Improving the QoS Performance of EDCF in IEEE 802.11e Wireless LANs

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Abstract—**The IETF is currently specifying QoS on the Internet, but providing QoS at the IP layer is sub-optimal without lower layers' support. With the growing popularity and acceptance of IEEE 802.11 wireless LANs, it is essential to focus on QoS enhancement at the MAC layer of the 802.11 standard. The EDCF, proposed by the IEEE 802.11e, is a contention-based MAC protocol supporting service differentiation through different interframe spaces, contention window limits, and persistence factors for different traffic priority classes. In this paper, we propose a retransmission scheme, known as Age-Dependent Backoff (ADB), to alleviate the delay and jitter of real-time packets by adjusting the persistence factors dynamically based on the ages of the real-time packets in the transmission queues and the lifetimes of the real-time packets. Simulation results indicate that using ADB in EDCF is efficient with low delay, jitter and drop rate for real-time traffic in a wide range of traffic loads.**

I. Introduction

The world of communications has undergone many changes over the last few years. One of the most important changes is the convergence of voice, video, and data communications under the roof of the Internet Protocol (IP) suite. IP was originally designed to support data services such as file transfer, e-mail and remote terminal, which are tolerant of delay and jitter. Voice and video services, as opposed to data services, require a certain minimum rate and suffer significantly from high delay and jitter. The development of IP-based multimedia networking applications has imposed more requirements on the IP network, creating a need for end-to-end Quality of Service (QoS) support. Although the Internet Engineering Task Force (IETF) is currently working on service differentiation at the IP layer, the result is sub-optimal without lower layers' support. In recent years, there has been a substantial increase in the deployment of IEEE802.11 wireless LANs for wireless Internet services. Wireless Internet Service Providers (WISPs) are mushrooming everywhere, deploying 802.11 hotspots in coffee shops, hotels, and airports. With the growing popularity and acceptance of 802.11 wireless LANs, it is essential to focus on service differentiation support at the 802.11 medium access control (MAC) layer.

To improve the current 802.11 MAC protocol to support applications with QoS requirements, the IEEE 802.11 Task Group E was formed proceeding with defining QoS enhancements for the 802.11 MAC protocol. The 802.11e draft introduces Enhanced Distributed Coordination Function (EDCF) and Hybrid Coordination Function (HCF), which are currently under discussion [1]. EDCF is a prioritization enhancement of the legacy 802.11 Distributed Coordination Function (DCF) using the Virtual DCF mechanism [2], and is the contention-based channel access mechanism for HCF. HCF is based on a polling mechanism similar to the legacy 802.11 Point Coordination Function (PCF) but allows the Hybrid Coordinator (HC), typically located at the Access Point (AP), to initiate contention-free Controlled Access Periods (CAPs) at any given time during a Contention Period (CP) after the channel remains idle for at least a PCF Interframe Space (PIFS) duration [3]. Since the success of 802.11 wireless LANs is based on DCF, a distributed mechanism with minimum management and maintenance costs, and the dynamic nature of ad-hoc networks makes it difficult to dynamically assign HC maintaining connection, reservation, and scheduling states, not to mention the complexity of handling overlapping coverage areas, we will focus on the improvement of EDCF in this paper.

In the legacy 802.11 DCF protocol [4], retransmission is attempted after a collision using Binary Exponential Backoff (BEB), the process of increasing the backoff range by doubling the contention window (CW) with every unsuccessful transmission. Doubling CW with every transmission retry reduces collision probability but causes large delay and jitter, which are problems for time-sensitive applications such as voice over IP and video conferencing. In the 802.11e EDCF protocol, the size of the new CW after an unsuccessful transmission is determined by expanding/reducing the size of the old CW [5] by a factor of a persistence factor (PF). In the event of a collision, a real-time station attempting retransmission should use a shorter backoff than on its first failed attempt in order to reduce delay and jitter. Retransmitted packets would therefore have a better chance of accessing the medium than new arrivals. In this paper, we propose an efficient retransmission mechanism called Age-Dependent Backoff (ADB), which dynamically adjusts PFs based on the ages of the real-time packets in the transmission queues and the lifetimes of the real-time packets.

