I J C T A, 9(9), 2016, pp. 3015-3027

Adaptive Bandwidth Allocation and Latency Guarantee for WLAN Networks using Fuzzy Logic Control

Bakeel Maqhat*, Mohad Dani Baba*, MdMahfudz MdZan*, Ruhani Ab Rahman*, Anwar Saif, Mat IkramYusof*, Mohd Asri Mansor* and Farok Hj Azmat*

ABSTRACT

The tremendous increase in user demands for multimedia applications with its various quality of service (QoS) requirements has become essential for the operators to accommodate the demand for real-time services in WLAN network. One of the challenging issues still open for research in IEEE 802.11 WLAN is the scheduling mechanism to fully support the various QoS requirements. Furthermore, the large contrast in the real-time and non-real-time traffic specification, the insufficient bandwidth allocation and not able to satisfy the latency requirement can lead to degradation or decreases the overall system performance. In this paper, an efficient scheduling scheme is proposed for IEEE 802.11n WLAN to supports small packet size and to guarantee the timing constraint of the real-time traffics. The proposed adaptive bandwidth allocation and latency guarantee for WLAN networks using fuzzy Logic control (ASEF) will provide fair resource allocations for the real-time and non-real-time traffics. The fuzzy system is used to control and dynamically assign the required bandwidth to the various service classes according to their delay constraint and throughput. The simulation results show that the performance of the proposed adaptive bandwidth allocation (ASEF) can satisfy the timing constraint of the real-time traffics and optimize the overall system utilization.

Keywords: IEEE802.11n, bandwidth allocation, scheduling, QoS, Fuzzy logic system.

1. INTRODUCTION

The widespread use of wireless Local Area Networks (WLAN) can be attributed to its unparalleled merits; low deployment cost and supports broadband bandwidth capability. Furthermore, the main reason is the convenience to access multimedia applications (such as internet gaming, video streaming, and voice) virtually anytime and anywhere, whilst the user mobility and connectivity is guaranteed. The original IEEE 802.11 is only introduced for Best Effort traffics. The standard IEEE 802.11 defines two schemes for Medium Access Control (MAC), the mandatory scheme named Distributed Coordination Function (DCF) and the optional scheme named Point Coordination Function (PCF). The IEEE 802.11 standard shortage is in providing the mechanism to support the real-time traffics. Hence, it is not easy to provide Quality of Service (QoS) supports or to guarantee the time-sensitive application. However, the increase in demand for multimedia applications has become a necessity to provide the supports for varying QoS requirements in WLANs. The IEEE 802.11e standard was modified and introduced an enhancement to the existing wireless technologies by adding Quality of Serviceto backup multimedia applications without losing backwards compatibility. IEEE 802.11e uses a medium channel access function known as Hybrid Coordination Function (HCF) which is composed of contention based Enhanced Distributed Channel Access (EDCA) and contention free Hybrid Coordinated Channel Access (HCCA).

* Computer Engineering Studies, Faculty of Electrical Engineerin, Faculty of Electrical Engineering, Universiti Teknologi MARA, 40450 Shah Alam, Selangor, Malaysia, *Email: bakeelyem@gmail.com*

The EDCA classifies the traffic flows into four Access Categories with eight priorities, where the higher chance is given to the higher priority traffic to be transmitted than the lower priority traffics. However, the low efficiency of PHY and MAC layers protocol have restricted functions to accommodate the higher data rate applications (video teleconferencing, high-definition television, online gaming, multimedia streaming, voice over IP, and file transfer)[1]

To this end, the IEEE 802.11n standard has defined several enhancement for MAC and PHY layers to resolve the encountered limitations and to ascertain a throughput of at least 100 Mbps at the MAC Service Access Point (SAP). However, IEEE802.11n standard does not specify the scheduling algorithm to guarantee QoS. This is done on purpose in order to allow the service providers and vendors to innovate in this area and propose their products. The time-sensitive applications such as VoIP, video conferencing and gaming require a higher bandwidth allocation, which increases delay and can reduce the efficiency of the overall system. The traffic scheduling is the key mechanism deployed in WLAN to ensure to achieve the required QoS support. Therefore, it is crucial to develop the appropriate bandwidth allocation schemes to guarantee the QoS demand.

This paper presents scheduling algorithm to support small size and time sensitive traffic and considering the QoS requirements by exploiting the A-MSDU attributes. Furthermore, this paper introduced a fair and efficient bandwidth allocation algorithm for WLAN networks to improve the system performance, in particular, in terms of the quality of service (QoS) efficiency. For this purpose, an embedded fuzzy expert system is developed for new deadline aware weights in resource allocation. The fuzzy expert system requires two input variables to calculate and control weights, namely, level factor for real-time traffic and throughput for the non-real-time traffic. This system can dynamically assign resources to real time and non-real time traffics queues effectively and maintains fairness to prevent the queue for non-sensitive applications to grow unlimitedly and at the same time considering the maximum latency for time sensitive applications.

The remainder of this paper is organized as follows. In section II, IEEE 802.11n concept and overview. The related works in section III. Theproposed scheme is presented in section IV. Simulation Model, results and dissection are presented in section V and VI respectively. Finally, conclusion is presented in section VII.

