Enhancing WLAN MAC Protocol performance using Differentiated VOIP and Data Services Strategy

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Summary
The application of Voice over-IP (VoIP) over WLANs has drawn a lot of attention from both industry and academia. Currently, the highly dense deployment of WLANs results in that the number of stations in a Basic Service Set (BSS) increases quickly, and thus the work on the capacity for VoIP is becoming a new research hot point. For many of today's 802.11 MAC implementations, the 802.11e requires a HW upgrade. However, replacing existing 802.11 HW devices to provide QoS is costly, and hence may not be desirable to many WLAN owners, especially, hotspot service providers with a huge number of deployed APs.
In recent years, various methods have been proposed to improve the capacity for VoIP in WLANs, and they can be divided into two approaches.

- Enhance the efficiency of VOIP in MAC layer
- Differentiate VoIP and data services

The first approach is achieved by header compression and frame aggregation.This strategy requires modification to existing protocols, and cannot be supported by current devices. This paper considers only the second method. It proposes a simple software upgrade based solution, called an Extended Dual queue Scheme (EDQ), to provide QoS to real-time services such as VoIP. The extended dual queue scheme operates on top of the legacy MAC. EDQ does not require any WLAN device hardware (HW) upgrade.
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Key words:
VOIP, WLAN, MAC, EDQ and QOS.

1. Introduction

Recent advances in wireless technology have equipped portable devices with wireless capabilities that allow networked communication even while a user is mobile. These devices include palmtop computers, Personal Digital Assistants (PDAs), portable computers, digital cameras and printers. To deal with this wireless connectivity need, various wireless communication standards have been developed [1]. Two major projects have been involved in standardizing the physical and the Medium Access Control (MAC) layers for wireless LANs, namely IEEE 802.11 [2] and ETSI HiperLAN.
Nowadays, the IEEE 802.11 WLAN technology offers the largest deployed wireless access to the Internet. This technology specifies both the Medium Access Control (MAC) and the Physical Layers (PHY) [15]. The PHY layer selects the correct modulation scheme given the channel conditions and provides the necessary bandwidth, whereas the MAC layer decides in a distributed manner on how the offered bandwidth is shared among all stations (STAs). This standard allows the same MAC layer to operate on top of one of several PHY layers.
It is well known that the length of a voice packet is much smaller than that of a data packet, and thus the efficiency of VoIP over WLANs is much lower than that of data services due to the fixed header overhead. Moreover, the VoIP service cannot provide saturated traffic, i.e., the voice station does not always have a packet available for transmissions, and thus VoIP service is less opportunistic than data service, which is generally considered to be saturated traffic, to contend the wireless resource. Therefore, it is challenging to improve the capacity for VoIP, which is defined as the number of voice stations supported simultaneously, in WLANs, especially in the scenario where both VoIP and data services exist.
In recent years, various methods have been proposed to improve the capacity for VoIP in WLANs, and they can be divided into two categories. One is to enhance the efficiency of VoIP itself in MAC layer by header compression [3] and frame aggregation [4]. The other is to differentiate VoIP and data services by enhancing the medium access control protocol so that voice stations obtain higher priority to access channel than data stations [5] [17]. Because the former requires modification to existing protocols, and cannot be supported by current devices, the paper deals with the second method.
The emerging IEEE 802.11e MAC, which is an amendment of the existing 802.11 MAC, will provide the QoS [6] [7] [14] [16]. The standardization of the IEEE 802.11e is still on-going at the final stage. The new MAC protocol of the 802.11e is called the Hybrid Coordination Function (HCF). The HCF contains a contention based channel access mechanism called Enhanced Distributed Channel Access (EDCA), which is an enhanced version of the legacy DCF, for a prioritized QoS support. With EDCA, a single MAC contains multiple queues with different priorities that access channel independently in parallel. Frames in each queue are transmitted using different channel access parameters.

