Impact of TCP Congestion Control Algorithms on IEEE802.11n MAC Frame Aggregation

Mustafa Samih and Emad Al-Hemiary
College of Information Engineering, Department of Network Engineering,
Nahrain University, Baghdad, Iraq
{mustafa,emad}@ie-nahrain.org

***Abstract*— The Media Access Control (MAC) introduced in the IEEE802.11n reduces the bottleneck in the legacy IEEE802.11 using different techniques such as frame aggregation and block acknowledgments. In this paper we investigate the interaction between MAC frame aggregation and Transmission Control Protocol (TCP). MAC layer interaction with TCP is very important as TCP determines the end-to-end transfer rate. Regarding the interaction with TCP a comparison is made through simulations among different TCP congestion control techniques, like Reno, NewReno, Vegas, Tahoe, and Selective Acknowledgments (SACK). Results show that with different aggregation sizes, NewReno and SACK outperform the other techniques. As an example a throughput of 80 Mbps is achieved with TCP NewReno and window size of 8 Kbyte. Due to the dependencies of Vegas on Round Trip Time (RTT) throughput remains constant versus offered load.**

***Keywords*— Frame Aggregation, IEEE802.11n, TCP, SACK**

INTRODUCTION

In recent years, the IEEE 802.11 standard has become the most widely used technology for wireless communication in local areas. The carrier sense multiple access with collision avoidance (CSMA/CA) protocol that is implemented by the 802.11 standard ensures that all wireless stations are sharing the wireless medium. This protocol is contention-based that enforces all stations to sense the medium before transmitting to avoid collisions and retransmissions. Distributed Coordination Function DCF is a form defined by the CSMA/CA as the access mechanism was adopted by the IEEE 802.11 standardizing forum. The DCF access mechanism is mandatory in IEEE 802.11a, IEEE 802.11b and IEEE 802.11g. The maximum theoretical data rate at an IEEE802.11b network is 11Mbps while it is 54Mpbs at IEEE802.11a and IEEE802.11g networks. Actually the achievable MAC throughput is much lower than that due to many overheads encountered in the operation of the DCF and channel conditions. With the increasing demand for multimedia communications over wireless LANs (WLAN) such as High Definition video streaming and Voice over IP [1], many enhancements were needed on the physical (PHY) and MAC layers of the IEEE802.11 standard [2].

The IEEE 802.11n provides a marked increase in throughput (from 20 Mbps to around 200 Mbps, in practice) as well as range of reception (through reducing signal fading) over the IEEE 802.11a/g standards currently in use. Multiple antennas, or MIMO (Multiple-Input, Multiple-Output), is the key innovation used in the physical layer [4], unlike the single-input- single-output (SISO) WLAN standards such as the legacy IEEE 802.11a/b/g [5], MAC efficiency is also improved through the use of frame aggregation and enhancements to the block acknowledgment protocol. These features together provide the bulk of the throughput enhancement over that achievable with IEEE802.11a and IEEE802.11g [3]. In this paper, we will study the Impact of various TCP congestion control mechanisms over the enhanced MAC layer design in the IEEE 802.11n systems. Specifically, the work in this thesis gets advantage of the work in [11, 23, 24] but adds extra knowledge as it discusses the interaction of TCP congestion algorithms with frame aggregation. It is interesting to predict the effect of flow control based on window scheme that adopted by TCP on frame aggregation. MAC layer interaction with TCP is very important as TCP determines the end-to-end transfer rate.This document is a template. An electronic copy can be downloaded from the conference website. For questions on paper guidelines, please contact the conference publications committee as indicated on the conference website. Information about final paper submission is available from the conference website.

