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Adaptive Segregation-Based MAC Protocol for Real-Time Multimedia Traffic in WLANs

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***Abstract*— Wireless local area networks (WLANs) have become very popular both in private and public sectors. Despite the fast expansion of WLANs in various environments, quality of service (QoS) issues for multimedia applications in WLANs are not yet resolved. Multimedia applications contain traffic that are sensitive to delay and jitter and therefore a best-effort protocol such as the legacy IEEE 802.11 is not suitable. The 802.11e protocol provides prioritization and classification of traffic to offer better QoS for real-time services. However, it leaves the design and implementation of many important optimization features to vendors. In this paper we introduce a mechanism to improve the delay and jitter of real-time traffic in WLAN nodes supporting multimedia applications. In our proposed mechanism, we segregate voice and video traffic from the best-effort traffic. We create a scheduler that schedules the access of real-time traffic and non real-time traffic to the medium with centralized polling and distributed contention respectively. We show that our proposed protocol performs better in terms of delay and jitter than the legacy 802.11 and 802.11e in a scenario where all wireless nodes carry multimedia traffic simultaneously.**

I. INTRODUCTION

The advancement of wireless communication technology has dramatically changed the way we communicate. Through it communications can be made anywhere and anytime, even on the move. This opens an array of exciting opportunities in business, residential, health-care, education, leisure, and many other sectors. In tandem with this advancement, interest and demand for WLAN multimedia applications and advance capabilities are growing rapidly, spurred by new devices and by the desire of WLAN users to extend the functionality of their networks. Voice over Internet Protocol (VoIP), video streaming, music streaming, and interactive gaming are among the most anticipated applications both in the residential and enterprise markets [1]. The biggest challenge to the industry is to support bandwidth-hungry multimedia applications over the wireless media with quality of service (QoS) support [2]. Multimedia applications contain different types of traffic and each type of traffic requires different level of QoS parameters. Non-prioritized, best-effort access effectively deals with contention from data applications such as file transfer, email, and Internet and intranet access while applications such as voice, video, and music streaming, and interactive gaming generate data streams that have strict latency and throughput requirements. To ensure a good user experience, traffic from different applications has to be managed and prioritized using QoS benchmark.

The legacy 802.11 standard has been very popular and widely used but it lacks QoS support for real-time traffic. The new WLAN standard known as 802.11e is designed to enhance the QoS of the legacy 802.11 and introduces priorities of traffic types to overcome QoS issues for real-time traffic. Although IEEE 802.11e supports some degree of QoS, research is still being carried out to further improve this standard. Analytical and simulation studies have been carried out in [3], [4], [5], [6] to evaluate the performance of the legacy 802.11 and 802.11e in implementing QoS for real-time traffic, but none has evaluated the delay and jitter performance of all nodes carrying multimedia traffic simultaneously. Two important QoS measurement parameters for real-time traffic are delay and jitter. To provide QoS to real-time traffic, packets must be sent below latency and jitter threshold requirements of each traffic type. High latency for real-time traffic such as multimedia will deteriorate the QoS performance.

In this paper we investigate the delay and jitter in WLANs for multimedia traffic with legacy 802.11a/b and 802.11e, and propose an adaptive segregation technique to improve the jitter and delay performance for multimedia traffic. We compare the results of our technique and show that it can improve the delay and jitter for multimedia traffic in WLANs.

A. The Legacy 802.11

The legacy 802.11 standard operates in two modes, contention free period (CFP) mode and contention period (CP) mode. The contention free period is known as the point coordination function (PCF) while the contention period is known as the distributed coordination function (DCF). DCF uses a carrier sense multiple access/collision avoidance (CSMA/CA) medium access control (MAC) protocol. Before transmitting, each node senses if the medium is idle for a period called DCF inter-frame space (DIFS). If the medium is idle for at least a DIFS, the station is allowed to transmit. If the medium is busy, the node then enters a back-off procedure where a slotted back-off time is generated randomly from a contention window (CW) size as:

