Utilizing IEEE 802.11n to Enhance QoS support in Wireless Mesh Networks

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***Abstract*—Wireless mesh networks (WMNs) have the potential in supporting multimedia applications with last-mile Internet access. To achieve this objective, shortcomings such as scarcity of the wireless link capacity and the lack of robust QoS scheduling must be overcome. In this paper, we consider WMNs utilizing the new IEEE 802.11n standard. Based on the standard's physical and medium access control (MAC) layer enhancements, we propose adapting the modulation and code scheme (MCS) index and aggregation frame length according to the online assessed link quality, performing frame aggregation by packing multiple small subframes. We also propose performing QoS bandwidth provisioning by optimally aggregating subframes according to their QoS constraints and fairness. Performance results show that incorporating link adaptation, frame aggregation, and QoS bandwidth provisioning can considerably improve the system performance in terms of MAC delay, achieved throughput and packet dropping ratio. The results also show that packet aggregation scheme has a crucial impact on system performance when aggregating small packet sizes.**

I. Introduction

Wireless Mesh Networks (WMNs) are currently emerging as a promising paradigm for broadband ubiquitous Internet access. Among other technologies, legacy IEEE 802.11 has been used in implementing WMNs. However, the IEEE 802.11 Working Group is currently working on a new standard, namely IEEE 802.11s [1], with the hope of realizing WMNs with reliable and inexpensive wireless services. Supporting a quality of service (QoS) to enable multimedia services is foreseen to be vital for the success of next generation WMNs. However, WMN's distributed nature imposes many challenges which require a tight control of systems available resources and a need for robust QoS scheduling.

In the related literature, there exists several attempts aiming at increasing the system capacity by utilizing network resources. The work in [2], [3], [4], and [5] propose increasing the system capacity through using multiple radios per node, where the radios assignments are based on schemes such as centralized, static, and dynamic channel assignments. Related works on voice over Internet Protocol (VoIP) over wireless local area networks (WLANs) [6], [7] and ultra-wideband (UWB) networks [8] proposed the introduction of a packet aggregation scheme. The technique trades off service time for packet length where the increase of medium access control (MAC) service time is mitigated by assembling multiple upper layer packets into a single MAC burst. In [9] the authors propose several performance optimizations aimed at improving the VoIP support in WMNs where header compression is exploited to improve the network capacity in terms of number of voice calls supported. In [10] an analytical model is developed in order to study the impact of packet aggregation on delay.

Nevertheless, there has been no proposals that improve QoS support in IEEE 802.11s based on utilizing the IEEE 802.11n [11] new defined physical and MAC enhancements. The IEEE 802.11n physical enhancements include adding multiple input multiple output (MIMO) antennas. Each with an RF chain that is capable of simultaneous receive or transmit traffic. Using different space time code structures, MIMO system can support data rate ranges between 6.5 to 600 Mbps. The IEEE 802.11n MAC enhancements include introducing a new MAC frame structures that can be used to aggregate multiple subframes.

In this work, we propose a novel scheme that exploits the IEEE 802.11n physical and MAC enhancements in order overcome the scarcity of the wireless link capacity and the lack of robust QoS scheduling in IEEE 802.11s. Our scheme entails the following components:

- *Link adaption*: based on online link assessment we adapt the appropriate data rate that best suits the instantaneous channel quality.
- *Adaptive aggregation frame length*: based on the link quality and the used modulation, we compute the probability of error bits, which can be used with the receiver correction threshold to determine a suitable aggregation frame length.
- *QoS bandwidth provisioning algorithm*: given the allocated bandwidth and the aggregation frame length, we provide an optimization program that distributes the bandwidth to guarantee the required QoS while maintaining fairness among connections.

Online interaction between OPNET Modeler and MATLAB is used to evaluate the incorporated enhancements. The perfromance results show that incorporating link adaptation increases the achieved data rate, which reduces the service time required

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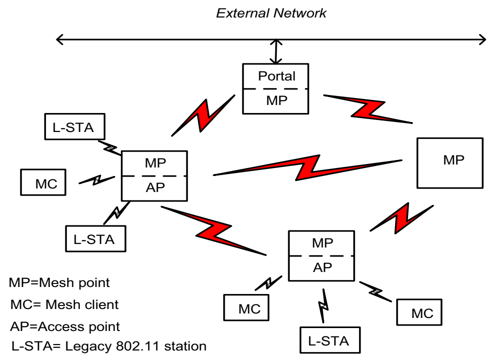


Fig. 1. Wireless mesh network's components and hierarchy structure

for a same amount of data. Exploiting frame aggregation technique helps reduce the MAC delay and increases the throughput, and consequently reduce the packet drop ratio. The proposed schemes indicates better improvement in case of traffics with small packet sizes in terms of MAC delay and achieved throughput.