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The extended dual queue scheme basically implements two queues in the device driver of the 802.11 WLAN devices. Therefore, these queues are conceptually located on top of the 802.11 MAC controllers running the legacy DCF. EDQ implements two queues in device driver, one for VoIP services and the other for Data services. The VoIP queue is always served prior to the data queue via strict priority queuing. It is shown that the extended dual queue approach provides good QoS to the VoIP packets in [8].

2 Related Works

2.1 IEEE 802.11 Legacy MAC

The IEEE 802.11 legacy MAC [9] defines two coordination functions, namely, the mandatory Distributed Coordination Function (DCF) based on CSMA/CA and the optional Point Coordination Function (PCF) based on poll-and response mechanism. Most of today's 802.11 devices operate in the DCF mode only. The author's overview how the DCF works here as the dual queue scheme proposed in [8] runs on top of the DCF-based MAC and the 802.11e EDCA is also based on it. The 802.11 DCF works with a single first-in-first-out (FIFO) transmission queue. The DCF CSMA/CA works as follows: when a packet arrives at the head of transmission queue, if the channel is busy, the MAC waits until the medium becomes idle, and then defers for an extra time interval, called the DCF Interframe Space (DIFS). If the channel stays idle during the DIFS deference, the MAC then starts the backoff process by selecting a random backoff counter. For each idle slot time interval, the backoff counter is decremented. When the counter reaches zero, the packet is transmitted. The timing of DCF channel access is illustrated in Figure 1. Each station maintains a Contention Window (CW), which is used to select the random backoff counter. The backoff counter is determined as a random integer drawn from a uniform distribution over the interval [0, CW]. If the channel becomes busy during a backoff process, the backoff is suspended. When the channel becomes idle again, and stays idle for an extra DIFS time interval, the backoff process resumes with the suspended backoff counter value. For each successful reception of a packet, the receiving station immediately acknowledges by sending an acknowledgement (ACK) packet. The ACK packet is transmitted after a short IFS (SIFS), which is shorter than the DIFS.

2.2 Dual Queue Scheme with Legacy 802.11 MAC

Simple dual queue scheme in [8] [10] provides a QoS for the VoIP service enhancement over 802.11 WLAN. The biggest advantage of this scheme is that it can be implemented in the existing 802.11 hardware. The dual queue approach is to implement two queues, called Real-time (RT) and Non Real-time (NRT) queues. Especially, these queues are implemented above the 802.11 MAC controllers, i.e., in the device driver of the 802.11 network interface card (NIC), such that a packet scheduling can be performed in the driver level. Packets from the higher layer or from the wireline port (in case of the AP) are classified to transmit into RT or NRT types. The port number as well as UDP packet type is used to classify a RT packet. Packets in the queues are served by a simple strict priority queuing so that the NRT queue is never served as long as the RT queue is not empty. It turns out that this simple scheduling policy results in a surprisingly good performance. It also implements the dual queue scheme in the HostAP driver [11] of Intersil Prism2.5 chipset [10].The MAC controller itself has a First In First Out (FIFO) queue (referred to as "MAC HW queue"). The performance of the dual queue scheme is compromised due to the queuing delay within the FIFO queue when the FIFO queue is large [8]. Unfortunately, the size of the MAC HW queue cannot be configured in many chipsets.

2.3 Modified Dual Queue Scheme

The MAC controller itself has a FIFO queue (referred to as "MAC HW queue"). The performance of the dual queue scheme is compromised due to the queuing delay within the FIFO queue when the FIFO queue is large [8]. Unfortunately, the size of the MAC HW queue cannot be configured in many chipsets. To handle this, MDQ implemented a NRT packet number controller, which restricts the number of outstanding NRT packets in the MAC HW queue. This modified scheme is referred to as Modified Dual Queue (MDQ). For the simulation of the modified dual queue in this paper, MDQ assumes that the number of NRT packets in the MAC HW queue is limited to two, thanks to the flow control unit. This number is the smallest, which can be practically implemented.