The subsequent sections of this paper are organized as follows. In Section II, we briefly review the legacy 802.11 DCF protocol and introduce the 802.11e EDCF protocol. In Section III, we describe the proposed ADB retransmission scheme for EDCF. The simulation model is discussed in Section IV and the performance evaluation is studied in

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Section V. Finally, a conclusion is provided in Section VI.

II. IEEE 802.11 DCF AND 802.11E EDCF

In this section, we describe DCF and EDCF, the two contention-based medium access schemes that are used by 802.11 and 802.11e wireless LANs respectively. The new 802.11e EDCF mode is conceived as a compatible extension of the legacy 802.11 DCF mode. Because 802.11e is a draft standard presently under review, many issues are still unsolved and are expected to change [6]. However, we expect that EDCF described here will not undergo any major modifications.

A. Legacy 802.11 DCF

The fundamental access method of the 802.11 MAC protocol is DCF, which supports asynchronous data transfer on a best-effort basis and is the only possible function in 802.11 ad-hoc networks. The 802.11 DCF MAC protocol operation is depicted in Fig. 1. For a station to transmit a MAC protocol data unit (MPDU), it must sense the medium to determine if another station is transmitting and must ensure that the medium is idle for the specified DCF Interframe Space (DIFS) duration. A station may transmit a pending MPDU when it determines that the medium is idle for a time interval greater than or equal to the DIFS period. If the medium is found to be busy, the station has to keep sensing the channel for an additional random time after detecting the channel as being idle for the DIFS duration. The additional random time period is selected from CW and the size of CW, bounded by the maximum value CW_{max}, is doubled after each unsuccessful transmission to reduce the collision probability. For each successful transmission, CW is reset to the minimum value CW_{min}. This is the so-called BEB algorithm. The backoff time, *backoff_time*, can be expressed as the following equation [7].

$$backoff\_time = randInt(0, min(CW_{min} \times 2^{retry}, CW_{max})) \times slot\_time \quad (1)$$

where $randInt(a, b)$ generates a random integer in the range from a to b uniformly, $min(c, d)$ gives the smaller value of c and d, *retry* is the number of retransmission attempts, and *slot_time* is a time duration specified by the physical layer parameters.

During the backoff, the station decreases its backoff counter by one if the medium is idle for a *slot_time* period and freezes the backoff counter when the medium is busy. When the backoff counter reaches zero, the station will transmit its MPDU immediately. When a destination station receives the MPDU successfully, it sends an Acknowledgement (ACK) frame back to the source station after a Short Interframe Space (SIFS) duration. DCF offers an optional means of transmitting data frames that require the transmission of Request To Send (RTS) and Clear To Send (CTS) frames prior to the transmission of the actual data frames. The RTS/CTS transmission scheme can alleviate the hidden terminal problem and can reduce the transmission time wasted as a result of a collision due to the longer frame size of the actual MPDU. The RTS and CTS frames include the information of how long it takes to transmit the next data frame and the corresponding ACK frame. The Network Allocation Vector (NAV) maintained by each station is an indicator of time periods when other stations close to the transmitting station and hidden stations close to the receiving station will not commence any transmissions. DCF with the RTS/CTS transmission scheme and the NAV settings of other stations are shown in Fig. 2.

