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Optimized WLAN MAC Protocol for Multimedia Applications

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***Abstract*— In this paper we propose an optimized WLAN MAC protocol for multimedia applications. The proposed MAC protocol consist of 3 main optimizing architectures, *traffic segregation*, *specific service interval (SI) for voice and video traffic* and *smaller contention window (CW)*. Traffic segregation segregates real-time traffic from best-effort traffic while specific service interval provides voice and video traffic with the SI based on its mean arrival rate. In the contention time frame, the CW is made smaller to reduce delay and increase throughput. Ns2 is used as a simulating tool to compare the results of our proposed protocol with the legacy 802.11 and the 802.11e protocols. We show that our protocol has significantly better throughput than the legacy 802.11 and the 802.11e. Fluctuations in the throughput is also shown to be lower in our proposed protocol. The technique implemented also provides easier and better management of real-time traffic to support guaranteed quality of service (QoS).**

I. INTRODUCTION

The issue of providing Quality of Service (QoS) for real-time traffic in multimedia applications in wireless local area networks (WLAN) is not fully resolved and is still an active research area. The introduction of more efficient medium access control (MAC) protocols such as the IEEE 802.11e has improved to a certain degree the performance of multimedia applications over legacy 802.11b, but has not achieved the desired optimization of the offered resources. This has been shown by many works published in [1] [2] [3] and [4].

In this paper we introduce a MAC protocol to improve and optimize real-time traffic in multimedia applications. Our protocol involves segregation of real-time and non real-time traffic into contention and contentionless time frames respectively. The duration of the contention period and the contentionless period depends on the amount of traffic types in the network. This segregation technique will provides easier management of the prioritized real-time traffic and at the same time does not eliminate entirely the bandwidth available to non real-time traffic, which was shown in our earlier works in [9].

In the contentionless frame, we introduce a specific service interval (SI) for VoIP traffic and for video traffic. The two dedicated SIs provide a more optimized and efficient medium access mechanism as traffic arrival rate is different for VoIP and video. Higher prioritization of voice traffic over video is maintained in our protocol. The negotiated transmission opportunity (TXOP) is calculated by the scheduler in the QoS access point (QAP) and the sum of all voice and video TXOP must comply with the voice delay threshold determined by the QAP. In the contention frame, we maintain the interframe space as in 802.11e but reduce the contention window (CW) size to reduce delay caused by a large CW. The number of traffic in the contention frame is reduced as only non real-time traffic contend for access in this period due to segregation. This results in fewer collisions and the shorter CW helps to further reduce delays when accessing the medium. A detailed explanation of the proposed protocol is provided in section II.

A. IEEE 802.11e

The IEEE 802.11e standard [5] was designed to enhance the performance of the legacy 802.11 standard [6]. It defines two mechanism, enhanced distributed channel access (EDCA) and hybrid coordination function (HCF) controlled channel access (HCCA). Both are backward compatible with the legacy 802.11 distributed coordination function (DCF) access mechanism. It introduces prioritization to traffic types and provides high priority to real-time traffic such as VoIP and video in order to maintain QoS. This prioritization is achieved through 4 access categories (ACs), AC_VO(for voice traffic), AC_VI(for video traffic), AC_BE(for best-effort traffic) and AC_BK(for back ground traffic). Different ACs have different priorities, serving different types of traffic. AC_VO possesses the highest priority and AC_BK the lowest. Multiple queues are used for the prioritization and separate handling of different traffic categories. Figure 1 shows the 4 access categories in 802.11e and mapping of these access categories to different traffic types is shown in Table I.