I. RELATED WORK

Frame Aggregation that is adopted in the MAC layer of IEEE802.11n reduces headers overhead and optimize channel access time by minimizing the number of backoffs and acknowledgment overhead. There are a number of publications which discuss this topic. The Impact of Multi-Rate Operation on A-MSDU, A-MPDU and Block Acknowledgment in Greenfield IEEE802.11n Wireless LANs: O. Abu-Sharkh [2] showed the impact of Multirate phenomenon on the recently introduced MAC mechanisms of IEEE802.11n such as block acknowledgment, A-MSDU and A-MPDU, his analysis is based on driver modification of certified IEEE802.11n device. Throughput Enhancement of IEEE 802.11 WLAN for Multimedia Communications: N. Huda [6] proposed a new MAC technique called Frame Aggregation and Block Acknowledgement (FABA) mechanism is the collecting of some features of above two important components, this technique improve the wireless network throughput, the simulation of FABA showed throughput of more than 160 Mbps when the PHY data rate 600 Mbps. Evaluations and Enhancements in 802.11n WLANs Error-Sensitive Adaptive Frame Aggregation: K. Chan[7] proposed an Error-Sensitive Adaptive Frame Aggregation (ESAFA) in which the aggregated frame size is set dynamically based on the frame error rate that is tolerable by particular multimedia traffic the simulation results shows that ESAFA outperforms the normal aggregation methods under different channel conditions. A Frame Aggregation Scheduler for IEEE 802.11n: Selvam [8] presented a simulation study on a frame aggregation scheduler; the proposed method dynamically chooses the aggregated frame size and aggregation method based on various parameters such as delay, by considering the expire time of first arrived packet in the sender queue where the deadline is the time to transmit an aggregated frame without violating the deadline of any of the aggregated packets. Prototype development of advanced hierarchical frame aggregation in fiber wireless access networks: Ghazisaidi [9] proposed and evaluated the performance of hierarchical frame aggregation techniques in Fiber-Wireless networks (FiWi), real channel conditions are used to transmit video traffic. On Maximizing VoIP Capacity and Energy Conservation in Multi-Rate WLANs: Kwan-Wu Chin [10] focuses on the effect of frame aggregation on VoIP in the power save mode used in WLANs, and then he proposed a scheduler so that VoIP efficiency increased with minimum power consuming. Finally, An Empirical Study on Achievable Throughputs of IEEE 802.lln Devices: V. Visoottiviseth, et al [11] used different 802.11n devices to measure the throughput using TCP and UDP traffics, their results shows enhanced performance over the legacy 802.11g.

II. FRAME AGGREGATION

By looking into figure 1 steps of a frame transmission, we can observe that channel inefficiently utilized by DCF. This inefficiency due to different reasons during the transmission procedure, transmission time is divided into a DIFS, a Contention Window backoff time, the frame transmission time, a SIFS, and the ACK frame transmission time [3, 12]. The physical protocol data unit (PPDU) transmission time also divided into two parts: 802.11 physical header and payload transmission time. This overhead in DCF leads to limitation in throughput especially when the payload size is small [1]. Throughput limits leads caused by overheads resolved through technique called frame aggregation implemented in IEEE802.11n MAC layer.

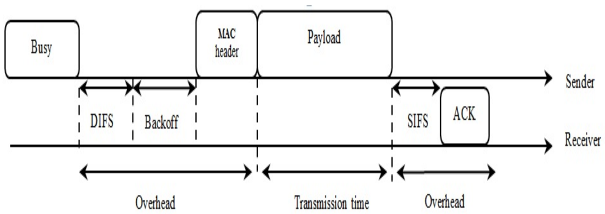

Fig. 1 CSMA/CA operation in legacy IEEE802.11

Aggregation technique is the main feature to improve IEEE802.11 MAC transmission efficiency. We have described the overhead in legacy IEEE 802.11 MAC. Frame aggregation can increase channel utilization and reduce channel access time. This technique concatenates multiple data packets from the upper layer (network layer) into one super frame for transmission. Overhead in legacy standards is reduced since multiple frames are transmitted in single superframe, transmissions time is reduced since the header overhead and interframe time is saved. Frame aggregation can provide higher throughput and transmission speed and that satisfies the modern multimedia applications for example HDTV and VoIP [1].

There are two methods of aggregation aggregate MAC service data unit (A-MSDU) and aggregate MAC protocol data unit (A-MPDU), the first method A-MSDU concatenate several MSDUs coming from upper layer into single super frame with single MAC header has one source address (SA) and one destination address (DA), which DA and SA parameter values mapped to the same receiver address (RA) and transmitter address (TA) values, the source and destination addresses are the same in all MSDUs this means that the super frame destined to one receiver [14]. Also all MSDUs must have the same TID value [13]. The super frame is appended with one Frame Check Sequence (FCS) which means that if any frame is lost in the aggregated frame the entire frame couldn't be recovered; the maximum length of an A-MSDU is 7935 bytes [14, 15, and 16]. A-MSDU frame structure is shown in figure 2.

