Enhanced Frame Aggregation Scheduler (EFAS) for data Transmission over IEEE802.11ac

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ABSTRACT

Wireless Local Area Networks (WLANs) based on the IEEE 802.11 standard are widely deployed in the home and enterprise segments across the globe. IEEE 802.11ac standard has brought several significant improvements compared to its predecessor IEEE 802.11n. It managed to break the Gigabits barrier with a combination of both refining older techniques and presenting new ones. The audio transmission over WLAN requires better Quality of Service (QoS). Over the time-varying characteristics of WLAN, better QoS requirement for the audio applications makes it a more challenging task. Presently IEEE802.11ac standard is in use for supporting WLAN with the main goal to provide higher throughput. This paper proposed an Enhanced Frame aggregation scheduler methodology in which proposed EFAS breaks the frames into smaller frames in the IEEE802.11ac standard and dynamically transmits the frame by analyzing the traffic. The evaluation of the proposed EFAS methodology is simulated in Network Simulator (NS2) by comparing it with the IEEE802.11ac standard method. The proposed methodology can be imposed at IEEE 802.11 MAC for achieving better audio quality.

1. Introduction

IEEE 802.11ac is the fifth generation in Wi-Fi networking standards. The IEEE802.11ac has a significant enhancement on the performance with respect to the conventional 802.11n standard. The IEEE 802.11ac standard for WLANs promises throughputs higher than 1 Gbps in the 5 GHz band Compared with the IEEE 802.11n standard [1], IEEE 802.11ac considers wider channels, modulation and coding rates with higher spectral efficiency and MU-MIMO capabilities, as well as channel bonding mechanisms [2], [3]. The IEEE 802.11ac Multiple Access Control (MAC) layer will basically follow the IEEE 802.11n standard, only extending it to accommodate the new MU-MIMO features, i.e., the ability to transmit multiple spatial streams from the Access Point (AP) to different STAs in parallel by using a multi-user beam forming scheme.

The EDCA enhances the original 802.11 DCF by the distinction of network traffic based on their quality class. EDCA differentiates incoming data traffic into four traffic classes viz.[4,5] Voice traffic, Video traffic, Best- Effort traffic, and Background traffic. To transmit distinct data to the lower PHY layer, EDCA provides different access categories to each type of traffic that acts as an individual queue. Network traffic is differentiated according to their quality class.

IEEE 802.11n WLAN has four Access Categories (AC) at its MAC layer. Each AC has individual buffers forming the queue. Each AC is reserved for different traffic and works as an independent queue. The queues of different traffic classes have different priorities. The queue with higher priority gets the transmission medium access first. Different access categories with their abbreviated name and priority are shown below in Table 1. In EDCA, video traffic is having their own queue buffers. Voice traffic (AC [VO]) is having maximum priority and then video traffic (AC [VI]) and background Traffic (AC [BK]) has the least priority.

Table I. Different Access Categories with their Priorities

S.No.	Data Traffic Class	Abbreviated Name	Priority
1	Voice Traffic	AC[VO]	1
2	Video Traffic	AC[VI]	2
3	Best-effort Traffic	AC[BE]	3
4	Background Traffic	AC[BK]	4

2. Frame Aggregation of IEEE 802.11ac

The techniques for MAC layer enhancement of IEEE802.11ac are AMPDU and AMSDU. The way of importing and exporting the data between the upper of the MAC sublayer and the higher layers considers the key difference between AMSDU and AMPDU. The Aggregation algorithm uses these techniques performed by using a single-block of the Acknowledgment (ACK) frame to exchange multiple MPDUs [14]. The main distinction between an MSDU and an MPDU is that the former corresponds to the information that is imported to or exported from the upper part of the MAC layer from or to the higher layers, respectively. Whereas, the latter relates to the information that is exchanged from or to the PHY layer by the lower part of the MAC layer.

