VARIABLE BIT RATE VOIP IN IEEE 802.11E WIRELESS LANS

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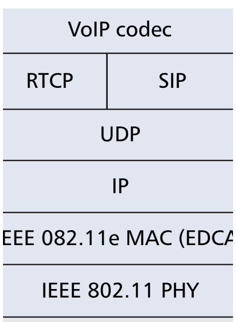

The authors propose two mechanisms to provide quality of service for variable bit rate VoIP in IEEE 802.11e contention-based channel access WLANs: access time-based admission control and access point dynamic access.

ABSTRACT

WLANs have become a ubiquitous networking technology deployed everywhere. Meanwhile, VoIP is one popular application and a viable alternative to traditional telephony systems due to its cost efficiency. VoIP over WLAN (VoWLAN) has been emerging as an infrastructure to provide low-cost wireless voice services. However, VoWLAN poses significant challenges due to the characteristics of contention-based protocols and wireless networks. In this article we propose two mechanisms to provide quality of service for variable bit rate VoIP in IEEE 802.11e contention-based channel access WLANs: access time-based admission control and access point dynamic access. Simulation results are conducted to study these schemes.

INTRODUCTION

In recent years, voice over IP (VoIP) has become an attractive alternative to the traditional public switched telephone network (PSTN). Providing quality of sservice (QoS) for VoIP on increasingly heterogeneous computer networks brings up many challenging issues. As an emergent component in computer networks, the wireless local area network (WLAN) has attracted wide attention from both academia and industry. Recently, VoIP over WLAN (VoWLAN) has become an infrastructure to provide wireless voice service with cost efficiency [1].

Typically, VoWLAN is implemented as a networking protocol stack [2], shown in Fig. 1. At the top of this protocol stack, a number of popular VoIP codecs, such as International Telecommunication Union (ITU) G711, G729, and G723, may be adopted at the application layer. A real-time transport protocol (RTP) stack packs various sizes of audio payload at various intervals. When a VoIP packet, also known as a payload, passes through RTP, User Datagram Protocol (UDP), IP, and medium access control (MAC) protocol layers, some extra bytes of overhead are added as a header or trailer. Finally, voice frames are transmitted according to MAC function over a radio link at the physical (PHY) layer.

VOICE MODEL AND VOIP

VoIP is a set of protocols to transport voice traffic over IP-based packet-switched networks with acceptable QoS and reasonable cost. VoIP traffic could be modeled as two types: constant bit rate (CBR) and variable bit rate (VBR). For CBR VoIP, a codec generates a constant audio payload during the whole voice conversation period. On the other hand, interactive voice conversations have two parties, and each party has many talk spurts and silent periods alternately. A silence suppression technique is adopted to stop sending RTP packets during silent periods [3]. One way to handle VBR VoIP is to regard VBR VoIP as CBR traffic at its peak throughput. Thus, the QoS MAC scheme studied in [4–6] can be directly applied. However, since each VBR VoIP call experiences silent/talk periods independently, some form of statistical multiplexing may be integrated into the QoS MAC to provide QoS for VBR VoIP with efficient bandwidth utilization. This is the primary objective of this research.

In the literature [7, 8] an ON-OFF voice conversation pattern is adopted by Brady's model or May and Zebo's model, as shown in Table 1. Both models belong to ON-OFF Markov modulated fluid (MMF) models. Here, p is the probability of one party of a voice conversation in the ON state. The May and Zebo model is adopted in this study. We assume that the two parties of a VoIP session reside in different WLANs, or one party resides in a WLAN and another party comes from a wired network. Thus, for each VoIP session, a downlink VoIP session comes from the access point (AP) to a station, and an uplink VoIP session comes from the corresponding station to the AP.

When there are N VoIP sessions, the AP needs to transmit frames of all N downlink sessions. On the contrary, uplink VoIP sessions are usually distributed among different wireless stations. This asymmetric nature of VoIP sessions requires that the AP have proportionally larger access time than any other station to achieve

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traffic balance. Otherwise, the service quality of downlink VoIP sessions will be worse than that of uplink VoIP sessions. This unbalanced downlink/uplink phenomenon is observed in [11]. The second objective of this research is to balance uplink/downlink VoIP sessions.