The remainder of this paper is organized as follows. Section II briefly reviews the IEEE 802.11s standard and highlights the WMN's main components, network hierarchy, and the utilized medium access protocol. Section II also describes the physical and MAC enhancements defined in the IEEE 802.11n draft. Section III introduces our proposed mechanisms for link adaptation, adaptive frame length aggregation, and QoS bandwidth provisioning. In Section IV, OPNET Modeler interfaced with MATLAB is used to evaluate the incorporated enhancements under different flow and packet size scenarios. Concluding remarks are given in the Section V.

II. PRELIMINARIES

In this Section, we first overview the WMN's main components, network hierarchy, utilized medium access control, traffic flows, and link quality assessment model. Next we present the physical and MAC enhancements of the new IEEE 802.11n amendment.

A. IEEE 802.11 based Wireless Mesh Networks

Generally, a mesh point (MP), Figure 1, is a node that can support mesh services, e.g., mesh path selection and forwarding. Mesh point may be collocated with one or more other entities (e.g., access point (AP), portal). Mesh point collocated with an access point is referred to as mesh access point (MAP). Such configuration allows this entity to provide both mesh functionalities and access point functionalities. Similarly, a mesh point collocated with a portal entity is referred to as mesh portal (MPP), where its functionalities are to interface the WMNs to other IEEE 802 LAN segments. A mesh client (MC) is associated with MAPs to gain access to the mesh network. This includes stations that can support mesh services and those which do not. In this work, we refer to both mesh client and legacy stations as mesh clients. We assume each MAP is equipped with two independent medium access control protocols, which utilize different channel frequencies, mesh clients tier (MCT) and backhaul tier (BHT) MAC. MCT MAC utilizes the IEEE 802.11e HCF controlled channel access (HCCA) to coordinate the medium access among the mesh access point and its associated clients. On the other hand, the BHT MAC utilizes the IEEE 802.11e enhanced distributed channel access (EDCA) for backhaul medium coordination.

The mesh traffic flow is described as follows: first after configuring the basic service set (BSS) utilized by each MAP, the MAP adopts the IEEE 802.11e reference scheduling algorithm, see Appendix A, to allocate transmission opportunities for the admitted connections, $txop_{i,j}$, where $i = \{1, 2, \dots C\}$ is i^{th} connection associated with j^{th} mesh access point, $j = \{1, 2, \dots, Z\}$. C and Z are the number of connections per MAP and the number of backhaul access points, respectively. During the allocated transmission opportunities, the MAPs poll the associated connections to deliver their traffics. The received data is then stored in k different queue classes at the MAPs. Next, the MAPs compute the total required $TXOP_j$ for connections belonging to station j,

$$TXOP_j = \sum_{i=1}^{C} txop_{i,j} \quad j = \{1, 2, \dots, Z\} \tag{1}$$

Then the MAPs commence the backhaul tier MAC procedure.

In this work, we consider wireless mesh networks that utilize the IEEE 802.11n links where each mesh point is equipped with multiple antennas. We also consider, following the IEEE 802.11n amendment, the space time block code (STBC) is the used coding structure. The WMN's link quality is assessed by exchanging physical frames called sounding frames (SF) over all transmitting and receiving antennas. After exchanging the SF frames, the mesh point estimates the channel state information,

$$H = \begin{bmatrix} h_{1,1} & \dots & h_{1,n_t} \\ \vdots & \ddots & \vdots \\ h_{n_r,1} & \dots & h_{n_r,n_t} \end{bmatrix} \tag{2}$$

where h is the channel fading coefficient between the $j^{th}, (j = 1, 2, \dots, n_t)$ transmitting antenna and the $i^{th}, (i = 1, 2, \dots n_r)$ receiving antenna,

$$h = (\frac{1}{l_p \times \text{shadow fading}}), \tag{3}$$

where l_p is the path loss and is modelled as

$$l_p = (\frac{\lambda}{4\pi d})^{\eta} \tag{4}$$

where λ is the wavelength, d is the physical distance between the transmitter and the intended receiver, η is the path loss exponent, n_t, and n_r denote the number of transmitting and receiving antennas, respectively. Given STBC as the coding structure, the link's SNR is computed as,