TABLE II
802.11AC AGGREGATION FRAME SIZES

	802.11n	802.11ac
A-MSDU length (bytes)	7935	11426
A-MPDU length (bytes)	65535	1048579

1) Aggregate MSDU:-

The principle of the A-MSDU is to allow multiple MSDUs sent to the same receiver to be concatenated in a single MPDU. This definitively improves the efficiency of the MAC layer, specifically when there are many small MSDUs. For an A-MSDU to be formed, a layer at the top of the MAC receives and buffers multiple MSDUs. The A-MSDU is completed when the size of the waiting MSDUs reaches the maximal A-MSDU threshold [6]. The maximum size of conventional and High Throughput (HT) is either 3839 bytes or 7935 bytes, respectively, while for VHT there is no constraint on the maximum size of AMSDU [7].

In Figure 1, we describe a simple structure of an AMSDU. Each MSDU consists of an MSDU header, which contains the Destination Address (DA), the Sender Address (SA) and the length of the MSDU, followed by the MSDU arrived from the Logical Link Control (LLC) layer and 0-3 bytes of padding. A major drawback of using A-MSDU is under error-prone channels. By compressing all MSDUs into a single MPDU with a single Frame Check Sequence (FCS), for any MSDUs that are corrupted, the entire A-MSDU must be retransmitted [8].

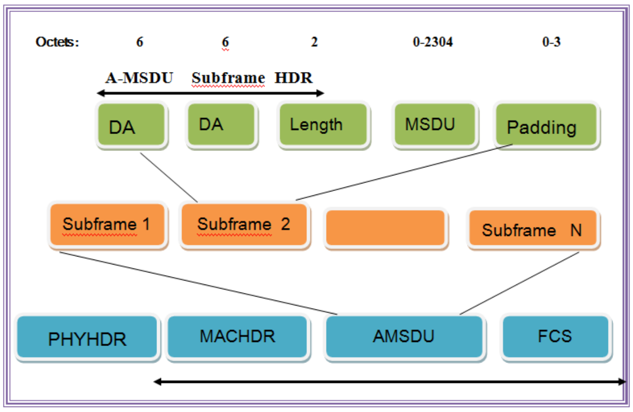

Figure 1: Basic structure of the A-MSDU subframe

2) Aggregate MPDU:-

The concept of A-MPDU aggregation is to join multiple MPDUs with a single leading PHY header. A key difference from A-MSDU aggregation is that AMPDU operates after the MAC header encapsulation process. The utmost number of MPDUs that it can hold is 64 because a Block ACK bitmap field is 128 bytes in length, where each MPDU is mapped using two bytes [9]. The basic structure is shown in Figure 2, 4 bytes of set fields (delimiters) are added before each subframe, and the padding changes from 0-3 bytes inserted at the end. The primary function of the delimiter header is to explain the location and length for each subframe inside AMPDU. The Cyclic Redundancy Check (CRC) byte used to check the authenticity of the previous bits. The padding field used to recognize the subframe at the destination while the delimiter signature assists the de-aggregation process.