$$back\_off\_time = rand[0; CW] \times slot\_time \quad (1)$$

Initially the *CW* is set to a minimum value, CW_{min}. It is doubled after each unsuccessful transmission attempt until it reaches a maximum value CW_{max}. If transmission is successful it is reset to CW_{min}. The back-off time is decremented by one slot when the medium is sensed idle for a DIFS. It is frozen if the medium becomes busy, and resumes after the medium has been sensed idle again for another period of DIFS [7]. Collision of packets occurs if the *CW* back-off time of two or more nodes reach zero at the same time. A positive acknowledgment is used to notify the sender that the frame has been successfully received. This is done after a short inter-frame space (SIF) time after receiving data. If an acknowledgment is not received within a time period of ACKTimeout, the sender assumes that there is a collision and schedules a retransmission by entering the back-off process again until the maximum retransmission limit is reached. Legacy 802.11 also provides a mechanism to handle hidden node problems with a four-way hand shake scheme known as request to send and clear to send (RTS/CTS).

PCF is a centralized polling scheme and was designed to support real-time traffic. It uses an access point (AP) as point coordinator to manage polling to the wireless nodes. With PCF enabled, the channel access time is divided into periodic intervals called beacon intervals. A beacon interval is composed of a CP and CFP. In PCF, an AP maintains a list of registered nodes and polls them according to the list. Nodes can only transmit when being polled and the size of each data packet is bounded by the maximum MAC packet size of 2304 bytes. One major problem faced by PCF is the link adaptation ability of the physical layer, which supports multirate and makes the transmission time of a packet variable. Since the legacy 802.11 has been comprehensively explained in the literature, we omit the details of it.

B. The 802.11e

The 802.11e standard defines a superset of features specified in the 1999 edition of the legacy IEEE 802.11 MAC protocol [8]. It introduces two main functional blocks, the channel access period (CAP) and traffic specification (TSPEC) management. Managing these two main blocks is the new coordination function called hybrid coordination function (HCF). HCF has two modes of operation, enhanced distributed channel access (EDCA) and HCF controlled channel access (HCCA). EDCA is a contention-based channel access function and operates concurrently with HCCA that is based on a centralized polling mechanism. The polling mechanism is controlled by the hybrid coordinator (HC) that is co-located with the quality of service access point (QAP). The HC performs bandwidth management including the allocation of transmission opportunity (TXOP) to QoS stations (QSTAs). A TXOP is a bounded time interval in which the QSTA is allowed to transmit a series of frames. It is defined by a start time and a maximum duration. If TXOP is obtained using the contention-based channel access, it is called an EDCA-TXOP. If a TXOP is granted through HCCA, it is called a HCCA-TXOP or a polled TXOP [8]. The duration of the EDCA-TXOP is distributed to non-AP QSTAs in the

TABLE I
MAPPING USER PRIORITIES TO ACCESS CATEGORIES

Priority	UP(User Priorities)	AC(Access Categories)	Designation
Lowest	1	AC_BK(0)	Background
	2	AC_BK(0)	Background
	0	AC_BE(0)	Best Effort
	3	AC_BE(1)	Best Effort
	4	AC_BE(2)	Video
	5	AC_VI(2)	Video
	6	AC_VO(3)	Voice
Highest	7	AC_VO(3)	Voice

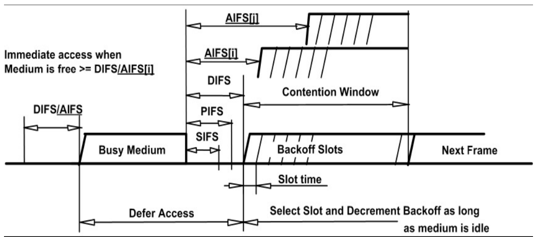

Fig. 1. Inter-Frame Space Relationship [8]

beacon frames along with other EDCA related parameters. The duration of a HCCA-TXOP is passed to the non-AP QSTAs directly by the HC as part of a QoS CF-Poll frame, which grants the HCCA-TXOP.