The main prioritization mechanisms introduced in the 802.11e EDCA parameters which differ from the legacy 802.11 are:

- Minimal CW value for a given AC ($CW_{\min}[AC]$): High priority traffic is assigned smaller $CW_{\min}$ to ensure it obtains more TXOPs than low priority traffic.
- Maximal CW value for a given AC ($CW_{\max}[AC]$)
- Arbitration Interframe Space $AIFS[AC]$: A period each AC has to wait before it starts its backoff procedure, and this period is different from a DCF interframe space

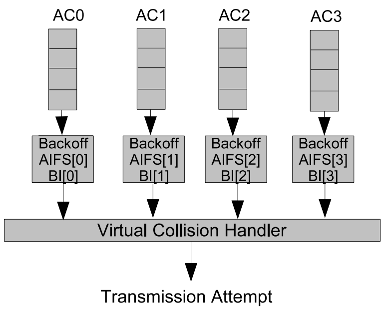

Fig. 1. Access Categories in 802.11e

TABLE I
MAPPING BETWEEN USER PRIORITIES AND ACCESS CATEGORIES

Priority	Access Category (AC)	Traffic Types
Lowest	AC[0]	Background
	AC[0]	Background
	AC[1]	Best-Effort
	AC[1]	Video
	AC[2]	Video
	AC[2]	Video
	AC[3]	Voice
Highest	AC[3]	Voice

(DIFS) used in the legacy 802.11. The $AIFS[AC]$for a given AC should be equal to a short interframe space (SIF) plus multiple time slots (i.e $AIFS[AC] = aSIFSTime + AIFSN[AC] * aSlotTime$). Where $AIFSN[AC] => 2$, such that $AIFS[AC] < DIFS$. In the legacy 802.11, $DIFS = aSIFSTIME + 2 * aSlotTime$. This ensure that the shortest $AIFS[AC] = DIFS$. Figure 2 shows the interframe relationship in the 802.11e EDCA MAC protocol.

- $TXOP_{\text{limit}}[AC]$: $TXOP_{\text{limit}}[AC]$provides a time limit for packets transmission in each AC. A transmission time limit is critical to provide real-time traffic in wireless nodes a deterministic delay. During an EDCA-TXOP, a wireless node may be allowed to transmit multiple data frames from the same AC with a SIFS gap between an acknowledgment (ACK) and the subsequent data frame transmission.
- Virtual Collisions: If the backoff counters of two or more collocated ACs in one station elapse at the same time, a scheduler inside the station treats the event as a virtual collision. The TXOP is given to the AC with the highest priority among the "colliding" ACs, and the other colliding ACs defer and try again later as if the collision occurred in the real-medium.

In the HCF control channel access (HCCA) time frame, access to the wireless medium is managed by means of polling by the QoS access point (QAP). QoS polling can take place during both the contention free period (CFP) and the contention period (CP). The central concept of HCCA is the controlled access phase (CAP), which is a bounded time interval formed by a concatenating series of HCCA TXOPs. Scheduling of the HCCA TXOP and formation of CAP are

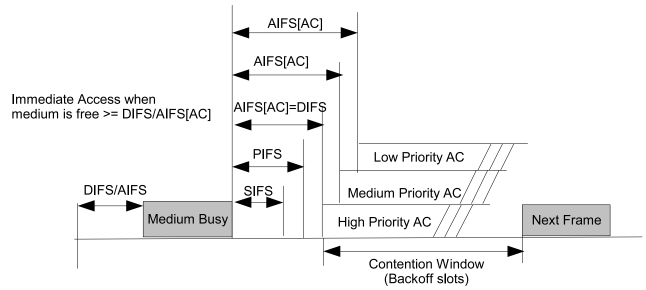

Fig. 2. Inter Frame Space (IFS) relationship used in 802.11e [7]

performed by the hybrid coordinator (HC) that is co-located with the QAP. When the HC needs access to the wireless medium (WM) to start a CFP or a TXOP in CP, the HC senses the WM to determine if it is idle for a PIFS period. After capturing the channel, the HC polls WNs in turn according to its polling list. In order to be included in the polling list of the HC, a WN must send a QoS reservation request using the special QoS management frame, and each individual flow needs one particular reservation request [8]. The QoS management frame contains the traffic specification (TSPEC), which includes the following parameters:

- Mean data rate (ρ) : average bit rate for packet transmission, in bits per second.
- Maximum service Interval (SI_{max}): maximum interval between the start of two successive schedule service periods (SPs)
- Delay bound (D): the maximum amount of time allowed to transport a MAC service data unit (MSDU) belonging to a traffic stream (TS) in the traffic specification (TSPEC).
- Nominal MSDU size (L):nominal size of a packet, in octets
- Minimum PHY rate (R): the minimum physical bit rate assumed by the scheduler for calculating transmission time, in bits per second.