$$SNR_{STBC} = \frac{\rho}{n_t} \|H\|_F^2 \tag{5}$$

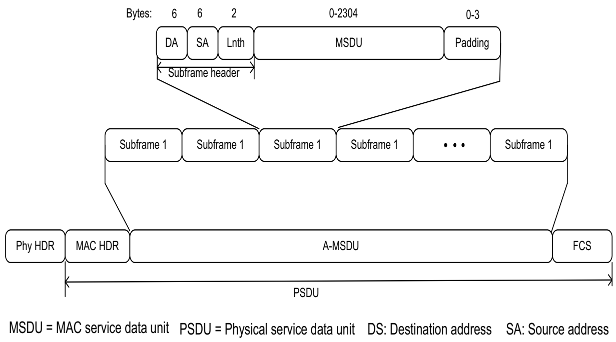


Fig. 2. A-MSDU aggregation frame structure

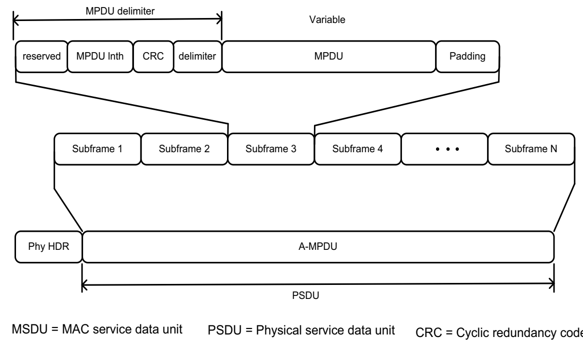


Fig. 3. A-MPDU aggregation frame structure

where ρ is the total transmitting power from all antennas per node. It follows that the maximum error-free data rate for space time block code channel is

$$C_{STBC} = r_s \log_2(1 + SNR), \quad (6)$$

where r_s is the spatial code rate defined as the average number of independent symbols transmitted from the n_t antennas over T symbol periods.

B. IEEE 802.11n physical- and MAC-layer enhancements

The IEEE 802.11n draft standard enhances the 802.11 physical and MAC specifications. In the following Sections we respectively detail these enhancements.

1) IEEE 802.11n physical-layer enhancements: The IEEE 802.11n high throughput (HT) physical specification defines 76 modulation and code scheme indexes for each channel bandwidth, i.e., 20 MHz and 40 MHz. MSC specifies the physical layer parameters that consists of modulation type (BPSK, QPSK, 16-QAM, 64-QAM) and coding rate (1/2, 2/3, 3/4, 5/6), number of coded bits per single carrier $N_{BPSC} = \{1, 2, 4, 6\}$, number of spacial streams $N_{ss} = \{1, 2, 3, 4\}$, and whether equal modulation (EQM) or unequal modulation (UEQM) per spacial streams is used. The IEEE 802.11n HT physical specification also defines two guard interval values $GI = 400ns$ and $800ns$. By combining different physical layer parameters values, the IEEE 802.11n can be configured to support up to 304 different data rates that range between 6.5 and 600 Mbps.

2) IEEE 802.11n MAC-layer enhancements: In addition to supporting large range of data rates, the IEEE 802.11n draft standard introduces other MAC enhancements, i.e., frame aggregation structure where multiple small subframes are carried in the same frame. In the following Sections three different aggregation techniques and their constraints (defined in the IEEE 802.11n draft standard) are introduced.

A-MSDU aggregation frame structure:

The basic principle of an A-MSDU aggregation is to send multiple MAC service data units (MSDUs) frames within a single MAC protocol data unit (MPDU) frame. Figure 2 depicts the A-MSDU aggregation structure. *An A-MSDU aggregation is constrained by the following*: 1) All MSDUs subframes must have the same traffic ID (TID). 2) The A-MSDU expiration occurs only when all A-MSDU's constituents expire. 3) All the destination addresses (DAs) and sender addresses (SAs) of all aggregated subframes must be the same, i.e., broadcasting or multi-casting is not allowed. 4) The maximum MPDU length that can be transported using A-MSDU aggregation technique is 4095 octets. 5) The MPDU constructed from multiple aggregated MSDUs subframes cannot be fragmented.

A-MPDU aggregation frame structure:

In general, the concept of the A-MPDU aggregation is to join multiple MPDU subframes in order to diminish the physical header overhead. Figure 3 depicts the A-MPDU aggregation structure. However, restriction of aggregating frames with matching TIDs is not an essential factor with A-MPDUs, but there are other constrains such as: 1) the maximum length that an A-MPDU can obtain is 65535 octets, 2) all the MPDUs within an A-MPDU are addressed to the same receiver address, and 3) the maximum number of aggregated subframes is 64.

A-MSDU and A-MPDU two-level frame aggregation structure:

The two-level frame aggregation comprises a blend of A-MSDU and A-MPDU over two stages. Over the first stage, multiple MSDUs form an A-MSDU with packets that have the same TID. Packets that have different TIDs move over the second aggregation stage where they are packed together with the A-MSDUs from the first stage by using A-MPDU aggregation.

III. Exploiting 802.11n capabilities to support QoS in IEEE 802.11s

Based on the IEEE 802.11n physical and MAC enhancements we propose incorporating the following mechanisms to enhance the IEEE 802.11s QoS support. The first two proposed enhancements are based on assessing the WMN's link quality to approximate the physical data rate, r_{phy}, and the aggregation frame length, AF_l. The third proposed enhancement is a QoS bandwidth provisioning mechanism implemented in the MAP to perform intelligent bandwidth distribution to further boost the QoS support. In the following Sections these three proposed enhancements are further detailed respectively.