3 Extended Dual Queue Systems (EDQ)

This section provides the details of the proposed extended dual queue system. 3.1 provide the operations of the extended dual queue system. 3.2 provide the pseudo structure of the EDQ.

3.1 Operations of the Extended Dual Queue System

Extended dual queue scheme provides a QoS for the VoIP service enhancement over 802.11 WLAN. The biggest advantage of this scheme is that it can be implemented in the existing 802.11 hardware. The dual queue approach is to implement two queues, called voice queues and data queues inside the AP.

The EDQ scheme is proposed to provide a QoS for the VoIP service enhancement over 802.11 WLAN. The biggest advantage of this scheme is that it can be implemented in the existing 802.11 hardware. The dual queue approach is to implement two queues, called VoIP queue and data queue. Especially, these queues are implemented above the 802.11 MAC controllers, i.e., in the device driver of the 802.11 network interface card (NIC), such that a packet scheduling can be performed in the driver level. Packets from the higher layer or from the wire line port (in case of the AP) are classified to transmit into VoIP or data types. The port number as well as UDP packet type is used to classify a VoIP packet. Packets in the queues are served by a simple strict priority queuing so that the data queue is never served as long as the VoIP queue is not empty. It turns out that this simple scheduling policy results in a surprisingly good performance.

3.2 Algorithm Structure of EDQ

```
Read incoming packet type at AP
set MAX priority to VoIP queue
set MIN priority to data queue
if (VoIP Packet)
        move current packet to VoIP queue
else
        move current packet to data queue
for (every packet transmission in AP)
        if (VoIP queue not empty)
                transmit the VoIP packet
        else
                transmit the data packet
```

4. Details of Extended Dual Queue Scheme

The 802.11 legacy MAC does not support the concept of differentiating packets with different priorities. Basically, the DCF is supposed to provide a channel access with equal probabilities to all stations contending for the channel access in a distributed manner. However, equal access probabilities are not desirable among stations with different priority packets. The EDQ is designed to provide differentiated, distributed channel accesses for packets with different priorities by enhancing the DCF.

The lengths of the voice packets are much smaller than that of other kind of data packets. We watch the every incoming packets length if it's too small our algorithm insert the packets into voice queue otherwise insert the packets into data services queue. Also our algorithm put the higher priority to voice queue. If voice queue contains packets it will be first processed by MAC controller. We have the benefit from the voice packet length, because it will also quickly processed by MAC controller. If the voice queue is empty after that data service queue packets are processed by MAC controller.

The VoIP service cannot provide saturated traffic; the voice stations do not always have a packet available for transmissions.

Here we set the Access Category (AC) uplink (STAs to APs) priority to RT and NRT packets. We set the user priority 3 to AC_VO (Voice), priority 2 to AC_VI (Video), priority 1 to AC_BE (Best Effort) and priority 0 to AC_BK (Back Ground).

We set the Access category (AC) downlink (APs to STAs) priority to RT and NRT packets. We set the user priority 7 to AC_VO (Voice), priority 6 to AC_VI (Video), priority 5 to AC_BE (Best Effort) and priority 4 to AC_BK (Back Ground).

In this section, the performance enhancement of EDQ is presented. Basically, a channel access function uses arbitration interframe space (AIFS [AC]), CWmin [AC], and CWmax [AC] instead of DCF Inter Frame Space (DIFS), CWmin, and CWmax, of the DCF, respectively, for the contention process to transmit a packet belonging to access category AC. AIFS [AC] is determined by

$$\text{AIFS [AC]} = \text{SIFS} + \text{AIFSN [AC]}.\text{Slot Time}$$

Where arbitration interframes space number AIFSN [AC] is an integer greater than one for STAs and an integer greater than 0 for APs. The backoff counter is selected from [0, CW [AC]]. The values of AIFSN [AC], CWmin [AC], and CWmax [AC], which are referred to as the DCF parameter set, are advertised by the AP via Beacons and Probe Response frames. The AP can adapt these parameters dynamically depending on the network condition. Basically, the smaller the AIFSN [AC] and CWmin [AC], the shorter the channel access delay for the corresponding priority, and hence the more capacity share for a given traffic condition. However, the collision probability increases when operating with smaller CWmin [AC]. These parameters can be used in order to differentiate the channel access among different priority traffic.