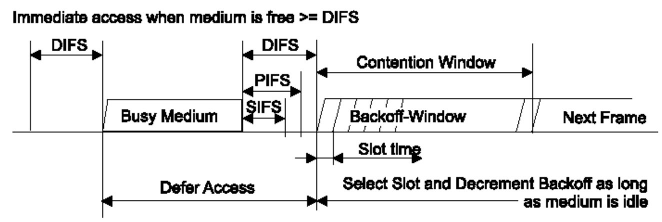

Fig. 1. 802.11DCF MAC protocol operation

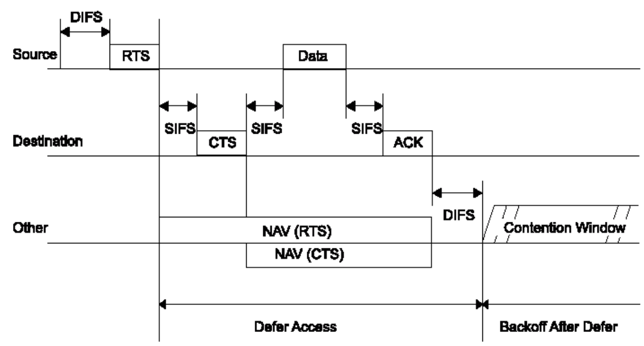

Fig. 2. RTS/CTS/data/ACK and NAV settings

B. New 802.11e EDCF

The new 802.11e EDCF medium access scheme is governed by a distributed mechanism very similar to legacy 802.11 DCF. Service Differentiation is achieved through the introduction of Traffic Categories (TCs). Each TC has a different transmission queue and each transmission queue has a different interframe space (Arbitrary Interframe Space – AIFS[TC]), a different set of contention window limits (CW_{min}[TC] and CW_{max}[TC]), and a different persistent factor (PF[TC]). Fig. 3 illustrates the service differentiation accomplished by using different AIFS values. Each TC within a station starts a backoff independently after detecting the channel being idle for an AIFS[TC] duration. In the EDCF retransmission scheme, the size of the new CW[TC] after an unsuccessful transmission is determined by expanding/reducing the size of the previous CW[TC] by a factor of PF[TC], whereas in legacy 802.11 DCF, CW is always double after every unsuccessful transmission, i.e. PF=2. As in legacy DCF, the CW[TC] never exceeds its maximum bound CW_{max}[TC]. A random backoff counter is chosen from the interval [0, CW[TC]] in the case of AIFS[TC]≥DIFS and from [1, CW[TC]+1] in the case of AIFS[TC]<DIFS [8].

When the backoff counter of a TC reaches zero, the station transmits a pending MSDU from the corresponding transmission queue. A short AIFS, a small size of CW limits, and a low PF value are associated with high priority packets, enabling them to start contenting the channel earlier and to complete the backoff sooner, thus offering a high probability of winning the contention race.

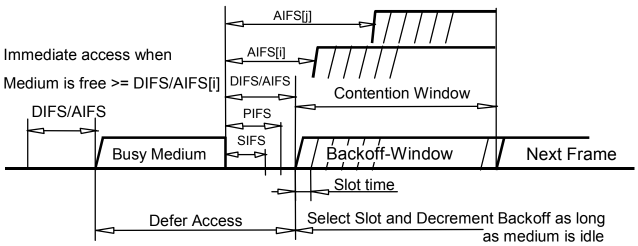

Fig. 3. Service differentiation by different AIFS values

A station may have up to eight transmission queues realized as virtual stations inside a station. Should the backoff counters of two or more parallel TCs in a single station count down to zero at the same time, a scheduler, which resides in the station, resolves the virtual collision by allowing the highest priority TC among the virtually collided TCs to transmit its MPDU [5]. The other virtually collided TCs execute the retransmission mechanism independently, as if a collision had occurred.

III. AGE-DEPENDENT BACKOFF FOR 802.11E EDCF

In this section, we discuss the backoff behavior of EDCF and propose a retransmission mechanism for EDCF to manage the contention for the medium among stations in 802.11e wireless LANs effectively.

A. *Backoff Behavior of 802.11e EDCF*

In legacy DCF, the BEB algorithm doubles CW with every transmission retry to reduce the collision probability in the next retransmission by providing greater transmission spacing among stations with pending MPDUs. CW becomes extremely large after successive transmission retries causing longer delay and jitter. To reduce delay and jitter, a smaller CW should be employed in the next retransmission providing a better chance for retransmitted packets to access the medium than for new arrivals [9].

EDCF utilizes a multiplier, PF, to govern the alternation of CW after an unsuccessful transmission. PF should be less than 1 for time-sensitive applications. However, collisions are a result of congestion and a wider CW is desirable to alleviate congestion. Reducing the CW size for every retransmission causes heavy congestion leading to more collisions.

