

Quality of Service- and Fairness-Aware Resource Allocation Techniques for IEEE802.11ac WLAN

SUMMERA NOSHEEN¹ AND JAMIL YUSUF KHAN¹, (Senior Member, IEEE)

School of Electrical Engineering and Computing, The University of Newcastle, Callaghan, NSW 2308, Australia

Corresponding author: Summera Nosheen (summera.nosheen@uon.edu.au)

ABSTRACT Increased demand for high-quality multimedia applications over IEEE802.11 networks has further motivated service providers to improve the QoS (quality of service) for all types of network services. IEEE802.11ac is one of the advanced WLAN standards that provides wideband transmission channels and higher data rates, which are vital to support very high-throughput services. However, due to the use of the CSMA/CA-based random access packet transmission technique, the QoS and service fairness criteria cannot always be maintained, particularly for variable bit rate services. This paper proposes new packet scheduling algorithms to support consistently high QoS and intra-service fairness for high data rate video streams. We name the proposed algorithms *FRA – TXOP* (Flow Rate Adaptive Transmission Opportunity) and *EAS* (Enhanced Access Scheduler). These algorithms offer appropriate QoS and fairness to video services in a CSMA/CA network. The *FRA – TXOP* algorithm allocates the transmission resources according to the channel congestion levels experienced by video flows to improve and maintain consistent QoS values. At the same time, the *EAS* algorithm offers service fairness among the video streams by appropriately scheduling them on the downlink channel. The *EAS* algorithm is implemented on top of the *FRA – TXOP* algorithm to achieve high QoS and service fairness among video streams. An NS-3 simulation model is developed to evaluate the performance of the proposed algorithms. The performance analysis shows that the proposed algorithms enhance the video flow QoS in terms of throughput and packet delay as well as improve the fairness among the video flows compared to other TXOP algorithms.

INDEX TERMS IEEE802.11ac, QoS, video flows, TXOP, resource allocation, service fairness.

I. INTRODUCTION

With the increasing use of high data rate services such as HD/UHD (high-definition/ultra-high-definition) video streaming, VR (virtual reality) applications, live sports streaming, multi-player online gaming, and other media time-sensitive applications in IEEE802.11 WLANs (wireless local area networks), the demand of QoS-based packet schedulers is also increasing. Cisco has predicted that video traffic will account for more than 82% of internet traffic by the year 2021 [1]. According to their prediction, the IEEE 802.11 WLANs will carry the majority of data rate traffic since that network is considered the de facto access network for the internet. It is also predicted that the consumption of video content will increase by 32% from 2016 to 2021. In this context, due to the explosive growth of video and streaming services, it is essential for WiFi networks to support

an appropriate level of QoS and inter/intra-service fairness for multimedia applications. Conviva [2] finds that when a video streaming application freezes due to reduced network throughput, one-third of the users feel the interruption to be intolerable and immediately stop following the live session. In bandwidth-limited networks with shared resources, offering appropriate QoS and adequate network transmission capacity is a challenging task. Thus, it is essential that traffic flows with stringent QoS requirements should be appropriately prioritised by adaptively allocating required network resources. To adaptively allocate transmission resources and maintain QoS and fairness among video flows, it is necessary to develop new quality-aware resource allocation techniques. This paper presents new resource allocation techniques for the IEEE802.11ac WLAN to achieve the above objectives.

The 802.11 standard [3] has evolved over the last two decades to introduce advanced PHY (physical) and MAC (medium access control) layer techniques to satisfy the necessary QoS requirements of time-sensitive VHT (very

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high throughput) services. The new PHY layer technologies such as beam-forming, adaptive modulation and coding, channel bonding, and MIMO (multiple-input-multiple-Output) techniques are beneficial to support VHT services. The DL-MU-MIMO (downlink multiuser-MIMO) technique introduced in the 802.11ac standard enables APs (access points) to support high data rate services with adequate transmission resources. The 802.11ac air interface can support up to eight different spatial streams by using 8 x 8 MIMO antennas. The maximum data rate of the standard can reach 7 Gbits/sec using the 160 MHz transmission band [4]. However, various packet overheads of the protocol stack, the CSMA/CA (carrier sense multiple access/collision avoidance)-based contention mechanism, and the frame-by-frame ACK (acknowledgement) technique limits the utilisation of available transmission capacity. The 802.11ac standard introduced several new MAC layer enhancements, including the frame aggregation scheme, the A-MSDU (aggregated MAC service data unit), the AMPDU (aggregated MAC protocol data unit), channel bonding, and the BACK (block acknowledgement) to address the lower channel utilisation issue. Some of these techniques were initially developed for the 802.11n standard and were further enhanced and adopted in the 802.11ac standard. These enhancements can support improved network throughput and reduce the overhead requirement. The BACK feature increases MAC layer efficiency by reducing the control frame overhead and mandatory inter-frame spacing durations. Multiple transmitted packets can be acknowledged using a single BACK frame instead of individual ACKs. Another key feature of the 802.11ac standard is the TXOP (transmission opportunity) feature developed for the 802.11e standard [5]. This feature provides a unique opportunity for a station after winning the channel access using the CSMA/CA protocol to transmit multiple frames back-to-back in a contention-free mode within a single TXOP period. The TXOP feature is useful for high data rates and time-constrained data transfer services since it reduces the packet collision probability. A TXOP duration limit is also introduced in the standard to maintain service fairness among competing users.

Nevertheless, it is challenging to offer high QoS to every end-user in a random-access network due to the non-deterministic resource allocation techniques used by the CSMA/CA MAC protocol. It is more likely that transmission resources will be unevenly distributed across the network due to the contention process and the backoff delays introduced by the CSMA/CA protocol. Although the TXOP feature of the 802.11ac MAC protocol can improve the throughput, video terminals still need to compete for channel access using the CSMA/CA protocol. The contention process can increase packet transmission delay and losses affecting the quality of transmission as well as service fairness.

Many researchers have worked to improve the QoS of individual video flows in WLANs [6]–[8], and [9]. The techniques [6]–[8] can maintain video streaming quality by dropping low-priority video frames and allocate additional

bandwidth to high-quality video frames. However, prioritising high-quality video frames in backlogged conditions can deteriorate end-users' visual quality due to the interruption in video streaming. Moreover, these approaches did not consider the fairness problem among the competing video streams. In [9], the proposed scheduler maintains the QoS of flows from different traffic classes while considering the application traffic's critical nature and its maximum allowable delay to prevent local network congestion. In addition, in [10], [11], the video scheduling mechanism measured the network congestion level using queue size and applied it to allocate the TXOP duration. The TXOP limit is extended by estimating the current queue size and taking into account all MAC overheads. However, for real-time network flows, the network congestion is determined by more than one factor, such as the number of packets residing in the MAC queue waiting for service and the current video transmission rate. In this paper, we focus on the adaptive scheduling of transmission resources so that all flows meet the necessary QoS and fairness requirements. To address these issues as mentioned above, we proposed two new resource allocation algorithms, namely, the *FRA – TXOP* (Flow Rate Adaptive Transmission Opportunity) and *EAS* (Enhanced Access Scheduler) algorithms [12]. The significant contributions of the proposed resource schedulers are as follows.

- The algorithms monitor channel conditions and prioritise time-sensitive traffic sources to allocate the appropriate transmission resources to maintain QoS and fairness.
- The *FRA – TXOP* algorithm detects network congestion to allocate appropriate network resources to each video flow, maintaining the QoS of video flows.
- The *EAS* algorithm offers equitable channel access to all video flows based on individual flow QoS requirements and improves inter-service fairness.
- The primary objective of the above algorithms is to reduce contention levels and wasted channel bandwidth to improve effective channel utilisation.

The paper is organised in the following manner. Section II reviews the related research literature and analyses different scheduling techniques proposed to improve the QoS of video streams. Section III describes the proposed algorithms. Section IV explains the traffic model with characteristics. Section V presents the network architecture and analysis of the simulation results. Section VI concludes the paper.

II. RELATED WORK

In recent years, network researchers have concentrated on developing new resource allocation techniques to support VHT traffic sources in IEEE802.11 networks. In this section, we review relevant techniques that have been proposed to improve the effective capacity of 802.11 networks by reducing the contention levels introduced by the CSMA/CA protocol.

Chang *et al.* [6] proposed the MPCA (multi-polling controlled access) technique to implement a TDMA (time

division multiple access)-like contention-free mechanism to support the QoS of video traffic by ensuring a tolerable video frame delivery delay. In the proposed mechanism, a station can send an additional traffic stream (*ADD-TS*) request to an HC (hybrid controller) to initiate data transmission. The HC examines the service load and admission control procedure to determine whether the request can be serviced. The HC broadcasts a *CP – Multipoll* frame to all accepted stations. However, the exchange of control signals at the start of every data transmission creates additional transmission overhead. The authors also proposed another algorithm to efficiently share transmission resources among video and non-video data streams. The algorithm adapts the video quality according to the packet collision rates. When the collision rate is low, the algorithm transmits all I, P, and B frames generated by the coder. As the network collision level increases, the algorithm reduces the transmission rate of I and P frames, and additional transmission resources are allocated to non-video traffic to maintain inter-service fairness. At higher collision rates, the video quality is further reduced to transmit only I frames. The techniques presented in [6] require the continuous exchange of control messages for every packet transmission. To overcome the high transmission overhead, Shamieh and Wang [13] proposed a solution to reduce transmission capacity consumption and time delay introduced by control messages. The authors employed a new technique to embed control data within the payload. The embedded data are hidden on a packet-by-packet basis. The reduction in the exchange of messages preserves the transmission bandwidth, which is utilised to improve the QoS of real-time multimedia flows. Nevertheless, the technique introduces data transmission errors due to an increase in payload size, causing a decrease in network throughput.