The 802.11 DCF is originally designed to provide a fair channel access to every station including the AP. However, since typically there is more downlink (i.e., AP-to-stations) traffic, AP's downlink access has been known to be the bottleneck to the entire network performance. Here our EDQ with DCF Scheme, which allows the differentiation between uplink and downlink channel accesses, can be very useful to control the network performance.

5 Performance Evaluation

In this section, an attempt is made to evaluate the performance of the original DCF, the modified dual queue (MDQ) scheme, the 802.11e EDCA using ns-2 simulator [12] and EDQ scheme. The following traffic models are used for simulations: two different types of traffic are considered for simulations, namely, voice and data. The voice traffic is modeled by a two-way constant bit rate (CBR) session according to G.711 codec [13]. The data traffic application is modeled by a unidirectional FTP/TCP flow with 1460-byte packet size and 12- packet (or 17520-byte) receiver window. This application corresponds to the download of a large file via FTP. The 802.11b PHY is used for simulations. The 11 Mbps transmission rate (out of 1, 2, 5.5, and 11 Mbps of the 802.11b PHY) is used in the simulations.
The queue sizes of 500 packets are used at the AP, which is large enough to ensure that there, is no buffer overflow in our simulation environments [8]. The network topology for the simulations is shown in Figure 1.

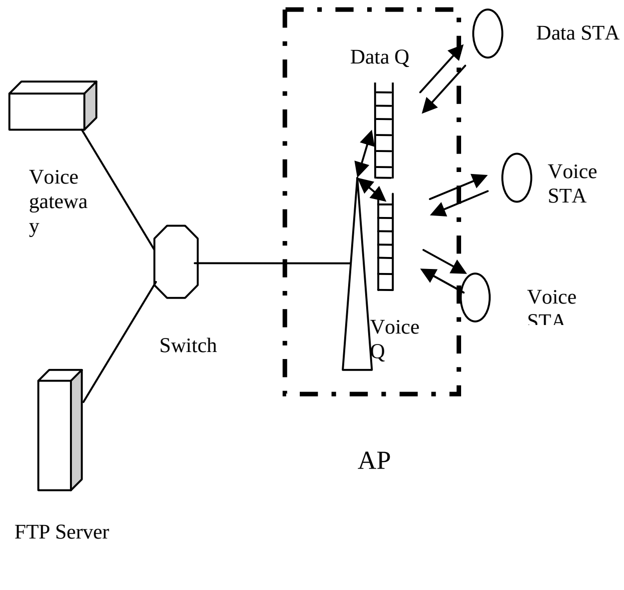

Figure 1: Network Model for the Simulations

5.1 Network Topology

Each station involving a VoIP session generates and receives only voice traffic. The other stations receive only TCP packets, and each of them treats only one TCP flow, i.e., the number of TCP flows corresponds to downstream TCP stations. This topology can be often found in the real WLANs with mixed VoIP and Internet traffic.

5.2 Comparison of Single Queue, MDQ, EDCA and EDQ

The authors simulate with a single VoIP session and various number of downstream TCP flows in order to compare the VoIP performance of MDQ, EDCA and EDQ. Figure 2, 3 presents the delay performance of these three schemes. As presented in [8], the downlink delay of the single queue increases linearly as the number of TCP flow increases, and hence cannot be used for VoIP in the mixed traffic environments. On the other hand, both MDQ and EDCA provide reasonable delay performance virtually independent of the TCP flow number. This is because both schemes provide higher priority to the VoIP packets over the TCP packets. The understanding of the detailed behavior of the MDQ scheme should be referred to [8]. The Figure 3 also shows that EDQ is totally independent of the number of TCP flow.