Delay is a key QoS metric for VoIP. From the ITU Telecommunication Standardization Sector (ITU-T) [12], the end-to-end delay requirement for interactive voice is 25 ms without echo cancellers, 150 ms with echo cancellers for excellent quality voice, and 400 ms with echo cancellers for acceptable quality voice.

IEEE802.11 WLAN and Related Work

The IEEE802.11 WLAN is being deployed widely and rapidly for many different environments, including enterprise, home, and public access networking [13]. In a broadcast network such as a WLAN, the MAC sublayer is responsible for arbitrating multiple stations accessing a shared transmission medium. There are two channel access functions defined in the IEEE802.11 MAC [14]: a mandatory distributed coordination function (DCF), which is based on carrier sense multiple access with collision avoidance (CSMA/CA) with binary exponential backoff, and an optional point coordination function (PCF), where the AP controls all the transmissions based on a centralized polling scheme. There are two types of 802.11 network: infrastructure (basic service set, BSS), in which an AP is present, and ad hoc (independent basic service set, IBSS), in which an AP is not present. In a long run time is always divided into repetition intervals called *superframes*. Each *superframe* starts with a beacon frame, and the remaining time is further divided into a contention-free period (CFP) and a contention period (CP). The DCF works during the CP, and the PCF works during the CFP. If the PCF is not active, a superframe will not include the CFP. However, the beacon frame is always sent whether or not the PCF is active. The beacon frame is a management frame for synchronizations, power management, and delivering parameters. In a BSS an AP sends beacon frames. In an IBSS any mobile station that is configured to start an IBSS sends beacon frames. As other mobile stations join that IBSS, each station, a member of the IBSS, is randomly chosen for the task of sending a beacon frame. Beacon frames are generated at regular intervals called target beacon transmission times.

The DCF defines a basic access mechanism and an optional request-to-send/clear-to-send (RTS/CTS) mechanism. In the DCF a station with a frame to transmit monitors the channel activities until an idle period equal to a distributed interframe space (DIFS) is detected. After sensing an idle DIFS, the station waits for a random backoff interval before transmitting. The backoff time counter is decremented in terms of slot time as long as the channel is sensed idle. The counter is stopped when a transmission is detected on the channel, and reactivated when the channel is sensed idle again for more than a DIFS. In this manner stations, deferred from channel access because their backoff time was

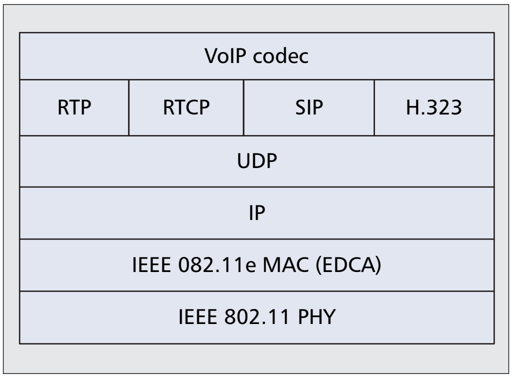

Figure 1. *VoIP over IEEE 802.11 WLAN protocol stack.*

Model	Mean ON period	Mean OFF period	*p*
Brady's model [9]	1 s	1.35 s	0.43
May and Zebo model [10]	352 ms	650 ms	0.35

Table 1. *Voice models.*

larger than the backoff time of other stations, are given higher priority when they resume the transmission attempt. The station transmits its frame when the backoff time reaches zero. At each transmission, the backoff time is uniformly chosen in the range $(0, CW - 1)$ in terms of time slots, where CW is the current backoff (contention) window size. At the very first transmission attempt, CW equals the initial backoff window size, CW_{min}. After each unsuccessful transmission, CW is doubled until a maximum backoff window size value, CW_{max}, is reached. After the destination station successfully receives the frame, it transmits an acknowledgment frame (ACK) following a short interframe space (SIFS). If the transmitting station does not receive the ACK within a specified ACK timeout, or it detects the transmission of a different frame on the channel, it reschedules the frame transmission according to the previous backoff rules.