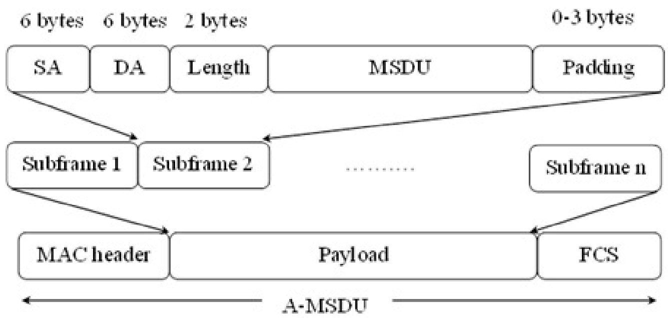

Fig. 2 A-MSDU frame Structure

The second method is A-MPDU which aggregates multiple MPDUs. The creation of A-MPDU is between the MAC layer and the Physical layer (PHY); A-MPDUs are created before sending to PHY layer for transmission. The difference between A-MPDU and AMSDU creation is that MAC does not wait for additional time for incoming packets from upper layer before the A-MPDU aggregation. MAC only uses the MPDUs already queued to create AMPDUs. Traffic Identifier (TID) might different in each MPDU in the same superframe. 65535 byte is the maximum size of A-MPDU. The delimiter field is used in the beginning of each MPDU to separate between MPDUs to help in deaggregation process, and padding bytes at the end of each MPDU to ensure that the size is multiple of 4 bytes. The CRC is added at each MPDU to check the integrity of each frame in the deaggregation process [1] , this give the advantage to A-MPDU over the A-MSDU when error happens in one frame does not affect other frames in the A-MPDU [13], A-MPDU become ready for transmission after adding the physical header. Figure 3 shows frame structure of A-MPDU.

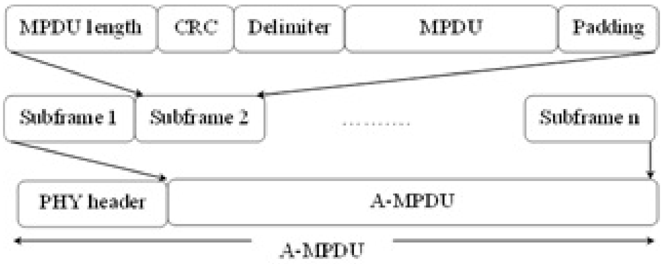

Fig. 3 A-MPDU frame structure

The last aggregation method called two level frame aggregations or Aggregate Physical Protocol Data Unit (A-PPDU) which aggregates MPDUs coming from MAC layer and each MPDU concatenates a number of A-MSDUs, the physical header is added to each PPDU to ensure error recovery [17].

III. BLOCK ACKNOWLEDGMENT

IEEE802.11e amendment first introduce block acknowledgment (BACK) to improve efficiency and channel utilization, this technique allows to transfer of multiple ACK that acknowledge received frames into single BACK frame instead of acknowledged each data frame independently [3]. BACK mechanism used only with A-MPDU not the A-MSDU that is because when an MSDU frame error the whole A-MSDU should be retransmitted to recover from error. The sender needs to retransmit only not acknowledged MPDUs. This technique is useful in error prone wireless environment especially in case of large frame transmission. BACK enhance the performance of IEEE802.11n MAC layer and adding further reliably to the upper layers [1].

IV. TCP CONGESTION CONTROL ALGORITHMS

Many network applications require data transmission over high speed wireless networks, these application demands high channel bandwidth between wireless stations. Keeping the stability and the reliably of internet requires congestion control mechanisms. TCP satisfy these requirements, which is a well-developed high speed transport protocol that works efficiently in high traffic congestion environment. One of the drawbacks of using TCP is the Additive Increase Multiplicative decrease (AIMD) congestion back-off algorithm that controls the window size, thus affects system throughput. Increasing the window size (W) by one segment every round trip time (RTT) for received acknowledgment and decreasing the window size into half for lost packets, that is how the AIMD works [18]:

Increase $W=W+1$ (1)

Decrease $W=W/2$ (2)

Reno modifies the Fast Retransmit operation to include Fast Recovery. Fast Retransmit, happens after receiving three duplicate acknowledgments for the same TCP segment (dup ACKs), the data sender assumes that a packet has been lost and without waiting for packet expiry time (timed out) retransmits the packet, that's leads to higher channel utilization and higher throughput. Fast recovery doesn't let the communication path (pipe) to be empty after implements Fast Retransmit to prevent the slow start algorithm. Fast Recovery is implemented by TCP sender after reaching threshold of dup ACKs, this threshold called tcprexmtthresh which is usually set to three [19]. Immediately after receiving three ACKs the sender reduces congestion window to half the current size. Reno triggers outgoing packets after receiving dup ACKs.