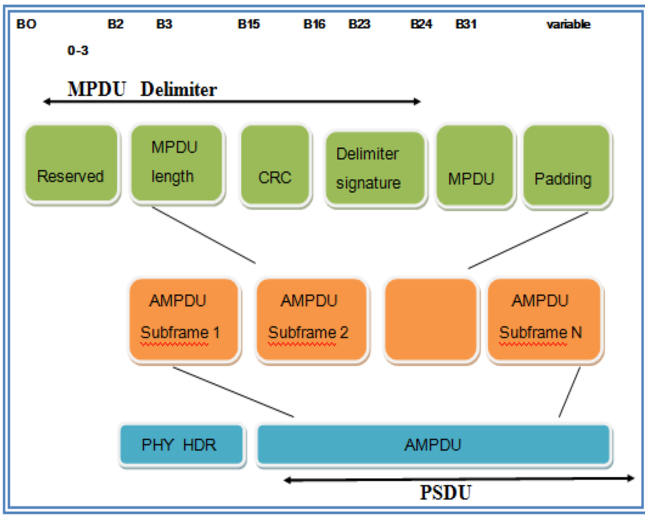

Figure 2: Aggregate-MPDU

3) Two-level aggregation:-

The Two-Level aggregation as shown in Figure 3 comprises a blend of A-MSDU and A-MPDU over two stages. In the first stage, MSDUs received by MAC from the upper layer is buffered for a short time until A-MSDUs are formed according to their traffic identifier, destination, source, and the maximum size of A-MSDU. The complete A-MSDUs and other non-aggregate MSDUs then enter the second stage to form an A-MPDU. Only complete A-MSDUs and MSDUs, not the fragments of AMSDUs or MSDUs, could be contained in an AMPDU. The entire aggregation scheme completes when A-MPDU is created [10]

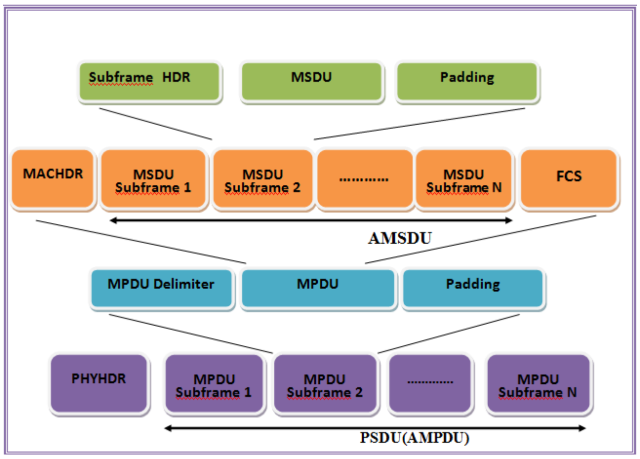

Figure 3: Basic structure of combination AMSDU/AMPDU subframe

3. Proposed (EFAS) Methodology

The frame aggregation is very effective in case of saturated traffic, otherwise, the frame has to wait for further packets which increases the delay in delivery. Here, we proposed the EFAS methodology which tries to make the best and efficient way to deliver the audio packet with minimal delay. The proposed EFAS method shown in flowchart describes the flow of methodology in which a new frame aggregation limit is fixed according to the load on traffic. Starting with the first frame, A-MPDU is constructed and NFA(ar) is fixed after the MAC traffic is being analyzed. Comparing the frame sequence with frame aggregation count, final A-MPDU is transmitted once the condition is true.

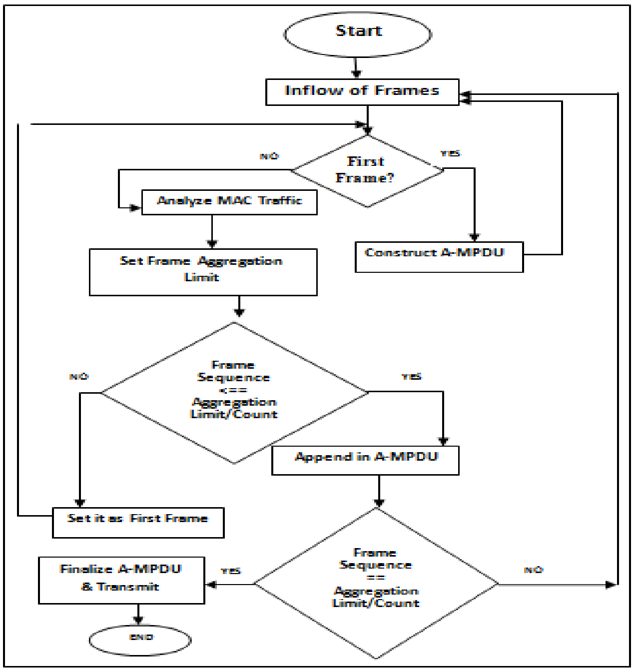

Figure 4: Flowchart of the proposed EFAS scheme.

