Research Article

Boosting VoIP Capacity via Service Differentiation in IEEE 802.11e EDCA Networks

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This paper considers the performance problem of VoIP over 802.11e WLANs caused by the unfairness between uplink and downlink as well as the inefficient EDCA. A novel medium access control scheme named BEDCA (Balanced EDCA) is presented, which provides service differentiation between the access point (AP) and the mobile stations (STAs) to enhance VoIP capacity. In BEDCA, the expression of AP’s contention window is obtained which is a relative constant value independent of the participating STAs. The minimum contention window of the STAs is traffic-aware based on the proposed algorithm. The performance improvement of BEDCA is verified through intensive simulations and the results show the capacity improvement of 82.1% compared to EDCA.

1. Introduction

In recent years, the wireless LAN (WLAN) has become the dominant Internet access method because of its attractive features, including low cost, high bandwidth, ease of deployment, and mobility support. Besides, the rapid growth of the mobile devices, coincidental with the proliferation of the emerging voice applications, has made the VoIP over WLAN become popular [1].

Distributed coordination function (DCF) is the fundamental 802.11 media access control (MAC) technique. While popularly being adopted in the current WLAN, the best-effort-based mechanism cannot meet the QoS requirements of the VoIP applications. As the VoIP applications are characterized by their real-time nature, they impose upper limits on delay and jitter in addition to usual requirements on packet loss and throughput, which are properly supported only in overprovisioned scenarios with DCF. To overcome this limitation, IEEE 802.11e [2], also known as the enhanced distributed channel access (EDCA), has been introduced to differentiate the channel access priorities of the different QoS requirements [3]. EDCA defines four traffic access categories (ACs), namely, voice, video, best-effort, and background. The MAC parameters can be set according to the expected access category of the incoming traffic. As a result, a high-priority data packet has a large chance to win the media contention for being transmitted than a low-priority one.

The EDCA service differentiation becomes less effective, however, when the VoIP streams dominate the traffic: a majority of STAs are sending/receiving VoIP packets. The problem is caused by the fact that there is no media access priority differentiation between the AP and STAs. As more STAs compete for the channel access for uplink traffic, the AP gets less opportunity to win the channel contention and thus its transmission attempts (downlinks) are suppressed. As VoIP applications generally require a balanced uplink/downlink throughput, the deficiency of the AP-QSTA access coordination in EDCA significantly throttles the VoIP capacity over WLAN. In addition, the fixed EDCA parameters perform poorly in matching the dynamical network conditions, in terms of the number of STAs and their corresponding traffic loads. In particular, when the WLAN is heavily loaded, the performance of the delay sensitive VoIP streams will be severely degraded.
The above VoIP over WLAN performance issues have inspired a number of research studies both at the TCP layer [4–8] and the MAC layer. In the paper, we focus on the latter. A natural idea is giving AP a channel access priority [9–11] to strike a balance between the downlinks and the uplinks. Studies also suggest that network collision reduction, either by adjusting the contention window size [12–15] or tuning the transmission opportunity coordination [16–20], can enhance the capacity of the VoIP traffic.

More recently, WiFox [15] and BDCF [21] are proposed. WiFox solves the problem of traffic asymmetry by adaptively prioritizing APs channel access over competing STAs. To achieve a fine balance between uplink and downlink traffic in order to optimize network throughput, WiFox works with the scheme wherein the percentage of downlink traffic being given priority is proportionately controlled based on the queue length at the AP. One issue in WiFox is that only increasing the chances for AP to contend for the channel may aggregate the collision rate in heavy load WLAN environment. As validated in Section 4, the obtained throughput of WiFox degrades when the network traffic load is heavy.

BDCF aims to achieve both the traffic balancing and the throughput enhancement by giving the AP’s minimum contention window size ($CW_{\text{min}}$) constant value while adjusting the STAs $CW_{\text{min}}$ depending on the number of STAs. The potential setback of BDCF, however, is that the scheme uses the number of the STAs as a parameter to give the STAs a relatively fixed contention window size. As for a given number of STAs, the MAC parameters in BDCF remain unchanged. However, even with fixed number of STAs, the network traffic load varies when the traffic load of the STAs changes, for example, with different sending rates. Thus, BDCF cannot fully adapt to the network dynamics. In this study, we argue that the traffic load of STAs, in addition to the STA number, is also an important factor that affects the channel contention and therefore should be taken into consideration when designing the VoIP capacity enhancement scheme. As we will show in Section 4, the performance of BDCF varies greatly with varying VoIP codecs when the number of the STAs remains unchanged. As a result, BDCF cannot effectively keep the balance between the uplinks and downlinks in the above mentioned network conditions.