In order to enhance IEEE 802.11 and provide QoS support over WLANs, the IEEE working group recently finalized the IEEE 802.11e standard [15]. IEEE 802.11e has a hybrid coordination function (HCF), which includes a contention-based channel access part and a centrally controlled channel access part. The contention-based channel access of the HCF is referred to as enhanced distributed channel access (EDCA), and the centrally controlled channel access is referred to as HCF controlled channel access (HCCA). Our proposed QoS MAC is based on contention-based IEEE 802.11e EDCA.

There is increasing interest in providing QoS to VoIP sessions in WLANs. In the literature [3, 8], the focus is on designing a centrally controlled polling-based MAC to provide QoS to VoIP in WLANs. In [11] the maximum number of VoIP sessions supported by contention-based MAC is evaluated in IEEE 802.11 (a/b) WLANs, and the unbalance problem of downlink/uplink

The ATAC algorithm includes a procedure at the AP and a procedure at stations. The intuition of the ATAC algorithm is to protect ongoing VoIP sessions by rejecting excessive newly arriving VoIP sessions if the channel access time is insufficient.

traffic as well as the relationship between system capacity and VoIP codec are also studied. In [16] the authors propose a generic approach that relates VoIP performance with the dynamics of priority MAC. This method improves VoIP capacity in WLANs. In summary, due to the limited capacity of WLANs, all ongoing VoIP traffic will be degraded if too many VoIP connections compete for media access. Therefore, effective admission control is crucial to providing QoS for VoWLAN. Moreover, multiplexing VBR VoIP traffic by exploring an ON-OFF VoIP traffic model can further improve the efficiency of VoWLAN. Taking these into consideration, we design a measurement-based admission control mechanism, Access Time Based Admission Control (ATAC), in this article. In addition, a MAC enhancement AP Dynamic Access (AP-DA) algorithm is proposed to address uplink/downlink VoIP traffic unbalance problem. Finally, the two new algorithms are combined (ATAC + AP-DA) to provide good QoS for VoWLAN. In this article we consider an infrastructure network (BSS). The proposed schemes are different from our previous work [5, 6], and consider VBR VoIP characteristics.

The rest of this article is organized as follows. We first propose the ATAC and AP-DA algorithms. Then we present simulations that demonstrate the effectiveness of the proposed QoS MAC mechanisms. Finally, we conclude this article in the last section.

Access Time Based Admission Control

The ATAC algorithm includes a procedure at the AP and a procedure at stations. The intuition of the ATAC algorithm is to protect ongoing VoIP sessions by rejecting excessive newly arriving VoIP sessions if the channel access time is insufficient. For each newly arriving VoIP session, the AP either accepts or rejects it based on estimated system utilization in the previous measurement interval. A newly arriving VoIP in terms of an admission request can come from uplink initiated by a local station or downlink initialed by a remote host. A measurement interval may consist of one or multiple superframes in IEEE 802.11 WLANs. At the end of each measurement interval, the AP calculates a system control parameter *TXOPBudget*, which is the additional amount of access time available for VoIP sessions. If *TXOPBudget* is depleted, the newly arriving VoIP session is rejected.