The following pseudo-code In figure 5 explains the step-by-step instructions of EFAS methodology. Starting with the inflow frame, the algorithm checks the number of frames in A-MPDU. If A-MPDU is empty then the next instruction adds a frame to A-MPDU by analyzing the voice buffer. After that the MAC traffic is being analyzed i.e buffer, then NFA(ar) limit is fixed as:-

NFA(ar)= Max.A-MPDU limit - Buffer size.

The conditional statement checks and compares the A-MPDU limit with NFA(ar), if A-MPDU is less than NFA(ar) then more frames are appended to A-MPDU else frame sequence is compared with NFA(ar) and final A-MPDU is transmitted after condition becoming true. Here, by considering a packet size of 1500bytes then in IEEE 802.11 ac the Maximum size for the A-MPDU frame is 42 and the Maximum Voice buffer size is 50. Figure 5. depicts the EFAS algorithm.

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Algorithm:-

Step 1: Start
Step 2: Store Inflow Frame
Step 3: Check A-MPDU
Step 4: If A-MPDU is empty
            Insert Frame in A-MPDU;
            Go to Step 2;
Step 5: Else
            Analyze MAC Traffic;
            Set New Frame Aggregation Limit(Ar)
            NFA(ar)= Max.A-MPDU limit - Buffer size
Step 5.A:   If A-MPDU Size <Ar
                Insert Frame in A-MPDU;
                A-MPDU Size ++;
            Else If   Frame Sequence == Ar
                      Finalize A-MPDU and Transmit;
            End
Step 6: Go to Step 2;
```

Figure 5: Algorithm for EFAS Scheme

4. Experimental Result

The evaluation of the proposed EFAS methodology is simulated in Network Simulator (NS2). The simulator supports a class hierarchy in C++ and a very similar class hierarchy in OTcl. The root of this class hierarchy is the Tcl Object in OTcl. NS2 provides substantial support for simulation of TCP, routing algorithms, queuing algorithms, and multicast protocols over wired and wireless (local and satellite) networks, etc. It is freely distributed, and all source code is available. NS simulator covers a very large number of applications, protocols, network types, network elements, and traffic models. These are called "simulated objects". The simulator is based on two languages: an object-oriented simulator written in C++ and an Otcl interpreter used to execute user's command scripts. In these scripts, the user can define a particular network topology, the specific protocols and applications that he wishes to simulate and the form of output he wishes to obtain from the simulator. We classify the network traffic in generic terms as below: Very high, Heavy, Medium, Light & Very light. The above-suggested network congestions scenarios are mainly classified by measuring the traffic. The network handling only one audio stream is considered as having no traffic congestion or having minimum network traffic load. Transmitting of two or three audio streams concurrently over the network is classified as moderate network congestion or intermediate traffic load that is in a permissible limit. Transmitting more than three audio streams concurrently is classified as heavy network congestion or the network load that is not suitable for the better audio QoS. The Proposed EFAS methodology is evaluated in comparison with the standard IEEE802.11ac protocol. After extensive simulation of the proposed EFAS methodology over the experimental setup, the results of performance metrics for measuring the QoS of audio such as end-to-end delay, jitter, and packet delivery ratio.