A. Link adaptation

As stated earlier that different combinations of the IEEE 802.11n physical layer parameters values can produce 304 different data rates. Based on the modulation and code scheme index parameters, the following formula can be used to compute the actual transmission data rate in Mbps, r_{phy},

$$r_{phy} = C \times BW \times N_{ss} \times N_{BPSC} \times C_r \times GI, \quad (7)$$

where C is a constant equal to 0.65, BW is the channel bandwidth factor that is equal to 20 when the 20 MHz channel bandwidth is used and is equal to 40 when the 40 MHz channel bandwidth is used, N_{ss} is the number of spacial streams equals to any value from 1 to 4, GI is the guard interval that is equal to 1 when GI=800 ns and is equal to 1.10769 when GI=400 ns is used, N_{BPSC} is the number coded bits per subcarrier, and C_r is the code rate. According to the modulation type, N_{BPSC} and C_r take different values as depicted in table I.

TABLE I
CHANNEL MODULATION PARAMETERS

Modulation Type	N_{BPSC}	C_r
BPSK	1	1/2
QPSK	2	1/2
QPSK	2	3/4
16-QAM	4	1/2
16-QAM	4	3/4
64-QAM	6	2/3
64-QAM	6	3/4
64-QAM	6	5/6

In essence, the objective is to adapt the link data rate according to the online link quality. The algorithm works as follows, 1) after exchanging SF frames, mesh points assess the link quality via computing the channel's SNR. The latter is then used to compute the link capacity as defined by equation (6). 2) Using the C_{STBC} as an upper bound, substitute different modulation types in equation (7) to find the link transmission data rate, i.e., $r_{phy} \leq C_{STBC}$. Since the C_{STBC} is a theoretical data rate, a guard data rate gap, Δ_r, can be considered, i.e., $r_{phy} + \Delta_r \leq C_{STBC}$.

B. Aggregation Frame Length adaptation

As explained earlier the IEEE 802.11n introduced three different aggregation techniques: A-MSDU, A-MPDU and two-level aggregations. Incorporating frame aggregation enhances the MAC performance by minimizing the MAC headers and medium access overheads that consumed to transmit the same amount of data. The disadvantage of considering large aggregation frames is the failure of frame decoding upon experiencing a bad link quality. Retransmitting large frames can dramatically decreases the system utilization.

In this Section we introduce a new algorithm that estimates the aggregated frame length according to the link quality, SNR, and to the receiver error correction threshold, f. The latter is the upper bound of the number of error bits that can be corrected by the receiver. Let P_f be the probability of f error bits occurring in a packet of N bits length . The basic idea is to choose an aggregation frame length such that P_f does not exceed a predefined threshold. Therefore, the P_f parameter can be optimized by the MAP to estimate the aggregation frame length, AF_l, based on the current link status. The following steps list the procedure to find the aggregation frame length:

- Assess the link's SNR using equation (5).
- Using the modulation type determined in Section III-A find the bit error rate (P_e), from the P_e versus SNR plotted curve for the selected modulation type. Given the aggregation frame length (N), P_e can be used to find the probability of having f error bits using the binomial distribution,

$$P_f = \binom{N}{f} P^f (1 - P_e)^{N-f}. \quad (8)$$

- To find the aggregation frame length, the MAP, in each iteration, adds a MSDU frame, increments the total accumulated number of aggregated bytes, AF_l, and computes the probability of having P_f. If the P_f value exceeds its predefined value, the MAP returns the previous AF_l as the appropriate frame length in bytes.

Given the aggregation frame length, AF_l, and $TXOP_j$, computed in equation (1), the MAP computes the useful time, U_{data}, of the $TXOP_j$ that is used to send useful data. U_{data} is the remaining time after subtracting all the physical and medium coordination overheads. To compute U_{data} first let O^H_{phy} denote the physical header overhead, $R^{AF} = AF_l - O^H_{phy}$ is the remaining aggregation farme length after subtracting the physical header. The U_{data} can be computed as follows,

$$U_{data} = TXOP_j - \left(\lceil P^{AF}_{TXOP_j} \rceil \times O^{AF}_T \right) - O^{TXOP_j}_{MC}, \quad (9)$$

$P^{AF}_{TXOP_j}$ is the possible number of aggregation frames transmitted in $TXOP_j$ period,

$$P^{AF}_{TXOP_j} = \frac{TXOP_j - O^{TXOP_j}_{MC}}{O^{AF}_T + \frac{R^{AF}}{r_{phy}}}, \quad (10)$$

where O^{AF}_T in equation (9) and (10) is the total overhead required to transmit one aggregation frame,

$$O^{AF}_T = 2 * SIFS + \frac{Ack}{r_{phy}} + \frac{O^H_{phy}}{r_{phy}}. \quad (11)$$