Procedure at the Access Point

In order to fully explore the multiplexing gain of the VoIP VBR traffic model and to smooth the access time fluctuation in wireless communication, we design a token bucket mechanism to allocate maximum allowable access time for all downlink VoIP sessions at the AP. The AP calculates the maximum allowable access time *TxLimit* for all downlink VoIP sessions. At the beginning of each measurement interval, the additional access time (tokens) *TxAddition* is poured into the token bucket at the AP. $TxAddition(t+1) = N \cdot p \cdot T \cdot \lambda \cdot T_{suc} \cdot SF(t)$, where N is the number of active downlink VoIP sessions at the AP; p is the probability of a downlink VoIP session at the ON state in May and Zebo's model; T is the measurement interval; λ is the VoIP codec sample rate; T_{suc} is the minimal required successful transmission time for a single VoIP uplink/downlink including *SIFS* and *ACK* overhead; and the last term, $SF(t)$, is the surplus factor in the tth measurement interval at the AP [5, 6]. In this formula, p and λ are decided by the specific VoIP codec, and T_{suc} is a constant for a particular VoIP codec in a WLAN. $T_{suc} = T_{frame} + SIFS + T_{ack}$, where T_{frame} is the transmission time for a VoIP frame; *SIFS* is the shortest interframe space defined in the IEEE 802.11 specification; and T_{ack} is the *ACK* frame transmission time. *SF* represents the ratio of over-the-air bandwidth reserved for *TxAccess* to bandwidth required for successful transmission, *TxSuccess*. Because of the collisions in contention-based MAC, and failed transmission caused by channel noise and signal fading, *SF* is always greater than 1. Moreover, *SF* is adjusted dynamically to compensate for the fluctuation of collision and noise in WLANs. Impacts of *SF* were studied in [6]. The AP measures *TxSuccess* and *TxAccess* of all downlink VoIP sessions in each measurement interval. *TxSuccess* includes all successful transmission times. *TxAccess* includes all channel access times regardless of success. To smooth the fluctuation of the ratio of over-the-air bandwidth in wireless transmission, the AP computes $SF(t+1)$ as $SF(t+1) = f \cdot SF(t) + (1 - f) \cdot TxAccess/TxSuccess$, where f is a damping factor in the exponential weighted average algorithm; $SF(t)$ is the SurplussFactor in the t-th measurement interval; *TxAccess* and *TxSuccess* are measured at the AP in the t-th measurement interval.

TxLimit is the tokens in the token bucket; that is, the maximum allowable transmission time for VoIP sessions. Whenever the AP accesses the channel, regardless of its successfulness, the same amount of access time (tokens) is subtracted from the token bucket. The AP cannot transmit any VoIP frame once the *TxLimit* is exhausted. $TxReminder(t)$ is the token left in the token bucket in the tth measurement interval. *TxLimit* for the $(t + 1)$th measurement interval is $TxLimit(t + 1) = \min\{Tx\text{Reminder}(t) + TxAddition(t + 1), N \cdot p \cdot T \cdot \lambda \cdot T \cdot SF_{\max}\}$. $SF_{\max}$ is the maximum value of *SF* and the maximum capacity of the token bucket. Intuitively, a larger $SF_{\max}$ means larger capacity of the token bucket. Consequently, the AP is more conservative on reservation access time for VoIP sessions.

At the end of the tth measurement interval, the AP calculates $TXOPBudget(t+1)$ as

$$TXOPBudget(t+1) = \max\left\{ \begin{matrix} aT - TxLimit(t-1) - \\ \sum_{k=1}^{N} TxLimit[k](t+1), 0 \end{matrix} \right\}.$$

The term $TxLimit(t+1)$ is the maximum allowable transmission time at the AP for all N downlink VoIP sessions in the $(t + 1)$th measurement interval; and

$$\sum_{k=1}^{N} TxLimit[k](t+1)$$

is the sum of the maximum allowable transmission time at stations for N uplink VoIP sessions in the $(t + 1)$th measurement interval. Whenever a station i transmits an uplink VoIP frame, it piggybacks the updated $TxLimit[i](t + 1)$ to the AP. The calculation of $TxLimit[i](t + 1)$ is explained in the next subsection. The term αT represents the total access time assigned to VoIP traffic in a WLAN.

When a VoIP session joins a WLAN, the AP decides to accept or reject that call according to the ATAC algorithm in Fig. 2. *TBT* is the transmission budget threshold, which is the minimal transmission time for a pair of uplink and downlink of a VoIP session: $TBT = 2 \cdot p \cdot T \cdot \lambda \cdot T_{suc} \cdot SF_{\max}$.