The following parameters have been set to define the simulation scenario that is defined in Table III.

Table III.

Parameters	Value
Network Simulator	NS 2.35
Channel	Wireless
Propagation Model	Two Ray Ground
MAC Protocol	IEEE 802.11ac
Simulation Time	200sec
Operating Frequency	2.4GHz
Packet Size	512byte
Transmitter Range	250m
Max Speed	100Km/h
Pause Time(sec)	2.0sec
Packet Rate	5 Packets/s
Traffic Source	CBR
Number of Nodes	10,20,30,40,50,60,70

Figure 6 shows the comparison of End-to-End delay in Standard IEEE802.11ac protocol and EFAS methodology at Very light load, Light load, Medium load, Heavy load, and Very high load.

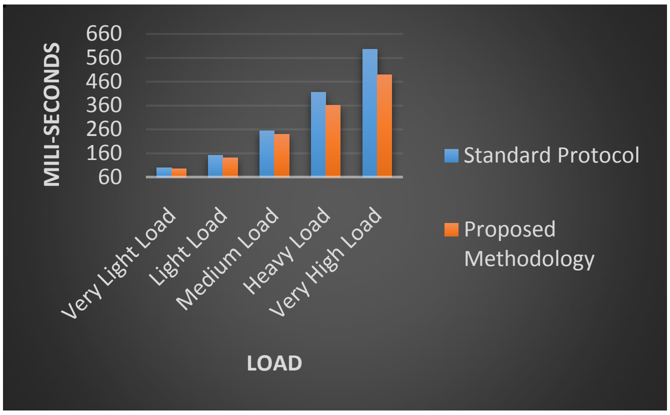

Figure 6. Comparison of End-to-End Delay of Standard Protocol and EFAS methodology at Very high, Heavy, Medium, Light & Very light load.

In Very high load and Heavy load, the proposed EFAS methodology performs better than standard IEEE802.11ac protocol due to swift transmission of data packets which in results reduced End-to-End delay from 596 millisecond, 416 millisecond to 489 millisecond, 360 millisecond respectively by EFAS methodology compared to Standard protocol. In medium load and light load, the proposed EFAS methodology reduces the End-to-End delay from 254 milliseconds, 152 milliseconds to 240 milliseconds, 141 milliseconds respectively with compared to standard protocol. In very light load, the End-to-End delay is closer to standard protocol because during very light load on traffic, the transmission of data packets is quick as in proposed EFAS methodology data packets are transmitted only by analyzing the load on traffic, because of less load on traffic channel the data packets experience ready to flow as channel remains available for transmission. Therefore. The end-to-end delay for proposed EFAS methodology is closer to standard protocol.

Figure 7 shows the comparison of Jitter in standard protocol and EFAS methodology at very high, heavy, medium, light and very light load.

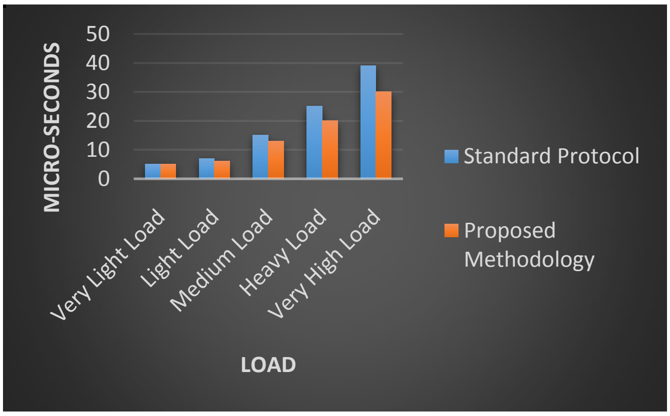

Figure 7. Comparison of Jitter of Standard Protocol and EFAS methodology at Very high, Heavy, Medium, Light & Very light load.

In our proposed EFAS methodology, the dropping of an audio frame from the audio access category due to the unavailability of audio buffers space has been rectified. The more audio buffers, in turn, leads to less re-transmission of arrived audio frames at MAC. In a very high load, the increased buffer space along with higher priority frame transmission policy leads to fluent transmission from the audio queue. The above features of the proposed methodology show better Jitter result in comparison with the standard protocol. In a very high load, the EFAS methodology reduces Jitter from 39 micro-sec to 30 micro-sec compared to standard protocol. In Heavy load and medium load, the proposed EFAS methodology performs better, where it reduces Jitter from 25 micro-sec, 15 micro-sec to 20 micro-sec, 13 micro-sec respectively. For very light load, the Jitter performance is the same. Figure 8 shows the comparison of PDR(packet delivery ratio) in standard protocol and EFAS methodology at very high, Heavy, Medium, Light and very light load.