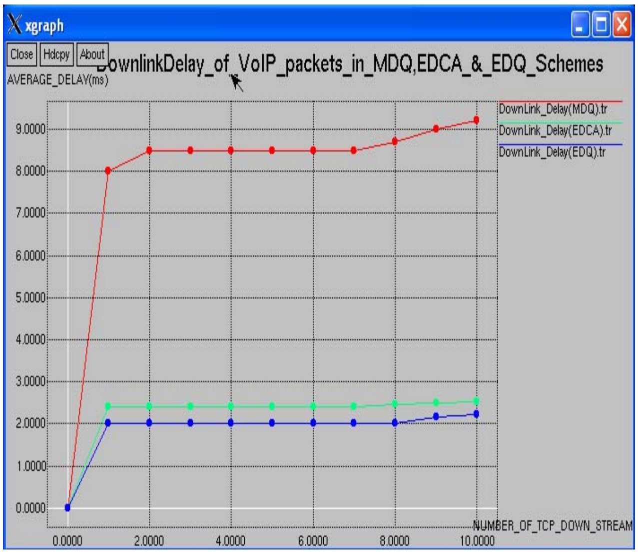

Figure 2: Down link Delay of VoIP packets in MDQ, EDCA and EDQ Schemes When Number of Down TCP Streams Increases.

Figure 2 shows that the voice delay of EDQ is reduced compare to MDQ and EDCA. The reason can be understood as follows: first, the EDQ uses smaller values of the channel access parameters than the MDQ, based on the legacy DCF, namely, CWmin [AC_VO] = 7 and CWmax [AC_VO] = 15 for EDCA AC_VO [3], and CWmin = 31 and CWmax = 1023 for the legacy DCF [2], respectively, in the case of the 802.11b PHY. Smaller channel access parameters imply a faster channel access.

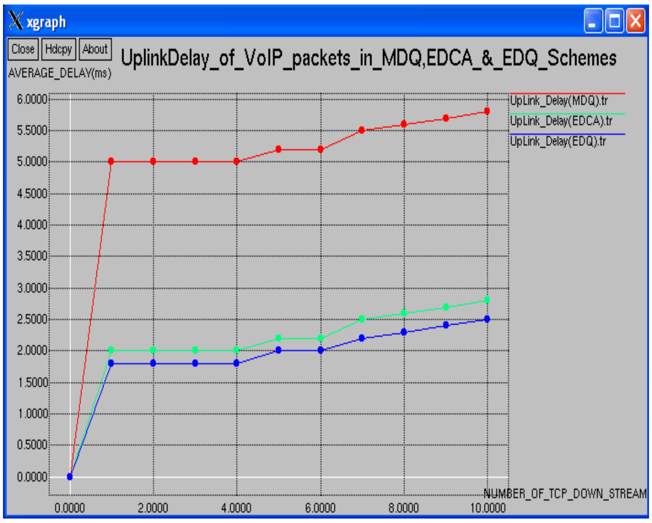

Figure 3: Up link Delay of VoIP packets in MDQ, EDCA and EDQ Schemes When Number of Down TCP Streams Increases.

From Figure 3 it should be also noted that the uplink delay performances of both EDCA and EDQ are the same since there is no difference between two schemes in case of uplink in simulation scenarios. That is, in simulations, a station transmits only a single type of traffic, i.e., either VoIP or TCP-ACK.