Wu *et al.* [7] proposed algorithms to mitigate sporadic frame drops caused by random packet losses. As packet re-transmissions using TCP (Transmission Control Protocol) may not achieve the stringent delay constraints of multimedia services, streaming quality can be maintained by appropriately dropping the lower-weight video frames to conserve bandwidth for higher-priority video frames. Their research claimed that selecting some high-quality video frames is more favourable than lowering the video quality using rate-adaptation strategies. The proposed techniques achieve a higher delivery ratio of the more critical I frames under tighter delay constraints to improve the average video quality. However, the end-users may lose interest due to a consecutive drop in low-quality video frames, which affects the video frame sequencing, and causes the video to be played with missing frames. Wu *et al.* [8] proposed a resource scheduling algorithm called PERES (partial reliability-based real-time streaming) for streaming HD real-time video over mobile devices. The proposed technique adjusts the timeout and the maximum number of retransmissions to minimise video distortion. The proposed technique balances the trade-off between delay and reliability. The authors implemented a partial reliable transfer and controlled the buffer level by

substituting the occupied sender buffers with high-priority video packets. However, prioritising high-priority and discarding low-priority video packets could interrupt the live video sequencing, and the user may lose interest in watching the video.

Chang *et al.* [10] proposed a scheduling policy for VBR (variable bit rate) video streams in the 802.11e HCCA method. Using an HCCA-based algorithm, the HC makes a reservation for intended stations by sending them a polling frame. When a reserved STA obtains the transmission resource using the EDCA parameters, it transmits the data held in the MAC queue as well as appends the remaining QS (queue size) value to the last transmitted packet. Using this information, the HC measures a definite length of TXOP duration. The HC also calculates the tolerable delay of each STA to update its TXOP timer. The simulation results indicated the satisfactory throughput and delay performance of the proposed algorithm. For example, for a video file, namely, Jurassic Park I, the proposed algorithm successfully maintained the estimated service interval of 141.69 msec. However, appending the QS on the last packet of each frame could create transmission overhead in the system as frequently added data are carried as on every previous packet of the video frame. Moreover, allocating transmission resources based solely on QS is not a realistic decision: in a real-time network, network congestion is determined by more than one parameter, such as the traffic arrival rates and current video transmission rate.

Kuo *et al.* [14] proposed a link adaptation mechanism for H.264/SVC-coded video streams by balancing between the video buffering ratio and the playback bit rate of a video stream. A lower MCS (modulation and coding scheme) value and a higher retry limit are used to generate robustness against packet loss and maintain playback smoothness. On the other hand, a higher MCS value and a lower retry limit are options for reducing the buffer ratio, which is the time spent on buffering in a live session, to decrease the total time of buffering and playing. The proposed technique achieved a low buffering ratio and maintained the playback bit rates. However, as shown by simulation results, when the RSSI (received signal strength indicator) is less than -80dBm, the proposed algorithm offers lower playback bit rates, causing degradation of video QoS.

At higher congestion levels, the proposed scheduling techniques in [6]–[8], and [14] reduce the packet collision rates by considering a decline in video quality, which may diminish the user interaction and interest with the streaming session. In contrast, our proposed scheduling algorithm reduces network congestion levels and packet collision rates by effectively maintaining higher transmission rates and packet delivery ratios.

Liu *et al.* [15] proposed an algorithm to allocate the transmission resources to video traffic flows based on an estimated deadline for each packet. The deadline estimation for a video packet can be used in two ways: first, to allocate bandwidth resources to a video frame before its deadline

expires; and second, to allocate transmission resources to other traffic sources when there is no strict deadline for video packets. This study focused on improving the performance of video transmission by optimising the CW (congestion window). The proposed technique provides guaranteed QoS for video packet transmissions. However, the deadline estimation method cannot offer guaranteed QoS but a reasonable throughput for background and best-effort services because the allocation of transmission resources to other traffic sources strictly depends on the estimated deadline of the video packet. In contrast, our proposed resource allocation technique does not set any strict deadline, such that background or best-effort applications can be deprived of channel bandwidth. Instead, our algorithm reduces the network backlog and packet drop to avoid the waste of channel bandwidth, which in turn can be utilised for non-multimedia sources.

Zhou *et al.* [16] proposed a Bi-level resource allocation algorithm to avoid stalling events among video users. A token-counter mechanism is adopted to calculate the priorities of video queues. A video terminal with the largest token counter has the highest priority and needs to be served on an urgent basis. The algorithm also offers throughput fairness to certain video users. The algorithm allocates a fixed TXOP duration to video users irrespective of network conditions such as traffic volumes and collision rates; the channel is always allocated for a fixed duration. In comparison, our proposed resource scheduling algorithm adapts the transmission capacities based on the network conditions and, at the same time, maintains QoS and inter-service fairness among different network users.

Zorba *et al.* [17] proposed a cross-layer queue length scheduler with heterogeneous traffic requirements for multi-user wireless downlinks. The BS scheduler follows an opportunity transmission strategy and selects a downlink user with the maximum SNR (signal-to-noise-ratio) value among other users. The IP packets are placed at the different priority queues in the proposed strategy before entering into the DLC (dynamic link control) queue. The high-priority packets are placed at the beginning of the queue, followed by the lower-priority packets. The DLC approach is applied to select the user with the best channel conditions to maximise the system's average throughput. When the network users satisfy the maximum allowed delay requirement, the DLC queue length is extended so that greater numbers of users enter the DLC queue. Conversely, if the maximum allowed delay requirements are barely satisfied, then the DLC queue's length is shortened. The proposed queue-based mechanism increases the system throughput by selecting the user with the best channel conditions; however, it restricts all network users' QoS. On the other hand, our proposed resource scheduler identifies the network users with deteriorating service quality and improves the QoS level by allocating extended transmission capacities in such a way that all network users receive similar QoS fairness.

A review of the above works shows that the aforementioned algorithms offer some QoS improvements, such as

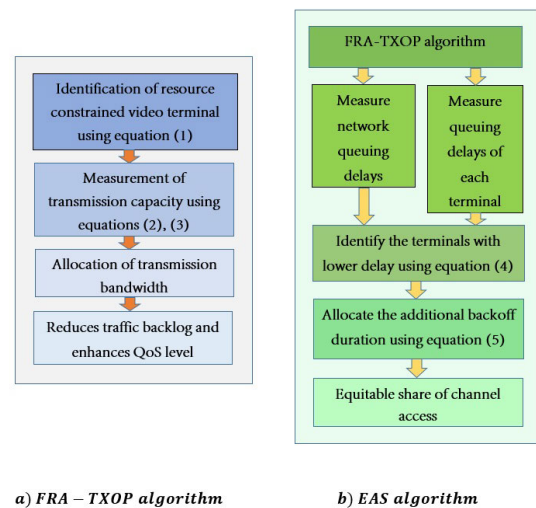


FIGURE 1. Technical building blocks of proposed resource schedulers.

providing high-throughput and reliable delivery of video streaming applications. However, none of the works considered both QoS improvements and inter-service fairness among competing video flows at the same time. Our proposed algorithms differ from those described above. The contribution of our work is two-fold: the proposed algorithms offer high-throughput and reduced latency for video traffic flows as well as maintain fairness among network users. The following section describes the proposed algorithms in detail.

III. QUALITY OF SERVICE-BASED VIDEO PACKET SCHEDULERS

This section introduces new MAC layer resource allocation algorithms for the IEEE802.11ac network to support high QoS and fairness to downlink video streams. A QoS fairness resource scheduler allows network resources to be shared equitably amongst all users in a network while maintaining the required QoS. The service quality of data-intensive applications degrades largely due to network congestion, which reduces the effective throughput of a network. Additionally, in a random-access network, not all transmission terminals receive an equitable distribution of network resources. To address these issues, a resource scheduler should identify resource-constrained transmitting devices and equitably allocate appropriate transmission resources. In a contention-based network such as the CSMA/CA network, different terminals may experience varying levels of congestion. To address the QoS and fairness issues, we propose two new resource scheduling algorithms for 802.11ac networks. The first algorithm is the *FRA-TXOP* algorithm, which identifies the transmission resource-constrained video terminals and allocates appropriate transmission resources to improve the QoS resource-constrained video streams. The second proposed algorithm, *EAS*, offers fairness to all competing video streams on the downlink. Both algorithms are implemented using the TXOP parameter of the CSMA/CA MAC protocol. A macro description of the proposed algorithms is shown in Figure 1, explaining the key features of the proposed resource

TABLE 1. List of notations used in the proposed video packet scheduler.