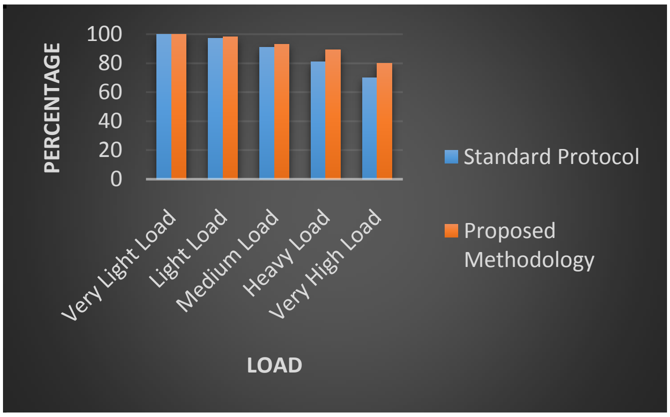

Figure 8. Comparison of Packet Delivery Ratio(PDR)of Standard Protocol and EFAS methodology at Very high, Heavy, Medium, Light & Very light load.

By predicting the current traffic flow and future audio injection rate, the decision is taken to borrow buffers from other queues. This provides buffer space in audio access categories, which can accept more audio frames, that have to be transmitted. In a very high and heavy traffic load, the EFAS methodology increases the packet delivery ratio on average by 10% and 8% respectively with standard protocol. Thus by saving the audio frames from being dropped out from the audio queue, the packet delivery ratio of the proposed EFAS shows better results than the standard protocol of IEEE802.11ac. In medium and light load on traffic, EFAS methodology increases the packet delivery ratio on average by 2% and 1% respectively compared to standard protocol.

As the traffic congestion decreases, audio buffer length becomes the same as the standard audio buffer length. Due to the same audio queue length in very light load, the proposed EFAS methodology demonstrates approximately equally to the standard protocol of the IEEE 802.11ac scheme. Therefore, in very light load, EFAS methodology has equal packet delivery ratio (PDR) compared to standard protocol.

Table IV. Summarize the result of proposed EFAS methodology in comparison with standard protocol.

Methodology	**Performance Metrics**	**Very high**	**Heavy**	**Medium**	**light**	**Very light**
EFAS	End-to-End Delay	489	360	240	141	95

	Jitter	30	20	13	6	5
	PDR	80	89	93	98	100
Standard protocol	End-to-End Delay	596	416	254	152	100
	Jitter	39	25	15	7	5
	PDR	70	81	91	97	100

5. Conclusion

In this paper, we compared the proposed EFAS methodology and IEEE 802.11ac a standard protocol. After exhaustive experimentations with the proposed methodology over the NS2 simulator. The simulation results of this experiment demonstrate that the proposed EFAS methodology performs better than the standard protocol. The simulation results of comparison have been presented in the form of graphs. The proposed methodology is able to reduce the average end-to-end delay and jitter as compared with standard protocol. Also, it shows an increase in the packet delivery ratio (PDR) with much lower delay against the standard protocol. EFAS considerably works better in the very high load on traffic. Therefore, the proposed EFAS methodology can be used for achieving better QoS during the audio transmission over WLAN. In the future, more work can be achieved on better QoS during video transmission over WLAN.

References

[1] IEEE 802.11n, Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications: Enhancements for Higher Throughput, 2009.