Procedure at Each Station

Each station maintains a set of local parameters to regulate uplink VoIP sessions it serves. Similar to the AP, each station has a token bucket. At the end of the tth measurement interval, SurplussFactor $SF[k](t)$ for the kth uplink VoIP session is computed. The additional tokens for the $(t + 1)$th measurement interval $TxAddition[k](t + 1)$ is calculated and added into the bucket as follows: $TxAddition[k](t + 1) = p \cdot T \cdot \lambda \cdot T_{suc} \cdot SF[k](t)$, and $TxLimit[k](t + 1) = \min\{p \cdot T \cdot \lambda \cdot T \cdot SF_{\max}, Tx\text{Reminder}[k](t) + TxAddition[k](t + 1)\}$.

The calculation procedure of $SF[k](t)$ for the kth uplink VoIP session is the same as the calculation procedure of $SF(t)$ at the AP, except that all measurements are local at each station. $TxLimit[k](t + 1)$ is the amount of tokens in the bucket for the kth uplink VoIP session in the $(t + 1)$th measurement interval at a station.

For each transmission of an uplink VoIP session, the same number of tokens is subtracted. Whenever an uplink VoIP frame is transmitted to the AP, the updated $TxLimit[k]$ is packed in the MAC header field. At the end of each measurement interval, the AP uses the latest value from each station to calculate $TxLimit[k](t + 1)$ and $TXOPBudget[i]$ as in the previous subsection. If the AP does not receive $TxLimit[k](t)$ from the kth uplink VoIP in the tth measurement interval, it uses the maximum token bucket value.

Although the AP and mobile stations measure and calculate their local parameters independently, the AP needs to collect channel access time from each station (VoIP uplinks) and itself (VoIP downlink) to calculate a system control parameter *TXOPBudget* at the end of each measurement interval. As explained in the previous subsection, the AP makes an admission decision based on *TXOPBudget* whenever a new VoIP request arrives. In other words, due to the independence of medium access at each station and the AP, collecting the latest measurement from each station is necessary for the AP to make an appropriate admission decision.

Parameter	Value
Data rate	11 Mb/s
Control rate	1 Mb/s
Slot	20 μs
SIFS	10 μs
Beacon interval	1000 ms
$AIFS[i]$	30 μs
$CW_{\min}[i]$	16
$CW_{\max}[i]$	256
$SF_{\max}$	2
δ	0.2

Table 2. *System parameters.*

If *TXOPBudget[1]>TBT*

{

- AP accepts this new VoIP call. It assigns a unique VoIP ID k, and puts the VoIP ID and the corresponding station ID into the beacon frame.

$$TxLimit(t+1) = TxLimit(t+1) + \frac{TBT}{2}$$

- When the corresponding station receives the beacon frame,

$$TxLimit[k](t+1) = TxLimit[k](t+1) + \frac{TBT}{2}$$

}

Else AP rejects this new VoIP call.

Figure 2. *Access Time Based Admission Control.*

AP Dynamic Access

It is important to realize that VoIP sessions access the shared channel according to contention-based MAC in our study. Thus, setting a maximum allowable access time quota does not guarantee that VoIP sessions actually gain that amount of access time. Downlink VoIP sessions are especially vulnerable due to the asymmetric traffics in WLANs. We propose an AP Dynamic Access (AP-DA) mechanism to balance downlink/uplink VoIP sessions.

In this algorithm, the AP transmits downlink VoIP frames in one of two modes: the basic EDCA access mode or the PCF access mode. In the EDCA access mode, the AP uses the original EDCA access parameters: $AIFS[i]$, $CWmin[i]$ and $CWmax[i]$. In this mode the AP is the same as any other stations in terms of the priority of access control. In the PCF access mode the AP uses the *PIFS* instead of $AIFS[i]$ to sense the channel as idle. As defined in the EDCA, *PIFS* is shorter than $AIFS[i]$. Thus, downlink VoIP frames access the channel without contention in the PCF access mode. The AP adaptively operates in one of the two modes based on the queue length of VoIP frames at the AP, as shown in Fig. 3.