2. IEEE 802.11N CONCEPT AND OVERVIEW

The IEEE 802.11n standard was ratified in 2009 with the main goal being to increase the effective throughput to at least 100Mb/s (with data rate reaching up to 600Mb/s) and not to simply build a radio capable of higher data rates. In the 802.11n, the enhancements for increasing the throughput are carried out in both the PHY and MAC layers. In this standard, new features dramatically enhance the data rates, communication reliability, coverage, and throughput. At the PHY layer, the data rate is mainly improved through the use of Multiple-Input Multiple-Output (MIMO) technology and channel bonding. MIMO uses multiple transmitter and receiver antennas (Tx/Rx) to improve the system performance by dividing the data into streams and then transmitting via several independent channels. Furthermore, the inefficiency imposed by the fixed overheads and contention losses at the MAC layer of the legacy 802.11 standards is mitigated in the 802.11n through the use of frame aggregation, block acknowledgment, and reverse direction data transfer. The key enhancement in 802.11n MAC layer is the frame aggregation technique whichcombines multiple frames into one large aggregated frame before being transmitted. Combining multiple frames into one large frame increases the channel utilization and improves the MAC throughput. [2]. The MAC layer defines two frame aggregation mechanisms, namely, the MAC protocol data unit aggregation A-MPDU and MAC service data unit aggregation A-MSDU. A-MPDU is robust against error due to packets retransmission and large aggregation size, whereas A-MSDU is effective in the error-free channel due to small headers. A-MSDU has poor performance in the erroneous channel because of the absence of packets retransmission.

Several works have been proposed to enable retransmission at the MSDU level, which makes it robust against error. Saif and Othman [3] introduced an MSDU frame aggregation called SRA-MSDU aggregation scheme to reduce the headers of the frames and supports the retransmission of the corrupted packets at the MSDU level. Thereby A-MSDU has become compatible and able to fulfil the QoS requirements for applications with small frame size such as VoIP, videos as well as interactive gaming. On the other hand, the aggregation scheme has a negative effect on the delay performance of the real-timeapplications, it causes additional delays, particularly when waiting for other packets in the queue to construct the aggregated frame. The delay imposed will severely affect the time constraint traffic such as VoIP, video, and online gaming. And due to scheduling is not addressed in the 802.11n standard, thus, schedulers that adapt with frame aggregation worth to be implemented in order to meet the traffic requirements and delay limits. Moreover, the scheduling and resource allocation algorithms should take into account the unfairness problem that may occur between the traffic classes during the aggregation.

3. RELATED WORKS

As a result of the non-existence of scheduling in the 802.11n standard; therefore it is left to the vendors for their interpretation. The current 802.11n scheduler inherits the priority mechanism of the legacy 802.11e Enhanced Distributed Channel Access (EDCA) scheduler. EDCA scheduler defines four access categories (ACs); voice (AC_VO), video (AC_VI), background (AC_BK), and best effort (AC_BE). Each AC has a separate queue and different channel access waiting time depending on the ACs. Therefore, higher priority classes will lead the lower priority classes in channel access speed. A number of scheduling and resource allocation algorithms have been studied to overcome the limitations that affect service class applications in WLAN. Several studies [4, 5], have tried to resolve the scheduling problem in WLAN by using contention windows according to EDCF. Nevertheless, others have tried to segregate real-time from non-real-time traffic and transmit in the contention-free period [6]. However, this scheme requires substantial overhead for the transmission of polling frames from Access Point to the stations that transmit real-time traffic. Ping et al. [7], have proposed a novel token-based scheme to eliminate collisions and subsequently increases channel utilization. The authors have reported that, by integrating voice and data traffic, the token-based scheduling can lead to a better result than DCF in terms of channel utilization. This work has been extended in [8], by the integrating the video, voice and data traffic over WLAN. This scheme can practically eliminate the contention and transmission opportunity (TXOP) idle time which leads to increase in channel utilization when compared with EDCA. However, this scheme requires substantial overhead for the transmission of polling frames from Access Point (AP) to the node. The problems in this scheme are not completely resolved which would affect the performance of this scheme, as it uses token-passing. Inan et al. [9] introduced an application-aware adaptive HCCA scheduler for IEEE 802.11e WLANs. This scheduling algorithm is based on the Earliest Deadline First (EDF) scheduling discipline to make polling order based on the computed deadlines of the traffic. According to traffic specification along with instantaneous buffer occupancy information in time sensitive application, this algorithm schedules multimedia traffic by associating each QoS station with a distinct service interval (SI) and TXOP. However, the drawback is the additional hardware/ firmware complexity that imposed by the algorithm which make it not easy to implement. A fuzzy expert system was introduced to adapt the packet size for VoIP traffic in ad hoc networks [10]. The model requires two variables input for a fuzzy system which are packet error rate and change of packet error rate. The results show that the expert system is capable of locating packet size values to the optimum level quickly along with the increment in the number of VoIP connections. A bandwidth allocation algorithm was proposed for the uplink traffic in mobile WiMAX called FADDR [11]. The algorithm uses fuzzy logic control which is embedded in the scheduler with an adaptive deadline-based scheme to guarantee a particular maximum latency for real-time traffics and maintain the minimum requirements for the non real-time traffics. The QoS-aware A-MPDU scheduler which is applied to real-time voice traffic by controlling the delay time of whole buffering for A-MPDU and configure the buffering separately depending on access category and IP address for the destination was introduced by [12]. Ramaswamy, et al.[13], have proposed Bi-Scheduler algorithm which separate frames depending on their access categories and schedules the VoIP traffic using A-MSDU aggregation, Whilst schedules the video and non real-time traffic using the A-MPDU aggregation. Nevertheless, the algorithm is not effective under low traffic load and the high sensitive traffic. Moreover, it may suffer from delay due to waiting for transmitting. An innovative frame aggregation scheduler have crystallized the approach of aggregate frames by calculating the deadline based on the earliest expiry time of a frame waiting in the queue, and selecting dynamically the aggregation scheme based on frame aggregation size and bit error rate by using optimal frame size from the lookup table [14]**.** However, this algorithm is restricted to deal with one type of traffic, thus other traffic will suffer from delay and eventually may affect the QoS requirements of other traffics. Moreover, schedulers by exploiting the A-MSDU attributes to enhance the system performance was developedin [15-17].

