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Improving the quality of Service of EDCF over DCF For Real Time Applications Using Probability Algorithm

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Abstract: IEEE 802.11 (802.11) WLAN standard is being accepted widely and rapidly for many different environments today Main characteristics of the 802.11 networks are their simplicity and robustness against failures due to the distributed approach. They are created and maintained by the IEEE LAN/MAN Standards Committee (IEEE 802). Quality of Service (QoS) support is the key for multimedia applications over WLAN. So, since the demand for QoS is nowadays is a topic of great concern for the transmission of the different services like video, voice, best-effort services controlling the maximum network applications, background services for which the priority assigned is lower than the one assigned to the standard. So in order to get the better Qos at our desired application we need to control the network parameters in such a way that we get a certain output and according to the output we will see that at which station we are getting the maximum output. And at that station we will send the access category at which we want the best output Qos. So, we have resolved this problem of attaining the good Qos at our desired station by giving the random values for the contention window and the arbitrary interframe spaces is done by calculating the probability of winning and the probability of collisions. Due to the increasing demand of high rates WLANS in the applications like video streaming and voice over IP. It's mandatory to improve its performance in every possible way for the ease of the user. To calculate the probability of winning of a station for random values of AIFS and CW using the proposed algorithm and to find out the probability of the collisions in a station for the random values of AIFS and CW using the proposed algorithm and also to analyze the result and check where the output coming better. Then the station at which we are getting the maximum output we will send that data for which we want the best QOS.

Keywords: QoS, SINR, MAC delay, RSSI downlink (DL) and uplink (UL), handoff, cellular communications

I. INTRODUCTION

With the increasing demand and penetration of wireless services, users of wireless network now expect quality of service and performance comparable to what is available from fixed networks. Media Access Control (MAC) protocol in wireless networks controls and manages the access and packet transmission through the shared channel in a distributed manner, with minimum possible overhead involved [2]. A MAC protocol should provide an efficient use of the available bandwidth while satisfying the Quality of Service (QoS) requirements of both data and real-time applications. Real-time services such as streaming voice and video require a certain quality of service such as low packet loss and low delay to perform well. To provide QoS for such kind of application, service differentiation is must. The IEEE 802.11 standard specifies two access mechanisms, the contention based Distributed Coordinator Function (DCF), and the centralized solution known as the Point Coordination Function (PCF). Presently, however, in most available products, only DCF is implemented. As both the medium access control (MAC) layer and the physical (PHY) layer of 802.11 are designed for best effort data transmissions, the original 802.11 standard does not take QoS into account. Hence to provide QoS support IEEE 802.11 standard group has specified a new IEEE 802.11e standard. This paper is organized as follows: Section II describes the 802.11 DCF and the 802.11e EDCF. In section III we analyses the performance of EDCA in supporting Real-time traffic and compare DCF and EDCF. Finally section IV concludes the paper.

II. MAC PROTOCOLS

The media access control (MAC) data communication protocol sub-layer, also known as the medium access control, is a sub layer of the data link layer specified in the seven-layer OSI model (layer 2). It provides addressing and channel access control mechanisms that make it possible for several terminals or network nodes to communicate within a multiple access network that incorporates a shared medium, e.g. Ethernet. The hardware that implements the MAC is referred to as a medium access controller. The MAC sub-layer acts as an interface between the logical link control (LLC) sub layer and the network's physical layer. The MAC layer emulates a full-duplex logical communication channel in a multi-point network. This channel may provide unicast, multicast or broadcast communication service. Distributed Coordination Function (DCF) is the currently used protocol that comes with an optional Point coordination Function (PCF) Protocol. Enhanced Distributed Coordination Function (EDCF) is the future protocol that promises to provide the QoS. The explanation of these protocols is as follows:

A. *Distributed Coordination Function (DCF)*

DCF is the basic and mandatory MAC mechanism of legacy IEEE 802.11 [11] WLANs. It is based on carrier sense multiple access with collision avoidance (CSMA/CA.). DCF is explained in this section as it is the basis for the Enhanced Distributed Channel Access