In this paper, we aim to address the above VoIP over WLAN performance issues and enhance the VoIP traffic capacity in a densely deployed 802.11 wireless network with one AP and $n$ active mobile STAs. We propose a balanced EDCA scheme, BEDCA, based on IEEE 802.11e. BEDCA modifies the access category of voice (AC_VO) only. As for other categories, we use the recommended configurations. Our scheme can be implemented solely at the AP side without the burden modifying the STAs. BEDCA differentiates the service between the AP and STAs by allocating a higher priority AC for the AP that benefits from $n$ times greater channel access probability compared to a STA. Our rigorous analysis reveals that AP’s minimum contention window size, $CW_{\text{min}}$, mostly determined by the MAC parameters, including the slot time, SIFS, DIFS and the transmission rate, and the VoIP codec, is relatively stable. Therefore, our proposed scheme suggests a traffic load based adaptive adjustment of the STAs minimum contention window size to enhance the VoIP capacity. We have validated the effectiveness of our proposed scheme through intensive simulations and our results show the capacity improvement of 82.1%.

The rest of the paper is organized as follows. Section 2 illustrates our system model. The algorithm of BEDCA is described in Section 3 and is verified by simulation results provided in Section 4. Finally, we conclude the paper in Section 5.

### 2. System Model

EDCA defines four ACs for different kinds of traffic to provide service differentiation. Each AC is assigned with its own MAC parameters and operates independently. The default EDCA parameters for each AC defined by the standard are listed in Table 1, where $CW_{\text{min}}$ and $CW_{\text{max}}$ are the minimum and maximum contention window sizes, respectively. AIFS and TXOP are newly defined parameters in 802.11e. AIFS, or AIFS-number, is used to calculate the AIFS, a time interval between frames being transmitted under the IEEE 802.11e EDCA MAC while TXOP is the period in which EDCA provides contention-free access to the channel. The proposed BEDCA only revises the contention window of AC_VO and leaves the other ACs and MAC parameters unchanged.

We consider a wireless LAN with one AP and $n$ STAs. Each STA is associated with the AP and shares the channel with other STAs by means of the modified EDCA access method with adaptive $CW_{\text{min}}$. We use AC[STA] for the uplink traffic at the stations and AC[AP] for the downlink traffic at the AP, respectively.

Let $\tau_A$ and $\tau_S$ be the transmission attempt probabilities for the AP and that of the STAs, respectively. The conditional transmission collision probabilities of the AP and the STAs, denoted as $p_A$ and $p_S$, can be determined by the probability of a collision seen by a transmitted packet on the channel [22] and expressed as

$$p_S = 1 - (1 - \tau_S)^n - (1 - \tau_A),$$

$$p_A = 1 - (1 - \tau_S)^n.$$  \hspace{1cm} (1)

As derived in [22], the initial contention window size, symbolized as $W_A$ or $W_S$ for AP or STAs, can be calculated as

$$W_A = \left( \frac{2}{\tau_A} - 1 \right) \left( 1 + p_A \sum_{i=0}^{m-1} (2p_A)^i \right)^{-1},$$

$$W_S = \left( \frac{2}{\tau_S} - 1 \right) \left( 1 + p_S \sum_{i=0}^{m-1} (2p_S)^i \right)^{-1}.  \hspace{1cm} (2)$$

### Table 1: Default EDCA parameters for each AC.

| AC         | $CW_{\text{min}}$ | $CW_{\text{max}}$ | AIFS | TXOP/ms |
|------------|-------------------|-------------------|------|---------|
| Background (AC_BK) | 15 | 1023 | 7 | 0 |
| Best-effort (AC_BE) | 15 | 1023 | 3 | 0 |
| Video (AC_VI) | 7 | 15 | 2 | 3.008 |
| Voice (AC_VO) | 3 | 7 | 2 | 1.504 |
where \( m \) is the maximum backoff stage which is a constant value.