4. ADAPTIVE SCHEDULER AND BANDWIDTH ALLOCATION USING FUZZY SYTEM

In this paper, the ASEF scheme introduced scheduling algorithm to dynamically distinguish the data of each service type class and place them on the queue based on their QoS requirements. Hence, the traffic vector variables that associate to each packet (P) can be defined as $P = (P_{io}, F)$. Where P_{io} is the Packet priority based on the traffic type, and is the packet lifetime and can be calculated as follows

$$F = T_{rvl} - t \tag{1}$$

Where t the current time and T_{rvl} is the arrival time of the packet in the queue. The key feature of the proposed Adaptive Scheduler based on Embedded Fuzzy (ASEF) system is the low-lifetime of the head of line packet, which allows strict priority traffic in which delay-sensitive data such as voice and video to be dequeued and prioritized for allocation before packets in other traffic classes are dequeued.

Firstly, the traffics belong to four ACs are mapped into one single queue in the sender side called the sending queue SQ, and packets in the SQ will be sorted based on their priorities. Accordingly, the packets with the highest priorities will occupy the top of the queue. Considering the inherited feature of real-time traffic, which is the packet lifetime, real-time packets will be classified into K levels. Based on this, all packets that have a lifetime within the duration equal to the time required to transmit an aggregated frame (T_{tx}) will be placed in the same level. Thus, each packet (P_i) will be designated within the same level of each head-of-level packet (P_{hol}), according to the governing equation:

$$\left| F_{hoL} - F_i \right| \le T_{tx} \tag{2}$$

Where; F_{HoL} is the lifetime of P_{HoL}, F_i is the lifetime of the packet which will place in the same level with P_{HoL}, T_{tx} is the absolute time required to transmit an aggregated frame and receive its block acknowledgment.

Each level will take the lifetime (F_{hoL}) value of the head-of-level packet (P_{hoL}) as a level factor (LF). Consequently, the level with the lowest factor LF will placed on the top of the queue and will be served first by inserting it into the aggregated frame which is referred as the superframe (every transmission session send one superframe). Furthermore, if there are many packets in one level, they will suffer from lack resources and cannot append them into current superframe in this transmission. Therefore, the level will be subjected to the second phase of the scheduling process, as it conducts selection process to choose among the competitive packets depends on their priorities. Accordingly, the packets of higher priorities will be selected to be transmitted. The amount of packets which remains in the existing level is kept in the level called Urgent Level (UL) which serve first in the next transmission to avoid resource consumption and, therefore, maintain system performance. Packets which are not received correctly according to block acknowledgement will be considered as corrupted and gain a high sending priority by placing it at the. The operations of the scheduling algorithm is illustrated in Figure. 1.

In the SQ, non-real time that has lower priority will occupy the lower part of the queue. In fact, the SQ is virtually split into two Queues; one for real-time and the other for non real-time. Therefore, non real-time queue has low chance to be selected for transmission (particularly in heavy real-time traffic load) due to the strong competition with real time traffic which has higher priority and packet lifetime. However, ASEF scheme introduced bandwidth allocation mechanism to satisfy fair resource sharing between real time and non-real time services, in addition to guaranteeing maximum lifetime for real-time.

4.1. Bandwidth Allocation using Fuzzy System

This paper proposes an embedded fuzzy system to dynamically compute the required bandwidth more accurately and with low complexity. The embedded fuzzy system works by selecting two input variables; first is level factor (LF) for real-time classes such as voice and video. Second variable is the throughput for non-real-time classes such as Best Effort (BE) and Background (BK), which is calculated as follows:

$$R_{tio} = Thp_{NRT} / (Thp_{NRT} + Thp_{RT}) \tag{3}$$

Where R_{tio} is the ratio of non real time throughput, Thp_{RT} Real time traffic throughput, and Thp_{NRT} non-Real time throughput.

Fuzzy inference control system output obtains the weight value (w_i) utilized to assign the real-time traffic (Voice and video access categories) amount of bandwidth. While, weight value (w_j) of non-real-time traffic (BK and BE access categories) is computed by subtracting outputted weight from one. Based on the weight value, scheduler is able to make a decision considering network constraints and QoS, so as the bandwidth is assigned for each access category traffics. The number of packets for real-time and non-real-time traffic which will be transmitted in every transmission session can be calculated by the following equations:

$$BW_{RT} = w_i \times Z_{max} \tag{4}$$

$$BW_{NRT} = w_j \times Z_{max} \tag{5}$$

$$Z_{max} = Z_{supr} - Z_{UL} \tag{6}$$

Where, w_i is the optimal bandwidth ratio for real time traffic which is obtained from the embedded fuzzy logic procedure, w_j is the optimal bandwidth ratio for non-real-time traffic, and Z_{max} is the maximum

Please again send picture.

Figure 1: Block diagram of the scheduling algorithm in ASEF Scheme.

size of superframe (Z_{supr}) after deduct the UL size (Z_{UL}). Accordingly, based on the fairness achievements, a queue with non real-time applications is allowed to transmit minimum number of packets on every transmission and served prior to real-time traffics when its lifetime have not been approached. The components and operations of the bandwidth allocation are illustrated in Figure 2.

The main target of this paper is to compute the optimal weight for each service class queue. Thus, the fuzzy logic system that enables the scheduler to allocate fairly the bandwidth for the real-time services within the delay bound and transmit non-real-time traffic ahead of real-time traffic when the lifetime is observed. Here, linguistic forms in the fuzzy system are categorised by groups of linguistic relations. This relationship forms a rule base of the fuzzy system which is transformed into a matrix equation. This indicates that the behaviour of the output will vary depending upon the behaviour of the input parameters. Figure 3 describes the proposed embedded fuzzy system.