B. *Age-Dependent Backoff Algorithm*

We now present the Age-Dependent Backoff (ADB) algorithm for high priority real-time traffic. The idea of ADB is to dynamically adjust PF based on the age of a real-time packet in the transmission queue and the lifetime of the real-time packet. The relationship between the new CW, *newCW*[*TC*], and the old CW, *oldCW*[*TC*], after a collision is shown in (2).

$$newCW[TC] = ((oldCW[TC] + 1) \times PF[TC]) - 1 \qquad (2)$$

where *PF*[*TC*] is given in (3).

$$PF[TC] = \frac{2}{LT[TC]} Age + 2 \qquad (3)$$

Age is the packet's age in the transmission queue and *LT* is the lifetime of the packet. The *newCW*[*TC*] never exceeds the parameter CW_{max}[TC] but can be less than CW_{min}[TC] to provide differentiation between retransmitted packets and new arrivals. Real-time packets are obsolete if they are not received by recipients within their lifetime. Packets with queuing delay longer than the lifetime will eventually be discarded by their applications and should not contend for the medium. Therefore, packets with $Age > LT$ are discarded before attempting transmission to save bandwidth and to prevent causing additional delay to other packets.

It can be seen in (2) and (3) that the new CW is expanded by a factor of PF between 1 and 2 in the first half of the packet's lifetime and is compressed by a factor of PF between 0 to 1 in the second half. The first half allows the backoff to increase gradually while avoiding high collision probability, but at the same time precluding a huge increase of delay and jitter. The second half decreases the backoff slowly from the expanded CW to raise transmission probability preventing packets from being dropped.

ADB requires minor modifications in the computation of CW to minimize the migration effort from the new 802.11e EDCF mode and provides backward compatibility to the legacy 802.11 DCF mode.

IV. SIMULATION MODEL

To evaluate the performance of EDCF with ADB, we simulate it on an ad-hoc network consisting of voice, video and data stations using OPNET Modeler 9.1 [10].

A. *Simulation Environment*

Our simulation environment is modeled on an 802.11e/b ad-hoc situation where different services such as IP telephony, video conferencing and best-effort File Transfer Protocol (FTP) may be active simultaneously in an independent Basic Service Set (BSS) and we assume that no hidden stations are present in the independent BSS. Table I shows the simulation parameters.

B. *Traffic Sources*

We use the G.729 coder [11] for our voice traffic model with the characteristic of an ON/OFF process, where voice users are either in talkspurt or silence. For efficient usage of wireless bandwidth, silence suppression is employed. Voice packets are only generated in talkspurt (ON), while no packets are generated in silence (OFF). Both the duration of talkspurt and of silence follow the exponential distribution with the mean duration equal to 1 and 1.35 seconds respectively [12]. Each voice station runs only one bi-directional voice session over UDP/IP.

TABLE I
PARAMETERS FOR OUR SIMULATION

Parameters	Value
Channel rate	11Mbps
Slot time	20μs
SIFS	10μs
DIFS (DCF)	$10\mu s + 2 \times 20\mu s = 50\mu s$
AIFS[1] (Voice)	$10\mu s + 2 \times 20\mu s = 50\mu s$
AIFS[2] (Video)	$10\mu s + 3 \times 20\mu s = 70\mu s$
AIFS[3] (Data)	$10\mu s + 5 \times 20\mu s = 110\mu s$
$[CW_{min}, CW_{max}]$ (DCF)	[31, 1023]
$[CW_{min}[1], CW_{max}[1]]$ (Voice)	[7, 31]
$[CW_{min}[2], CW_{max}[2]]$ (Video)	[15, 63]
$[CW_{min}[3], CW_{max}[3]]$ (Data)	[15, 255]

We consider low-quality video conferencing characterized by a relatively low bit-rate of 128kbps for the uplink and the downlink. We assume that the video frame rate is 20 frames/sec and the video frame length is exponentially distributed with mean 800 bytes. A video station runs one bi-directional video session over UDP/IP.