Let us consider a scenario when a node, either the AP or one of the STAs, attempts to transmit a packet. A successful transmission happens if and only if no other nodes transmit. Thus, the probability of a successful transmission is

\[
P_s = \tau_A (1 - \tau_s)^n + n \tau_s (1 - \tau_s)^{n-1} (1 - \tau_A).
\]  

(3)

Similarly, the probability of an idle slot \( P_i \), when no one transmits, can be expressed as

\[
P_i = (1 - \tau_s)^n (1 - \tau_A).
\]  

(4)

The collision probability \( P_c \) of a specific time slot can be derived:

\[
P_c = 1 - P_i - P_s.
\]  

(5)

The system throughput can be estimated by the fraction of time the channel is used for successful packet transmissions \([22]\). Let \( E[P] \) be the average payload size. The average successfully transmitted bytes in a given time unit are \( P_s E[P] \). Considering that the length of a time unit can be any of the following three: idle time, successful transmission time, and failed transmission time (because of the collision), the normalized system throughput, denoted as \( S \), can be determined by

\[
S = \frac{P_s E[P]}{P_i \delta + P_s T_i + P_c T_c},
\]  

(6)

where \( \delta \) is the duration of the idle slot, \( T_i \) is the average time of a successful transmission, and \( T_c \) is the time duration of a failed transmission. Note that the value of \( \delta \) is defined in a given standard (802.11a/b/g/n).

### 3. BEDCA Scheme

In this section, we provide the theoretical analysis that backs the intuition of the proposed BEDCA scheme. Our study suggests that the optimal \( CW_{\text{min}} \) of the AP holds a constant value regardless of the traffic load dynamics at the STAs. Therefore, the problem is simplified to finding an appropriate \( CW_{\text{min}} \) of the AC_VO that maximizes the system throughput and maintains a balance between the uplinks and the downlinks.

#### 3.1. Theoretical Analysis

We start the discussion by explaining how to obtain the optimal \( CW_{\text{min}} \) to maximize the throughput \( S \) while maintaining the balance between the uplinks and the downlinks. Note that (6) can be transformed to

\[
S = \frac{E[P]}{T_i - T_c + (P_i (\delta - T_i) + T_c) / P_c}.
\]  

(7)

In practice, \( T_i \) and \( T_c \) are determined by the parameters of the PHY and MAC layers (802.11a/b/g/n) and the average packet size. For the sake of the simplicity, we adopt their estimation [22] as

\[
T_i = T_{\text{data}} + \text{SIFS} + \delta + T_{\text{ack}} + \text{DIFS} + \delta,
\]

\[
T_c = T_{\text{data}} + \text{DIFS} + \delta,
\]  

(8)

where \( T_{\text{data}} \) is the frame transmission time. Notice that \( T_{\text{data}} \) value varies due to the payload sizes variation. For a certain voice codec, however, the length of the data is static. As a result, \( T_i \) and \( T_c \) can be considered as the known constants. To maximize \( S \), it is equivalent to maximize

\[
F(\tau_s, \tau_A) = \frac{P_i}{P_i (\delta - T_i) + T_c}.
\]  

(9)

To balance the uplink and the downlink traffic, we differentiate the services between the AP and STAs where AP is allocated \( n \) times access probability compared to a STA expressed as \( \tau_A = n \tau_S \).

By plugging \( P_i, P_s \), and \( P_c \) in (9), we can obtain an approximation of \( F(\tau_s, \tau_A) \) after utilizing second-order Taylor series expansion given \( \tau_A \ll 1 \):

\[
F(\tau_s) = \frac{2n \tau_s (1 - n \tau_s)}{\delta - 2n (\delta - T_c) + n^2 (\delta - T_c) \tau_s^2}.
\]  

(10)

By setting the first derivative of (10) to zero and utilizing the quadratic function, we can obtain that

\[
\tau_s^{\text{opt}} = \frac{T_c}{n\sqrt{T_c}}.
\]  

(11)

Thus,

\[
\tau_A^{\text{opt}} = n \tau_s^{\text{opt}} = \frac{T_c}{T_c}.
\]  

(12)

Based on the above result, we can derive the collision probability \( P_c \) under \( \tau_A^{\text{opt}} \) as

\[
P_c = 1 - P_i - P_s
\]

\[
\approx 1 - e^{-\sqrt{T_c} \tau_s^{\text{opt}} (1 - \frac{\delta}{T_c} + \frac{\delta}{T_c})}.
\]  

(13)

Let us take 802.11b parameters as an example. The default parameter values are reported in Table 2.