4.2. Fuzzy System Proceedings

As mentioned earlier, the embedded fuzzy system uses two variables, the input and the output respectively for different classes of fairness and avoid certain class starvation. This approach is very useful for wireless communications due to their non-linear environment which is hard to quantify the information precisely and develop a mathematical model. In this system, an innovative fuzzy logic system is designed for the linguistic information where the most important design is to utilize the expert information for the rule base creation. In this system, an individual based inference method with Mamdani's design [18] were utilized, where the inference system rules are jointed into one value. The fuzzy system consists of the following stages: fuzzification, fuzzy reasoning inference and defuzzification.

The role is to dynamically analyse all input traffic and combine them into one overall fuzzy set. Firstly the fuzzification process handles two input variables, LF and R_{tio} for the overall system. Then Reasoning inference mechanism which contains the rule base to manipulate the input variables as shown in Figure 4. At this point, the actual decision is made representing the human expert process which performs to the linguistic behaviour to obtain the output value.

Lastly, the defuzzification phase calculates crisp numerical values to obtain the required weight, which provides an indication of the priority for the scheduler. Three linguistic levels have been defined for input

Please again send picture.

Figure 2: Block diagram of the bandwidth allocation in ASEF Scheme

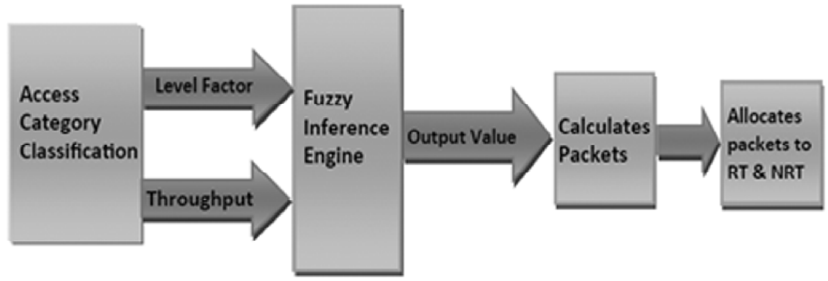
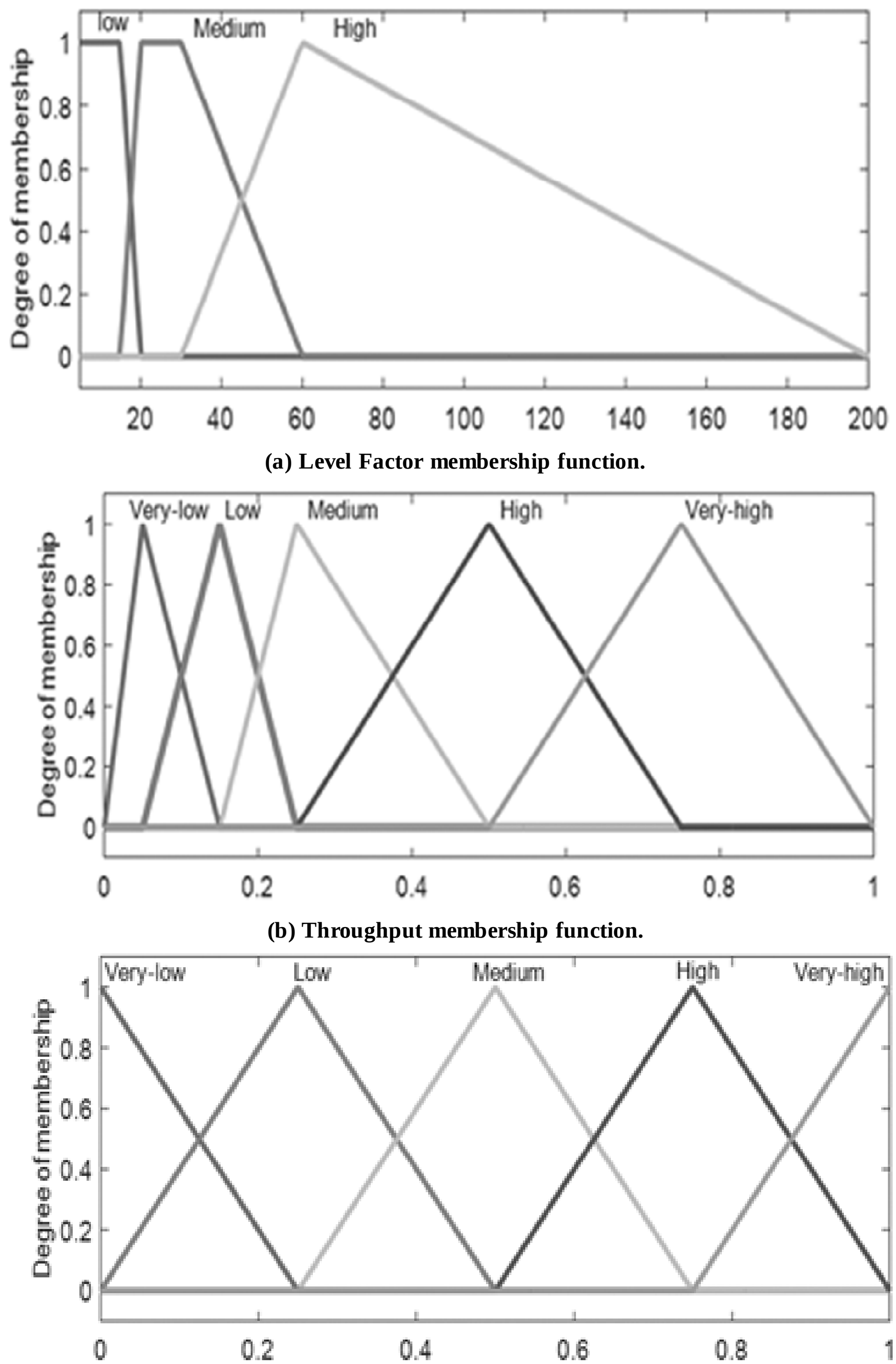


Figure 3: Model of Fuzzy System.