N is the number of active downlink VoIP sessions served by the AP; $(1 + \delta)N$ and $(1 - \delta)N$ are the upper and lower bounds of the queue length threshold in this adaptive algorithm. Note that transmissions in both modes are controlled by the proposed ATAC algorithm.

Performance Evaluation

We simulate the proposed QoS MAC mechanisms with discrete event simulation. The following performance metrics are applied in the experiments:

- Delay of downlink/uplink VoIP sessions
- Throughput of downlink/uplink VoIP sessions
- System utilization
- The number of VoIP sessions served

In the simulations only VBR VoIP sessions are considered. We adopt the IEEE 802.11b WLAN. The system parameters are listed in Table 2. For the voice model, we use May and Zebo's ON/OFF voice model. For VoIP sessions, we employ the G711 a-Law codec with 160 bytes/20 ms payload. An ideal channel without noise is assumed. The capture effect is not considered. Initially there are two VoIP calls in a WLAN. Every 5 s, two new VoIP calls arrive. Any VoIP session continues in the simulation if it is served by the WLAN. The total simulation time is 200 s. In this article we compare three schemes: IEEE 802.11e EDCA, the proposed ATAC, and ATAC + AP-DA. As our ongoing research project, other VoIP over WLAN schemes are compared to our proposed scheme.

If AP is at EDCA mode
{ If ($Q_{len} > (1 + \delta) \cdot N$) AP changes to the PCF mode;}
Else{ If ($Q_{len} < (1 + \delta) \cdot N$) AP changes to the EDCA mode;}

Figure 3. *AP Dynamic Access algorithm.*

In our simulations, since only VoIP sessions over one hop of a WLAN is considered, we set the one-hop delay limit to 20 ms. We set SF_{max} to 2 based on the simulation experiments.

Figure 4 shows downlink VoIP throughputs of the three schemes. The throughputs fluctuate constantly due to the nature of VBR VoIP sessions. However, the moving tendency and average value can be observed clearly. In Fig. 4a (the EDCA scheme) the downlink VoIP throughput increases gradually to around 1 Mb/s, and then decreases to a very small value around zero until the end of the simulation. In Fig. 4b (the ATAC algorithm) the downlink VoIP throughput increases gradually to around 0.5 Mb/s, and then hovers around this value until the end of the simulation. Figure 4c (the ATAC + AP-DA algorithm) is similar to Fig. 4b. Without admission control as in the EDCA scheme, all VoIP sessions are served, and they compete for limited bandwidth in the WLAN, which causes large numbers of collisions; eventually the AP can barely transmit any VoIP frames successfully. The proposed mechanism only accepst 19 VoIP sessions based on availability of system bandwidth, and each VoIP session is able to acquire sufficient access time, so the overall throughput is stable.