For a certain codec, for example, G.711 codec, which has its data length of 160 bytes, the optimal attempt access probability of the AP is 0.3276. Similarly, the collision probability \( P_c \) is around 0.12. In addition, when \( n \) is large (while it is unlikely that \( n \) becomes infinitely large in the real world, we merely perform the mathematical derivation here to compute the optimal values as a guidance), the \( P_A \) depicted in (1) can be calculated as

\[
P_A = 1 - (1 - \tau_s^{\text{opt}})^n = 1 - \left(1 - \frac{\tau_A^{\text{opt}}}{n}\right)^n \approx 1 - e^{-\tau_A^{\text{opt}}}. \]  

(14)
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Table 2: Relevant network parameters of 802.11b.

| Parameter       | Default |
|-----------------|---------|
| PHY_header      | 192 bit |
| ACK_frame       | 112 bit + PHY_header |
| Rate            | 11 Mbit/s |
| Slot            | 20 us |
| SIFS            | 10 us |
| DIFS            | 50 us |

By substituting the obtained values of $p_A$ and $r_A^{opt}$ into (2) and using the numerical analysis approximation, $W_A$ can be determined to be a constant value of 4.

The above results reveal that the AP’s optimal contention window size of AC_VO is independent of the number of STAs, once the 802.11 CSMA parameters and the VoIP codec are determined. Note that the optimal value can be easily computed by combining (12), (14), and (2). As we have shown above, to boost the VoIP capacity, we only need to adjust the STAs’ contention window size.

3.2. The Process of the Algorithm. Enlightened by the above conclusion, we develop an AP oriented node access scheme in a way that the $CW_{min}$ adjustment is transparent to STAs. The $CW_{min}$ adjustment is based on the target value of $p_A$ derived in (13). The detailed steps are described in Algorithm 1.

We assume the time domain is divided into a number of equal time periods, for example, the 802.11 beacon period of 100 ms. In each time period, the AP calculates the current of collision probability $f_{curr}^j$ as the ratio of the transmission failures to the total data transmissions as

$$f_{curr}^j = \frac{E(data\_collisions^j)}{E(data\_sent^j)},$$

(15)

where $E(data\_collisions^j)$ is the number of collisions observed during period $j$ and $E(data\_sent^j)$ is the total number of the transmitted packets in the same period. To minimize the bias against the transient collisions, we use an estimator of exponentially weighted moving average (EWMA) to smooth the estimated values with a smoothing factor $\alpha$.

Then the derived network collision probability is compared with the target value. If the collision probability is larger, indicating a saturated network condition, the attempt probability $r_s$ is decreased in a multiplicative way with the factor $\beta$ to suppress the STAs’ network accesses. Otherwise, the $r_s$ is increased additively with the factor $\epsilon$ to encourage the STAs’ network accesses. Note that the value of $r_s$ remains unchanged if the collision probability is equivalent to the target value.

Given $r_s$, it is straightforward to determine the values of $CW_{min}$ for STAs according to (2). Finally, the computed $CW_{min}$ values at the AP are broadcasted to the STAs along with the beacon messages.

4. Evaluation

In this section, we validate our proposed BEDCA scheme by comparing its performance with the default EDCA of 802.11e, WiFox algorithm [15], and the BDCF [21] scheme. Our experiments are conducted on the ns-2 (version 2.28) with the EDCA extension from the TU Berlin.

4.1. Simulation Testbed. Our network testbed, illustrated in Figure 1, consists of one AP and 70 STAs. The AP is attached to a gateway node through the Ethernet with 125 Mbps, ensuring that it is not the bottleneck of the testbed. The network is located in a 500-meter by 500-meter square area, where the AP is located at the center and the STAs are uniformly distributed. To provide a stable and evenly distributed workload to each STA, we deploy the same number of workstations around the AP (connected through Ethernet), each of which serves as a peer station for the corresponding STA. During the simulation, each STA sends/receives its data packets over the uplink/downlink TCP/UDP flow to/from its peer station.

To select a proper VoIP codec for our experiments, we list five common ones as illustrated in Table 3. We finally pick G.711 encoder in the experiment as it is one of the most often used codecs in prior work [16, 23, 24].