(a) Level Factor membership function.

(b) Throughput membership function.

(c) Output weight membership function.

Figure 4: Fuzzy membership functions for inputs and output.

Table 1
Fuzzy system rule base.

Rules No	*Level factor*	*Throughput*	*Outputw_i.*
0	Low	Very low	Very high
1	Low	Low	Very high
2	Low	Medium	Very high
3	Low	High	Very high
4	Low	Very high	Very high
5	Medium	Very low	High
6	Medium	Low	High
7	Medium	Medium	High
8	Medium	High	Medium
9	Medium	Very high	Medium
10	High	Very low	High
11	High	Low	Medium
12	High	Medium	Low
13	High	High	Low
14	High	Very high	Very low

Algorithm 1 Pseudo code of ASEF scheme

if *SQ is not null* **then**
In the SQ Sort packets in ascending order base on their priority
#classified the SQ into K levels
get first packet from the top of SQ (P_{hol})
for i = 1: N **do**
get the next packet (P_i)
if | $F_{hoL} – F_i$| T_{tx} **then**
put P_i in the Level$_k$
end if
LF of Level$_k$ = F_{hoL} and K=++
end for
while (not end of SQ) and (not exceed the T_{agg} *Eq.* (7)) and (not exceed Z_{max}) **do**
first input of fuzzy system = *LF*
second input of fuzzy system= R_{tio} Eq. (3)
calculate RT traffics packets $BW_{RT} = w_i \times Z_{max}$
if UL is not null **then**
place the UL's packets into superframe.
if (No. of Level$_k$ packets > BW_{RT}) then
select the highest priority packets
and place them into superframe
move the remaining packet to the UL
end if
calculate NRT traffics packets $BW_{RT} = w_j \times Z_{max}$
place these packets into the superframe
end while
end if
if *channel is idle and superframe is not null* **then**
send the superframe
end if

variable of real-time packets namely (low, medium, high) and five levels for R_{tio} of non-real-time packets and the output variable w_i which are (very low, low, medium, high, very high). As the utilized fuzzy system consider two variables as an input, three membership function for the first variable and five membership functions for the second variable, subsequently the rule base composed of fifteen rules (see Table 1). The dynamically normalized scale for the LF input variable is formed from 0 to 200 millisecond and for the input variable R_{tio} and output variable w_i formed from 0 to1.

In order to utilize the bandwidth to reach the highest overall system throughput taking into account the delay requirement for the service class traffic, ASEF scheme considers real-time traffic with a level factor, and non-real-time traffic with itsthroughput ratio to obtain the optimal bandwidth for each service class. The weight value w_i is dynamically determined by fuzzy system. This approach employs intelligent strategy to allocate the right bandwidth to every queue in the system maintaining the overall system capacity. In every transmission the scheduler starts serving the UL first then select packets according to bandwidth that assigned by fuzzy system for every queue. The superframe is transmitted as soon as it reaches the aggregation delay limit (T_{agg}). The T_{agg} can be calculated by the following equation:

$$T_{agg} \leq LF / \left(T_{tx}\right) \tag{7}$$

The pseudo-code of ASEF scheme is presented in algorithm 1.

4.3. Verification of the Fuzzy System by Example

The verification of the fuzzy system is performed by the following example to monitor the fuzzy inference process in the system, and to shows the values of the different inner variables. The example contains a set of if-statements with the activation degree of each logical rule and linguistic labels, for the given set of input values. Then shows the crisp value obtained by the inference process that is finally assigned to the output variable.

Suppose the scheduler specifies the LF with the value of 16 ms and thenonreal time throughput ratio (R_{tio}) with the value of 0.7. The fuzzy system read these variables as its inputs, considering the membership functions of Figure.4. We can make out the linguistic values for two input variables [16, 0.7] and can be read as follows:

Let the LF be 16 ms which after fuzzification is in linguistic form low at grade membership of 0.8 and medium at grade membership of 0.2 (See Figure.4 (a)). Let the R_{tio} be 0.7 which is after fuzzification, high at grade membership of 0.3 and very high at grade membership 0.7 (Figure.4 (b)).

Applying fuzzy reasoning with rule base from Table 1 and Figure. 2 we can read as

- If the LF is low (0.8) AND R_{tio} (0.30) THEN the weight value (w_i) is very high at grade membership of 0.2 [Rule 3].
- If the LF is low (0.80) AND R_{tio} very high (0.7) THEN w_i is high at grade membership of 0.79 [Rule 4].
- If the LF is medium (0.2) AND R_{tio} high (0.3) THEN w_i is medium at grade membership of 0.2 [Rule 8].
- If the LF is medium (0.2) AND R_{tio} very high (0.7) THEN w_i is medium at grade membership of 0.2 [Rule 9].

From above rules and using Mamdami's inference we can infer that the output weight value (w_i) will be very high at grade of membership 0.79 and medium at grade membership of 0.2. The final value of output weight of time is calculated after defuzzification which gives a crisp value of 0.74. It will be used as weight for real time service to calculate the number of packets. For every two input variables; LF and R_{tio}, the intelligent fuzzy system generates one crisp value w_i. More examples can be found in Table 2.