$O^{TXOP_j}_{MC}$ in equation (9) and (10) is the MAC coordination overhead that is required by the MAC protocol to peform traffic class (TC) predetermined arbitration IFS (AIFS), random backoff (bk), request to send (RTS), clear to send (CTS) and short interframe space (SIFS). $O^{TXOP_j}_{MC}$ is given by

$$O^{TXOP_j}_{MC} = AIFS[TC] + \overline{bk} + RTS + CTS + 2 * SIFS \quad (12)$$

$\overline{bk}$ is the moving average and computed as follows

$$\overline{bk} = \alpha \times \overline{bk} + (1 - \alpha) \times bk \quad (13)$$

where α is a smoothing factor parameter.

The U_{data} then is passed to the optimization program, described in next Section, to find the optimum bandwidth distribution among different classes.

C. Bandwidth Provisioning Scheme

We propose an adaptive bandwidth provisioning scheme (modified from high speed downlink packet access (HSDPA) [12]), a scheme proposal for the MAP, which enables QoS guarantees and fairness by optimally distributing the U_{data}, among the contending connections.

Given r_{phy} computed in Section III-A, U_{data} computed in III-B, in addition to other locally known information such as the queue size, q^k, and the mean MSDU size, m^k, of the k class, we device an optimization program that can optimally distribute the bandwidth among traffic classes. Let $b^k_{req}(A)$ represent the time in seconds required to empty the k^{th} queue class using the A^{th} aggregation technique and computed as follows:

$$b^k_{req}(A) = \begin{cases} \frac{q^k\left(m^k+sm^H\right)+O^H_{MAC}}{r_{phy}} & \text{if } A = AMSDU; \\ \frac{q^k\left(m^k+DL+O^H_{MAC}\right)}{r_{phy}} & \text{if } A = AMPDU. \end{cases} \quad (14)$$

where sm^H is the subframe header (DS, SA, frame length, and padding), O^H_{MAC} is the MAC header, and DL is the MPDU delimiter. Also, let a^k represent the actual allocated time in seconds to the k^{th} traffic class. The following optimization program can optimally distribute the bandwidth,

$$\max_{1 \le j \le k} \sum_{j=1}^{k} w^j \times a^j \times b^j_{req}(A) \quad (15)$$

subject to

$$\sum_{j=1}^{k} a^j = U_{data} \quad (16)$$

$$lb^k \le a^k \le b^k_{req}(A) \quad (17)$$

where w^i is a class prioritization weight assigned to different traffics, and lb^k is the lower bound time assigned to k^{th} traffic flow and computed as follows: let $\overline{F^k_{tx}}$ denote the average number of transmitted frames of the k^{th} class during each $TXOP_j$ period.

$$\overline{F^k_{tx}} = \alpha \times \overline{F^k_{tx}} + (1-\alpha)F^k_{tx}. \quad (18)$$

Also let $\overline{F_{tx}}$ represent the total number of frames transmitted from all classes each $TXOP_j$ period.

$$\overline{F_{tx}} = \alpha \times \overline{F_{tx}} + (1-\alpha)F_{tx}. \quad (19)$$

The lower bound bandwidth of each class is computed as

$$lb^k = \beta \times \left(\frac{1 - \frac{\overline{F^k_{tx}}}{\overline{F_{tx}}}}{\sum_{i=1}^{k}(1 - \frac{\overline{F^k_{tx}}}{\overline{F_{tx}}})} \times U_{data} \right) \quad (20)$$

where β is the percentage of bandwidth that the k^{th} traffic class is voluntarily relinquishing to other flows. Although the optimization program equation (15) may favor flows with higher queue size, the fairness is guaranteed by equation (17). Upon computation of a^k by the optimization program for the k classes, the MAP starts transmitting data from k classes based on their bandwidth shares, a^k.

IV. PERFORMANCE EVALUATION

The OPNET Modeler is used to evaluate the performance of the proposed enhancements. A WMN topology similar to the one in Figure 4 that incorporates the proposed enhancements is modelled. In this topology, each MAP utilizes IEEE 802.11e HCCA MAC protocol reference scheduler, detailed in Appendix A, to coordinate medium access of u_i users, where $i = \{1, 2, \ldots, w\}$ and w is the number of mesh backhaul access points, in this work we consider $w = 5$ and u_i is varied in each scenario to examine different system performance aspects. Using OPNET built-in Standard Models (OSM), each mesh client is configured to generate one of the following traffic types, i.e., either voice (vo), video (vi), file transfer protocol (FTP), or hypertext transfer protocol (HTTP). The traffic specifications of each traffic type are detailed in table II. Traffic flows of the same traffic type may generate traffic with different data rates that is confined in the specified range per each traffic type.