5.3 Performance Comparison when Number VOIP Session Increases

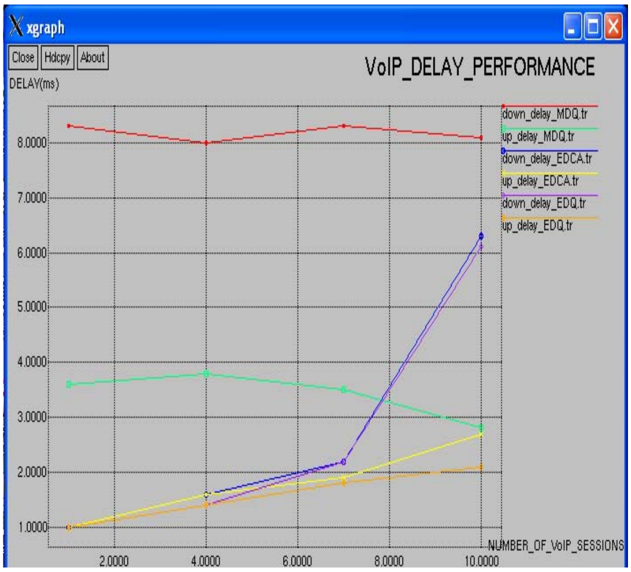

Figure 4: VoIP delay performances of MDQ, EDCA and EDQ

Figure 4 presents the delay performance comparison for the MDQ EDCA and EDQ as the number of VoIP sessions increases. Here we simulate with 10 downstream TCP flows and various number of VoIP sessions. The EDQ provides a better VoIP delay performance than the EDCA scheme with multiple VoIP sessions while both of them still provide acceptable delay performances. However, it is observed that the delay of EDQ and EDCA, especially, the downlink delay, increases as the VoIP session number increases. This must be a negative effect of small EDQ access parameters, i.e., these small values result in some collisions among VoIP packets from different STAs.

5.4 Aggregated TCP Throughputs Of MDQ, EDCA and EDQ

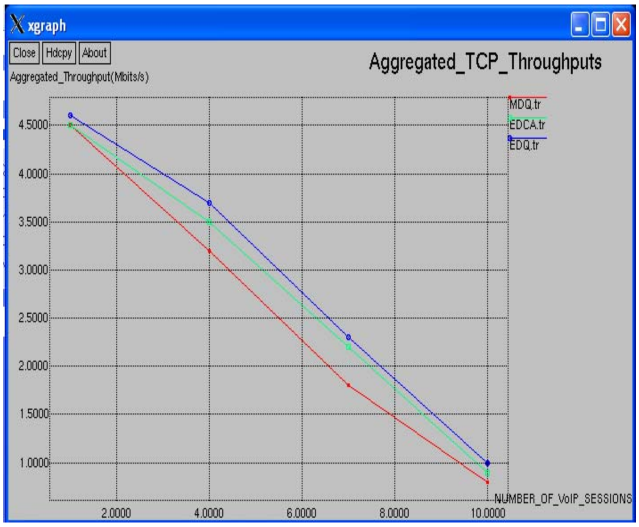

Figure 5 Aggregated TCP Throughputs of MDQ, EDCA and EDQ Schemes in downstream TCP flows

Figure 5 shows the aggregated throughput performance of downstream TCP flows, which are measured at the AP, with MDQ, EDQ and EDCA. It is observed that the EDQ provides a better throughput performance than the MDQ and EDCA. It takes a shorter time for the EDQ to transmit a VoIP packet due to a lower channel access delay. As a result, the EDQ allows more time resource for TCP packet transmissions. Moreover, TCP under EDQ can get more transmission opportunities than that under the MDQ and EDCA because it contends in parallel with VoIP under EDQ. On the other hand, with MDQ scheme, TCP packets are not served when a VoIP packet exists in the RT queue by strict priority queuing. This is the reason why the TCP throughput with the EDQ is a bit larger than that with the MDQ.