Table 2
Examples of input variables and their crisp values

Input		*Output*	
LF	R_{tio}	w_i	w_j
15	0.9	0.9	0.1
25	0.2	0.75	0.25
40	0.4	0.51	0.49
80	0.1	0.62	0.38
120	0.7	0.18	0.82
190	0.8	0.12	0.88

5. SIMULATION MODEL

Several simulations were conducted in this paper by using the network simulator (NS2) in order to evaluate the different traffics scheduling in terms of packet loss, throughput and average delay. The scheduling scheme deals with multimedia traffic. Specifically the real-time and non-real-time traffics which are directed to their corresponding access categories queue; voice and video for real time and BK and BE for non-real-time. Moreover, this paper uses the simulation scenarios number 17 of the point-to-point usage model [19]. The scenario consists of a single-hop WLAN in which the transmission power of all the High Throughput STAs is high enough to ensure no hidden terminals in the network. All the stations are operating over a 20 MHz. Furthermore, the network traffic is composed of constant bit rate (CBR) based on UDP. The traffic includes the VoIP with the packet size of 120 bytes and traffic rate 96kbps, Video conferencing (512 bytes) [19]. For non real-time traffic; Best-efforts (1000 bytes) with traffic rate 1.07Mbps, and Background (500 bytes), the number of stations increased from 15 to 150 for every 10 units. The simulation parameters and traffic characteristic are listed in Table. 3.

Table 3.
Simulation parameters.

parameter	*Value*
T_{SIFS}	16μs
T_{PHYHDR}	20μs
T_{IDLE}	9μs
T_{DIFS}	34μs
CW_{MIN}	16
Basic rate	54Mbp
VoIP lifetime	30ms
Video lifetime	100ms

6. RESULTS AND DISCUSSION

The average delay in the ASEF scheme and RSA-MSDU is shown in Figures. 3 and 4. The proposed ASEF scheme achieves smallest average delay due to its ability to transmit the packets at a sufficient time before their expiration time. Thus, the packets will not suffer a long queuing delay. The lower the contending station, the smaller the average delay as the stations will have frequent access to the medium and the frames will not suffering a long queuing delay. The performance gain for the ASEF scheme over the RSA-MSDU is about 56% and 50% for VoIP under data rate of 150Mbps and 300Mbps respectively. For video conferencing the gain is about 40% in case of 150Mbps while it's about 29% at high data rates.

The packet loss ratio of ASEF scheme compared to RSA-MSDU is shown in Figures. 5-8. With a small number of competing network stations, the packet loss will be small because the network will have a low number of superframes compete to transmit. The packet loss will rise with the increasing number of stations. Moreover, the Figures show the outstanding performance of our proposed scheme in reducing the packet loss ratio at the high noise and the high traffic load. Figure.5 shows, an enhancement in VoIP packet loss of about 25% and 30% for both data rates, while Figure.6 shows the video conferencing enhancement which is about 44% and 41%. Best-efforts and Background traffic follow the same behavior as the other traffics and scores an enhancement of 58% and 50% under 150Mbps and 300Mbps, respectively, see Figures. 7 and 8.

Figures. 9-12 show the throughput performance of the proposed ASEF scheme scheme under a different number of stations. In a large number of stations, the collisions occur repeatedly and impact the system performance. Nevertheless, the system throughput of the ASEF scheme reaches about 32 Mbps and 43Mbps whereas the SRA-MSDU hardly reaches 13Mbps and 20Mbps at high traffic load under 150Mbps and 300Mbps respectively, see Figure. 9.

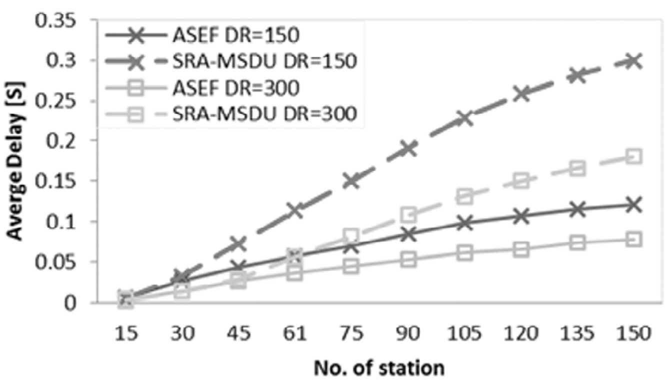


Figure 5: VoIP average delay.

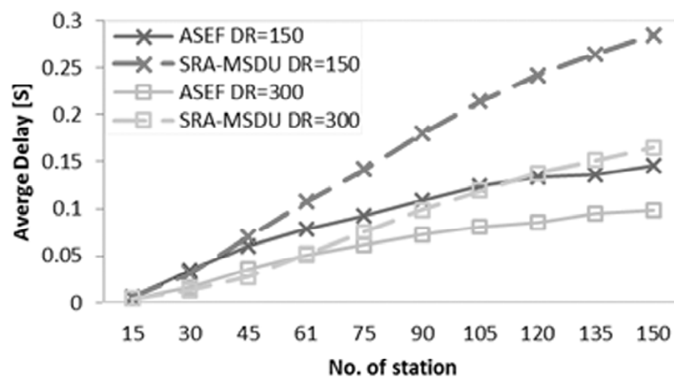


Figure 6: VideoConf. average delay.