TABLE II
TRAFFIC SPECIFICATIONS

TSPEC	VoIP (G.729A)	Video (MPEG-4)	FTP	HTTP
Mean data rate	26-64 kb/s	1-1.5 Mb/s	400-600 kb/s	350-600 kb/s
delay bound	100 ms	150 ms	-	-
Nominal MSDU	160 octets	1024 octets	1200 octets	500 octets
Max service interval (MSI)	25 ms	30 ms	-	-

The backhaul topology consists of MAP uniformly distributed around the MPP with distance d ranges between 100 to 300 (meters). IEEE 802.11e EDCA MAC is utilized as mesh backhaul point's medium access coordination protocol. The traffic received, from associated users, by each MAP, is queued in four queue classes. After that the traffic is forwarded to the mesh portal. Before every MPP and MAP communication, OPNET simulation model sends the network status parameters to the MATLAB. After performing the related computation the latter returns the values of the optimum bandwidth share per each traffic class. These returned values are then used by OPNET Modeler to respectively send the data from specified classes queue .

In the first scenario we associate each MAP with four different traffic types. Each MAP is associated with four users. Figure 5 shows the affect of the incorporated enhancements on the medium access delay, defined as the time between inserting the packet in the queue until it starts transmission. Due to shortening the wasted bandwidth consumed by the MAC headers and MAC medium accesses, sending data using A-MSDU or two-level aggregations tend to have lower MAC delay compared with the case without data aggregation. Larger aggregated frames promote higher MAC service savings, compared with A-MSDU, the two-level aggregation allows packing of mixed traffic types. These aspects make the two-level aggregation technique produces lower MAC delay compared to those techniques which send data with A-

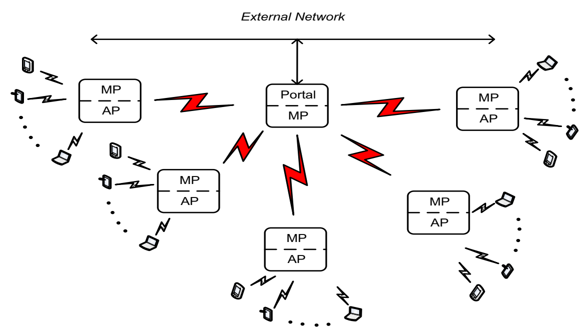


Fig. 4. Wireless mesh network model

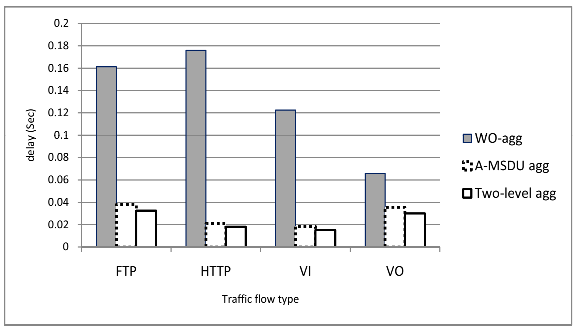


Fig. 5. Medium access delay for different traffic classes using different aggregation techneques.

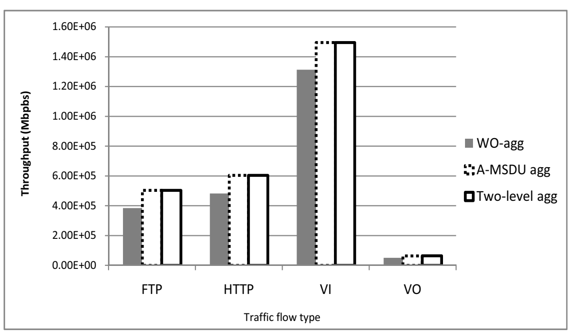
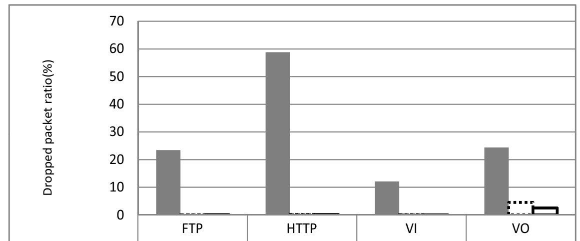


Fig. 6. Throughput comparison of different traffic classes using different aggregation techniques.