The EDCA mechanism provides differentiated, distributed access to the wireless medium (WM) for QSTAs using eight different user priorities (UPs). The EDCA mechanism defines four access categories (ACs) that provide support for the delivery of traffic with UPs at the QSTAs. These access categories are AC_VO (for voice traffic), AC_VI (for video traffic), AC_BE (for best-effort traffic), and AC_BK (for background traffic). AC_VO possesses the highest priority, while AC_BK has the lowest priority. Each AC has its own queue and parameter set. The EDCF parameter set includes *Minimum Contention Window Size* (CW_{min}), *Maximum Contention Window size* (CW_{max}), *Arbitration Inter-Frame Space (AIFS),* and *Transmission Opportunity limit* (*TXOPlimit*). CW is set as CW_{min} at the very beginning. A successful transmission will reset CW to CW_{min}. Instead of a DIFS, a station needs to defer for *AIFS*. The ACs is derived from the user priorities (UPs) as shown in Table I. The differentiation in priority between the AC is realized by setting different values for the AC parameters, which are arbitrary inter-frame space number (AIFNS), contention window size and transmission opportunity (TXOP) limit. Figure 1 shows the inter-frame space relationship used in DCF and EDCF.

HCCA is another component of HCF and similar to PCF, it uses polling access to the wireless medium. However, unlike PCF, QoS polling can take place during CP and scheduling of packets is based on admitted TSPECs. The central concept of HCCA is controlled access phase (CAP), that is a bounded time interval and formed by concatenating a series of HCCA TXOPs. Scheduling of HCCA TXOP and formation of CAP are performed by the HC. When the HC needs access to the

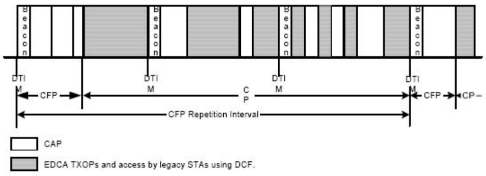

Fig. 2. Relationship between CAP/CFP/CP

wireless medium (WM) to start a CFP or a TXOP in CP, the HC shall sense the WM to determine it to be idle for a PIFS period. The HC then transmits the first frame of any permitted frame exchange sequence, with the duration value set to cover the CFP or the TXOP. The first permitted frame in a CFP after a target beacon transmission time (TBTT) is the beacon frame [8]. Figure 2 shows an example of the frame sequence exchange during the CAP.

C. Proposed Adaptive Segregation of Real-Time Traffic

In this work we segregate real-time multimedia traffic from non-real-time traffic and show that significant improvement is made on delay and jitter of real-time multimedia traffic. As in both the legacy 802.11 and 802.11e, we divide the operating mode into contention free period (CFP) and contention period (CP). CFP is used mainly for real-time traffic while CP is used by best-effort traffic. This will provide real-time traffic with a more deterministic delay, jitter and throughput while reducing the number of contending traffic in the CP. The real-time traffic is allowed to contend for access in two scenarios, first if the access point (AP) does not poll for a duration of a DIFS period, and second, if the real-time traffic destination node is not the AP and is in the same basic service set (BSS). This will make the MAC protocol more flexible and robust while maintaining compatibility with 802.11e and reduce overhead which is relatively high in centralized mode. A scheduler in the AP will notify the nodes that are allowed to contend based on the destination address in its polling list. The allocation of the contention free period and the contention period is made dynamically and depends on the traffic load of real-time traffic. We introduce a scheduler within the access point (AP) that calculates the transmission opportunity (TXOP) for each node and the allocation of contention free period (CFP) and contention period (CP) for each cycle of its superframe. The TXOP is calculated based on the number of MAC service data unit (MSDUs) in the current queue of each QSTA. First the scheduler allocates the duration of the maximum CFP and CP equal to CFP_{max} and CP_{max} respectively. This will provide a maximum of $CFP_{max} + CP_{max}$ per cycle of CFP and CP that will cap the access delay to below the total duration of CFP_{max} and CP_{max}. Next, the scheduler calculates the minimum of all the maximum service interval (SI) for all admitted streams depending on the class of the admitted traffic streams. Based on the SI, mean data rate (ρ_i), and the physical data rate (R_i), the scheduler calculates the maximum TXOP of each node as:

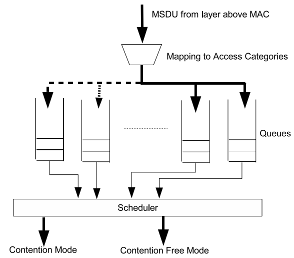

Fig. 3. Proposed MAC Architecture

$$TXOP_i = max(\frac{SI \times \rho_i}{R_i}) \tag{2}$$

The duration of CFP is self adjusting and depends on the load of the real-time traffic. As the load of traffic increases, the duration of CFP increases until it reaches the maximum value of 90 msec. Any new nodes requesting to associate will be turned down. All successfully received packets are not acknowledged as no retransmission is provided. This is because retransmission of real-time packets introduces delay to the next consecutive packet. The proposed basic MAC structure is shown in Figure 3. When the MSDUs arrive from the layer above MAC, it is placed in the appropriate queues according to its mapping categories. The scheduler monitors the packets in the queues. In the non-AP quality of service station (non-AP QSTA), the scheduler simply schedules the packets according to the traffic class and schedules it appropriately either to the contention mode or the contention free mode MAC protocol. In the access point quality of service station (QAP), the scheduler calculates the service interval, TXOP, of each traffic stream based on the negotiated transmission specifications and the duration of CP and CFP.

1) Analytical analysis: Markov chain model shown by Bianchi [9] and Jiang Zhu and Abraham Fapojuwo [10] on DCF and EDCF has shown that throughput performance in contention mode is related to the number of nodes, the maximum backoff time and the contention window size. It has been shown that the probability τ that a station transmits in a randomly chosen slot time can be expressed as:

$$\tau = \sum_{i=0}^{m} b_{i,0} = \frac{b_{0,0}}{1-p} = \frac{2(1-2p)}{(1-2p)(W+1) + pW(1-(2p)^m)} \tag{3}$$

where:

τ= Probability that a station transmit in a randomly chosen time;

i = Backoff stage from 0 to m;

m = Maximum backoff stage;

$b_{j,0}$ = Transitions form one state j to state 0;

p = Collision probability;

W= Contention window size.

At steady state, each remaining station transmits a packet with probability τ. This yields the conditional collision probability as in (4);

$$p = 1 - (1-\tau)^{n-1} \quad (4)$$

The throughput S is calculated based on the ratio of the average amount of payload information successfully transmitted and the length of a slot time:

$$S = \frac{E[payload\,information\,transmitted\,in\,a\,slot\,time]}{E[slot\,time\,length]} \quad (5)$$

The average amount of payload information successfully transmitted in a slot time is equal to $P_{tr} \times P_s \times E[P]$, where P_{tr} is the probability that there is at least one transmission in the considered slot-time, P_s is the probability that the transmission occurring on the channel is successful and $E[P]$ is the average packet payload size. P_{tr} and P_s are expressed as:

$$P_{tr} = 1 - (1-\tau)^n \quad (6)$$

$$P_s = \frac{n\tau(1-\tau)^{n-1}}{P_{tr}} = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^n} \quad (7)$$

where n is the number of nodes.

The average length of a slot time, $E[slot\,time\,length]$, is obtained considering that, the slot is empty with probability $1 - P_{tr}$; it contains a successful transmission with probability $P_{tr}P_s$; and it contains a collision with probability $P_{tr}(1-P_s)$. Hence, (5) becomes

$$S = \frac{P_s P_{tr} E[P]}{(1-P_{tr})\sigma + P_{tr}P_sT_s + P_{tr}(1-P_s)T_c} \quad (8)$$

where:

T_s= The average time the channel is sensed busy (i.e., the slot time lasts) because of a successful transmission;

T_c= The average time the channel is sensed busy by each station during a collision;

σ= The duration of an empty slot time.

Rearranging (8) we obtain;

$$S = \frac{E[P]}{T_s - T_c + \frac{\sigma(1-P_{tr})/P_{tr}+T_c}{P_s}} \quad (9)$$

As T_s,T_c, $E[P]$ and σ are constants, the throughput S is maximized when the following quantity is maximized:

$$\frac{P_s}{(1-P_{tr})/P_{tr} + T_c/\sigma} = \frac{n\tau(1-\tau)^{n-1}}{T_c^* - (1-\tau)^n(T_c^*-1)} \quad (10)$$

where $T_c^* = T_c/\sigma$. Taking the derivative of (10) with respect to τ, and setting it equal to 0, the following equation is obtained:

$$(1-\tau)^n - T_c^*\{n\tau - [1-(1-\tau)^n]\} = 0 \quad (11)$$

Under the condition $\tau << 1$:

$$(1-\tau)^n \approx 1 - n\tau + \frac{n(n-1)}{2}\tau^2$$

TABLE II

PARAMETERS OF DCF USED IN SIMULATIONS

Parameter	SIFS	DIFS	Slot time	CW_{min}	CW_{max}
802.11b PHY (sec)	10μ	15μ	20μ	31μ	1023μ