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(EDCA), working of DCF which we discuss in this paper. The 802.11 MAC works with a single first-in-first-out (FIFO) transmission queue [1]. The CSMA/CA constitutes a Distributed MAC based on a local assessment of the channel status, i.e. whether the channel is busy or idle. If the channel is busy, the MAC waits until the medium becomes idle, then defers for an extra time interval, called the DCF Inter-frame Space (DIFS). If the channel stays idle during the DIFS deference, the MAC then starts the back-off process by selecting a random back-off counter (or BC).For each slot time interval, during which the medium stays idle, the random BC is decremented. If a certain station does not get access to the medium in the First cycle, it stops its back-off counter, waits for the channel to be idle again for DIFS and starts the counter again. As soon as the counter expires, the station accesses the medium. Hence the deferred stations don't choose a randomized back-off counter again, but continue to count down. Stations that have waited longer have the advantage over stations that have just entered, in that they only have to wait for the remainder of their back-off counter from the previous cycle(s). Each station maintains a contention window (CW), which is used to select the random backoff counter. The BC is determined as a random integer drawn from a uniform distribution over the interval [0, CW].The larger the contention window is the greater is the resolution power of the randomized scheme. It is less likely to choose the same random BC using a large CW .However, under a light load; a small CW ensures shorter access delays .The timing of DCF channel access is illustrated in Fig. 1. An acknowledgement (ACK) frame is sent by the receiver to the sender for every successful reception of a frame. The ACK frame is transmitted after a short IFS (SIFS), which is shorter than the DIFS. As the SIFS is shorter than DIFS, the transmission of ACK frame is protected from other station's contention. The CW size is initially assigned CWmin and if a frame is lost i.e. no ACK frame is received for it, the CW size is doubled, with an upper bound of CWmax and another attempt with backoff is performed. After each successful transmission, the CW value is reset to CWmin. All of the MAC parameters including SIFS, DIFS, Slot Time, CWmin, and CWmax are dependent on the underlying physical layer (PHY) [5]. DIFS is determined by SIFS+2*Slot Time, irrespective of the PHY

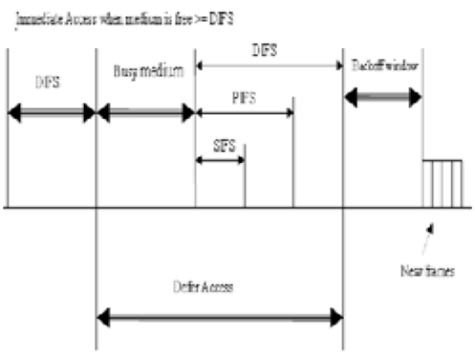

Fig. i: Timing relationship for DCF

A. Enhanced Distributed Coordination Function (EDCF)

EDCF is designed to provide prioritized QoS by enhancing the contention-based DCF. It provides differentiated, distributed access to the wireless medium for QoS stations (QSTAs) using 8 different user priorities (UPs).Before entering the MAC layer, each data packet received from the higher layer is assigned a specific user priority value. How to tag a priority value for each packet is an implementation issue. The EDCA mechanism defines four different first-in first-out (FIFO) queues, called access categories (ACs) that provide support for the delivery of traffic with UPs at the QSTAs. Each data packet from the higher layer along with a specific user priority value should be mapped into a corresponding AC according to table II. Note the relative priority of 0 is placed between 2 and 3.This relative prioritization is rooted from IEEE 802.1d bridge specification [7]. Different kinds of applications (e.g videoconferencing traffic, online traffic and back off traffic) can be directed into different ACs. For each AC, an enhanced variant of the DCF, called an enhanced distributed coordination function (EDCF), contends for TXOPs using a set of EDCF parameters from the EDCF Parameter Set element or from the default values for the parameters when no EDCF Parameter Set element is received from the QAP of the QBSS with which the QSTA is associated.