	FTP	HTTP	VI	VO
WO-agg	23.47507565	58.84854432	12.10179421	24.40300949
A-MSDU agg	0.015926103	0.081779522	0.027462468	4.530585541
Two-level agg	0.063704412	0.098135427	0.032039546	2.502453386

Fig. 7. Packet drop ratio of different traffic classes using different aggregation structures

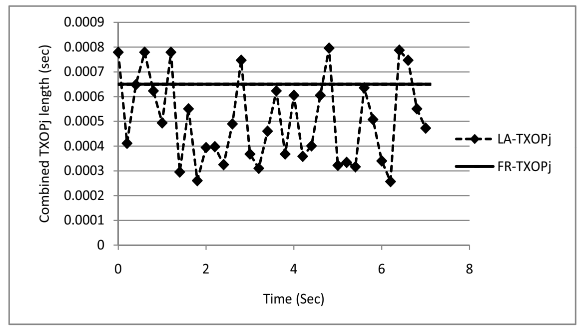


Fig. 8. The allocated transmission opportunity difference between link adaptation (LA-$TXOPj$) and using fixed physical transmission rate (FR-$TXOPj$)

MSDU aggregation structure. In addition to achieving lower MAC delay, the saved bandwidth from aggregation can also be used to deliver more traffic flows. Figure 6 depicts the throughput performance different among A-MSDU, two-level, and without aggregation techniques. Schemes incorporating data aggregation outperform those which do not. As they achieve higher throughput, schemes implementing aggregation technique tend to have lower packet dropping ratios (percentage of the total dropped to the total received) as shown in Figure 7. Good link quality can be opted to send at higher data rates, which can shorten the service time required to transmit the same amount of data. This means nodes can finish earlier before their allocated transmission opportunity. The saved bandwidth can either be dedicated for further data transmission by the transmission opportunity's owner or can be added to the contention period time. Figure 8 shows a snapshot of 7 seconds of the adaptive link's transmission opportunity compared to that of fixed data rate for video traffic flow. The horizontal line in Figure 8 represents the transmission opportunity value of video traffic flow using fixed physical rate (100 Mbps), similar to IEEE 802.11n physical data rate, while the dots represent the transmission opportunity values using link adaptation. Dots below the fixed data rate line represent the saved bandwidth that has been added to the system resources.

In the second scenario we examine our scheme under different network loads by increasing the number of network traffic flows. Figure 9 relates the MAC delay to the increasing number of traffic flows, where the ratio between different traffics, i.e. voice, video, FTP, HTTP, is set to 1:1:1:1. As the combined transmission opportunities by MAP grows with increasing the number of flows, the possibility of more bandwidth savings, contributed by shortening the MAC service time, also increases. According to the exchanged traffic specifications, each admitted traffic flow is granted with an adequate transmission opportunity, $txop_{i,j}$. Due to the bursty nature of traffics, the allocated $txop_{i,j}$ might not be sufficient for some traffic flows with higher QoS constraints or bad link qualities. Combining transmission opportunities of larger number of traffic flows permits flexible data packing management which

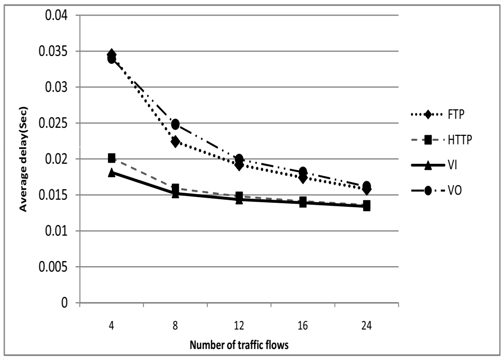


Fig. 9. Medium access delay versus increasing network load using two-level aggregation structure

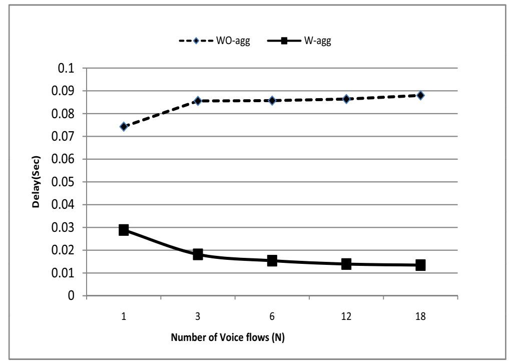


Fig. 11. The MAC delay versus increasing voice network traffic load with and without using aggregation technique

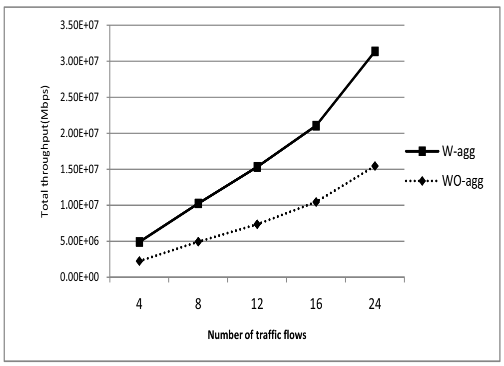


Fig. 10. The total throughput of all traffic flows with and without using aggregation technique