NewReno is a modified version of Reno, The modification happens in sender Fast Recovery phase when receiving ACKs that acknowledges a number of packets not all packets that's waiting to retransmit in fast recovery phase. These partial ACKs in Reno lead TCP to exit Fast Recovery phase and implements AIMD. NewReno doesn't exit fast recovery when receiving partial ACK, instead these ACKs indicate TCP that the packets followed the acknowledged packets in sequence are lost and should be retransmitted, also NewReno may use delayed ACKs instead of immediate ACKs. In Tahoe Triple duplicate ACKS are treated the same as a timeout. Tahoe will perform "fast retransmit", reduce congestion window to 1 maximum segment size, and back to slow start state [19].

TCP Vegas depends on packet delay, rather than packet loss, as a signal to select the rate of data packets to send. TCP Vegas detects congestion at an early stage based on increasing RTT values of the packets in the connection unlike other schemes like Reno, NewReno, etc. the algorithm relies mainly on accurate calculation of the Base RTT value [20]. In SACK the receiver decides which packets, messages, or segments in a stream are acknowledged. With selective acknowledgments, the data receiver can inform the sender about all segments that have arrived successfully, so the sender need retransmit only the segments that have actually been lost [21].

V. SIMULATION AND NETWORK TOPOLOGY

In this section we will introduce the model used in our simulation. We use NS 2 which is a discrete event simulator and building network topology using Tool Command Language (TCL) [22]. We build simple network topology using the frame aggregation model in [1] which implements the MAC and PHY layers of IEEE802.11n, our network consists of one wired node, one Access point, and one wireless node as shown in figure 4, TCP traffic is generated at wired node and sent over 1000 MB wired link to the access point, then the access point perform aggregation and send A-MPDUs to the wireless node which is a fixed node within (100x100) m area also Request To Send/Clear To Send (RTS/CTS) mechanism is not used.

Table1. Simulation Parameters [15]

Parameter	**Value**
Interface Queue length	100 packets
SlotTime	9 µs
SIFS	16 µs
Routing Protocol	*DSDV
Offered load	(10 – 144) Mbps
Maximum Aggregation Size	(4 – 64)KByte
Packet size	(500-1500)Byte
TCP congestion window	(2048, 8192) Byte
Block ACK threshold	2 Frames

*DSDV ROUTING PROTOCOL IS A PARAMETER NEEDED FOR SIMULATION

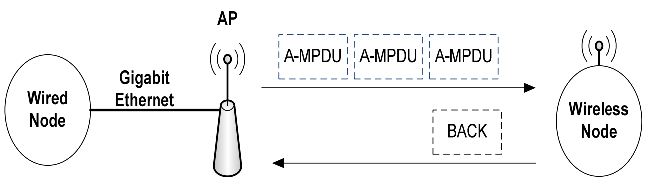

Fig.4 Simulation Scenario

VI. RESULTS AND PERFORMANCE EVALUATION

We simulate the network shown in figure 4 using different parameters and the results are collected. Figure 5 shows a compression between MAC layer average throughput and the maximum size of A-MPDU for different packet sizes, the network load is 144 Mbps, 2048 Byte TCP congestion window and NewReno is used.

The A-MPDU with a larger packet size shows higher throughput due to lower overhead generated by packets headers to be concatenated inside A-MPDU. Figure 6 show Average end to end delay, we observed that when using A-MPDU aggregation scheme, the average end to end packet delay is decreased rapidly. The reason behind this is multiple packets can be received at once and the overhead produced due to back-off mechanism used in DCF and additional header transmission is reduced. Many network applications can benefit from these network enhancements, especially in applications use File Transfer Protocol (FTP) and delay sensitive applications, such as Voice over IP.

