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QoS Proposes by EDCA using WLANs

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Abstract: Wi-Fi or wireless local area networks (WLANs) are among the most popular wireless internet access technologies used. The major challenge faced by WLANs is the provision of **Quality of Service** (QoS) for real-time applications at high congestion periods. IEEE 802.11e draft presents the only comprehensive QoS infrastructure for WLANs which proposes **Enhanced Distributed Channel Access** (EDCA) that propounds the prioritization of medium access for different traffic classes. EDCA while making perceptible improvements for real-time applications are neglects non-real-time applications by allocating their share of the bandwidth to the former in an inefficient manner. In our study, we have found that there are considerable design improvements possible for IEEE 802.11e-based QoS propositions.This paper proposes a novel medium access and transmission mechanism for IEEE 802.11 WLANs that is specifically designed to ensure QoS for triple play services. Based on our study regarding the trafficcharacteristics of triple play services, we propose a mechanism that adaptively uses the medium access and transmission parameters according to the traffic characteristics of the applications, for better utilization of the available bandwidth. Then the mechanism also proffers higher medium access priority to the access point as compared to the stations in order cope with the issue uplink/downlink traffic asymmetry. Simulation-based analysis of the proposed mechanism to the compares its performance with EDCA. The proposed mechanism offers promising results in terms of packet loss, packet delay and throughput, and thus ensures QoS for voice, video and data transfer applications.

Keywords: Quality of service (QoS); IEEE 802.11e; Triple play services; Medium access control (MAC); Medium access priority; Traffic asymmetry.

I. INTRODUCTION

The internet user is familiar with the term Wi-Fi. Our homes, restaurants, airports and even trains, buses and commercial airplanes provide Wi-Fi connectivity. The large-scale use of smartphones and the popularity of voice chat applications have also made Wi-Fi a widely used network for communications. The major utility of Wi-Fi is cheap internet access and support for high data rates, but this explosive increase in Wi-Fi usage has its repercussions that a shared bandwidth(up to 600 Mbps) is divided between all of the stations running different applications simultaneously due to which degradation in performance occurs especially for real-time applications. Therefore, it is required by the network to provide appropriate services to delay sensitive applications and guaranteed required bandwidth to the triple play services of voice, video and internet access. The IEEE 802.11e draft presents the sole brief quality of service (QoS) framework for IEEE 802.11 wireless local area networks (WLANs) by introducing enhanced distributed channel access (EDCA) , which characterizes traffic into high and low priority traffic classes. EDCA implementations divide traffic into four major categories of voice, video, best effort and background traffic. Each traffic category gets different medium access priority with voice getting the highest, video second highest, best effort second least and background getting the least priority to access the medium. This medium access priority is enforced by awarding different contention window sizes to different traffic categories with highest priority traffic category getting the lowest contention window size and different sizes of arbitration inter-frame space (AIFS) with high priority traffic transmitting with shorter inter-frame space.

II. BACKGROUND AND RELATED WORK

There are two generic approaches for the provision of QoS: (1) integrated services and (2differentiated services. The differentiated services approach has been followed in IEEE 802.11 WLANs up till now which advocates priority-based QoS for different traffic types described in and Integrated services involve end-to-end reservation mechanisms and are generically designed for wired networks,but some work has been initiated in this regard in the wireless domain also like in DARE, a distributed end-to-end reservation protocol that uses reservation techniques for QoS in wireless mesh networks . The basic protocol followed by the integrated services approach is the Resource Reservation Protocol (RSVP) . characteristic-based medium access adaptability involving other medium access and transmission parameters like TXOP limit and inter-frame spacing for comprehensive solution to the problem.

III. TRAFFIC CHARACTERISTICS

In order to utilize the bandwidth in the most efficient manner, it is essential to understand the traffic characteristics and behaviour of the traffic generated by triple play applications. This section briefly describes the traffic characteristics of the applications widely used by triple play services.

_ No ON-OFF cycles
_ Short ON-OFF cycles
_ Long ON-OFF cycles

In no ON-OFF cycles, there is no steady phase and all of the data is transferred in the buffering phase while in short ON-OFF cycles, there is a steady state phase in which there is an ON period for 2.5Mb of data followed

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by an OFF period in which no data is transmitted by the streaming source. Long ON-OFF cycles streaming strategy is similar to short ON-OFF cycles except for the ON period which is for more than 2.5 Mb of data.