Table 1: Details of Access Classes

Priority	Access Category (AC)	Designation
1	0	Videoconferencing
2	1	Online
3	2	Backoff
4	3	Voip

Table shows the implementation model with four transmission queues, where each AC behaves like a virtual station: it contends for access to the medium and independently starts its back-off after sensing the medium idle for at least AIFS period. In EDCA a new type of IFS is introduced, the arbitrary IFS (AIFS), in place of DIFS in DCF. Each AIFS is an IFS interval with arbitrary length as follows:

AIFS [AC] = SIFS + AIFSN [AC] × slot time

Where AIFSN [AC] is called the arbitration IFS number and determined by the AC and the physical settings, and the slot time is the duration of a time slot. The timing relationship of EDCA is shown in Fig 3. The AC with the smallest AIFS has the highest priority. The values of AIFS[AC], CWmin[AC], and CWmax[AC], which are referred to as the EDCA parameters, are announced by the AP via beacon frames. The purpose of using different contention parameters for different queues is to give a low-priority class a longer waiting time than a high-priority class, so the high-priority class is likely to access the medium earlier than the low-priority class. An internal collision occurs when more than one AC finishes the back-off at the same time. In such a case, a virtual collision handler in every QSTA allows only the highest-priority AC to transmit frames, and the others perform a back-off with increased CW values.

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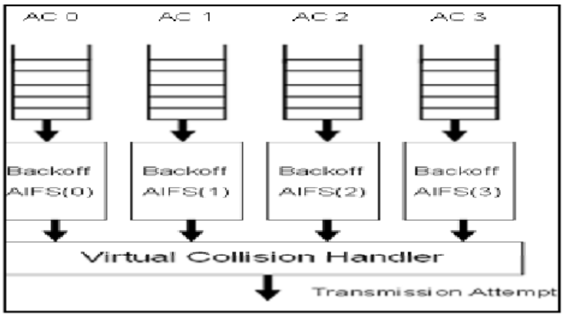

Fig. ii: Implementation model

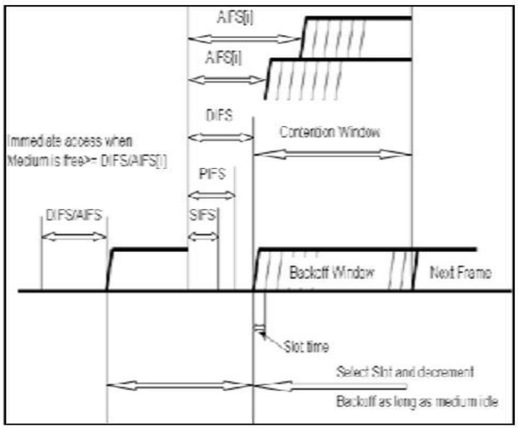

Fig.iii: Timing relationship for EDCF

TXOP-Transmission opportunity is defined in IEEE 802.11e as the interval of time when a particular QSTA has the right to initiate transmissions. There are two modes of EDCA TXOP defined, the initiation of the EDCA TXOP and the multiple frame transmission within an EDCA TXOP. An initiation of the TXOP occurs when the EDCA rules permit access to the medium. A multiple frame transmission within the TXOP occurs when an EDCAF retains the right to access the medium following the completion of a frame exchange sequence, such as on receipt of an ACK frame. The TXOP limit duration values are advertised by the QAP in the EDCA Parameter Set Information Element in Beacon frames. During an EDCA TXOP, a STA is allowed to transmit multiple MAC protocol data units (MPDUs) from the same AC with a SIFS time gap between an ACK and the subsequent frame transmission. A TXOP limit value of 0 indicates that a single MPDU may be transmitted for each TXOP. This is also referred to as contention free burst (CFB).In this paper, we only investigate the situation where a station transmits one data frame per TXOP transmission round.