5.5 Jitter Performance Comparison

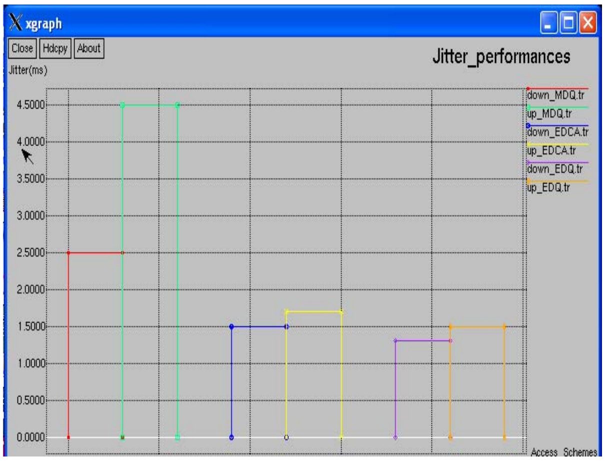

Figure 5 (a): Jitter performances of three access schemes for one VoIP Session

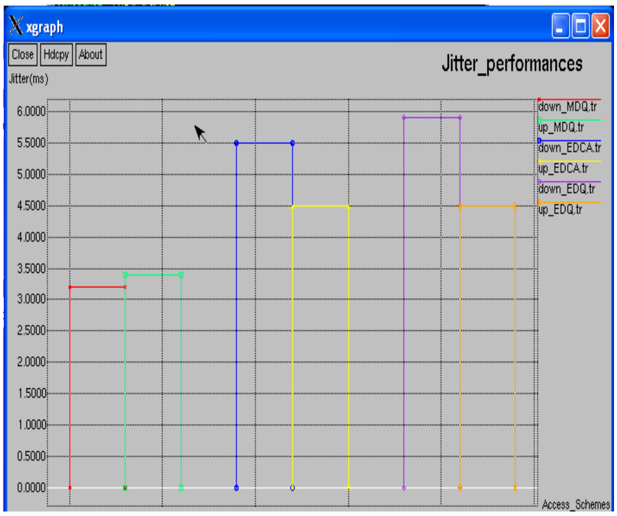

Figure 5 (b): Jitter performances of three access schemes ten VoIP Sessions

Figure 5 shows the jitter performance for both 1 and 10 VoIP session cases. It can be imagined that there are two major factors, which increase the VoIP jitter in the 802.11 WLAN, namely, contention/collision with other stations and random delay inside the queue. First, when one VoIP session exists, i.e., Figure 5 (a), EDQ schemes demonstrate better jitter performances than MDQ and EDCA schemes, thanks to their smaller channel access parameter values. The reason is explained as follows with 1 VoIP session, TCP flows can use a large fraction of the total bandwidth and hence more TCP stations contend for channel. EDQ schemes, which use small channel access parameter values, can reduce the contention with TCP stations and AC_BE in the AP. Accordingly, the jitter becomes smaller.

In Figure 5 (b), the result is quite different from 1 VoIP session case. In this situation, CWmin value of EDQ schemes is not large enough for collision avoidance. Accordingly, many collisions can occur, thus increasing the jitter considerably. However, the jitter of downlink VoIP packets in EDQ remains small because downlink can perfectly avoid the contention with TCP stations. The reason for the increase in jitter of downlink VoIP packets in all schemes is that VoIP packet generation times of each VoIP sessions are randomized in simulations, and hence a VoIP packet arriving at the AP queue experiences random queuing delay. From the jitter performance evaluation thus far, it can be concluded that when there are a smaller number of VoIP sessions, the jitter performance of the EDQ is better than that of the MDQ and EDCA while they perform about the same when there are many VoIP sessions.

6. Conclusions

In this paper, the authors have comparatively evaluated the Extended dual queue (EDQ) scheme, based on the legacy 802.11 DCF/MDQ, and the emerging 802.11e EDCA in terms of their QoS provisioning capability. They have presented an extended version of the originally-proposed dual queue scheme by considering a practical implementation limitation.

From extensive simulations considering the VoIP delay/jitter and TCP throughput, it is found that the EDQ surely provides a better performance than the MDQ and EDCA scheme, thanks to the flexible channel access parameter control of the EDQ depending on the underlying network condition, e.g., the traffic load. It is concluded that the EDQ scheme is practically a good solution in order to provide QoS for VoIP services when the 802.11e is not available or where the HW upgrade is not desirable.

In future, the authors would like to focus on the development of the algorithms in order to improve the capacity of VoIP by deploying multiple APs in the WLAN.

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