All of the tests are performed using the 802.11b physical maximal transmission rate of 11 Mbit/sec with basic access except for the evaluation of the VoIP capacity which is conducted with both 802.11b and 802.11g (54 Mbit/sec). In the experiments, the parameters are set as shown in Table 4. The maximum backoff stage, $m$, as validated in [22], does not practically affect the system throughput as long as $m$ is greater than four or five. Thus, in our experiments, we give $m$ the value of 7. The parameter of $\alpha$, when chosen in the range of [0.75, 0.9], can achieve good performance in terms of the goodput and delay as evaluated in the previous work [25]. So in the following simulation scenarios we set $\alpha$ to 0.875. $\beta$ and $\epsilon$ are given with the empirical values of 5/6 and 0.01, respectively, in our experiments as this gives a good tradeoff between goodput and downlink/uplink ratio. The run time of each simulation is 60 s including the initial bias (5 s).
Figure 1: Simulation topology.

Table 3: Characters of different kinds of codecs.

| Codec   | Bitrate/kbps | Payload/byte |
|---------|--------------|--------------|
| G.711   | 64           | 160          |
| G.723.1 | 5.3          | 20           |
| G.729A  | 8            | 20           |
| GSM     | 13           | 33           |
| iLBC    | 13.3         | 50           |

Table 4: Parameters of BEDCA algorithm.

| Parameter | Value |
|-----------|-------|
| α         | 0.875 |
| m         | 7     |
| β         | 5     |
| ε         | 6     |
|           | 0.01  |

We use the following three metrics in our evaluation.

(1) **Gain of goodput**: the average throughput of the successful transmissions for a certain duration.

(2) **Downlink/uplink throughput ratio**: the throughput ratio between the downlink traffic and the uplink traffic.

(3) **VoIP capacity**: the VoIP quality is evaluated on the basis of $R$-score defined in ITU E-model, which is a popularly used QoS metric for VoIP calls [9]. $R$-score takes into account one-way delay, loss rate, and the type of the encoder. The one-way packet delay is measured as the time from the source sending a packet to the destination receiving it while the packet loss rate is the average proportion of packets lost during the measurement period [26]. For the G.711 encoder, the $R$-score is expressed as [27]

$$R = 94.2 - 0.024d - 0.11(d - 177.3)H(d - 177.3)$$

$$- 30 \ln (1 + 15e),$$

where $d$ is the one-way delay (in milliseconds), $e$ depicts the packet loss probability, and $H(x)$ is the step function:

$$H(x) = \begin{cases} 0 & x < 0 \\ 1 & x \geq 0 \end{cases}$$

(17)

The values of the $R$-score are typically categorized as shown in Table 5. In general, the acceptable limit of $R$-score is 60 in WLANs [27, 28]. Thus, the VoIP capacity is the maximum number of calls whose $R$-score is above 60 simultaneously for a certain channel bit rate.

| R-score range | MOS | Speech transmission quality |
|---------------|-----|-----------------------------|
| 100–90        | 4.50–4.34 | best                       |
| 90–80         | 4.34–4.03 | high                       |
| 80–70         | 4.03–3.60 | medium                     |
| 70–60         | 3.60–3.10 | low                        |
| 60–0          | 3.10–1.00 | poor                       |

4.2. Experimental Results. We first present the throughput improvement of BEDCA as shown in Figures 2 and 3. Figure 2 plots the throughput performance of four different schemes as the number of STAs changes from 1 to 25. We select the 64 kbps G.711 codec as the workload. Combining the uplinks and downlinks, each STA at the 64 kbps G.711 codec should produce 128 kbps throughput.

As we can see in Figure 2, when the numbers of STAs are one and five, all four schemes have the throughput around 0.128 Mbps and 0.64 Mbps, respectively. When the number of STAs is getting larger, however, the throughput of the EDCA scheme suffers the performance degradation because of the growing number of the message collisions caused by the larger traffic volume from the STAs. A similar phenomenon can be identified in Figure 3, which shows the average throughput when different codec schemes are used. It is obvious that the performance difference between EDCA and other three schemes is the largest for the G.711 workload and getting smaller when the codec scheme becomes GSM and G.7321. The reason is that both GSM and G.7321 have much less traffic volume than that of G.711 and therefore incur much less message collisions.

Besides, the figures also show that WiFox performs worse compared with BDCF and BEDCA in terms of network throughput. As mentioned before, the reason lies in the fact that WiFox alleviates the traffic asymmetry problem only by dynamically controlling the priority of AP over associated stations. This will aggravate the network collisions, leading to degraded network throughput in heavy load network environment, for example, when the VoIP conversations are above 10 with G.711 codec.