TABLE III

PARAMETERS OF EDCF USED IN SIMULATIONS

Traffic	Transport Protocol	CW_{min}	CW_{max}	AIFSN
Voice	UDP	3	7	30μ
Video	UDP	7	15	30μ
Best Effort	UDP	15	1023	50μ

This yields the following approximation solution:

$$\tau = \frac{\sqrt{[n+2(n-1)(T_c^*-1)]/n} - 1}{(n-1)(T_c^*-1)} \approx \frac{1}{n\sqrt{T_c^*/2}} \quad (12)$$

From the analytical analysis, first we can deduce that the probability of collision, p increases as the number of nodes, n increases as shown in (4). Second, at saturation the maximum throughput is independent on the number of nodes. Introducing traffic segregation as in our proposed protocol, we reduce the number of traffic streams contending for access in the CP and therefore reduce the rate of collisions. Saturation point is reached with more number of nodes as only best-effort traffic contend for access. Therefore more nodes with best effort traffic can access the medium before total throughput decreases.

II. SIMULATION SCENARIO

Our main objectives for the simulation are to investigate the performance of the legacy 802.11 and 802.11e and compare them with our proposed segregation technique in providing parametrized QoS. We create a worst case scenario with all wireless nodes transmitting voice, video and best-effort traffic simultaneously to a base station, which we refer to as the access point (AP). 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 [11]. For video traffic the inter-arrival time is 16 msec with a packet size of 1280 bytes. The best effort service inter-arrival time is 1.5 msec and has a packet size of 500 bytes. With these parameters the data rate for voice, video and best effort are 64 kbps, 640 kbps and 1.07 Mbps, respectively. Our simulations use the MAC protocol parameters as in the IEEE standards [7] [8] as shown in Table II and Table III.

The transmission rate is set at 11 Mbps for all the simulated protocols. The 11 Mbps is chosen because all protocols support this transmission rate. The packets access delay, jitter and throughput are measured to provide a snapshot of QoS performance. Initially we start with a single node transmitting to an AP. The traffic load is incremented by increasing the number of nodes after each simulation. We use ns2 [12] [13] as our simulation tool and tailor it to our needs for traffic segregation in the MAC protocol. We let our simulations run for 100 sec and for a maximum of 22 nodes as delays observed beyond 22 nodes for multimedia traffic are too high and not relevant for this study.

III. Simulation Results

Our results show that with real-time multimedia traffic running simultaneously in each node, our proposed segregation technique performs better compared to the legacy 802.11 and 802.11e. Access delay experienced by voice traffic shows that with more than 10 nodes and with each node carrying multimedia traffic, the proposed segregation protocol performs 2-6% better in terms of delay and jitter. It shows that for up to 20 nodes, voice traffic access delay is below 150 msec. This conforms to the International Telecommunication Union (ITU-T) recommendation [14] that one-way delay should be kept lower than 150 ms for acceptable conversation quality. For less than 10 nodes, our proposed protocol performs equally well as in the 802.11e, keeping access delay of voice traffic below 100 msec. For the legacy 802.11, access delay of voice traffic reaches more than 150 msec with only 7 wireless nodes carrying multimedia traffic which is shown in Figure 4. It can only support not more than 5 voice traffic in our simulated multimedia scenario. For video traffic, it is also shown in our investigation that the proposed protocol performs slightly better than 802.11e. Access delay for the proposed protocol is lower than both 802.11e and legacy 802.11, as shown in Figure 5. The legacy 802.11 fails to carry video traffic with QoS support for multimedia applications with more than 6 nodes. For best effort traffic in multimedia applications, all three protocols show similar results with less than 5 nodes. As the number of node increases, the proposed protocol shows slightly better delay performance, but as number of node increases further beyond 10, delay in both the proposed protocol and the 802.11e is higher than the legacy 802.11. For more than 15 nodes, delay experienced by the best-effort traffic in all the three protocols is too high at 2 sec. This is shown in Figure 6. Another important parameter that we use for comparison is the jitter experienced by each type of traffic. Figure 7 shows that voice jitter in the proposed protocol for 15 nodes does not exceed 135 msec, while in 802.11e and legacy 802.11, jitter can reach 150 msec and 3.0 sec respectively as shown in Figure 8 and Figure 9. For video traffic, jitter in the proposed protocol for 15 nodes is below 1.0 sec and for 802.11e it is below 1.2 sec with some bursts of jitter reaching 1.6 sec. This is shown in Figure 10 and Figure 11. The total throughput of the proposed protocol is shown to be comparable with 802.11e as shown in Figure 12. Converting these results to the number of multimedia nodes that can be supported by WLANs, it is clear that our proposed protocol can provide QoS support to more wireless nodes carrying multimedia traffic than the legacy 802.11 and 802.11e.