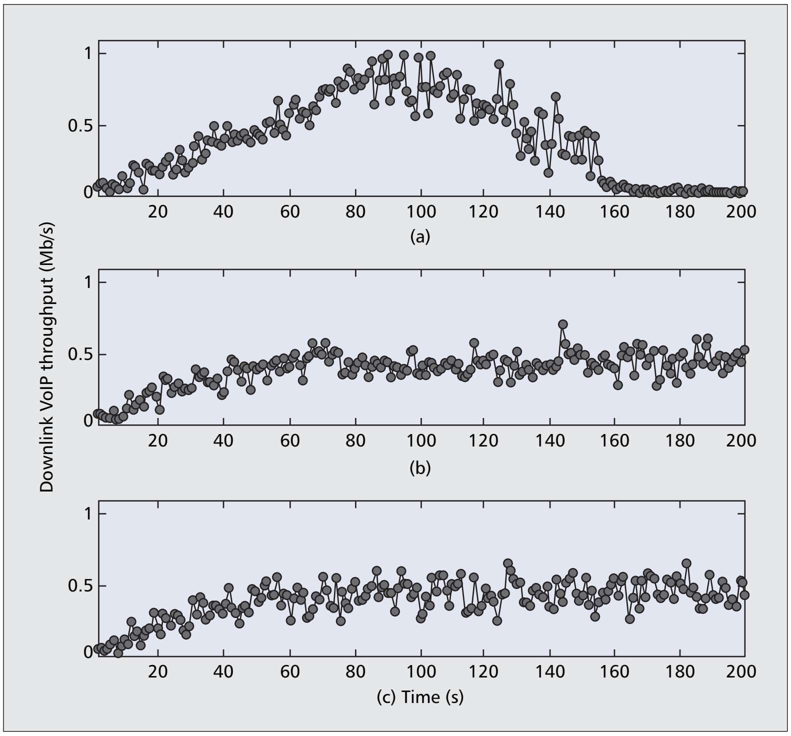

Figure 4. *Downlink VoIP throughput: a) EDCA; b) ATAC; c) ATAC + AP-DA.*

Figure 5 shows the average delay of downlink VoIP frames in three schemes. Figures 5b (ATAC) and 5c (ATAC + AP-DA) have similar results. Both significantly outperform Fig. 5a (EDCA) in terms of average delay. Due to space limitations, we skip the results of uplink VoIP throughput and delay.

Figure 6 shows the cumulative fractions of downlink/uplink VoIP delays. In Fig. 6a (EDCA) the delays of about 50 percent of downlink VoIP frames and about 70 percent of uplink VoIP frames are less than 50 ms. This result undoubtedly indicates that the EDCA scheme does not provide proper QoS for VoIP. Figure 6b (ATAC) shows that the delays of almost 100 percent of both downlink and uplink VoIP frames are less than 6 ms. Moreover, there are more uplink VoIP frames concentrated at lower delay values than downlink VoIP frames. For example, there are about 60 percent downlink VoIP frames and 90 percent uplink VoIP frames that have a delay less than 2 ms. Figure 6c (ATAC + AP-DA) is similar to Fig. 6b except that the downlink and uplink delays are more equally distributed. Figure 6 demonstrates that the proposed ATAC scheme is capable of providing QoS for VoIP, and the AP-DA algorithm improves the balance between downlink/uplink VoIP sessions.

In the proposed admission control scheme calculating parameters at the end of each measurement interval involves some computing overhead. There are two factors worth consideration regarding how the computing overhead affects the performance of VoWLAN. First of all, the AP and each station calculate their

TxLimit and *TxAddition* independently. In other words, the AP and each station maintain a local token bucket to control their channel access based on local parameters. This distributed approach can fully utilize the computing capacity of each station and therefore alleviate the performance bottleneck at the AP. Second, a measurement interval may consist of multiple superframes as claimed earlier, and the length of measurement interval can be configured in a VoWLAN system. A trade-off exists between the computing overhead and QoS. If a longer measurement interval is configured, fewer calculations are required, and therefore the computing overhead is lower. However, a longer measurement interval implies slow responses to any changes in the VoWLAN system and in turn causes less desirable QoS. On the other hand, a shorter measurement interval means fast response to any fluctuation in the VoWLAN system, and therefore achieves better QoS at the cost of higher computing overhead.