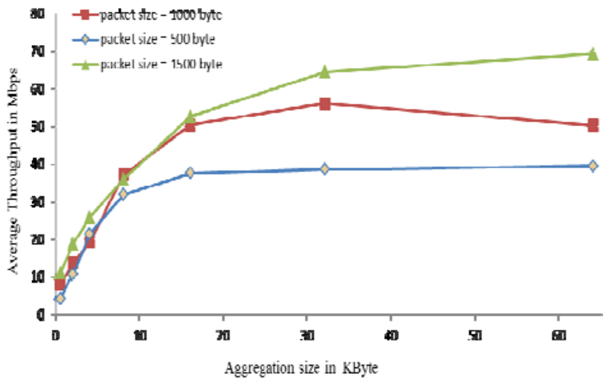

Fig. 5 Average network throughput versus aggregation size

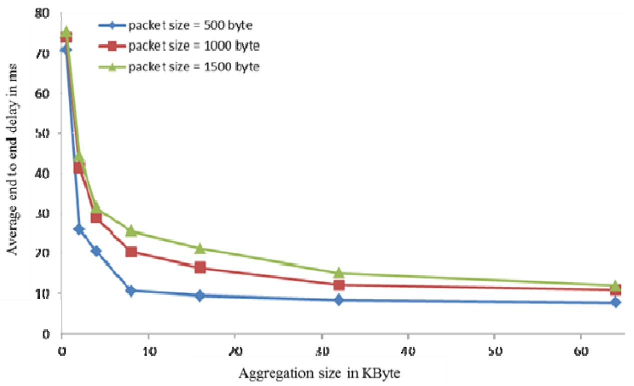

Fig. 6 Average end to end delay versus aggregation size

A comparison is made among different TCP congestion control algorithms with their interaction with frame aggregation, each algorithm is simulated using different A-MPDU sizes, and the TCP congestion window used is 8192 Byte for all algorithms. As can be seen from results in table 2 that NewReno and SACK outperform other algorithms, NewReno enhances the fast recovery phase when receiving duplicate ACKs, unlike Reno the NewReno can inject a new unsent packet when perform fast retransmission this accommodates aggregated frame at the data link layer and can concatenate many new unsent packets waiting for transmission, NewReno can use delayed acknowledgment (DACK) instead of immediate ACKs this feature can further increase performance, in our simulation we used DACK by 100 ms (in our simulation), if packet timed out then NewReno back to slow start. SACK performs well with frame aggregation because the receiver only acknowledged received packets so the sender only retransmits the unacknowledged packets also unsent queued packets can be sent during fast retransmission. In Reno we see degradation in performance that is because it follow a reserved strategy; Reno exits fast recovery when receiving partial ACKs and go back to slow start, it does not utilize the fast recovery phase. Finally the performance of Vegas degrades when used with frame aggregation because considered as delay-based algorithm. Vegas reduces its sending rate based on accurate calculation of round trip time of packet, due to that a number of aggregated packets may suffer high end to end delay this may lead to fixed window size that is restricts TCP transmission rate to MAC layer. As a result the use of TCP Vegas is not suitable with frame aggregation. By default TCP packet timeout is higher than MAC frame timeout this suits all congestion control algorithms benefits from this to work with frame aggregation.

Table 2. Throughput in Mbps for different TCP congestion control algorithms

Algorithm	A-MPDU size 64KB	A-MPDU size 32KB	A-MPDU size 16KB	A-MPDU size 8KB	A-MPDU size 4KB
Tahoe	78.1	69.1	54.7	36.5	18.1
Reno	61.2	58.3	50.6	33.9	18.1
NewReno	78.9	69.7	55.1	36.7	18.3
NewReno DACK=100	80.9	70.4	55.7	36.9	19.7
SACK	79.3	70	55.2	36.7	18.3
Vegas	17.4	17.4	17.4	17.4	17.4

In figure 7 and figure 8 we tested the network performance using different network loads and TCP NewReno with a congestion window of 2048 Kbyte, 32 Kbyte represents throughput saturation point the results shows approximately linear increase in throughput and decrease in end to end delay, these results shows the improvements and advantages of using frame aggregation over the normal frame transmission of legacy standards where throughput is saturated at certain level of network load. , the work in this paper get advantage of the work in [11, 23, 24] but adds extra knowledge as it discusses the interaction of TCP congestion algorithms with frame aggregation.

