Receiver-Driven Queue Management For Achieving RTT-fairness in Wi-Fi Networks

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Abstract — **In this paper, we introduce a novel queue management technique for the buffer space at the base station of infrastructure 802.11 networks that considers mobile user's receiving characteristics. The maximum amount of the base station buffer that can be used by a given flow is updated proportionally to RTT, measured at the mobile nodes and sent to the base station my the mean of link layer acknowledgements. In this way, the proposed scheme remains transparent to high-level protocols. The proposed approach makes possible the implementation of algorithms able to provide RTT-fairness in wired-cum-wireless networks. Results show the advantage of the proposed queue management scheme when compared to that of traditional drop-tail queue management.[1]**

Keywords – active queue management, AQM, wireless fairness, buffer management, RTT fairness

I. INTRODUCTION

The IEEE 802.11 standard (WiFi) has become the de facto standard technology for wireless access in the last mile, specially the so called infrastructure mode in which a base station (BS) is connected to the Internet, allowing mobile nodes (MN) to communicate with fixed users and servers. In this scenario, the wireless link is usually the bottleneck of the end-to-end path between the fixed sender located in the Wide-Area Network (WAN) and the mobile receiver. This makes the BS buffer a key point for enhancing the performance of data transfer to the mobile receiver.

Roughly 80% of the bytes transferred on the Internet are transmitted by the transport level Transmission Control Protocol (TCP). The sending rate of TCP is governed by its congestion control mechanism that uses a pattern of acknowledgements sent by the TCP receiver to adjust the transmission rate of the TCP sender. Although this adjustment can bring equal share of the available bandwidth when all the TCP connections sharing a channel have the same characteristics, it can produce unbalanced sharing otherwise. Specially, a TCP sender will receive a lower rate of acknowledgement messages from a receiver located far away than it receives from a TCP receiver closer to the TCP sender. This makes the transmission window of the connection with the receiver closer to the sender to grow much faster than that of the receiver far away, grabbing a larger portion of the channel capacity [6].

Providing fairness among TCP flows in WiFi networks is a known problem and several studies have analyzed this issue [12-19]. RTT unfairness is particularly relevant to mobile users in infrastructure WiFi networks since most of the traffic comes from TCP senders located on the Internet to the mobile users, limiting the receiving rate of mobile users located far away from their communicating peers. Solutions proposed so far to this problem demand changes in TCP making difficult, if not infeasible, their deployment on a large scale.

This paper proposes a scheme for the introduction of active management of the BS queues in infrastructure WiFi networks called Receiver-Driven Queue Management (RDQM) for improving the performance of the transmissions to mobile users. The criteria used in the management of the queues can be oriented to specific goals and in this paper the proposed scheme aims at providing RTT fairness. However, the queue management scheme can be employed to achieve other goals. The advantage of this management scheme resides in the fact that it requires no changes in high-level protocols. Moreover, it is totally transparent to the wired network.

To handle the problem of RTT-fairness in wired-cum-wireless environment, the RDQM algorithm implemented at the BS limits the maximum buffer space that can be allocated for an incoming flow proportionally to its associated RTT values. It is shown that the RDQM scheme can ameliorate the RTT fairness problem since it addresses the causes of the problem by distributing the space at the bottleneck buffer according to the RTT values. Moreover, RDQM operates at the link/network layer which makes it transparent to and consistent with any TCP version operating on top of it, demanding no change in TCP.

The rest of the paper is organized as follows: Section II reviews related work. Section III presents the core of the proposed approach including description of main architecture as well as specification of the algorithm implemented as the RTT-unfairness solution; Section IV presents performance evaluation results confirming the design assumptions; and Section V concludes the paper drawing some considerations for future work on the topic.

[1] This work has been partially supported by the Italian National Project: Wireless multiplatfOrm mimo active access netwoRks for QoS-demanding muLtimedia Delivery (WORLD), under grant number 2007R989S

II RELATED WORK

Previous works dealt with throughput fairness problem. In [15], the unfairness between upstream and downstream transmission is demonstrated by showing measured values. Manipulation of the receiver advertised window to ensure the upstream/downstream fairness is proposed in [16]. In [17], buffer resizing and the prioritization of TCP-ACK packets are adopted. In [18], the per-station fairness is calculated as network cost for setting the ECN bits.

In [19], the authors proposed a fair sharing mechanism that employs a token-bucket filter, aiming at overall fairness between the uplink and the downlink rather than on a per-flow fairness. Other works [20, 21] proposed solutions to the fair allocation problem by adjusting the contention window parameters dynamically.

The RTT-fairness problem is also addressed by several TCP variants for wired networks such as TCP Libra [6] and TCP Hybla [7]. In these protocols, the sending rate is as a function of the RTT values to provide flows with equal opportunity. Nonetheless, TCP variants require modification of the TCP sender code and, thus, their dissemination on a large scale can be infeasible

Availability of different queue management techniques provides an attractive way to control the performance of individual flows at the network routers. Despite the good performance in wired networks, many active queue management techniques show poor performance when adopted in wireless networks. For example, RED needs to be enhanced with ECN marking to bring performance advantages [12]. Moreover, the burden of processing its requirements makes it impractical for the most common light-weight software running in the Access Points (AP).

In [13], the authors proposed a management scheme which admits flows into dedicated virtual queues served in a round-robin fashion. This mechanism guarantees fairness between uplink and downlink flows as well as between multiple flows following in the same direction.