Notations	Description
T_B	observation period
N_b	number of beacon intervals in an observation window
M_v	number of active video users
CI_q	congestion index value of the q_{th} video flow
$Q_{q,i}$	MAC queue size of the q_{th} video terminal in the i_{th} beacon interval, expressed in number of packets
\overline{Q}_q	average MAC queue length of the q_{th} video terminal over N_b , expressed in number of packets
$FR_{q,i}$	past flow rate of the q_{th} video terminal in the i_{th} beacon interval, expressed in packets per second
\overline{FR}_q	average past flow rate of the q_{th} video terminal over N_b , expressed in packets per second
$T_{k,i}$	packet delay of any network flow k in the i_{th} beacon interval, expressed in seconds
\overline{T}_q	average packet delay of the q_{th} video terminal over N_b , expressed in seconds
$TXOP_D$	default TXOP duration (3.008 msec) defined in IEEE802.11ac protocol [18]
$TXOP_E$	length of extended TXOP duration in μsec
$TXOP_q$	measured TXOP duration of the q_{th} terminal in μsec
N_{CIH}	number of video terminals with a high congestion index
σ	standard deviation of queuing delays
$T_{backoff_2}$	length of any additional backoff duration in seconds
$T_{backoff_{2,q}}$	intended additional backoff delay for the q_{th} video flow in seconds

schedulers. As indicated in Figure 1(a), the *FRA – TXOP* algorithm identifies and mitigates traffic backlog, thereby improving video QoS levels. Similarly, as mentioned in Figure 1(b), the *EAS* technique is developed on top of the *FRA – TXOP* algorithm; it offers equitable channel access to different network terminals, improving the individual user's visual experience. These protocols are described in detail in the following sections. Table 1 lists the notations used to describe the proposed video schedulers.

A. FRA – TXOP ALGORITHM

This algorithm utilises the TXOP and BACK features of the 802.11ac MAC standard. The *FRA – TXOP* algorithm adapts the TXOP duration of a network terminal according to its experienced congestion level. The *FRA – TXOP* algorithm identifies congestion levels of each contending network terminal and allocates appropriate transmission resources to improve the QoS of different flows. This algorithm utilises a new parameter known as the congestion index (CI) to determine the congestion level experienced by each terminal. The CI is the ratio of the MAC queue length measured over a T_B duration to the past flow rate measured over the same period T_B , as shown in Figure 2. The T_B observation period consists of N_b beacon intervals.

The queue length and the flow rates are expressed in the number of packets. Equation (1) is used to measure the CI of the q_{th} video terminal. The congestion of resource-constrained terminals can be easily detected by equation (1). Since the video packets will arrive in the queue, the queue length of a congested terminal will gradually increase if the terminal cannot obtain sufficient transmission capacity. Figure 2 shows an example in which queue lengths and flow rates are measured over the beacon interval in a staggered manner. We utilised the staggered measurement technique

because in a random-access network, an appropriate delay needs to be introduced to experience the effect of network congestion due to the variable resource allocation technique. However, a longer observation period could affect the congestion mitigation response time, which may reduce the effectiveness of any resource allocation algorithm. As indicated in (1), CI is the congestion index measurement of the q_{th} terminal, $Q_{q,i}$ and $FR_{q,i}$ are the MAC queue size and past flow rate of the q_{th} video terminal at the i_{th} beacon interval, respectively. The mean values of these two entities are calculated over an

$$CI_q = \frac{\sum_{i=1}^{N_b} \frac{Q_{q,i}}{N_b}(T_B)}{\sum_{i=1}^{N_b} \frac{FR_{q,i}}{N_b}(T_B - 1)} = \frac{\overline{Q}_q(T_B)}{\overline{FR}_q(T_B - 1)} \quad (1)$$

observation window of N_b beacon intervals. \overline{Q}_q is the average MAC queue length of the q_{th} video terminal at observation period T_B , and $\overline{FR}_q(T_B - 1)$ is the average flow rate for the past interval $T_B - 1$.

In 802.11 networks, the data terminal will attempt a transmission and wait for an ACK from the receiver. If no ACK is received, a collision due to the unavailability of transmission capacity is assumed, and re-transmission is scheduled. When the collision level in a network increases, the available channel capacity decreases due to wasted channel bandwidth caused by the re-transmissions. Terminals with higher CI values indicate that those terminals do not receive adequate channel capacity to support their QoS requirements. Threshold values can be used to identify the congested terminals. The algorithm utilises condition (2) to define the TXOP duration. A CI threshold value of 1 is utilised to differentiate between a congested and non-congested terminal. If

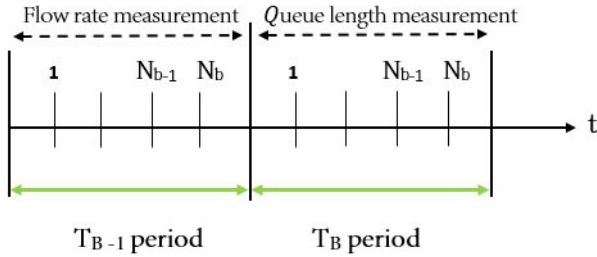


FIGURE 2. Congestion index (CI) measurement procedure.

$CI_q \leq 1$, then the corresponding terminal is considered “non-congested” since the short-term average transmission capacity is higher than the average queue length. In this case, the default TXOP duration ($TXOP_D$) is assigned. IEEE802.11ac protocol defines standard TXOP duration limits for different services [18].

However, if the $CI_q > 1$, then the corresponding terminal is not receiving sufficient transmission capacity; it is said to be “congested”. To address this congestion problem, the *FRA – TXOP* algorithm adapts the allocated TXOP duration using the equation (3). The TXOP duration is adaptively adjusted based on the congestion level. According to equation (3), the TXOP duration is extended according to the congestion level, as indicated by the CI. When a terminal is lightly congested, there will be a slight extension in the TXOP duration to relieve the congestion level and vice versa. The allocated TXOP duration is adjusted to transmit a higher number of packets per TXOP period.

In equation (3), the rate of increase is moderated by the number of contending video terminals M_v . M_v represents the number of contending video terminals in the network. The parameter M_v offers the degree of fairness by limiting the rate of increase. For example, if the terminal experiences a CI value of 1.5, its TXOP duration will be $1.4 \times TXOP_D$. A highly congested terminal of $CI = 3$ will receive a $1.6 \times TXOP_D$ duration. The *FRA – TXOP* algorithm offers a peak transmission rate, which could alleviate the congestion problem and maintain the necessary QoS.

$$TXOP_q = \begin{cases} TXOP_D & \text{if } CI_q \leq 1 \\ TXOP_E & \text{if } CI_q > 1 \end{cases} \quad (2)$$

$$TXOP_E = \left[1 + \frac{1}{M_v} \text{Max}(2, CI_q) \right] \times TXOP_D \quad (3)$$

B. EAS ALGORITHM

The second QoS-enabled resource allocation algorithm is *EAS*, which was developed to improve the fairness of the *FRA – TXOP* algorithm. The *EAS* algorithm is implemented on top of the *FRA – TXOP* algorithm. Although the *FRA – TXOP* algorithm can identify resource-constrained terminals and allocate additional resources, the channel access delay in a CSMA/CA network can also increase due to the contention process. This delay can be reduced if fewer terminals are trying to access the channel at a given time. To identify the terminals that are experiencing longer

access delays, the algorithm compares each terminal’s queuing delay with the average network queuing delay, as shown in equation (4). The average network packet delay and standard deviation (σ) of all video flows are calculated over N_b beacon intervals. The $T_{k,i}$ is the packet delay of any network flow k at the i_{th} beacon interval. For all values of t , the difference of average network delays and standard deviation value will always be greater than zero.

The variable \bar{T}_q is the mean packet delay in seconds of the q_{th} terminal, which can be computed by taking the average of all instantaneous packet delays over an observation window of N_b intervals. The packet delay value equals the packet size sent in bits over a transmission rate in bits per second.

$$T_{backoff_{2,q}} = \begin{cases} T_{backoff_2} & \text{if } \bar{T}_q \leq \frac{\sum_{k=1}^{N_b} T_{k,i}}{N_b} - \sigma \\ 0 & \text{if } \bar{T}_q > \frac{\sum_{k=1}^{N_b} T_{k,i}}{N_b} - \sigma \end{cases} \quad (4)$$

$$T_{backoff_2} = N_{CIH} \times \frac{\frac{\sum_{k=1}^{N_b} T_{k,i}}{N_b} - \sigma}{\bar{T}_q} \quad (5)$$

Equation (4) compares the queuing delay of each terminal with the difference of average network queuing delay and its standard deviation. By identifying the terminals based on this comparison, the algorithm introduces an additional backoff delay for the terminals experiencing lower queuing delays. The identified video terminals need to wait for an additional backoff period of $T_{backoff_2}$ before they can attempt channel access using the CSMA/CA procedure. In contrast, video terminals with higher average video packet delay will be allowed to access the channel immediately by discriminating amongst the contending flows. Equation (5) calculates the $T_{backoff_2}$ duration for low-delay video terminals, where the variable N_{CIH} indicates the number of video flows with high CI values ($CI > 1$). The parameter N_{CIH} controls the length of the additional backoff period ($T_{backoff_2}$). When more video terminals are congested, the low-delay terminals need to wait slightly longer time so that network resources can be shared uniformly. Conversely, if the value of N_{CIH} is small, the low-delay experiencing terminals will wait for a short time to access the channel. Therefore, this approach enables equitable channel access time amongst the competing video streams. The justification for introducing an additional back-off access delay is to offer priority to congested terminals, facilitating fairness among competing video streams. This additional backoff delay can be easily integrated with the CSMA/CA protocol, as given by equation (6). Equation (6) explains the traditional packet transmission delay in the CSMA/CA protocol and an additional backoff period introduced by the *EAS* algorithm.