III. SIMULATION EVALUATION

Due to the increasing demand of high rates WLANS in the applications like video streaming and voice over IP. It's mandatory to improve its performance in every possible way for the ease of the user. Quality of Service (QoS) support is the key for multimedia applications over WLAN. So, since the demand for QoS is nowadays is a topic of great concern for the transmission of the different services like video, voice, best-effort services controlling the maximum network applications, background services for which the priority assigned is lower than the one assigned to the standard. So in order to get the better Qos at our desired application we need to control the network parameters in such a way that we get a certain output and according to the output we will see that at which station we are getting the maximum output to calculate the probability of winning of a station for random values of AIFS and CW using the proposed algorithm, to find out the probability of the collisions in a station for the random values of AIFS and CW using the proposed algorithm ,to analyse the result and check where the output coming better. Then the station at which we are getting the maximum output we will send that data for which we want the best QOS.

The proposed work is about the evaluation of the performance by calculating the probability of winning of a station and the probability of collisions in the station. This is calculated by taking into account the Contention window and the Arbitrary Inter frame Spaces. In the earlier researches the size of the AIFS and the CW is kept constant or taken for the fixed values only.

But in this thesis, I am taking the random values for the Contention Window and the arbitrary interframe spaces and checking the results. I have taken four cases, the station where we are getting the maximum probability of there we will send the priority data which we desire. I have used a general algorithm to calculate the probability of winning of a station and another algorithm for calculating the probability of the collision.

A. *Probability of the winning*

The probability of the winning means the maximum probability of which station to win that means at which station stations there are minimum number of the collision and hence there will be successful transmission of the access category.

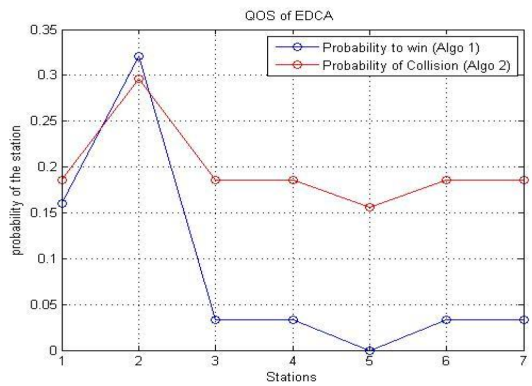

Fig.iv: Probability of the winning and collision of different stations independently with collisions

A. *Probability of the collisions*

The probability could be computed using a simple trick. It is clear that the contest of the stations ends by either winning of a station or a collision.

Thus, utilizing the complementary probability, we can put down

$1 = P^1_{win} + P^2_{win} + : : : + P^k_{win} + P_{coll}$

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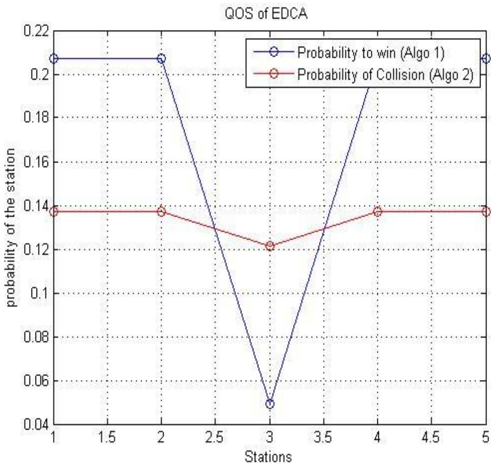

Fig.v: Probability of the winning and collision of different stations independently with large number collisions

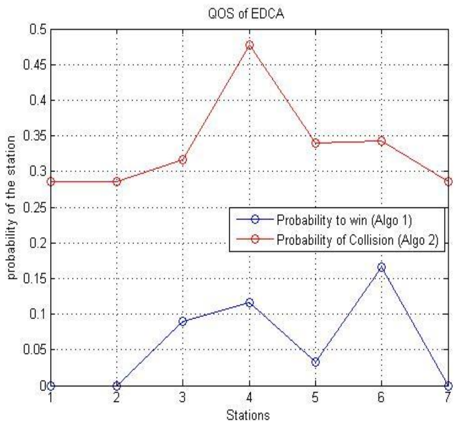

Fig vi: - Probability of the winning and collision of different stations independently with more collisions