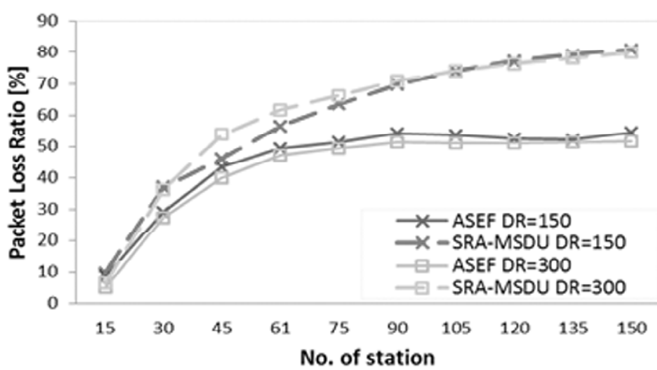


Figure 7: VoIP packet loss ratio

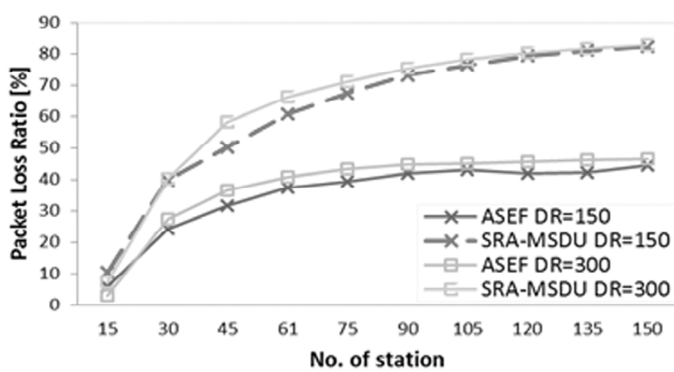


Figure 8: Video Conf. packet loss ratio

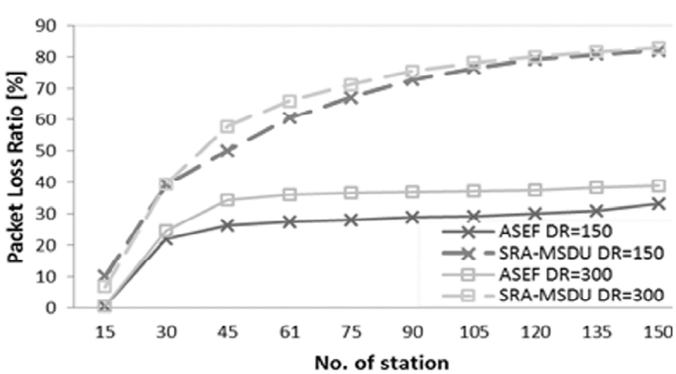


Figure 9: Best-efforts packet loss ratio

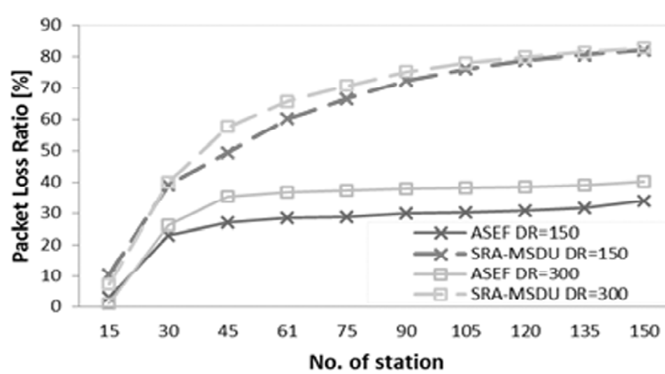


Figure 10: Background packet loss ratio

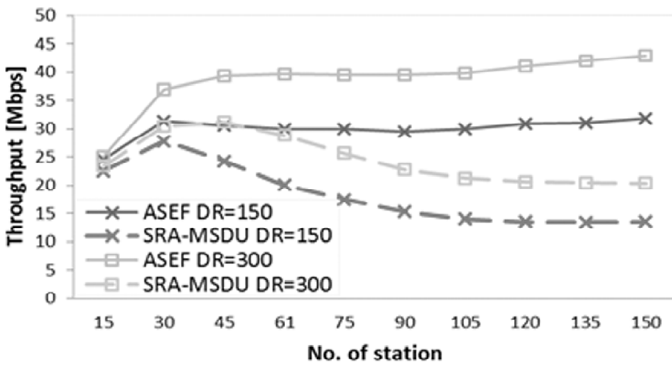


Figure 11: System throughput.

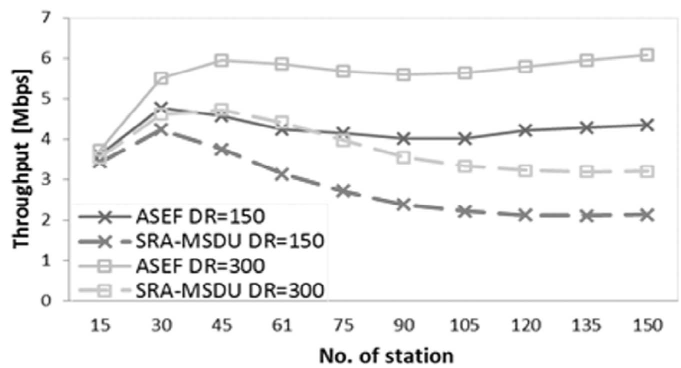


Figure 12: Video Conf. throughput.

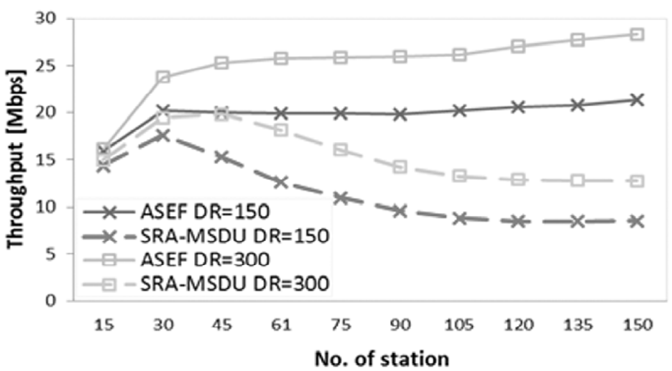


Figure 13: Best-efforts throughput.