In CSMA/CA networks, the packet transmission delay is determined by network load conditions and the protocol parameters. Equation (6) shows the approximate average packet transmission delay in a CSMA/CA network. A packet

may require N transmission attempts, of which $N-1$ attempts will be unsuccessful. The equation shows that a packet transmission delay largely depends on the value of N . The packet transmission delay and effective channel capacity can be increased by reducing the number of collisions.

$$T_{packet} = 2T_{DIFS} + T_{backoff} + T_f + T_{SIFS} + T_{BACK} + (N-1)T_{collision} + T_{backoff_2} \quad (6)$$

where T_{packet} is the packet transmission delay, $T_{backoff}$ is the total mandatory backoff delay, T_f is the frame transmission delay, T_{DIFS} , and T_{SIFS} are the DCF inter-frame spacing and short inter-frame spacing durations respectively. T_{BACK} is the transmission delay of the BACK frame, and $T_{collisions}$ is the average delay caused by the re-transmission procedures. $T_{backoff_2}$ is the additional backoff delay introduced by the EAS algorithm to low packet delay terminals to offer inter-service fairness.

The proposed algorithms measure the MAC queue sizes and flow rate information of video terminals and send these values to the access point by piggybacking with data frames on each beacon interval. The beacon interval of 50 msec is used in the simulation work, as listed in Table 2. The information is transferred every 50 msec and is averaged over a T_B interval, indicating a fixed-size duration. When all terminals send their data to the AP, it calculates CI values for all active video flows. Next, a sorting process is used to identify the terminal with the highest CI value. As indicated in Figure 3, A is the computing load equalling the time taken to calculate the CI values and complete the sorting process. B is the time needed to determine the TXOP period based on equations (2) and (3). C is the time taken to send the information regarding TXOP selection from an access point to the network terminals by using the standard beacon frame of an 802.11 WiFi LAN. One byte is required to express this Boolean value ‘‘Send TXOP decision’’ to specified video terminals, which signals the selection of a default or extended TXOP period. This information is embedded inside the ‘‘Frame body’’ field of the beacon frame. The information can be easily carried in the frame body as it has enough capacity (2312 bytes) to contain these data. The address of the congested terminal is embedded inside the ‘‘Address 1’’ field of the beacon frame. Therefore, the total computational delay $T_{computational}$ equals the summation of T_B , A , B , and C durations, which is necessary to process the allocation of transmission resources as shown in Figure 3.

The proposed techniques allocate the channel resources based on congestion levels; however, the algorithms utilise the traditional CSMA/CA procedure to reserve the TXOP duration. In the literature, we can find many theoretical analyses to allocate channel resources in random access-based wireless networks, which can be quickly adopted in this work. One of the analytical models based on the IEEE 802.11e EDCA procedure is referenced as [19]. The analytical model explains the effects of the significant QoS features of EDCA, including the impact of contention and the back-

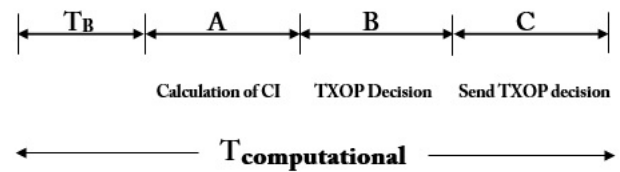


FIGURE 3. Computational delay of the proposed TXOP allocation procedure.

off process. Siris *et al.* [20] presented an analytical model describing the effects of congestion in the scheduling of transmission resources. The analytical framework measures the user’s throughput by considering several factors such as minimum contention window, physical transmission rates, and congestion levels in CSMA/CA-based wireless networks. How much the congestion of a wireless terminal costs with regard to the duration of a successful transmission has also been interpreted. The station with a lower transmission rate can have a longer transmission duration and unacceptable service quality. This analysis can be directly applicable to our research work; it explains how the backlogged conditions deteriorate the video flow rates and QoS levels in CSMA/CA-based wireless networks. The research work also provides solutions to improve video transmission rates, QoS, and fairness levels.

The proposed resource schedulers can be immensely useful in dense networks including cooperate offices, metropolitan areas, seminar halls, lecture rooms, auditoriums, sports stadiums, etc., where several users are streaming multimedia content as HD, UHD, or 3D (three-dimensional space) videos. Different users may experience different levels of congestion and compete for shared resources. The proposed resource scheduling techniques can resolve video stability problems, and resource underutilisation as congestion may waste channel bandwidth. Specifically, these techniques maintain the end-user’s quality of experience by effectively analysing the backlog terminals and allocating need-based transmission resources to eliminate congestion. The proposed algorithms can be deployed on real-time streaming servers to improve the QoS of streaming services such as watching a live cricket match, a movie/series on Netflix, or HTTP live streaming.

IV. TRAFFIC MODEL AND CHARACTERISTICS

In this work, the MPEG4 [21] video model is used to generate video traffic streams on the downlink. The MPEG4-based video traffic generator can generate video streams of different target bit rates. The MPEG4 stream structure consists of three different types of frames I, P, and B. These frames can be grouped and referred to as GOP (group of pictures), as shown in Figure 4. The I frame, referred to as an intra-coded picture, contains a complete image, whereas the P (predicted) frame only contains the changes in the image from the previous frame. The B (bidirectional) frame increases the coding efficiency. The B frames are coded based on a forward prediction from a previous I or P frame and a backward prediction from a succeeding I or P frame. The GOP length is determined by the number of frames between two I frames.

TABLE 2. List of simulation parameters.

PHY layer parameters	WLAN Standard: IEEE802.11ac
Transmission frequency	5 GHz
Transmission bandwidth	40 MHz
SNR(min)	26.7 dB
Transmit power	27.01 dBm
Transmit & Receive gain	1 dBi
Energy Detection Threshold	-90 dBm
Noise level	2.8 dBm
Propagation path loss model	Free space path model
Guard interval	400 nsec
Beacon interval, N_b	50 msec, 20
MCS, $Datarate_{(max)}$,	7, 150 Mbps
Antenna(s)	1 x 1
Application layer parameters	Flow rate: Peak. 18 Mbps, Min. 8 Mbps, Avg. 12 Mbps
HD video streaming	payload size = 2000 Bytes frames per second = 25, GOP length = 12 Standard: MPEG4
Data traffic	Burst rate: 10 packets/sec Packet size: Mean. 1300 Bytes, Min. 900 Bytes, Max. 1500 Bytes
Downlink MAC parameters	EDCA(AC-VI), CW[7,15] Default TXOP length ($TXOP_D$) = 3.008msec BACK count = No. of packets transmitted in a TXOP
Uplink MAC parameters	EDCA(AC-BE), CW[15,1023]
Re-transmission limit	7
Transport Protocol	UDP
No. of Video, Data transmitters	varies (1-10)
No. of AP	1
Simulation length	400 seconds

I frames are independent, whereas other frames depend on past or next frames. The GOP length is set as 12 frames. The bit rate of a stream depends on the number of frames per second and the GOP length. The video packet arrival rate is determined by the video frame rate, image content, and video coder characteristics. In our work, we used a VBR MPEG-4 coder operating at a fixed frame rate, but each video frame’s image content could vary; hence, a flow data rate will be variable. As listed in Table 2, the peak video flow rate is 18 Mbps; however, the video coder will maintain an average flow rate of 12 Mbps and is determined by the trace file’s video data. The packet arrival rate at the MAC layer will depend on the average video stream rate and MAC layer packet size. The mean packet arrival rate at a video terminal can be measured by the following equation (7):

$$N_{p,vid} = \frac{VF_j}{L} \tag{7}$$

where $N_{p,vid}$ is the average number of video packets per video frame, VF_j is the number of bits in the j^{th} video frame, and L is the packet size in bits. The video trace files given in [22] are used to simulate high-quality video flows. A MPEG4 video trace file includes different visual scenes may consist of one or more video objects. Each video object is characterised by temporal and spatial information in the form of shape, motion, and texture. Each video object can be encoded in scalable (multilayer) or non-scalable form (single layer), depending on the application, and represented by the VOL (video object layer). The VOL provides support for scalable coding. A video object can be encoded using spatial or temporal scalability, going from coarse to fine resolution. Depending on parameters such as the available data rate and computational power, the desired resolution can be made available to the decoder. Video stream characteristics, along with other key simulation parameters, are listed in Table 2.