Table 1: Traffic characteristics corresponding to different VOIP codecs

Codec	Bit rate (kbps)	Packet size (bytes)	Packet interval (µs)	Inter-packet interval (ms)
ITU G.711	64	284 364	2.27 2.91	20 30
ITU G.729	6.4 8 11.4	134 144 154	1.07 1.15 1.23	10 20 30

Web-browsing:

Most of the internet traffic consists of the web-browsing traffic, and nowadays, many applications rely on the hypertext transfer protocol (HTTP) as their application layer protocol for web-browsing. HTTP has multiple versions among which HTTP 1.0 and HTTP 1.1 are the most popular. HTTP 1.0 can download back-to-back requested objects with one TCP connection per object but there is a limit to the number of connections and any left objects are retrieved from the web source after one of the ongoing connections terminate. HTTP 1.1 is similar to HTTP 1.0 except that it allows persistent connections and requests are pipelined.

Webpage is a hypertext document containing text of hypertext markup language (HTML) which links to other objects stored somewhere on another file or server.The hypertext document is called the main object while the linked objects are called in-line objects. A typical HTTP session consists of ON-OFF periods with ON periods corresponding to the period between the request and the retrieval of the last object. The OFF period is the silent interval when no data is transmitted.

IV. PROPOSED MECHANISM

The proposed mechanism maximizes the utility of bandwidth resources and distributes the channel efficiently between stations running different applications while prioritizing medium access for time-critical applications.Furthermore, to increase the downlink throughput, the proposed mechanism grants the access point greater medium access priority as compared to the stations for most of the traffic categories. In addition to the classification of different traffic and their subsequent prioritization over each other, the proposed mechanism utilizes the traffic behaviour and selects an appropriate transmission style for the type of traffic that is about to be transmitted by the station or the access point. Figure 1 shows the architecture of the proposed mechanism while the following subsections discuss the system in detail.

1. Voice: To accommodate the bursty nature of voice traffic, reduced inter-frame space (RIFS) introduced in the IEEE 802.11n standard is used by the proposed mechanism as RIFS is more suitable for traffic that has considerable inter-frame interval but require higher medium access priority. No frame aggregation is used by the mechanism for voice data traffic.

2. Video: video traffic consists of a stream of packets so the proposed mechanism uses frame aggregation for the entire period of TXOP limit which means no inter-frame space between the frames.

3.Web-browsing:Web-browsing traffic has a burstynature (small data intervals and large silent intervals)so SIFS is used.

4. File-sharing: frame aggregation is used by the proposed mechanism for file-sharing traffic, for the entire TXOP limit period as file-sharing traffic consist of streams of back-to-back packets which are unordered. However, file-sharing traffic is insensitive to unordered packet delivery.

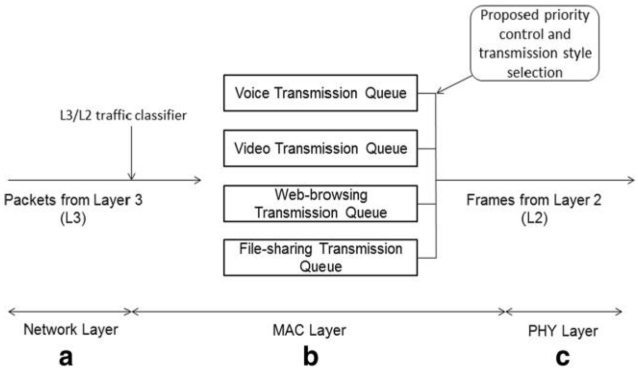

Figure1:Architecture of the proposed mechanism.
(a) Packet processing at the network layer.
(b) Assignment of transmission queues (TxQ) and subsequent proposed priority control and frame transmission style selection at MAC layer.
(c) Processing of datagrams at PHY layer.

Test and simulation environment:

The proposed medium access and transmission mechanism has been implemented using the open source simulation tool NS-2 . We have used the TKN EDCA Model for NS-2 [36] for the EDCA simulations, but to simulate the proposed mechanism, we had to modify the TKN EDCA Model according to the propositions mentioned in Section 4 by (1) changing the medium access priorities for different traffic, (2) awarding higher medium access priority to the access point for some of the traffic, (3) associating different transmission styles with different traffic queues and (4) adaptively changing the TXOP limit for each traffic queue. By integration of the modified TKN EDCA Model with the 802.11 MAC layer of NS-2, the proposed mechanism was successfully implemented. Data samples were taken for simulation runtimes of up to 5,000 during which most of the nodes try to transmit simultaneously, but there are silent intervals for some nodes as well.