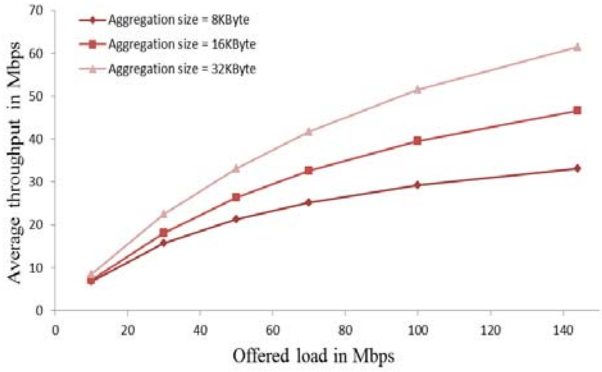

Fig. 7 Average network throughput versus offered load

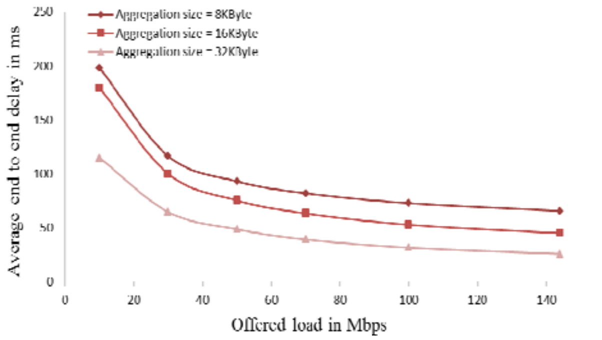

Fig. 8 Average end to end delay versus offered load

VII. CONCLUSIONS

The work in this paper analyses and simulates MAC frame aggregation techniques in the IEEE802.11n WLAN standard and their interaction with TCP congestion control techniques. The simulated network; modelled using NS-2, is composed of a single wireless node and an access point. Different parameters in the MAC and transport layers are considered. With A-MPDU frame aggregation, TCP window size lower than 8 KBs affects network throughput, while it is saturated to a maximum of 80 Mbps for larger values of window size. Network throughput is saturated for a maximum aggregation size between 32 KB and 64 KB. TCP NewReno and SACK outperform other TCP techniques where the maximum throughput achieved is 80 Mbps with a TCP window size of 8 KB and maximum aggregation size of 64 KB. TCP Vegas is not suited with frame aggregation because it depends on RTT to control the rate of transmission, as a result any delay-based TCP congestion control algorithm (like Vegas) does not recommended to use with IEEE802.11n. Packet size also affects performance, where larger packet size means lower overhead on the A-MPDU. A-MPDU aggregation scheme reduces the average end to end packet delay significantly. The reason behind this is multiple packets can be received at once and the overhead produced due to back-off mechanism used in DCF. Throughput in NewReno is enhanced furthermore when using DACK instead of immediate ACK, where the maximum throughput achieved is about 81Mbps. Due to the reserved strategy that Reno implements the throughput is less than other algorithms because Reno exits fast recovery when receiving partial ACKs and go back to slow start.

VIII. REFERENCES

[1] C. Wang, H. Wei, "IEEE 802.11n MAC Enhancement and Performance Evaluation", Mobile Networks Applications, Springer Science, 2008.

[2] O. Abu-Sharkh and M. Abdelhadi, "The Impact of Multi-Rate Operation on A-MSDU, A-MPDU and Block Acknowledgment in Greenfield IEEE802.11n Wireless LANs", In Proceedings of IEEE Wireless Advanced, pp.116-121, 2011.

[3] E. Perahia and R. Stacey, Next Generation Wireless LANs Throughput, Robustness, and Reliability in 802.11n, Cambridge University Press, 2008.

[4] T. Paul and T. Ogunfunmi, "Wireless LAN Comes of Age: Understanding the IEEE802.11n Amendment". Magazine of Circuits and Systems, published by IEEE Circuits and Systems Society, volume 8, issue 1, pp. 28-54, 2008.

[5] M. Kim and C. Choi, "Joint Rate and Fragment Size Adaptation in IEEE 802.11n Wireless LANs", In Proceedings of the 8th Annual IEEE Consumer Communications and Networking Conference - Wireless Consumer Communication and Networking, pp. 942 - 847 2011.