V. SIMULATION MODEL AND RESULTS

A network simulation model was developed in NS-3 [23] to evaluate the performance of proposed protocols. The network architecture is shown in Figure 5. The simulation model transmits multiple downlink video streams using the AP. Each video server supports multiple video streams sent to different receiving stations. The AP also supports data packet transmission in the network on the uplink. Video streams use UDP and IP protocols to transmit data over the 802.11ac link. For this simulation, a high-SNR channel is simulated to study the effect of packet collisions. The simulation results are averaged over 30 different seed values.

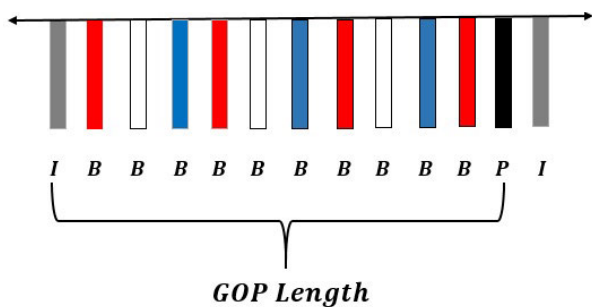


FIGURE 4. A typical GOP length consist of 12 video frames.

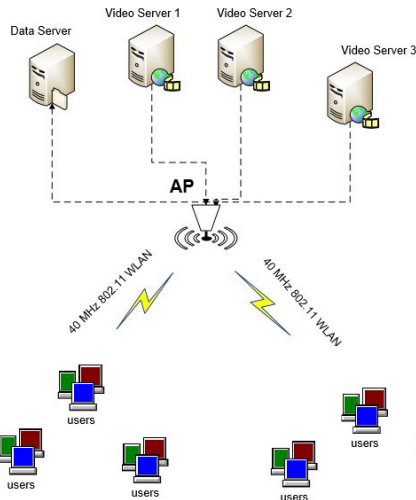


FIGURE 5. Network architecture.

A. PERFORMANCE EVALUATION OF FRA-TXOP AND EAS PACKET SCHEDULERS

In this section, the simulation results and performance analyses of proposed algorithms are presented. The performance of proposed algorithms is compared with that of a similar protocol known as the *Bi – level* resource allocation algorithm published in [16]. The algorithm allocates a fixed-size TXOP duration based on the priorities of video streams. The *Bi – level* algorithm is a resource allocation technique designed to offer a QoS guarantee and inter-service fairness among the competing video flows. Therefore, this algorithm is closely applicable to our proposed techniques. The *Bi – level* algorithm is also implemented in our simulation model, along with the proposed protocols. The performance of algorithms is evaluated using video and data traffic generators in a high-SNR channel so that the MAC protocol performance can be investigated and compared. Figure 6 presents the simulated video traffic characteristics. This figure shows the transmission data rate of several video streams for 120 seconds of the simulation time. Each video flow generates different transmission data rates, which are controlled by video traffic characteristics. The plot shows three separate streams that generate traffic patterns independent of each other.

Figure 7 shows the aggregated downlink throughput of four different packet transmission algorithms for a different number of concurrent video flows. These throughput values were obtained for the MCS-7 link. The plot shows that the standard *Fixed – TXOP* algorithm offers the lowest throughput, whereas the *FRA – TXOP* algorithm provides the highest throughput. Both the proposed protocols offer higher throughput than the default and the *Bi – level* algorithms. The *EAS* algorithm provided a slightly lower throughput than the *FRA – TXOP* algorithm only due to the use of an additional backoff duration to provide fairness among the competing

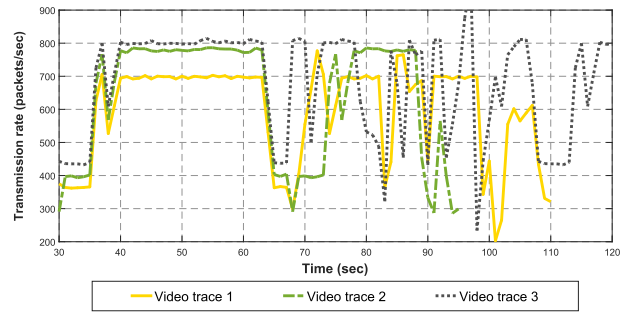


FIGURE 6. Transmission data rates for video traffic flows.

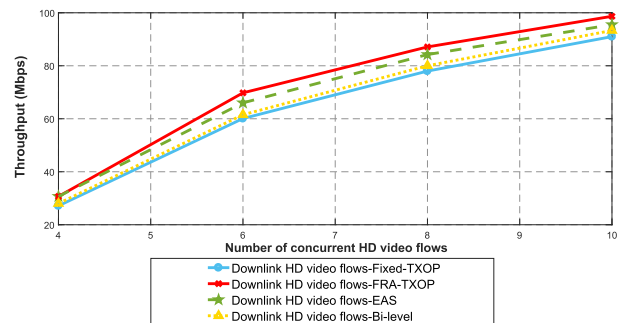


FIGURE 7. Downlink HD video flows throughput analysis of the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

video terminals, as discussed in section IIIB. The *EAS* algorithm is implemented on top of the *FRA – TXOP* algorithm.

The main reason for the improved throughput is explained by the collision statistics presented in Table 3. The table lists the number of re-transmissions caused by packet collisions for the different number of concurrent flows (ρ). The number of re-transmissions in the network increases with the increasing network load. Since we used a high-SNR channel, all re-transmissions in the network are in response to packet collisions. As indicated, both proposed protocols offer lower numbers of packet transmissions than the standard and the *Bi – level* packet transmission protocols. The proposed algorithms reduce the number of re-transmissions, thereby successfully decreasing packet delay and improving overall channel utilisation. Table 3 shows that for eight video flows, the *FRA – TXOP* and *EAS* algorithms attained 25.8% and 24.5% fewer re-transmissions than the *Fixed – TXOP* MAC protocol. Compared with the *Bi – level* algorithm for the same number of video flows, the *FRA – TXOP* and *EAS* algorithms reduced the packet re-transmission rates by 15% and 13.35%, respectively. Each throughput performance of

TABLE 3. Comparison of video downlinks MAC re-transmissions for the four algorithms.

ρ	No. of Retransmission for HD video downlinks			
	Fixed-TXOP	FRA-TXOP	EAS	Bi-level
4	86	54	54	80
6	765	355	376	700
8	3445	2556	2601	3002
10	80852	55686	55765	76543

the four algorithms is analysed in Figure 8. As shown in the plot, the proposed *FRA – TXOP* and *EAS* techniques achieve uniform throughput compared to the *EDCA* and *Bi – level* algorithms for 8 and 12 concurrent users. The referenced *Bi-level* algorithm offers throughput fairness to some users. As illustrated, a throughput drop is observed for certain users; for example, for eight concurrent flows, the 3rd, 5th, and 6th flows could not achieve the same throughput as other users because each TXOP allocation is a fixed-sized process. The *Bi – level* algorithm fails to fulfil the run-time transmission capacity requirements of these flows. The congested terminals need large transmission capacities instead of using a default TXOP duration to establish the required packet flow rate. Similarly, for a twelve-simultaneous-flow scenario, the *Bi – level* algorithm cannot offer equitable throughput to the 1st, 7th, and 10th streams. The *EDCA* algorithm performs the worst among all the algorithms in offering uniform throughput values. It lacks the capability to identify the dynamic traffic requirements due to the random allocation of transmission resources.

Figure 9 illustrates the standard deviation measurement concerning mean throughput values for twelve simultaneous flows. As indicated, the *FRA – TXOP* and *EAS* techniques offer a shorter standard deviation than the other two algorithms, confirming the similarity in individual user’s throughput. The *EAS* algorithm achieves a lower standard deviation than the *Bi – level* and *EDCA* techniques but a slightly higher value than the *FRA – TXOP* algorithm. This result is due to the inclusion of additional backoff delays to video terminals experiencing a lower queueing delay, as described in Section IIIB. The channel utilisation for 8 and 12 simultaneous flows is plotted in Figure 10. The channel utilisation is measured by dividing the achievable aggregated throughput by the theoretical maximum throughput: an aggregated throughput is obtained by combing all individual users’ throughput values as provided in Figure 8. The *FRA – TXOP* and *EAS* algorithms achieved the highest system utilisation. The *Bi – level* algorithm attained more channel utilisation than the *EDCA* algorithm, albeit lower than that of the proposed techniques.