[2] E. H. Ong, J. Kneckt, O. Alanen, Z. Chang, T. Huovinen, and T. Nihtila, “IEEE 802.11ac: enhancements for very-high throughput WLANs,” 2011 IEEE Personal Indoor and Mobile Radio Communications.

[3] M. Park, “IEEE 802.11ac: dynamic bandwidth channel access,” 2011 IEEE International Conference on Communications.

[4] Hazra PK, De A. Performance analysis of IEEE 802.11e enhanced distributed channel access. The International Journal of Advanced Manufacturing Technology. 2011; 2(4):507–29.

[5]Abdel-Rahman MJ, Shankar HK, Krunz M. Adaptive Cross-layer Protocol Design for Opportunistic WLANs over TVWS. IEEE International Symposium on Dynamic Spectrum Access Networks; 2014. p. 519–30.

[6] Charfi, E., Chaari, L., Kamoun, L. (2013) PHY/MAC Enhancements and QoS Mechanisms for Very High Throughput WLANs: A Survey. In proceedings of IEEE Communications Surveys and Tutorials, pp. 1–22.

[7] Technology, “Information Technology Specific requirements- Part 11: Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) specifications, IEEE P802.11-REVmcTM/D4.0"” 2015, pp. 1-3730.

[8] Skordoulis, D., Ni, Q., Chen, H.H., Stephens, A.P., Liu, C., Jamalipour, A., “IEEE 802.11n MAC frame aggregation mechanisms for nextgeneration high-throughput WLANs”, Journal of IEEE Wireless Communications, Vol. 15, No.1, 2008, pp. 40-47

[9] Charfi, E., Chaari, L., Kamoun, L. (2013) PHY/MAC Enhancements and QoS Mechanisms for Very High Throughput WLANs: A Survey. In proceedings of IEEE Communications Surveys and Tutorials, pp. 1–22

[10] Wang, C. Y., Wei, H. Y. (2009) IEEE 802.11n MAC Enhancement and Performance Evaluation. Mobile Networks and Applications 14, 760–771.

[11] Mayank Patel, Naveen Choudhary (2017). Time Restraint Tractable Aggregation Scheme for Beneficial Video Transmission over WLAN. Journal of Engineering and Applied Sciences.12(23). 10.3923/jeasci.2017.7170.7179, pp. 7170-7179 (Scopus)

[12] Mayank Patel, Naveen Choudhary (2017). Adroit Buffer Augmentation for Reliable Video Transmission over WLAN. Indian Journal of Science and Technology. 10(22). DOI: 10.17485/ijst/2017/v10i22/113962, pp. 1-13. (WOS)

[13] Mayank Patel, Naveen Choudhary (2017) "Performance Evaluation of MPEG-4 and H.264/SVC Video Codec Over WLAN", International Journal for Research in Applied Science and Engineering Technology (IJRASET), ISSN : 2321-9653, Volume 5 Number 6, DOI : 10.22214/ijraset.2017.4153, pp. 899-905.

[14] Naveen Choudhary, Mayank Patel (2014). “QoS Enhancements for Video Transmission over High Throughput WLAN: A Survey “International Journal of Research and Scientific Innovation (IJRSI). ISSN: 2321 - 2705 , Volume I Issue VII , Page 43-50

[15] Mayank Patel, Naveen Choudhary (2015). “A Novel Adaptive Cross-Layer Mapping to Support Real Time Multimedia Applications over IEEE 802.11n” Review Of Business And Technology Research (RBTR), ISSN 1941-9414, Volume 12 Number 2,Page 103-107

[16] Mayank Patel, Naveen Choudhary (2019). “A-Dynamic Cross-Layer Mapping to Support Real Time Multimedia Applications over IEEE802.11n”. Paper present in IEEE International conference on Energy, Communication, Data Analytics and Soft Computing (ICECDS) with catalog no. CFP17M55-PRJ: 978-1-5386-1886-8, Poonamallee on 1st August, 2017. DOI :10.1109/ICECDS.2017.8390107, pp. 3474-3478