow with $SI_{max} = 80$ ms and M flows with $SI_{max} = 160$ ms

doi: 10.1049/iet-com:20070488

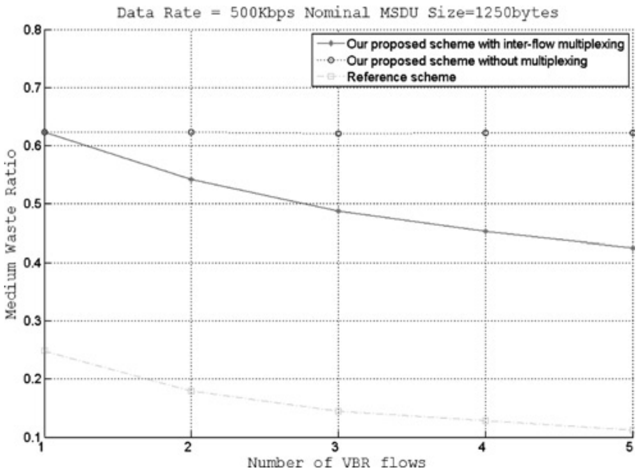

Figure 7 *Medium waste ratio comparison*

All SI_{max} are equal to 80 ms

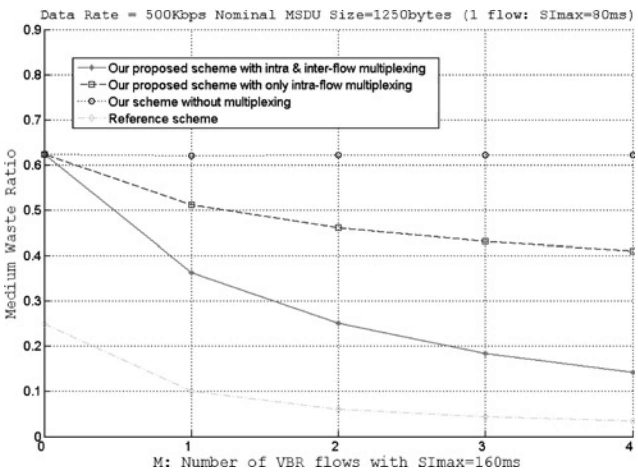

Figure 8 *Medium waste ratio comparison*

One flow with $SI_{max} = 80$ ms and M flows with $SI_{max} = 160$ ms

scheme does not. Besides, the medium waste ratio becomes smaller if more VBR flows are multiplexed, as illustrated in Figs. 7 and 8. Tables 6, 7, and Figs. 3–8 are all for VBR traffic with the following characteristics: variable packet size, mean data rate = 500 kbps and nominal MSDU size = 1250 bytes.

6 Conclusion

We have presented in this paper an effective TXOP allocation scheme for real-time VBR flows with various delay bounds. Our proposed scheme takes advantage of both intra- and inter-flow multiplexing gains to reduce the allocated TXOP durations when maintaining the QoS guarantee in terms of both delay bound and packet loss probability. The effectiveness of our proposed scheme is verified with computer simulations. In real systems, it is likely that there are only a limited number of possible applications, and therefore one can pre-compute the QoS parameter of each type of application so that admission control can be performed in real time. An interesting further research topic, which is currently under study, is to allocate an aggregate TXOP to multiple VBR flows which have different packet loss probability requirements.

7 References

[1] IEEE 802.11 WG: 'IEEE Standard 802.11-1999, Part 11: Wireless LAN MAC and Physical Layer Specifications', Reference number ISO/IEC 8802-11: 1999(E), 1999

[2] IEEE 802.11 EG: 'IEEE 802.11e/D8.0, Wireless MAC and physical layer specifications: MAC QoS enhancements'

[3] MANGOLD S., CHOI S., HIERTZ G.R., *ET AL.*: 'Analysis of IEEE 802.11e for QoS support in Wireless LANs', *IEEE Wirel. Commun.*, 2003, **10**, (6), pp. 40–50

[4] ANSEL P., NI Q., TURLETTI T.: 'FHCF: a fair scheduling scheme for IEEE 802.11e WLAN', *INRIA Res. Rep.*, 2003, **4883**

[5] HIGUCHI Y., FORONDA A., OHTA C., YOSHIMOTO M., OKADA Y.: 'Delay guarantee and service interval optimization for HCCA in IEEE 802.11e WLANs'. Proc. IEEE WCNC, 2007

[6] ANNESE A., BOGGIA G., CAMARDA P., GRIECO L.A., MASCOLO S.: 'Providing delay guarantees in IEEE 802.11e networks'. Proc. WWIC, 2005

[7] FATTAH H., LEUNG C.: 'An overview of scheduling algorithm in wireless multimedia networks', *IEEE Wirel. Commun.*, 2002, **9**, (5), pp. 76–83

[8] FAN W.F., GAO D., TSANG D.H.K., BENSAOU B.: 'Admission control for variable bit rate traffic in IEEE 802.11e WLANs'. Proc. IEEE LANMAN, 2004

[9] HUANG Y.W., LEE T.H., HSIEH J.R.: 'Gaussian approximation based admission control for variable bit rate traffic in IEEE 802.11e WLANs'. Proc. IEEE WCNC, 2007

[10] KIM H.S., SHROFF N.B.: 'Loss probability calculations and asymptotic analysis for finite buffer multiplexers', *IEEE/ACM Trans. Netw.*, 2001, **9**, (6), pp. 755–768

[11] ADDIE R.G., ZUKERMAN M.: 'An approximation for performance evaluation of stationary single server queues', *IEEE Trans. Commun.*, 1994, **42**, (12), pp. 3150–3160

[12] MAGLARIS B., ANASTASSIOU D., SEN P., KARLSSON G., ROBBINS J.D.: 'Performance models of statistical multiplexing in packet video communication', *IEEE Trans. Commun.*, 1988, **36**, (7), pp. 834–843

[13] NORROS I.: 'On the use of fractal Brownian motion in the theory of connectionless networks', *IEEE J. Sel. Areas Commun.*, 1995, **13**, (6), pp. 953–962

[14] CHOE J., SHROFF N.B.: 'A central limit theorem based approach for analyzing queue behavior in high-speed networks', *IEEE/ACM Trans. Netw.*, 1998, **6**, (5), pp. 659–671

[15] CHOE J., SHROFF N.B.: 'A new method to determine the queue length distribution at an ATM multiplexer'. Proc. IEEE Infocom, 1997

[16] CHOE J., SHROFF N.B.: 'On the supremum distribution of integrated stationary Gaussian processes with negative linear drift', *Adv. Appl. Probabil.*, 1999, **31**, (1), pp. 135–157

[17] GAO D., CAI J., NGAN K.N.: 'Admission control in IEEE 802.11e Wireless LANs', *IEEE Netw.*, 2005, **19**, (4), pp. 6–13

[18] GARCIA A.L.: 'Probability and random process for electrical engineering' (Addison Wesley, 1994)

[19] BORJESSION P.O., SUNDBERG C.E.W.: 'Simple approximation of the error function Q(x) for communication applications', *IEEE Trans. Commun.*, 1979, **27**, (3), pp. 639–643

[20] KOZAMERNIK F.: 'Media streaming over the Internet – an overview of delivery technologies', *EBU Tech. Rev.*, 2002

doi: 10.1049/iet-com:20070488