The *FRA – TXOP* and *EAS* algorithms reduce the number of re-transmissions due to the lower number of TXOP reservation attempts since the algorithms adaptively allocate the TXOP length based on congestion experienced by each stream. Congested terminals receive longer TXOP durations, as explained by equation (3). Figure 11 shows the average MAC queue lengths of two video flows for the *Fixed – TXOP* and *FRA – TXOP* algorithms. The plots clearly show that the average MAC queue length is much shorter for the *FRA – TXOP* algorithm due to the adaptive allocation of channel resources. Figure 11(b) illustrates that from 240 sec of simulation time onward, for the *Fixed – TXOP* algorithm, the queue length increases significantly; this occurs because irrespective of the network condition, the traditional algorithm allocates fixed-size resource periods throughout the simulation time. On the other hand, the *FRA – TXOP* algo-

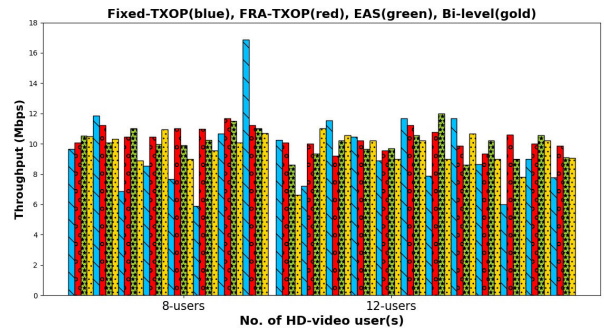


FIGURE 8. Individual downlink HD video flows throughput analysis of the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

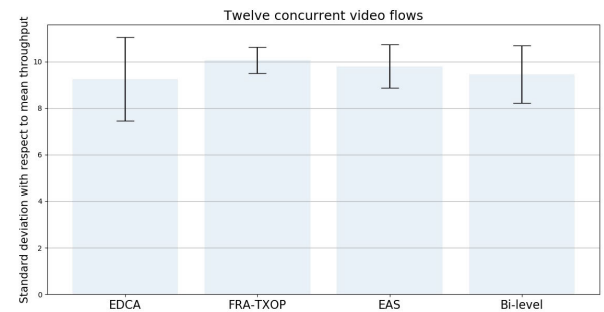


FIGURE 9. The standard deviation measurement concerning mean throughput values for twelve simultaneous flows.

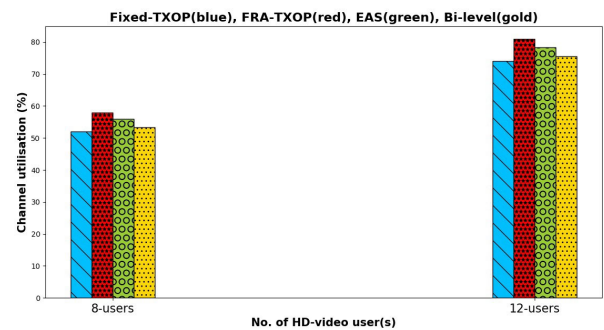


FIGURE 10. Analysis of system utilisation for the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

rithm adjusts the resource capacities, i.e., the TXOP duration, based on network traffic loads.

The allocation of the TXOP duration of two video flows using the *FRA – TXOP* algorithm is shown in Figure 12. In this simulation model, a default TXOP duration of 3008 μ sec is used. As indicated in the plot, the TXOP durations are dynamically adapted with respect to backlog conditions. For example, for video flow 1 between 144 sec and 164 sec, a default value of the TXOP period is allocated due to the low congestion levels. On the other hand, as soon as a backlog condition is identified, the TXOP duration is adapted appropriately according to the experienced congestion level, as shown between 180 sec and 188 sec. For video flows 1 and 3, the extended TXOP duration grows as long as 5800 μ sec.

Figure 13 compares the PDR (packet delivery ratio) of all algorithms for the different numbers of video flows. The PDR

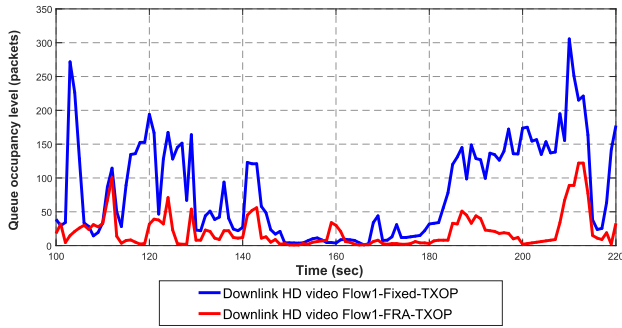


FIGURE 11. Queue occupancy-level analysis of HD video Flow 1 and Flow 3.

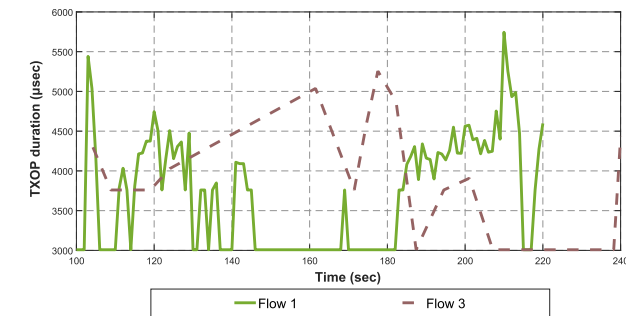
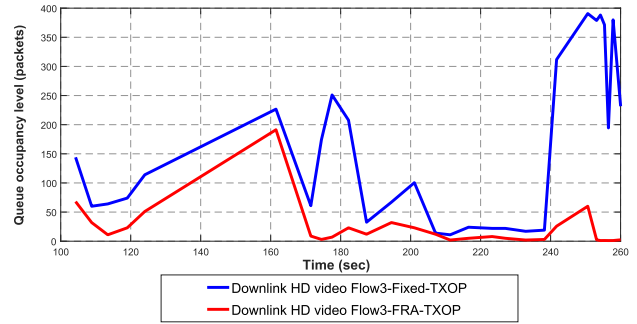


FIGURE 12. Adapted TXOP durations with respect to time.

FIGURE 13. Analysis of PDR (%) delay for the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

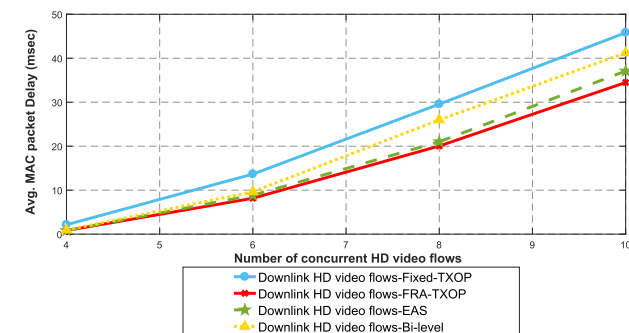
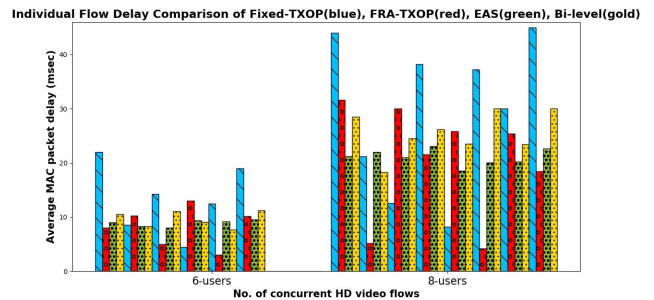


FIGURE 14. Analysis of MAC packet delay for the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

is the ratio of the number of successfully delivered video packets to the total number of generated video packets at

FIGURE 15. Individual HD video flow packet delay distribution.

the transmitter. The plot shows that the proposed protocols consistently offer higher PDR values compared to the *Fixed – TXOP* and *Bi – level* algorithms for the different numbers of video flows. Figure 14 illustrates the average packet delay of four 802.11ac packet transmission algorithms for a different numbers of video flows. As expected, the proposed algorithms offer the lowest packet delays for all load conditions. The proposed algorithms provide shorter delays due to the allocation of appropriate transmission capacities under different loading conditions.

Figure 15 shows the packet delay distribution of individual video flows for 6 and 8 consecutive downlink video flows. This graph is presented here to show packet delay fairness. As indicated in the plot, for the *Fixed – TXOP* algorithm, different individual flows experience different delay values. In contrast, the *EAS* algorithm’s plot shows that all flows experience lower and consistent packet delays. The consistency is much higher for the *EAS* algorithm than the *FRA – TXOP* algorithm. The *EAS* algorithm achieved higher fairness due to the introduction of an additional backoff delay for low average packet delay terminals, as explained earlier in section IIIB. The *EAS* algorithm pays a minimal price to achieve this fairness, as shown in Figure 14: a slight increase in the packet delay value compared to the *FRA – TXOP* algorithm.

Figure 16 shows the $T_{backoff2}$ delays for several flows. As shown in the plot, the maximum backoff delay increased to 8 msec. This additional backoff delay maintained an equitable video frame delay for all users compared to the standard 802.11ac MAC represented by the *Fixed – TXOP* values.

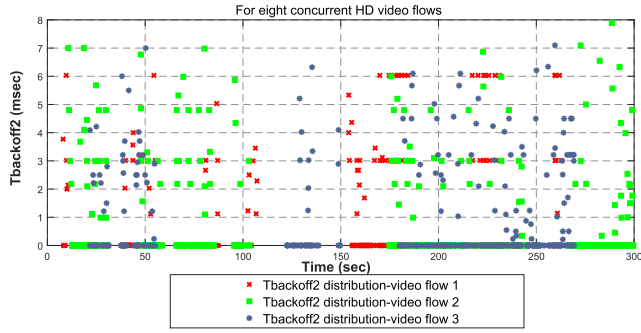


FIGURE 16. Distribution of $T_{backoff_2}$ intervals of three HD video flows.

The *EAS* algorithm also outperformed the *Bi – level* algorithm, where most of the flows experienced longer video frame delays. Note that a video frame consists of multiple MAC packets and TXOP durations.