As for the best-effort FTP traffic, we assume that the upload and the download streams are identical with 200kbps of data traffic in each stream. We also assume that the inter-request times of file transfers follow the exponential distribution and the file size is 1024 bytes. TCP is used for transporting the FTP traffic.

V. PERFORMANCE EVALUATION

The main performance parameters for voice traffic are packet delay, jitter and loss. The total codec delay for G.729 is 25ms [11], and the end-to-end delay for IP telephony ranges from 300 to 1000 ms. User tolerance of delay varies significantly. Demanding users require delay of 200 ms or less, while more patient users are satisfied with delay of 300 to 800 ms [13]. Since the wireless LAN represents only a single hop of an end-to-end connection, we consider 25 ms as the maximum acceptable value for the voice packet transfer delay over the wireless LAN and let *LT* for voice packets be 25 ms.

We assume that 75 ms is the maximum allowable video packet delay at a single wireless hop. *LT* for video packets is assigned to be 75 ms.

We conceived an ad-hoc network as an independent BSS with 10 voice stations, 4 video stations and *n* best-effort FTP client and server stations. Since 802.11e EDCF is designed to be backward compatible to 802.11 DCF, we assume that half of the FTP client and server stations are using the legacy 802.11 DCF protocol while the other half are using the new 802.11e EDCF protocol with PF equal to 2.0.

Voice packet delay, jitter and drop rate are shown in Fig. 4, 5 and 6 respectively. We define the jitter as the variance of delay and the drop rate as the percentage of packets with delay longer than their lifetime. Since ADB dynamically adjusts the value of PF based on the ages and the lifetime of the voice packets, avoiding long delay and at the same time preventing high collision when the traffic load is heavy, the voice packet delay, the jitter and the drop rate are improved considerably.

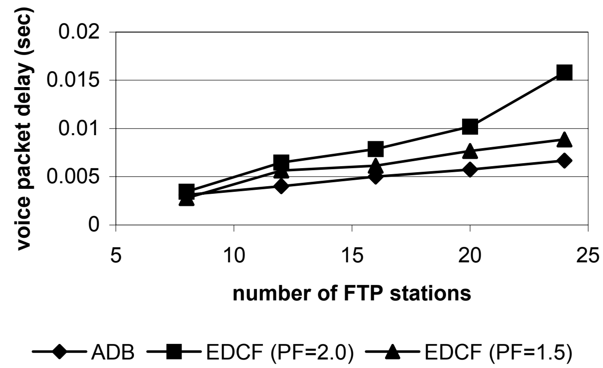

Fig. 4. Voice packet delay

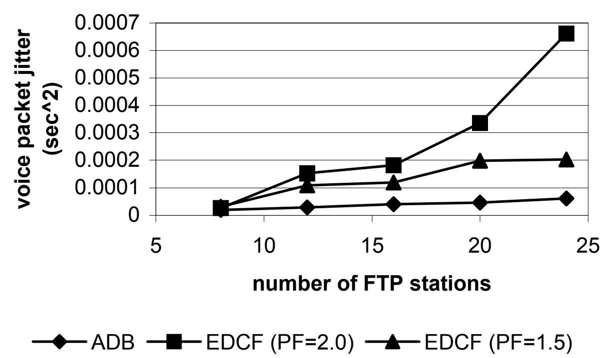

Fig. 5. Voice packet jitter

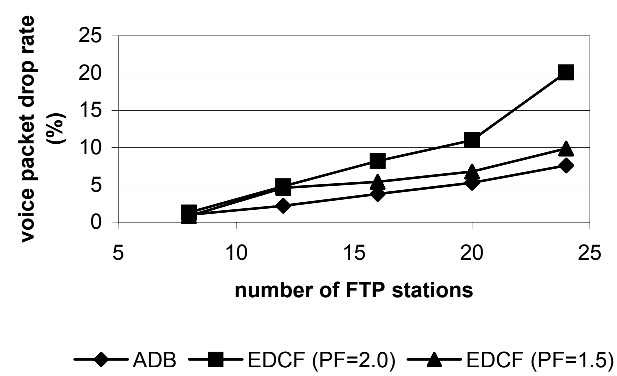

Fig. 6. Voice packet drop rate

Fig. 7, 8 and 9 show that ADB provides significant improvements in video packet delay, jitter and drop rate. As the number of FTP stations increases, the improvements become more noticeable and pronounced.