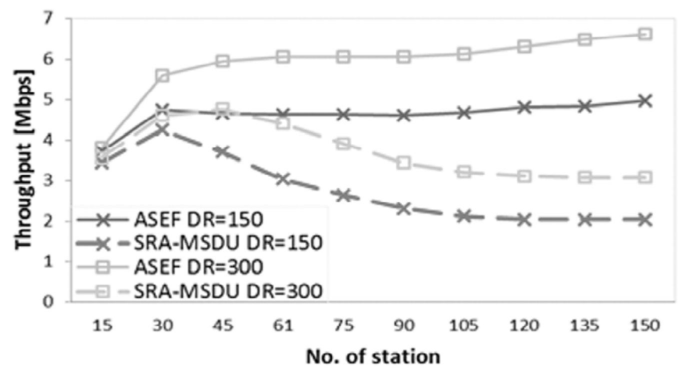


Figure 14: Background throughput.

The throughput gain reaches more than 50% and 48% for video conferencing under different data rate. Whereas, the non-real-time traffic achieves higher than 73% and 68% for best-efforts and background respectively at a data rate of 150Mbps, and about 64% and 59% in high data rate, see Figures. 11 and 12.The throughput gain reaches more than 65% and 58% in both data rate.

7. CONCLUSION

This paper has proposed different traffic class's scheduler that relies on traffic priority and takes into account the packet's lifetime of the time constraint traffic in order to satisfy their QoS requirements. Moreover, the proposed adaptive scheduler based on embedded fuzzy system (ASEF) is fully dynamic by employing the fuzzy logic system. The algorithm dynamically allocates the optimal bandwidth for real-time and non-real-time applications by means of a fuzzy system that grants the optimal bandwidth for different traffics, based on level factor and throughput. The simulation results show that the ASEF scheduling algorithm improves the system performance in terms of delay packet loss ratio and throughput for the real-time and non-real-time applications.

ACKNOWLEDGMENT

The authors would like to thanks Faculty of Electrical Engineering, UiTM for the research support and RMI for the research grant no: 600-RMI/DANA 5/3/REI.

REFERENCES

[1] D. Skordoulis, N. Qiang, C. Hsiao-Hwa, A. P. Stephens, L. Changwen, and A. Jamalipour, "IEEE 802.11n MAC frame aggregation mechanisms for next-generation high-throughput WLANs," *Wireless Communications, IEEE,* vol. 15, no. 1, pp. 40-47, 2008.

[2] C.-Y. Wang, and H.-Y. Wei, "IEEE 802.11n MAC Enhancement and Performance Evaluation," *Mobile Networks and Applications,* vol. 14, no. 6, pp. 760-771, 2009.

[3] A. Saif, and M. Othman, “SRA-MSDU: Enhanced A-MSDU frame aggregation with selective retransmission in 802.11n wireless networks,” *Journal of Network and Computer Applications,* vol. 36, no. 4, pp. 1219-1229, 2013.

[4] Y. P. Fallah, P. Nasiopoulos, and H. Alnuweiri, “Scheduled and Contention Access Transmission of Partitioned H.264 Video Over WLANs.” pp. 2134-2139.

[5] R. Haywood, S. Mukherjee, and P. Xiao-Hong, “Investigation of H.264 Video Streaming over an IEEE 802.11e EDCA Wireless Testbed.” pp. 1-5.

[6] H. Zen, D. Habibi, A. Rassau, and J. Wyatt, “Adaptive Segregation-Based MAC Protocol for Real-Time Multimedia Traffic in WLANs.” pp. 461-466.

[7] W. Ping, and Z. Weihua, “A token-based scheduling scheme for WLANs supporting voice/data traffic and its performance analysis,” *Wireless Communications, IEEE Transactions on,* vol. 7, no. 5, pp. 1708-1718, 2008.

[8] P. Koutsakis, “Token and Self Policing-Based Scheduling for Multimedia Traffic Transmission over WLANs,” *Vehicular Technology, IEEE Transactions on,* vol. PP, no. 99, pp. 1-1, 2011.

[9] I. Inan, F. Keceli, and E. Ayanoglu, “An adaptive multimedia QoS scheduler for 802.11 e wireless LANs.” pp. 5263-5270.

[10] T. Frantti, “Fuzzy packet size optimization for delay sensitive traffic in ad hoc networks.” pp. 2633-2637.

[11] A. M. Alsahag, B. M. Ali, N. K. Noordin, and H. Mohamad, “Fair uplink bandwidth allocation and latency guarantee for mobile WiMAX using fuzzy adaptive deficit round robin,” *Journal of Network and Computer Applications,* vol. 39, pp. 17-25, 2014.

[12] S. Seytnazarov, and Y.-T. Kim, “QoS-aware MPDU aggregation of IEEE 802.11 n WLANs for VoIP services.” pp. 2-4.

[13] V. Ramaswamy, A. Sivarasu, B. Sridharan, and H. Venkatesh, “A bi-scheduler algorithm for frame aggregation in IEEE 802.11 n,” *arXiv preprint arXiv:1401.2056,* 2014.

[14] T. Selvam, and S. Srikanth, “A frame aggregation scheduler for IEEE 802.11 n.” pp. 1-5.

[15] B. Maqhat, M. D. Baba, and R. A. Rahman, “A-MSDU real time traffic scheduler for IEEE802. 11n WLANs.” pp. 286-290.

[16] B. Maqhat, M. D. Baba, R. Ab Rahman, and A. Saif, “Scheduler algorithm for IEEE802. 11n wireless LANs,” *International Journal of Future Computer and Communication,* vol. 3, no. 4, pp. 222, 2014.

[17] B. Maqhat, M. Dani Baba, R. A. Rahman, and A. Saif, “Performance analysis of fair scheduler for A-MSDU aggregation in IEEE802. 11n wireless networks.” pp. 60-65.

[18] P. J. King, and E. H. Mamdani, “The application of fuzzy control systems to industrial processes,” *Automatica,* vol. 13, no. 3, pp. 235-242, 1977.

[19] al, A.P.S.e., Technical report, IEEE 802.11n working document 802.11-03/802r23, May 2004.