QoE fairness is further elaborated in Figure 17 in terms of throughput and packet delay values. The QoE can be described in terms of QoS and fairness. A high-QoE condition can be achieved when consistent high QoS and fairness can be obtained for all competing flows. In this work, the QoS fairness values are evaluated using the standard Jain fairness index (JFI) [24]. The JFI values are calculated using equation (8).

$$JFI = \frac{\left[\sum_{m=1}^{M_v} (X_{videoflow})_m \right]^2}{M_v \times \sum_{m=1}^{M_v} (X_{videoflow})_m^2} \quad (8)$$

where the variable $X_{videoflow}$ represents the mean throughput or packet delay of a video flow and M_v is the total number of active HD video terminals, as explained earlier in section III. As shown in Figure 17a, the *EAS* algorithm achieves higher packet delay fairness with a smaller standard deviation (σ). On the other hand, the *Bi – level* and *Fixed – TXOP* algorithms show higher σ values, indicating lower delay fairness, i.e., different flows experience different QoS outcomes. The plot also shows that the *EAS* algorithm, which is used in conjunction with the *FRA – TXOP* algorithm, offers the highest and most consistent delay fairness across all load conditions, which confirms the results presented in Figure 14. Figure 17b plots the throughput fairness indexes. This plot shows that the *FRA – TXOP* and *EAS* algorithms achieve higher throughput fairness with a small deviation from the respective mean values. The results indicate that the proposed algorithms can fairly share the available bandwidth among different video users. Conversely, the *Bi–level* and *Fixed–TXOP* algorithms cannot maintain a fair throughput share; in particular, under high-traffic conditions, the fairness value is significantly reduced. The *Fixed – TXOP* and *Bi – level* algorithms cannot respond to variable network traffic load conditions, resulting in lower QoE conditions. It has been shown in previous results that the proposed algorithms offer high packet delivery ratios,

which is another indication of a high QoS. Figures 7-17 show the effectiveness of the *FRA – TXOP* and *EAS* algorithms for providing a consistent high QoS and fairness for concurrent downlink flows under different loading conditions.

After evaluating the performance of the proposed algorithms using objective QoS measures, a subjective study is performed to assess the significance of the proposed algorithms in terms of user engagement with the perceived video quality. User engagement is a qualitative measurement of the user’s interest and interaction in a video session. A user viewing a video clip continuously is a *view*. For instance, a view could be watching a movie trailer clip, an episode of a TV series on streaming services, or a live cricket game. The view-level engagement metric of interest depends on the video play duration and the rate at which the video was encoded, resulting in how frequently the receiving equipment needs to use the buffering option. Subjective results are achieved by measuring the quality of video streaming on the receiver side. In our work, subjective evaluation is performed in terms of time required to deliver video frames, the GOP (group of pictures) distribution delay, and the PSNR (peak signal to noise ratio). Although the PSNR is an objective measurement, it can be related to the subjective quality of a video flow [25]. These subjective parameters are discussed in the following sections.

- **Delay of a video frame** is the time required to deliver an entire video frame at the receiver end. A video frame consists of a number of MAC packets.
- **GOP delay distribution** is the time required to deliver a video message on the user’s screen. A low-GOP delay will enable uninterrupted video content delivery at the receiver. We measured the video message delay in terms of the time required to deliver a GOP, which is a collection of successive frames within a video stream. A typical GOP length is shown in Figure 2.
- **Peak signal to noise ratio (PSNR)** The PSNR value of a video flow is calculated by using equation (9) [26].

$$PSNR = 20 \log_{10} \frac{MAX_{Bit\ rate}}{\sqrt{\mu_{exp} - \mu_{current}}} \quad (9)$$

where $MAX_{Bit\ rate}$, μ_{exp} and $\mu_{current}$ are the maximum transmitted bit rates, expected throughput, and current achieved throughput of a video flow, respectively.

The instantaneous video frame delay distribution of the *Fixed – TXOP* and *EAS* algorithms are presented: in Figure 18. The plot shows that for four concurrent HD downlink flows, the *EAS* algorithm achieves closer frame delay distributions than the *Fixed – TXOP* algorithm. Figure 18b shows that for most of the time, video frame delays of the *EAS* algorithm vary between 20 and 40 msec, whereas for the *Fixed – TXOP* algorithm, the delay figures mostly vary between 25 and 45 msec. The *EAS* algorithm achieves much improved performance for eight consecutive flows, as shown in Figure 19b, where most of the video frames experience delays of between 20 and 250 msec. In contrast,

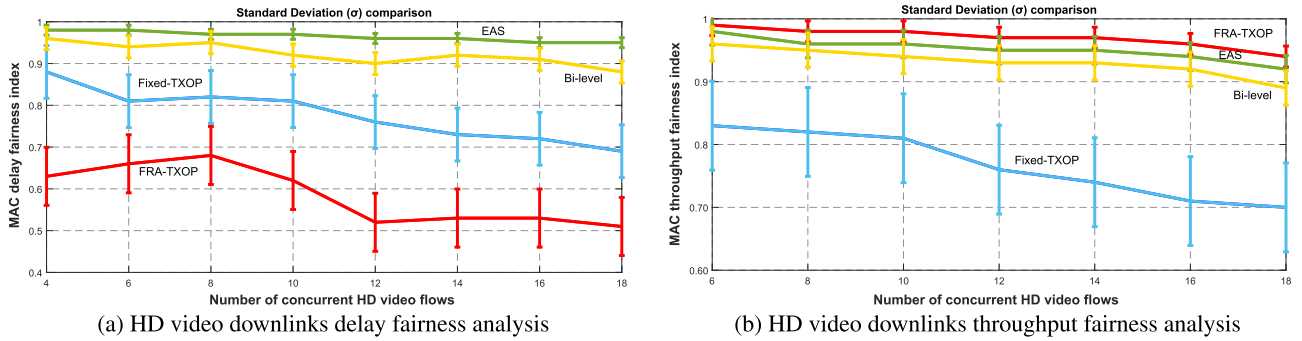


FIGURE 17. Traffic load vs QoS fairness comparison of the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

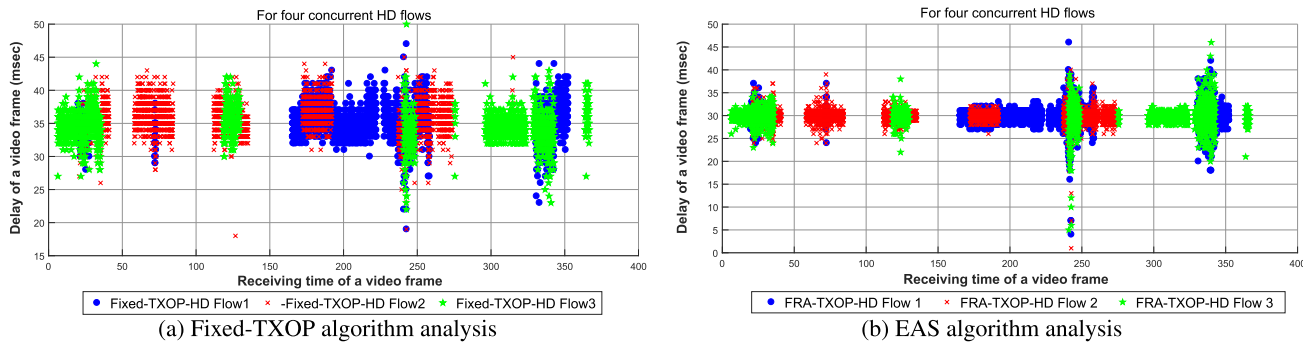


FIGURE 18. Four concurrent HD video flows frame delay comparison.

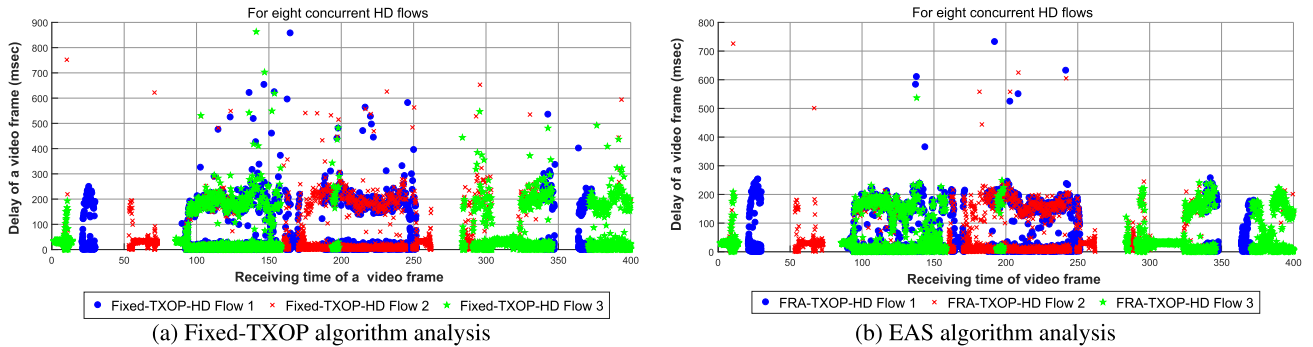


FIGURE 19. Eight concurrent HD video flows frame delay comparison.

for the *Fixed – TXOP* algorithm, the delay distribution varies from 30 and 500 msec.