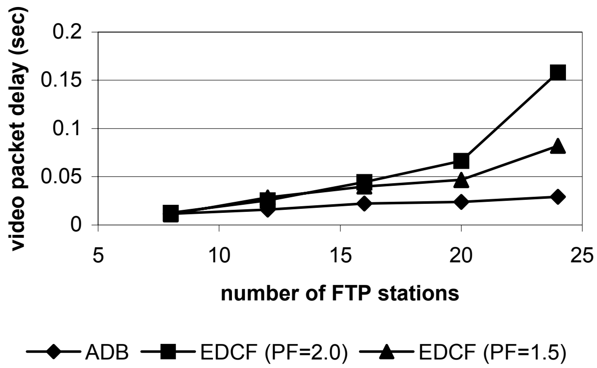

Fig. 7. Video packet delay

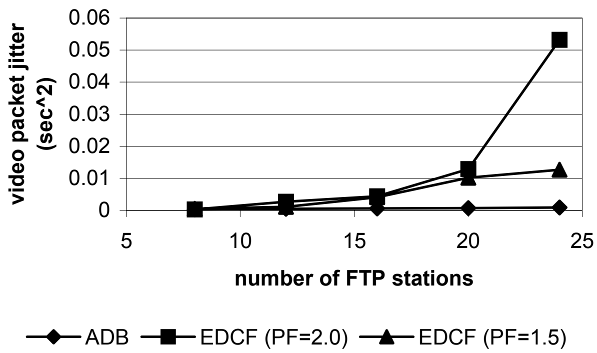

Fig. 8. Video packet jitter

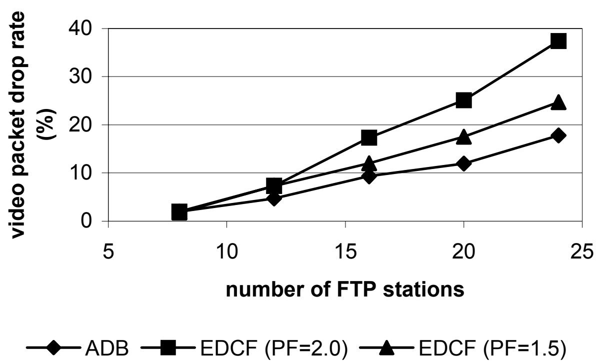

Fig. 9. Video packet drop rate

Fig. 10 demonstrates that ADB enhances the performance of the voice and video traffic without sacrificing the throughput of the best-effort FTP traffic and prevents the FTP traffic from starvation.

VI. Conclusion

In this paper, we propose the ADB algorithm to govern the alternation of CW based on the ages and the lifetimes of the real-time packets. ADB requires minor changes in the computation of CW minimizing the migration effort from the new 802.11e EDCF mode and provides backward compatibility to the legacy 802.11 DCF mode. Simulation results indicate that ADB offers considerable improvements in delay, jitter and drop rate of the real-time packets without causing starvation of the best-effort traffic.

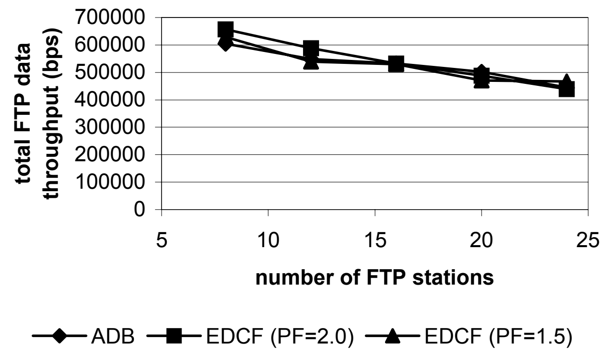

Fig. 10. Total FTP data throughput

VII. References

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