IV. Conclusion

This work has investigated the performance of the legacy 802.11, 802.11e and has proposed an adaptive segregation MAC protocol for multimedia traffic in WLANs. It has been

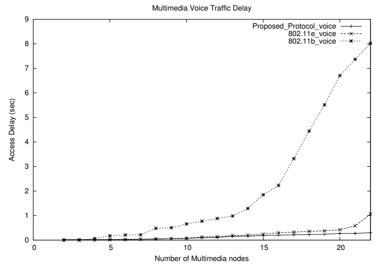

Fig. 4. Access Delay of Multimedia Voice Traffic

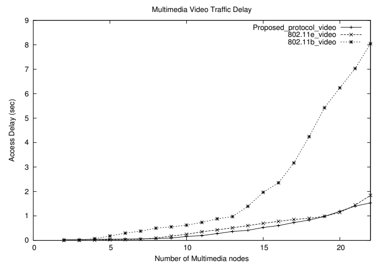

Fig. 5. Access Delay of Video Traffic

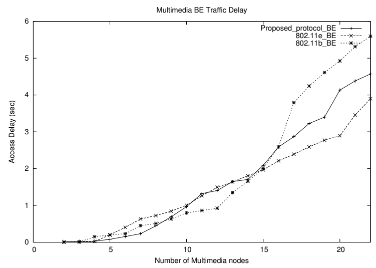

Fig. 6. Access delay of Multimedia Best Effort Traffic

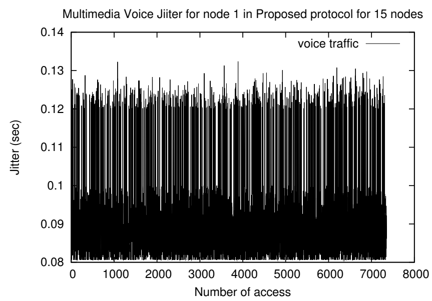

Fig. 7. Voice Jitter of node 1 for 15 nodes in the Proposed Protocol

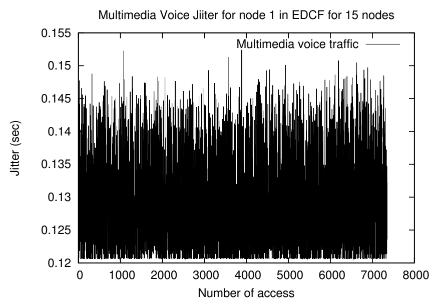

Fig. 8. Voice Jitter of node 1 for 15 nodes in the 802.11e

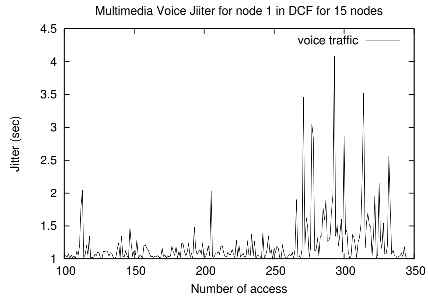

Fig. 9. Voice Jitter of node 1 for 15 nodes in the legacy 802.11

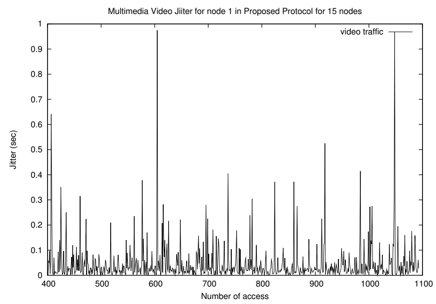

Fig. 10. Video Jitter of node 1 for 15 nodes in Proposed Protocol

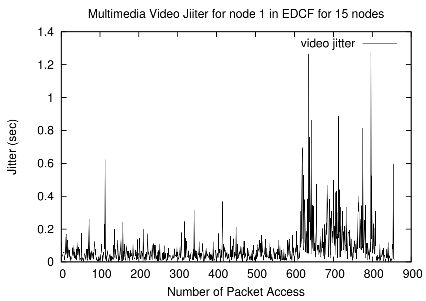

Fig. 11. Video Jitter of node 1 for 15 nodes in 802.11e

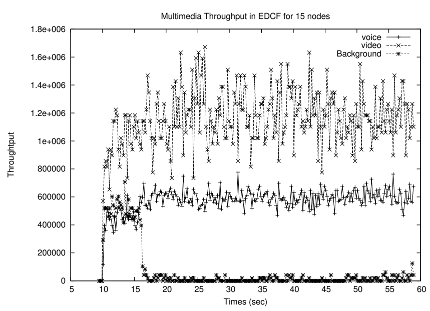

Fig. 12. Total Throughput for 15 nodes of voice, video and best-effort traffic in EDCF

shown that the 802.11e MAC protocol provides a certain degree of QoS in the worst case scenario where all wireless nodes carry multimedia traffic simultaneously. By segregating real-time traffic as in the proposed protocol, the performance of real-time multimedia traffic is improved. Our proposed protocol outperforms the legacy 802.11 and 802.11e in terms of access delay and jitter. The proposed protocol also provides easy implementation of the scheduling mechanism, where 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 node admission control and reservation of bandwidth at the access point to provide guaranteed QoS. The flow of real-time traffic in the proposed protocol is more deterministic and therefore can be easily monitored and managed. Although some improvement on the delay and jitter of the 802.11e MAC protocol has been achieved in this paper, further work is necessary to provide guaranteed QoS for multimedia traffic. Extending the work on the proposed MAC protocol, our future work will include implementation of admission control, bandwidth reservation mechanism and efficient management of the MAC protocol in the network.