Figure 20 shows the GOP distribution delay for a different number of flows in terms of the cdf (cumulative distribution function). Figure 20a depicts the GOP distribution delay for a medium-loaded network. The plot indicates that the *EAS* algorithm delivers more than 95% of GOPs in less than 340 msec. At the same time, the standard *Fixed – TXOP* algorithm delivers less than 30% of GOPs within the same delay. Figure 20b shows that for a high-load condition with eight HD flows, the *EAS* algorithm received more than 90% of GOPs for less than 600 msec. On the other hand, the *Fixed – TXOP* algorithm received approximately 75% of GOPs within the same delay. The GOP delivery delay plays

a significant role in measuring the user’s visual quality experience [27]. Lower GOP delay and jitter values ensure a high QoE, including fairness. In comparison, longer GOP delays and jitter could reduce the subjective quality of video streams.

Figure 21 shows the average packet arrival rate measured over 10 GOPs for different video flows. The figure also shows the respective PSNR values for the proposed algorithms along with the reference and default algorithms. It can be observed that the *FRA – TXOP* and *EAS* algorithms achieve higher PSNR values than the *Fixed – TXOP* and *Bi – level* algorithms. These results show that the proposed algorithms can deliver high-quality video flows than the referenced schemes. For the proposed algorithms, most of the video flows achieve PSNRs of 30 dB or above. According to the reference [25],

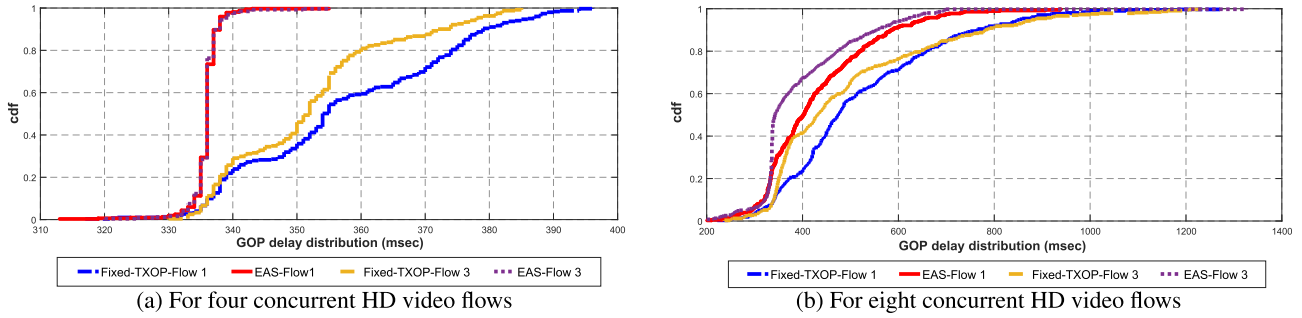


FIGURE 20. GOP delay distribution analysis.

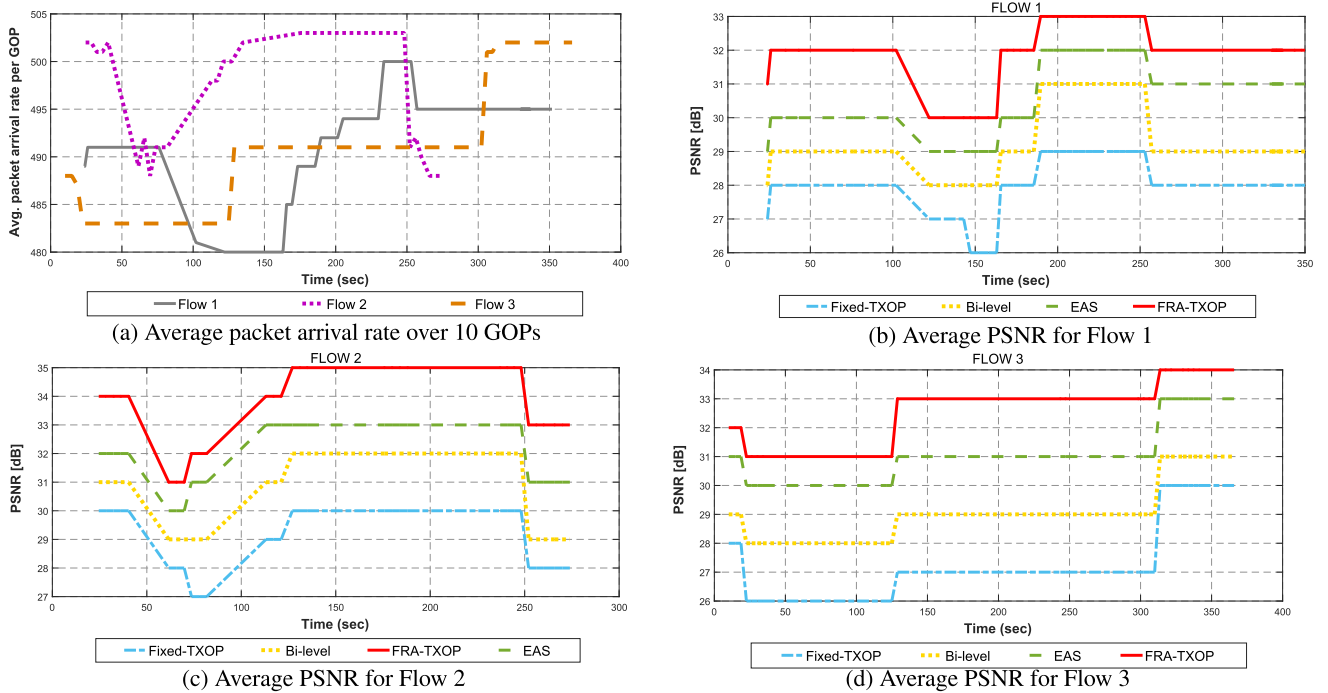


FIGURE 21. Average packet arrival rates and PSNRs for the Fixed-TXOP, Bi-level, FRA-TXOP and EAS algorithms.

any flows experiencing PSNR > 31 dB will achieve high perceptible quality. The plot shows that the *EAS* algorithm experiences slightly lower PSNR values than the *FRA – TXOP* algorithm. This reduction in the PSNR is caused by slightly lower throughput than that of the *FRA – TXOP* algorithm. This is one of the minor penalties that the *EAS* algorithm pays to maintain high fairness for all video flows.

VI. CONCLUSION

The proliferation of HD video services has motivated the demand for new scheduling techniques to guarantee high-quality video delivery in IEEE802.11ac wireless networks. This paper addresses the issue of QoS degradation for video deliveries caused by congestion due to overloaded nodes. The proposed *FRA – TXOP* algorithm identifies the backlog video flows and adapts the transmission resources according to congestion levels. The *EAS* algorithm works in conjunction with the *FRA – TXOP* technique. The *EAS* algorithm identifies video flows with a longer delay than the average packet

delay and allocates them appropriate bandwidth resources to improve the quality of the viewer’s experience. We presented extensive simulation results to analyse the performance of the proposed protocols. A comprehensive analysis of the results has shown the effectiveness of the proposed algorithms. The proposed techniques can easily be integrated with the current IEEE802.11ac MAC standard without any modifications to other layers. There is no additional development cost as the default 802.11ac protocol stack is utilised for the implementation. In the future, we are planning to extend our research to examine the QoS and fairness issues of the IEEE802.11ac protocol (i.e., Wi-Fi 6).

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SUMMERA NOSHEEN received the B.S. degree in computer engineering from the University of Arid Agriculture Rawalpindi, Pakistan, and the M.Sc. degree in computer engineering from the COMSATS Institute of Information Technology, Islamabad, Pakistan, in 2005 and 2007, respectively. She is currently pursuing the Ph.D. degree with the School of Electrical Engineering and Computing, The University of Newcastle, Australia. Her research interests include wireless networks, quality of service, quality of experience, and MAC layer resource allocation. She received the University of Newcastle International Postgraduate Research Scholarship and the University of Newcastle Research Scholarship in 2017.



JAMIL YUSUF KHAN (Senior Member, IEEE) received the Ph.D. degree in electronic and electrical engineering from the University of Strathclyde, Glasgow, Scotland, in 1991. Since receiving his Ph.D., he has worked as a Researcher at the University of Strathclyde for the European Research Programme - RACE. In 1992, he moved to New Zealand to work as an academic with the Institute of Information Science and Technology, Massey University. He moved to the University of Newcastle, Australia, in 1999. He is currently an Associate Professor with the School of Electrical Engineering and Computing, where he leads the Telecommunications Network Research Group. He is also the Programme Convenor of the Telecommunications and Computer Systems Engineering programmes with the School of Electrical Engineering and Computing. His main areas of research interests are communication network architecture and protocols, wireless networks, wireless sensor networks, cooperative and cognitive networks, M2M communications, energy harvesting networks, smart grids and smart cities, vehicular networks, and cyber-physical system design. So far, he has published 170 internationally refereed publications in the form of books, book chapters, journals, and international conference papers. He is a Senior Member of the IEEE Communications Society and a member of the ACM. He actively participates in various IEEE and other professional forums, serving as a journal editor and a member of editorial boards. He has also served and is serving as the Chair/Vice-Chair and TPC member of many conference technical committees.

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