As discussed previously, the balance between the uplink and the downlink streams is a key factor that determines the quality of the VoIP services. When the balance is reached, the throughput ratio of the uplink to the downlink should be...
equal to one. Figure 4 plots the throughput ratio comparison among the four different schemes. To simplify the illustration, our ratios are always computed by comparing a smaller value to a larger value. As a result, the ratios are always less or equal to one. Our simulation is performed with 10 pairs of VoIP flows.

As shown in Figure 4, the ratios of BEDCA are closest to one for all codecs and indicate the better uplink/downlink balance compared to the other three schemes. In comparison, EDCA has the poorest performance with the ratios being around 0.3 for G.711 and GSM codes. The reason is that there is no service differentiation between the AP and the STAs. When the network load is heavy, the STAs are most probably to win the channel content and thus the downlinks (sent by the AP) are suppressed, eventually causing the buffer overflow at the AP, as observed in Figure 6. Since BDCF as well as WiFox favors the AP channel access, the AP can easily win the channel contention. As a result, they achieve better ratio comparison with EDCA. In addition, WiFox outperforms BDCF. The reason lies in that WiFox prioritizes APs channel access over competing STAs according to the dynamic traffic load at the AP in terms of the queue length. BDCF gives AP the priority by assigning the STAs’ contention window size $n$ times larger than that of the APs, which is a relative constant value with neglecting the network traffic load. When the traffic becomes much heavier, for example, G.711 codec is used, however, the STAs with BDCF start to have trouble to acquire the channel accesses (this phenomenon is also observed in Figure 7) and thus its ratio performance degrades to a lower value.
Next, we are curious to compare the VoIP capacity of the four schemes. Our capacity measuring methodology is described as follows. For each scheme, first we start with one pair of VoIP flows. We record the average packet delay by measuring the difference between the packet transmission and reception time and estimate the packet loss rate by dividing the number of received packets by the total number sent. Based on the average packet delay and the loss rate, we determine the $R$-score defined in (16). We make sure that the $R$-score is no less than 60, which is above the low quality of the VoIP service as shown in Table 5. Second, we add one more pair of VoIP flows and measure the $R$-scores for all VoIP pairs. We make sure that none of the pairs has the $R$-score below 60, which implies that the system can service all VoIP flows with the guaranteed quality. We repeat the above procedure until we find that at least one of the flows has the $R$-score lower than 60. We record the previous number of pairs (when the system does not have any flow’s $R$-score under 60) as the VoIP capacity.

It is worth noting that BEDCA, which works at the MAC layer, is based on 802.11e. Its performance is only determined by the operation mechanism of the EDCA access method, regardless of the amendments for the PHY layer as well as the MAC layer of other 802.11 standards (802.11a/b/g/n). Thus, in the paper, we take the PHY parameters of 802.11b and 802.11g as an example to perform the experiment.

Figure 5 clearly shows that BEDCA always outperforms the other three schemes. Particularly, with 802.11g scheme, the VoIP capacity of BEDCA is 51 while that of EDCA is 28. In other words, BEDCA can accommodate as much as 82.1% more VoIP flows (in 802.11g) than EDCA. It can be explained that the higher channel access priority assigned to the AP effectively avoids the downlink data collision and therefore reduces the downlink data loss caused by the AP buffer overflow. Figure 6 plots the CDF of the packet loss rate with 50 pairs of 802.11g VoIP flows. As shown in the figure, the loss rate of nearly 50% of packets, belonging to the downlink flows, is distributed in the range of [0.09, 0.2] while the loss rate of the other 50% packets, belonging to uplink flows, is below 0.07. The loss rate of the downlink flows is so high that the $R$-score of these flows becomes low, resulting in poor VoIP capacity. Comparatively, the packet loss rate distribution of BEDCA is quite concentrated and the highest loss rate is below 0.12. Another reason for the enhancement of BEDCA is that the dynamic updating technique of the STAs can adapt to the various network traffic loads.