During a TXOP, the wireless medium is accessed by only one QoS station (QSTA), either an AP or a WN. A downlink TXOP consists of a burst of QoS data frames transmitted from the QAP to a WN while an uplink TXOP is initiated when the QAP polls a WN, which takes control of the medium for the TXOP limit specified in the poll message. If the traffic stream (TS) of a polled WN is not backlogged, or if the head-of-line packet does not fit into the remaining TXOP duration, the WN sends a QoS Null frame to the QAP. The basic frame exchange between an AP and a WN in 802.11e HCCA is shown in Figure 3.

The calculation of the scheduled service interval (SI) is carried out in two simple steps as follows: First, the scheduler calculates the minimum of all maximum SIs for all admitted streams. Let this minimum be SI_{min}. Second, the scheduler chooses a number lower than SI_{min} that is a submultiple of the beacon interval. This value is the scheduled SI for all WNs with admitted streams [6]. This is shown in Figure 4. If

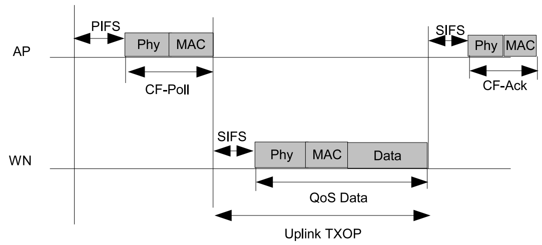

Fig. 3. Basic Frame Exchange in CFP

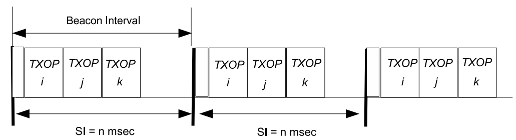

Fig. 4. Service Interval scheduled for TS

a new TS is admitted with a maximum SI smaller than the current SI, the scheduler needs to change the current SI to a smaller number than the maximum SI of the newly admitted stream. Therefore, the TXOP duration for the current admitted streams needs also to be recalculated with the new SI.

II. Proposed Optimization Technique

Our optimization technique comprises an online and an offline procedure. The online procedure consists of three main optimizing architectures : *segregation of traffic, specific SI for voice and video traffic,* and *smaller CW in contention time frame.* as shown in Figure 5. The offline procedure handles the polling list, calculation of service interval and allocation of TSPEC in each WN. Both of these main optimizing architectures are located and implemented in the QAP. Each of the WNs require only the basic scheduler for traffic segregation and TSPEC negotiation request as most of the main optimizing tasks are being handled in the QAP.

First, we introduce a scheduler which segregates real-time traffic (VoIP and video) and best-effort traffic into contentionless and contention time frames respectively. This is to provide better and easier management of different traffic types and provide real-time traffic with deterministic delay and jitter, therefore providing guaranteed QoS. It has been shown in [9] that the segregation technique improves delay and jitter significantly for VoIP and video traffic in multimedia applications and does not eliminate best-effort traffic.

Second, we introduce two different SIs in the contentionless time frame, one for VoIP and the other for video. This will reduce packets dropped at the queuing buffer of each WN, especially for video traffic, as its inter-arrival rate is much higher than the VoIP inter-arrival rate. The scheduler allocates SI_{voice} and SI_{video} as follows:

$$SI_{voice} = t_{vo\_arr} + (t_{phy} + \frac{h_{mac} + L_{voice}}{\rho_{data}}) \tag{1}$$

$$SI_{video} = t_{vi\_arr} + (t_{phy} + \frac{h_{mac} + L_{video}}{\rho_{data}}) \tag{2}$$

where;
t_{vo_arr} - mean arrival rate of voice traffic
t_{vi_arr} - mean arrival rate of video traffic
t_{phy} - time to transmit a preamble and a PLCP header
h_{mac} - length of MAC header
L_{voice} - length of voice header
ρ_{data} - data mean bit rate