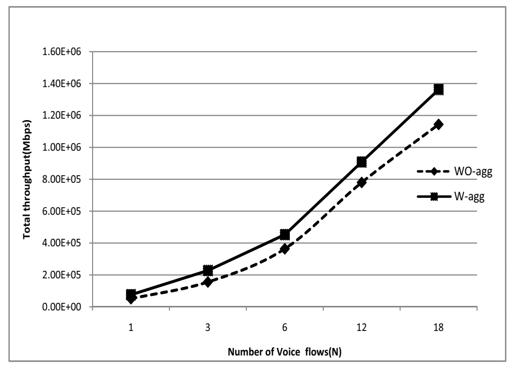


Fig. 12. The achieved throughput for increasingly voice network traffic load

saves bandwidth for backlogged QoS flows. Consequently, the saved bandwidth is dynamically reallocate to backlogged flows based on QoS requirements. Larger combined transmission opportunities also leads to accommodating more data per each transmission opportunity period. Figure 10 shows the total aggregated throughput for all traffic flows with and without data aggregation. The performance gap between the results with and without data aggregation expands as the number of traffic flows increases due to the advantage of combing more transmission opportunities.

In addition to the effect of combining transmission opportunities, the data aggregation technique is also affected by the packed packet sizes. In the third scenario, we evaluate the performance of our scheme for different packet sizes and the number of traffic flows. In this scenario we increase the voice traffic flows number, i.e., in each simulation run the traffic flows load changing as follows: 1, 3, 6, 12, 18 voice traffic flows. As each frame causes MAC service overhead, transmitting small packet sizes severely wastes the scarce bandwidth. On the other hand, aggregating small frames allows lager number of frames accommodation. These properties cause a large performance difference between with and without data aggregation in terms of MAC delay as shown in Figure 11. Figure 12 shows the total achieved throughput under small packet size scenario. Increasing the transmission opportunity via increasing the number of traffic flows and using small packet sizes can achieve higher throughput.

V. CONCLUSION

In this paper we proposed an enhancement scheme which exploits the IEEE 802.11n physical and MAC enhancements to improve the IEEE 802.11s QoS support which constrained by link bandwidth deficit and QoS scheduling. The proposed scheme includes: 1) adapting the appropriate data rate based on the instantaneous channel quality, which is done by opting the proper modulation and code scheme index, 2) an adaptive aggregation frame length based on the instantaneous link quality to maximize the advantage of large frame aggregation and to preclude large frame retransmissions, 3) a rational optimization program that distributes the bandwidth to guarantee the QoS requirements, while maintaining fairness among connections. Online interaction between OPNET Modeler and MATLAB is used to evaluate the incorporated scheme. Performance results demonstrate that incorporating link adaptation increases the achieved throughput. Exploiting frame aggregation technique, on the other hand, reduces the MAC delay, and thus, decreases the packet drop ratio. As shown by the perfromance results, packet aggregation techniques have a crucial impact on system performance when aggregating small packet sizes.

APPENDIX A
IEEE 802.11E REFERENCE SCHEDULING ALGORITHM

To provide Qos support the IEEE formed a Task Group "E" (TGe) to extend IEEE 802.11 MAC protocol, IEEE 802.11e draft standard. The new medium access control (MAC), called hybrid coordination function (HCF), uses a contention-based channel access method called enhanced distributed channel access (EDCA) that operates concurrently with an HCF controlled channel access (HCCA) method. In HCCA method, the hybrid coordinator (HC) uses point interframe space (PIFS) to attain control of the channel and then to allocate $txop_i$ to admitted connections according to the polling list. During the admission stage the connections sends the traffic specifications (TSPEC) to the MAP. The latter then computes the scheduling service interval,

$$si = \frac{T_B}{\lceil \frac{T_B}{\min(MSI_{i,j})} \rceil}, \tag{21}$$

where $i = \{1, 2, \ldots C\}$ denotes the i^{th} connection, C is the total number associated connections with the MAP, T_B is the beacon interval, and $MSI_{i,j}$ is maximum service interval of i^{th} connection. According to the defined TSPEC of i^{th} connection, the MAP scheduler computes the $txop_{i,j}$ lengths. To illustrate that, let $r_{i,j}$ represent the mean data rate required by the i^{th} connection. It simply follows that the number of arrival packets $n_{i,j}$ during the (si) interval is approximately computed as

$$n_{i,j} = \lceil \frac{r_{i,j} si}{m_{i,j}} \rceil, \tag{22}$$

where $m_{i,j}$ is the mean MSDU size of the i^{th} connection. Given the estimated number of packets, the MAP can compute the $txop_{i,j}$ that allocated to the i^{th} connection as follows

$$txop_{i,j} = \max(\frac{n_{i,j} m_{i,j}}{r_{phy}} + O, \frac{m_{max}}{r} + O), \tag{23}$$

where r_{phy} is the physical layer rate and m_{max} is the maximum MSDU size, o represents the physical and MAC overheads.

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