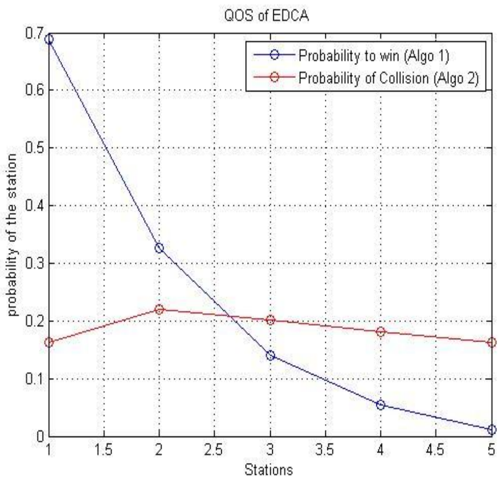

Fig vii. - The probability of winning is maximum

IV. CONCLUSION AND FUTURE WORK

Thus we can conclude from the above results that the optimized results are given in figure vii. But if want to use two stations at a time for the better performance out of five stations then we could use the situation shown in figure viii. In this situation the probability of winning is maximum as compared to all other cases. Hence we can transmit the required parameters from the station 1 and 2 for better results with the situation as is produced in. Otherwise the situation shown in table VII is better as compared to all other cases. So we have concluded by our analysis that if we want to send the data at which we want to obtain the best results we will send the data at that station where we are getting the maximum probability of winning n lesser probability of collision. It will help us to attain the better qos for the desired data i.e if we want to attain the maximum qos for the video and voice we will send the video and voice from that station at which we are getting the maximum output

REFERENCES

[1] M. Barry, A. T. Campell, A. Veres, “Distributed Control Algorithms for Service. Differentiation in Wireless Packet Networks”, Proc. IEEE INFOCOM 2001.

[2] Liljana Gavrilovska, Ramjee Prasad, “Ad hoc networking towards seamless communications”, Published by Springer, 2006.

[3] S. Choi, K. G. Shin, “A cellular local area network with QoS guarantees for heterogeneous traffic”, Proceedings IEEE INFOCOM’97, pp. 1032-1039, Kobe, Japan, April 1997.

[4] IEEE 802.11e/D11.0, Draft Supplement to Part 11, "Wireless Medium Access Control (MAC) and physical layer (PHY) specifications. Medium Access Control (MAC) Enhancements for Quality of Service (QoS)", October2004

[5] IEEE Std. 802.11b, Supplement to Part 11, "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications: Higher-speed Physical Layer Extension in the 2.4 GHz Band", IEEE Std. 802.11b-1999, 1999.

[6] IEEE Std. 802.11-1999, Part 11,"Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY)Specifications", Reference number ISO/IEC 8802- 11,1999(E), IEEE Std. 802.11, 1999 edition, 1999

[7] Jiang Zhu and Abraham O. Fapojuwo, Senior Member, IEEE. “A New Call Admission Control Method for Providing Desired Throughput and Delay Performance in IEEE802.11e Wireless LANs.” IEEE TRANSACTIONS ON WIRELESS COMMUNICATIONS, VOL. 6, NO. 2, FEBRUARY 2007.

[8] Jalel Ben-Othman Laboratoire CNRS-Prism University of Versailles, Lynda Mokdad Laboratories LACL University of Paris 12, Mohamed Ould Cheikh LAMSADE Laboratory University of Paris-dauphine. “Q-HWMP: improving end-to-end QoS for 802.11s based mesh networks. This full text paper was peer reviewed at the direction of IEEE Communications Society subject matter experts for publication in the IEEE Globecom 2010 proceedings.

[9] Pierre Ferr´e, Angela Doufexi, Andrew Nix and David Bull University of Bristol, Centre for Communication Research Woodland Road, Bristol, BS8 1UB, UK. “Throughput Analysis of IEEE 802.11 and IEEE 802.11e MAC.” WCNC 2004 / IEEE Communications Society 0-7803-8344-3/04/$20.00 © 2004 IEEE.