Table 2 TXOP limits for different traffic categories:

Traffic category	Relative TXOP limit
Voice	Low
Video	High
Web browsing	Moderate
File sharing	Moderately high
Network control	Lowest

Simulation topologies and parameters

As mentioned earlier, the proposed mechanism has been simulated using six different simulation topologies each with a different number of stations. In order to make the results easy to compare and evaluate, different stations having different traffic were considered as traffic from two distinct applications that exist on the same station contend for the medium in the same way as traffic that exist on two different stations when subjected to differentiated service-based QoS implementations like EDCA. The distribution of traffic stations for each network topology.

Voice traffic is generated with the parameters of ITU G.711 in order to exercise the proposed mechanism with the heaviest of voice codecs. ITU G.711 codec has 64 kbps of bit rate, 284 bytes of packet size, 2.27 µs of packet interval, and 20 ms of inter-packet interval. Video traffic parameters depend upon the type of browser and the container as mentioned earlier so there are no specific values for video traffic;

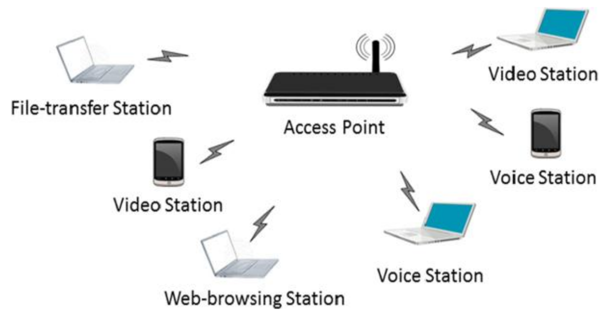

Figure 2 A general WLAN environment exhibiting the 6-station simulation topology

Throughput:

Voice traffic throughput for EDCA and the proposed mechanism closely follow each other, both at the uplink and the downlink when the network load is low (4–5transmitting stations). However, when the network load is moderate (6–7 transmitting stations), the throughput of the proposed mechanism at the uplink closely follows that of EDCA, but at the downlink, the proposed mechanism faces a slight degradation in throughput as compared to EDCA, this slight depreciation is because of the relatively higher medium occupancy by other applications at that instant. At high network loads (8–9 transmitting stations), sharp depreciation in the throughput of EDCA is seen both at the uplink and downlink while the proposed mechanism offers a gradual depreciation in throughput as depicted in Figure 3a. Video traffic's throughput for EDCA and the proposed mechanism closely follow each other both at the uplink and downlink as shown in Figure 3b. But as the network load increases, abrupt depreciation is experienced by EDCA, and the proposed mechanism betters EDCA at each value of the network load because of the better adaptiveness of the proposed mechanism for video traffic. Webbrowsing traffic observes less throughput at the uplink for the proposed mechanism than EDCA when the network load is low, but when the network load gets higher (6–9 transmitting stations), the proposed mechanism experiences higher throughput than EDCA and the throughput depreciation with increasing network load is also gradual in the case of the proposed mechanism as shown in Figure 3c.

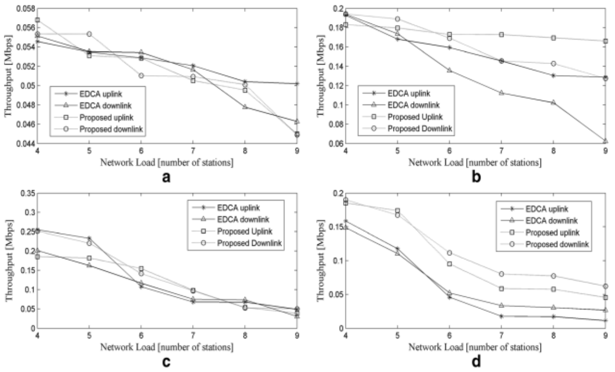

Figure 3 Average station throughput over the network load. (a) Voice traffic. (b) Video traffic. (c) Web-browsing traffic. (d) File-sharing traffic.

Discussion and performance comparison

We evaluate the performance of the proposed mechanism by providing the percentage enhancement it offers for throughput, packet delay and packet loss in comparison to EDCA. We discuss our findings as follows.