Besides, as shown in Figure 5, both BDCE and WiFox perform worse than BEDCA but better than EDCA. The reason can be easily concluded from the above experimental results. On the one hand, they adjust the MAC parameters to enhance the network throughput. On the other hand, the two algorithms give the AP a priority to avoid the high packet loss rate of the downlink flows. However, WiFox only prioritizes AP’s channel access, leaving the STAs’ MAC parameters unchanged. This may aggregate the collision in heavy load WLAN environment, causing degraded throughput as shown in Figures 2 and 3, consequently poor VoIP capacity compared to BEDCA. Meanwhile, in BDCE, the contention window of STAs depends on the number of the participating STAs, which is $n$ times that of AP. This gives AP too high priority when $n$ is large, for example, 50, which causes high one-way packet delay of the uplink VoIP flows. As shown in Figure 7, the one-way packet delay of 50% packets belonging to the downlink flows in BDCE is smaller than that of BEDCA. However, the other 50% packets belonging to the uplink flows have such a high one-way packet delay, which is up to 300 ms, making an unacceptable VoIP quality.

Furthermore, we can also see that WiFox outperforms BDCE from Figure 5. Again, it is because that WiFox adjusts the MAC parameters more dynamically according to the network traffic load, which can balance the uplink and downlink traffic more efficiently as shown in Figure 4.

In wireless networks, packet losses may be caused by collisions as well as other reasons like degraded signal quality and handoffs. However, they are not quite easy to distinguish...
Table 6: Packet error rate of GE model with varying $P_B$.

| $P_B$ | $P$         |
|-------|-------------|
| 0.4   | 0.1660      |
| 0.5   | 0.2060      |
| 0.6   | 0.2460      |
| 0.7   | 0.2860      |

at AP side. For simplicity, when calculating the network collision rate, we omit the retransmissions due to the transmission errors. To study the effects of the wireless channel errors on the quality of VoIP, we conduct experiments with the Gilbert-Elliott (GE) model. The GE model is one of the well-known channel models used to measure the burst error pattern [29]. It is based on a Markov chain with two states $G$ (for good or gap) and $B$ (for bad or burst). In the good state ($G$), losses occur with low probability $P_G$, while in the bad state ($B$) they happen with high probability $P_B$. Let $P_{GB}$ be the probability of the state transitioning from a good state to a bad state, and let $P_{BG}$ be the transition from a bad state to a good state. The average packet loss rate produced by the Gilbert channel is $P = (P_G P_{BG} + P_B P_{GB})/(P_G P_{BG} + P_B P_{GB})$. In the experiment, we set $P_G$, $P_{GB}$, and $P_{BG}$ as 0.01, 0.96, and 0.94, respectively, where $P_{GB} = 1 - P_{GB}$ and $P_{BG} = 1 - P_{BG}$ [30]. $P_B$ is varied from 0.4 to 0.7 with 0.1 intervals to vary the packet error rate $P$ as shown in Table 6. In the experiments, the NOAH routing is used.

The experimental results are shown in Figure 8. As expected, the VoIP capacity of all the four algorithms degrades gradually with the increasing packet error rate. The reason is obvious that the packet losses caused by link error increase the retransmissions both at the AP and at the STAs, causing degraded throughput and increased packet delay. Besides, BEDCA still outperforms the other three schemes in this condition. BEDCA assumes that all the retransmissions are caused by collisions. Thus, the increasing packet error rate will lead to a higher estimation of the network collision rate. As depicted in Algorithm 1, BEDCA will give the STAs a larger contention window size correspondingly. However, we find that although BEDCA can achieve a better balance between the uplink and downlink traffic as shown in Figure 4, the uplink flows are still the dominant in the previous experiments. Thus, the larger $CW_{min}$ of STAs will not cause the unbalance between the uplink and downlink VoIP flows. In addition, in the implementation, we limit the maximum value of STAs’ $CW_{min}$ to 31 to keep the traffic categories in EDCA. Similarly, in WiFox, the packet losses cause a larger queue length at AP which will give AP more chances to access the channel. However, even in this condition, the uplink flows still suppress the downlink flows. Thus, WiFox is also sustainable to the link errors and performs better than BDCF and EDCA. Both BDCF and EDCA give the AP as well as the STAs constant MAC parameters for a certain number of participating STAs. Thus, their performance degradation is solely caused by the network retransmissions due to the link error. Prioritizing AP makes BDCF outperforms EDCA.