REFERENCES

[1] Wi-Fi Alliance, “Wi-Fi Certified for WMM - Support for Multimedia Application with Q0S in Wi-Fi Networks”, Wi-Fi Networks, September 1, 2004.

[2] Aura Ganz, Zvi Ganz, and Kitti Wonthavarawat “Multimedia wireless networks”, Prentice Hall, 2004.

[3] Bianchi, “Understanding 802.11e contention-based prioritization mechanism and their coexistence with legacy 802.11 stations”, IEEE Network, pp. 28-34, 2005.

[4] Deng and Yen, “QoS provisioning system for multimedia transmission in IEEE 802.11 WLAN,” IEEE J. Sel. Areas Comm., vol.23, No.6, pp.1240-1252, 2005.

[5] S.Mangold, S.Choi, P. May, O.Klein, G. Hiertz and L.Stibor, “IEEE 802.11e wireless LAN for quality of service,” IEEE Wireless Communication, December 2003.

[6] Q. Ni, Romdhani, Turletti, and Aad, “Journal of Wireless Communication and Mobile Computing, vol.4, no. 5, pp. 547-566, 2004.

[7] IEEE, “Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications,” 1999.

[8] IEEE, ”Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications, Amendment 8: Medium Access Control (MAC) Quality of Service Enhancements”, 11 November 2005.

[9] Giuseppe Bianchi, “Performance Analysis of the IEEE 802.11 Distribute Coordination Function”, IEEE Jour. Sel. Areas in Comm., vol.18. NO.13, March 2000.

[10] Jiang Zhu and Abraham O. Fapojuwo, “A New Call Admission Control Method for Providing Desired Throughput and Delay Performance in IEEE 802.11e Wireless LANs”, IEEE Trans. on Wireless Communications, Vol. 6, No 2, February 2007.

[11] International Engineering Consortium, “WLAN network capacity analysis,” www.iec.org, 2006.

[12] K. Fall and K. Varadhan, “The ns manual,” 2005.

[13] Cincconetti, Lenzini, Mingozzi and Stea, “A software architecture for simulating IEEE 802.11e HCCA,” Department of Engineering and Information, University of Pisa, Italy.

[14] Telecommunication Standardization Sector of ITU “Series G: Transmission System and Media, Digital System and Networks,” International Telecommunication Union (ITU-T), Recommendation G.114, Mei 2003.