Throughput performance

Voice traffic throughput has no significant improvement or depreciation at the uplink and at the downlink as voice traffic has low bit rate, due to which all implementations are able to provide the required throughput. Although a slight depreciation in performance can be observed at the downlink for moderate network load but there is no significant anomaly shown by the application in practical as the average throughput drop for the downlink of voice traffic at moderate network load is only 3.92% and as mentioned in Section 3, the major factors that influence voice traffic performance are packet loss and packet delay.

Packet delay performance

The packet delay performance of the proposed mechanism for voice, video, web-browsing and file-sharing applications is shown in Figure 4a,b. Voice traffic experiences up to 10% to 35% improvement in packet delay performance at the uplink while there is some depreciation in performance as well because the proposed mechanism while maintaining fairness among different applications slightly suppresses the medium access ability of voice applications as compared to EDCA. But the delay for the proposed mechanism is always less than the threshold value as mentioned earlier. At the downlink, there is no significant performance enhancement or depreciation for voice traffic.

V. RESULTS AND ANALYSIS

We compare the performance of the proposed mechanism with EDCA through three performance metrics of average throughput, average packet delay and average packet loss.

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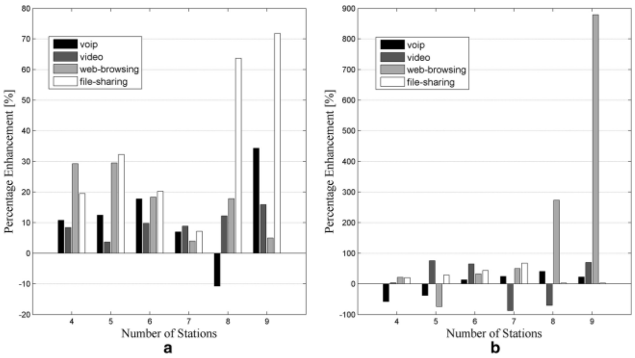

Figure 4 Percentage packet delay enhancement. (a)Uplink.(b)Downlink.

VI. CONCLUSION

In this paper we have presented an improved medium access and transmission mechanism for IEEE 802.11 WLANs which ensures provision of QoS for triple play services in a WLAN environment, first, by prioritizing medium access of the stations and the access point according to the volume of the traffic that they have to transmit and according to the sensitivity of the application generating the traffic to packet delay, packet loss and throughput. Secondly, the proposed mechanism makes the transmission style of the nodes adaptive according to the traffic characteristics of the type of traffic a node is about to transmit .The research aimed for the provision of QoS for triple play services of voice, video and internet access (web-browsing and file-sharing).

VII. FUTURE WORK

We are proposed solution can be implemented on any station by making changes to its wireless adapter's driver and on the access point by adding some functionality in its firmware.

REFERENCES

[1] Specific Requirements Part 11: Wireless Physical Layer (PHY) Specifications Amendment 8: MAC Quality of Service(QoS), IEEE Standard 802.11e, 2005
[2] T Nilsson and J Farooq, A Novel MAC scheme for solving the QoS parameter adjustment problem in IEEE 802.11e EDCA, in Proc. 2008 IEEE international symposium on a world of wireless, mobile and multimedia networks, Newport Beach, CA, 23–26 June 2008, pp. 1–9
[3] Arpit Gupta, Jeongki Min and Injong Rhee, WiFox: Scaling WiFi Performance for Large Audience Environments, in CoNEXT'12 Proceedings of the 8th international conference on Emerging networking experiments and technologies, Cote d'Azur, France, 10–13 December 2012, pp. 217–228
[4] W Stallings, Integrated and differentiated services, in High-Speed Networks and Internets: Performance and Quality of Service, ed. by MJ Horton, 2nd edn.(Prentice-Hall, Upper Saddle River, New Jersey, 2002), pp. 469–502 NG Corporation, Implementing the VOIP Network. Tech. Brief. Series onVOIP and Converged Net. 3(6) (2005).
[5] John Williams, Troubleshooting IP Video Quality of Service,http://www.jdsu.com/ProductLiterature/IPVIDEOQOS_WP_ACC_TM_AE.pdf. Accessed15 Dec 2013
[6] MadWifi, http://www.madwifi-project.org/. Accessed 27 Jan 2014
[7]. DD-WRT, http://www.dd-wrt.com/. Accessed 27 Jan 2014. 8.OpenFlow, https://www.opennetworking.org/. Accessed 27 Jan 2014.