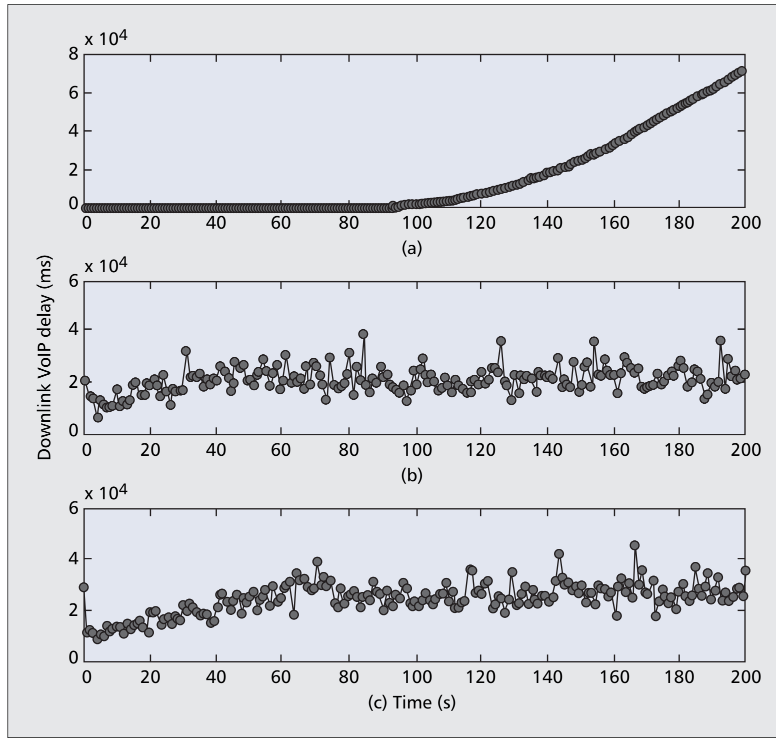

Figure 5. *Downlink VoIP delay: a) EDCA; b) ATAC; c) ATAC + AP-DA.*

Conclusions

In this article we propose two QoS MAC schemes for VBR VoIP in WLANs: the Access Time Based Admission Control algorithm and the AP Dynamic Access (AP-DA) algorithm. The ATAC algorithm significantly improves QoS of VBR VoIP traffic, in terms of average delay and delay distribution, over the existing standard IEEE 802.11e EDCA. The AP-DA algorithm provides a solution for balancing downlink/uplink VoIP traffic. Overall, the proposed QoS MAC enhancements provide a satisfactory solution to QoS for VBR VoIP in WLANs.

Our future work includes the optimization of SF_{max} and the analysis and optimization of δ.

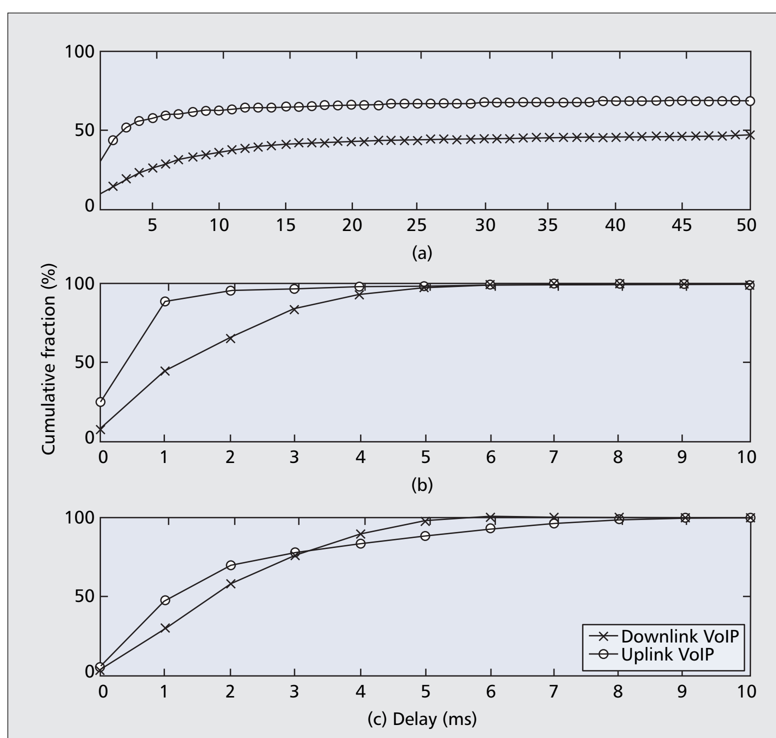

Figure 6. *Cumulative fractions of downlink/uplink VoIP delay: a) EDCA; b) ATAC; c) ATAC + AP-DA.*

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