In a scenario where overlapping SI occurs between SI_{voice} and SI_{video}, two methods are used to resolve this issue. First, the higher priority traffic is given access and second, if the scheduled SI time of the first TS can provide more than 50% of the total transmitted packet before the second scheduled SI, the first packet is given access (poll) regardless of its priority. This is shown in Figure 6 which shows a scheduled SI for two WNs and a QAP with uplink and downlink packet exchange. Video traffic has typically shorter SI than voice as it has higher arrival rate. The first scheduled video packet of WN_1 overlaps with the scheduled voice packet of WN_2 after more than 50% of the video packet has been transmitted. In this scenario, the video packet is granted access by the QAP. In the second scenario where the scheduled voice packet of WN_1 overlaps with a video packet, access is granted to the voice packet as it has higher priority.

Let us define the time to transmit an SDU that belongs to a TS i, t_{N_i} and the time to transmit a CF-Poll, t_P , including the interframe space as follows:

$$t_{N_i} = SIFS + (t_{phy} + \frac{L_i + h_{mac}}{\rho_{data}}) \tag{3}$$

$$t_P = PIFS + (t_{phy} + \frac{h_{poll}}{\rho_{poll}}) \tag{4}$$

where;
$SIFS$ - the short interframe space
$PIFS$ - the idle time the QAP needs to wait before polling
h_{poll} - length of polling header
ρ_{poll} - mean rate of polling header

The total time of transmission for n number of WNs in a given SI is as in (5). This time must be less than the delay threshold of voice traffic, D_{vo_thr} to maintain guaranteed QoS for voice traffic as in (6).

$$\sum_{i=1}^{n}(((t_{N_i})_{voice} + t_p) + ((t_{N_i})_{video} + t_p)) \tag{5}$$

$$\sum_{i=1}^{n}(((t_{N_i})_{voice} + t_p) + ((t_{N_i})_{video} + t_p) < D_{vo\_thr} \tag{6}$$

The scheduler serves each TS on a periodic basis, where the period for voice service interval is SI_{voice} for all voice TS and video service interval is $SI_{voice}$for all video TS. The duration a WN is allowed to transmit voice and video frames after it acquires the channel $(TXOP)$ are set as (7) and (8) so as to comply with (6).