Furthermore, we conduct experiments when hidden nodes exist with 802.11b physical parameters. In the wireless network infrastructure, the hidden node problem occurs when the STAs are visible from AP, but not from other STAs communicating with the AP. To construct this scenario, in this experiment, the 72 STAs ($n_0, n_1, n_2, \ldots, n_{70}, n_{71}$) are uniformly distributed around the AP with a radius of 90 meters. The transmission radius and the carrier sense range are set as 100 meters and 150 meters, respectively. The RTS/CTS exchange is disabled. Through numerical calculations, we get the following. (1) From $n_0$ to $n_{12}$, the STAs can hear each other. (2) The distance between $n_0$ and $n_{23}$ is larger than 150 meters. That is to say, $n_{23}$ is hidden from $n_0$. (3) $n_{34}$ is hidden from $n_0$, $n_1$. (4) The nodes in the range from $n_{18}$ to $n_{23}$ (cluster$_1$) are hidden from all the nodes in the range from $n_{6}$ to $n_{12}$ (cluster$_2$). Based on these analyses, our simulation experimental methodology is as follows. (1) We start one/two pair of VoIP conversations in cluster$_1$. (2) We add one more pair of VoIP flows in cluster$_2$ and measure the $R$-scores for all VoIP pairs. Step (2) will be repeated until we find that at least one of the flows has the $R$-score lower than 60. The previous number of pairs is recorded as the VoIP capacity.

The experimental results are shown in Figure 9. According to the figure, the VoIP performance degrades when there are hidden nodes. It is because the fact that hidden STAs cannot hear each other, the carrier sensing mechanism may fail to prevent packet collisions. When a collision occurs, the binary exponential backoff (BEB) algorithm in EDCA will compute a new random backoff time to retransmit the frame. Since the BEB algorithm assigns a larger backoff counter to the failed STA, the corresponding STA cannot compete with the channel immediately even when the channel is idle. This causes performance degradation over wireless networks. Besides, BEDCA performs best, followed by WiFox and BDCF. EDCA performs worst. Again, the larger $CW_{min}$ of the STAs caused by the increased collisions in BEDCA does not either destruct the balance between the uplink and downlink flows or induce server collisions. Meanwhile, the dynamical adjustment of the MAC parameters and the prioritization of AP guarantee the performance of VoIP flows over wireless networks.
Finally, we conduct experiments on how the proposed scheme would deal with mixed flows, where both VoIP flows and data flows exist. In the experiments, the data flows act as best-effort traffic. The rate of the data flow is 1 Mbps and the packet size is 1000 bytes. The experiments consider the legacy 802.11e MAC and the BEDCA. In the experiments, we take the 802.11b parameters and the G.711 codec as an example. Here we evaluate the maximum number of VoIP conversations and the obtained data throughput.

Figure 10 depicts the data throughput achieved by the data flow for an increased number of voice conversations when there are only one uplink data flow and one downlink data flow. Figure 11 describes the obtained throughput of data flows when there are 5 pairs of VoIP flows. From the figures we can see that the throughput of data decreases linearly with the number of voice flows. Besides, the data throughput, for a given number of VoIP conversations, is higher when using BEDCA as compared to EDCA. The reason lies in that the STAs’ contention window of the AC_VO is fixed at (3, 7) in EDCA while that is varied with the network conditions in BEDCA which is always larger than 7 in the experiments. That is to say, BEDCA performs a less aggressive behavior compared with EDCA in the mixed flows scenario and gives the data flows higher throughput.

Figure 12 shows the VoIP capacity; we find that it reduces slowly with increasing number of data flows. As shown in the figure, when there are 20 data flows (10 uplink data flows and 10 downlink data flows), BEDCA can still support 11 voice flows while the maximum number of conversations with EDCA is 7 (in 802.11b).
5. Conclusion

In the paper, we propose a novel access scheme for 802.11e WLANs with the goal to improve the QoS of VoIP over WLAN by differentiating the service between the AP and STAs. We first present the theoretical analysis that reveals the relation between the throughput and the access probability. Then we derive the expression of the optimal access probability that maximizes the throughput where we find that the optimal access probability of AP is a relative constant value independent of the number of the STAs. Finally, a practical scheme is developed that enables the STAs to achieve the optimal access probabilities by adjusting their CW min parameters. We validate our algorithm through simulation which reveals a margin improvement of VoIP capacity. For future work, we propose implementing BEDCA in the off-the-shelf commercial IEEE 802.11 AP and conduct experiments in real world wireless environments. Besides, other parameters that indicate the network traffic load will be taken into consideration, like idle slots, to further enhance the performance of BEDCA.

Conflict of Interests

The authors declare that there is no conflict of interests regarding the publication of this paper.

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