[6] T. Huda, "Throughput Enhancement of IEEE 802.11 WLAN for Multimedia Communications", In Proceedings of AH-ICI. First Asian Himalayas International Conference, pp. 1-6, 2009.

[7] K. Chan, "Evaluations and Enhancements in 802.11nWLANs – Error-Sensitive Adaptive Frame Aggregation", Master's Projects, the Faculty of the Department of Computer Science San Jose State University, 2009.

[8] T. Selvam and S. Srikanth , "A Frame Aggregation Scheduler for IEEE 802.11n", In proceedings of IEEE National Communications Conference NCC, pp.1-5, 2010.

[9] N. Ghazisaidi, "Prototype development of advanced hierarchical frame aggregation in fiber-wireless access networks", Journal of Electronics Letters, Institution of Engineering and Technology publications, Volume 47 Issue 2 pp. 124 – 126, 2011.

[10] K. Chin, "On Maximizing VoIP Capacity and Energy Conservation in Multi-Rate WLANs", journal of Communication Letters, IEEE Communications Society publications, volume 14 , Issue 7 , pp. 611 – 613, 2010.

[11] V. Visoottiviseth et al,"An empirical study on achievable throughputs of IEEE 802.11n devices", In Proceedings of 7th International Symposium on Modeling and Optimization in Mobile, Ad Hoc, and Wireless Networks, WiOPT. , pp.1-6, 2009.

[12] G. Bianchi, "Performance Analysis of the IEEE 802.11 Distributed Coordination Function", IEEE Journal on Selected Areas in Communications, Volume 18, issue 3,pp. 535 – 547, 2000.

[13] D. Skordoulis and Q. Ni, "IEEE 802.11N MAC Frame Aggregation Mechanisms for Next Generation High Throughput WLANs", Journal of Wireless Communications, published by IEEE Communications Society, Volume 15, Issue 1, pp. 40-47, 2008.

[14] B. Kim, et al, "Effect of Frame Aggregation on the Throughput Performance of IEEE 802.11n", In proceedings of Wireless Communication and Networking Conference WCNC, pp. 1740 - 1744, 2008.

[15] IEEE 802.11n Amendment 5, LAN/MAN Standards Committee of the IEEE Computer Society, Approved 11 September 2009.

[16] T. Sivanthi and U. Killat, "Performance Analysis of Inflight Video Streaming over IEEE 802.11n" , In Proceedings of Consumer Communication and Networking Conference CCNC, pp.1-5, 2010.

[17] G. Lee, et al," Robust wireless transmission utilizing PPDU-based aggregation technique for next generation wireless LANs", Journal of Communications letters, IEEE Communications Society publications, Volume 14, Issue3,pp. 205–207, 2010.

[18] H. Jamal and K. Sultan, "Performance Analysis of TCP Congestion Control Algorithms", International Journal of Computers and Communications, Volume 2, Issue 1, 2008.

[19] K. Fall and S. Floyd, "Simulation-based Comparisons of Tahoe, Reno, and SACK TCP", Lawrence Berkeley National Laboratory.

[20] D. Kim, Cano, et al, "A comparison of the performance of TCP-Reno and TCP-Vegas over MANETs", In Proceedings of 3rd International Symposium on Wireless Communication Systems ISWCS, pp. 495 - 499 , 2006.

[21] M. Smith and K. Ramakrishnan, "Formal specification and verification of safety and performance of TCP selective acknowledgment", IEEE/ACM Transactions on Networking, Association for Computing Machinery , Volume 10, Issue 2, pp. 193 – 207, 2002.

[22] E. Altman and T. Jimenez, Network Simulator 2 for Beginners, Lecture notes 2003-2004.

[23] R. Jiang, et al, "Interactions Between TCP and the IEEE 802.11 MAC Protocol", In Proceedings of IEEE Conference Publications the DARPA Information Survivability Conference and Exposition, pp. 273 – 282, vol.1, 2003.

[24] D. Skordoulis, "Adaptive Delayed Channel Access for IEEE 802.11n WLANs", In proceedings of 4th IEEE International Conference on Circuits and Systems for Communications ICCSC, pp. 167 – 171, 2008.