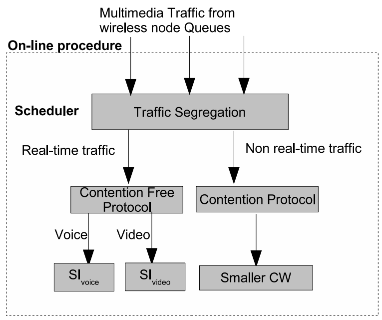

Fig. 5. Main optimizing architectures implemented in QAP

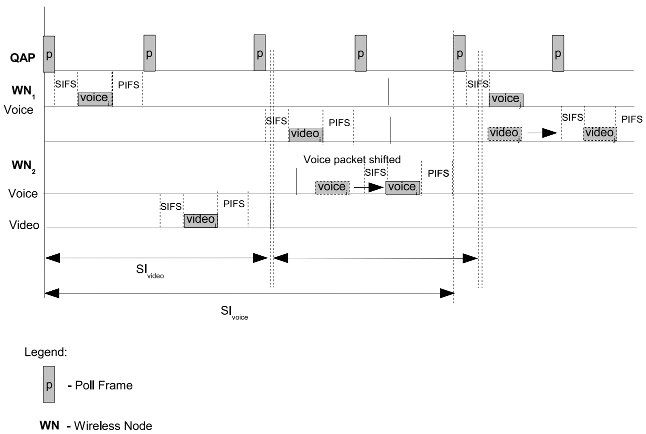

Fig. 6. Frame Exchange of Voice and Video Packets

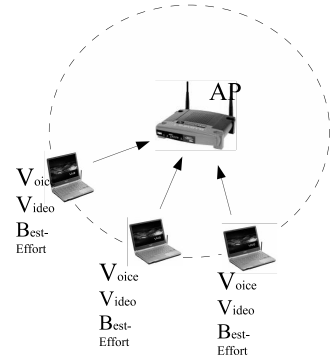

Fig. 7. Simulated environment with WNs carrying multimedia traffic

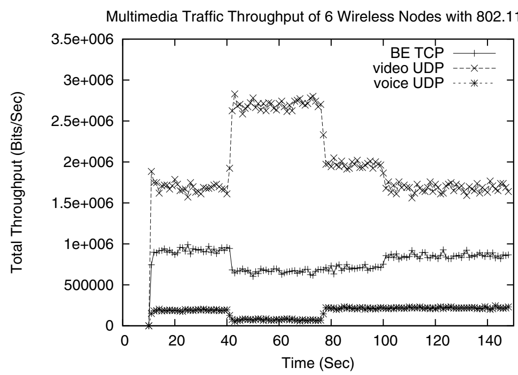

Fig. 8. Total throughput of 6 wireless nodes multimedia traffic simulated with 802.11b

$$TXOP_{voice} = (\frac{\rho}{L_{voice}} \times SI_{voice}) \times t_N + t_p \quad (7)$$

$$TXOP_{video} = (\frac{\rho}{L_{video}} \times SI_{video}) \times t_N + t_p \quad (8)$$

In the contention time frame, we introduced a smaller backoff CW. There will be no competition from high priority traffic as only the best-effort and background traffic contend for access. Each WN contends for the medium with the same rules as the DCF in the legacy 802.11. The interframe space is set to 2 and the CW_{min} and CW_{max} is $\frac{aCW_{min}}{2}$ and $aCWmin$ respectively to reduce delay caused by backoff. As in the 802.11e EDCA, the aCW_{min} are not fixed and depend on the physical layer transmission mode. This is contained in the physical management information base (MIB) attributes tables and assigned by a management entity or by a QAP [5].

III. Simulation Scenarios

Our main objectives for these simulations was to investigate the performance of the legacy 802.11 and 802.11e protocols and compare them with our proposed protocol. We create a worst-case WLAN scenario with all WNs communicating with an AP carrying three types of traffic, VoIP, video streaming and best-effort as shown in Figure 7. In line with the traffic characteristics used in real wireless network environments and digitized with the G.711 coding standard, the inter-arrival time of voice traffic is made 20 msec with a packet size of 160 bytes [10]. For video traffic the inter-arrival time is 16 msec with a packet size of 1280 bytes. The best effort inter-arrival time is set to 1.5 msec and has a packet size of 500 bytes. The voice and video traffic use a constant bit rate user datagram protocol (UDP) while the best-effort traffic use a transmission control protocol (TCP). Transmission rate is set to 11 Mbps for all the simulated protocols. We use ns2 [11] [12] as our simulation tool and tailor it to our needs for simulations of our MAC protocol. Simulations of the legacy 802.11 and the 802.11e use the parameters as in [5] and [6] .

IV. Simulation results

Our simulation results show that our proposed protocol improves significantly the performance of voice and video traffic in multimedia applications as compared to the legacy 802.11b and 802.11e, when simulated in a worst case scenario where all 3 types of traffic run simultaneously in each node. In Figures 8, 9 and 10, we display results obtained from simulation of 6 nodes (5 WNs and 1 AP) carrying multimedia traffic (Because of space limitation results with different numbers of nodes are not able to be displayed). Results obtained from 6 nodes are chosen as it showed the threshold level video traffic supported in 802.11e and clearly showed the improvement made in our proposed protocol. of In Figure 8, the total throughput of voice and video traffic fluctuates significantly. This fluctuations of throughput reflects the poor QoS provided by the access

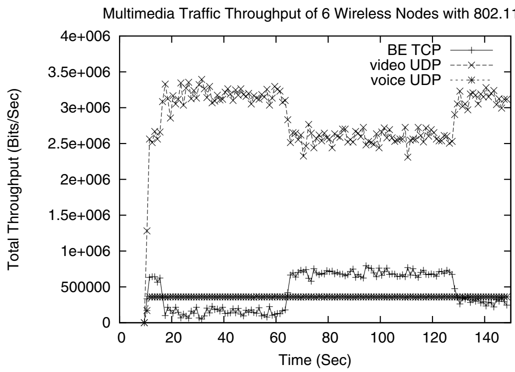

Fig. 9. Total throughput of 6 wireless nodes sending multimedia traffic simulated with 802.11e

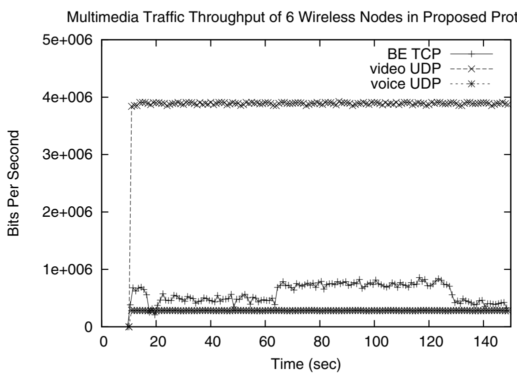

Fig. 10. Total throughput of 6 wireless nodes sending multimedia traffic simulated in our proposed protocol

mechanism in the legacy 802.11b. As the 802.11b has no differentiation and prioritization of traffic, all traffic is treated as equal priority. The best-effort traffic using TCP protocol competes to access the channel at the same priority level as real-time traffic and thus voice traffic suffers as a result. For the 802.11e protocol, throughput performance of voice and video traffic is better than 802.11b. Voice traffic total throughput is maintained throughout the simulation and with 6 nodes, the total throughput is about 380 Kbits/sec. This assured that QoS in all voice traffic is maintained. Video traffic, being lower priority than voice traffic, suffers fluctuations in throughput with the 802.11e protocol as shown in Figure 9. This shows that the 802.11e protocol, while providing QoS to voice traffic, fails to provide QoS to lower priority traffic with high interval rate and large packet size as in video. In our proposed protocol, we show that the QoS for both voice and video traffic is maintained. The total throughput of voice traffic is maintained at 380 Kbits/sec while the total throughput of video traffic is at 4 Mbits/sec, with no major fluctuations. The number of traffic concurrent multimedia streams that can be supported with QoS in our simulated environment can be summarized as in table II.

TABLE II

NUMBER OF TRAFFIC SUPPORTED WITH GUARANTEED QOS

	802.11	802.11e	Proposed MAC
Voice traffic	3	8	9
Video traffic	2	4	6

V. CONCLUSION

This work has investigated the performance of the legacy 802.11 and 802.11e protocol and has proposed a MAC protocol to optimize transmission of multimedia traffic in WLANs. It has been shown that the legacy 802.11 protocol fails to provide QoS to real-time multimedia traffic and that 802.11e provides QoS to voice traffic but fails to provide QoS to video traffic with more than 5 WNs and an AP. By introducing segregation of traffic and two different SIs for voice and video traffic, the MAC protocol can be optimized and performance of real-time traffic improved in multimedia applications. The proposed MAC protocol also provides easy implementation of the scheduling mechanism, whereby scheduling is implemented after the traffic has been mapped to its access categories. Other advantages of the proposed protocol are that it is easier to implement admission control and bandwidth reservation to provide guaranteed QoS to WNs already associating with the AP. The flow of real-time traffic in the proposed protocol is more deterministic and therefore can be easily monitored and managed. Although improvement of throughput performance and optimization have been achieved in this paper, further work is necessary. Extending the work on the proposed MAC protocol, our future works will include implementation of admission control, a bandwidth reservation mechanism, and efficient management of the MAC protocol in the network.

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