The work in [14] proposes two alternative approaches. The first one, called PMS-AF (Selective Packet Marking with ACK filtering), is based on packet priority marking by the sender to assist its discard at the AP. The second, called LAS (Least Attained Service), guarantees fairness between all flows giving higher priority to the least served flows.

All approaches overviewed above either depend on a specific network infrastructure [12] or require modifications at the sender [14]. The closest approach to the one presented in this paper is [13]. However, as many other proposals, it operates as a standalone module and it doesn't use any parameters provided by other network nodes.

II. RECEIVER-DRIVEN QUEUE MANAGEMENT

The proposed Receiver-Driven Queue Management (RDQM) scheme aims at optimizing the performance of data transfer over Wi-Fi infrastructure networks. RDQM is implemented at the BS node and it manages the outgoing queue of the wireless interface (by either enqueuing or discarding packets). RDQM uses feedback information provided by the mobile receiver which is encapsulated into the link layer acknowledgment frame. In the next subsections the architecture of RDQM and its employment to ameliorate the RTT-unfairness problem are described.

A. Architecture

In networks with mobile users and wireless access links, end-to-end transmissions (connections) transverse heterogeneous channels (see Fig.1).

In RDQM architecture the operation of the buffer management at the base station is driven by information fed by the mobile user, such as RTT, bandwidth-delay product, or any other metrics easily measurable at the Mobile Node (MN) end.

Information is provided by the MN to the BS in the *duration* field of the 802.11 link layer header which is a field with 14 bits (Fig. 2) unused by the IEEE 802.11 standard. The IEEE 802.11 MAC [4] employs a stop-and-wait ARQ scheme requiring the receiver to positively acknowledge every successfully received frame. As a result, the MNs can send any signaling information destined to the BS. In this way, unnecessary changes of existing MAC protocol are not needed, favoring coexistence with the standard implementation and incremental deployment of the mechanism.

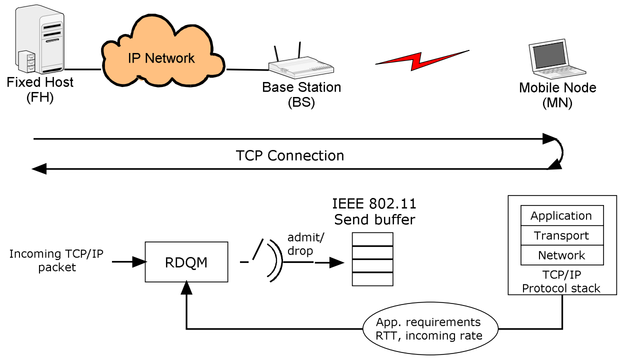

Figure 1. Receiver-Driven Queue Management (RDQM) architecture.

Octets:	2	2	6	4
	Frame Control	Duration	ADDR1	FCS

Bit 15	Bit 14	Bits 13-0	Usage
0	0 - 32 767		Duration
1	0	0	Fixed value during CFP
1	0	1 - 16 383	Feedback Information
1	1	0	Reserved
1	1	1 - 2007	AID in PS Poll Frames
1	1	2 008 - 16 383	Reserved

Figure 2. Reserved portion of IEEE 802.11 link layer ACK frame.

B. Achieving RTT-fairness

A first-come-first-served policy (drop tail) can yield TCP connections with large RTT to starvation since packet belonging to connections with small RTT can monopolize the buffer space. To promote RTT fairness among TCP connections, the RDQM divides the buffer space among the

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flows proportionally to their RTT. Initially, the RTT is taken as that experienced by the SYN TCP segment [3]. After that, the RTT value is measured during connection lifetime [1, 10].

For every packet the BS receives from the wired network, it verifies whether or not the number of enqueued packets of the same flow, cur_i, exceeds the maximum number of allowed packets in queue for that flow, max_i. In case it exceeds, the packet is dropped, otherwise it is enqueued. Such scheme corresponds to a partial sharing scheme with thresholds defined dynamically [11].

RDQM tries to allocate buffer space to accommodate the bandwidth-delay product of that flow estimated as the product of the measured RTT by the wireless link available bandwidth. Such policy is based on the well-known principle for TCP New-Reno [5] that states that for a connection to fully utilize the path between a sender and a receiver [2] the buffer space at a bottleneck link should be at least equal to the bandwidth delay product between the sender and the receiver.

Algorithms 1 to 3 present the details of the proposed approach. Algorithm 1 describes the procedures implemented at the mobile node which essentially inserts the measured RTT values into outgoing LL-ACK packets. Algorithm 2 corresponds to the procedure implemented at the BS which either accepts or discards incoming packets. Algorithm 3 updates the queue thresholds, *i.e.*, cur_i, and max_i. Algorithm 2 is triggered by the arrival of TCP data packet at the base station from the fixed sender while Algorithm 3 is performed for every incoming link layer ACK from the mobile node.

Algorithm 1: On LL-ACK packet transmission at MN

```
If TCP data packet received
    Encapsulate feedback info into outgoing LL-ACK
    Send LL-ACK
Endif
```

Algorithm 2: On TCP DATA packet arrival at BS

```
Get Flow ID
If packet belongs to a new flow Then
    Create new entry for incoming flow
    Set Flow Packet Limit equal to MAX_def
    Set Flow Packet Count equal to zero
    Set Flow RTT equal to RTT_def
Else
    If Flow Packet Count is greater than Flow Packet Limit Then
        Drop incoming packet
    Else
        Increase Flow Packet Count by one
        Accept packet
    Endif
Endif
```

Algorithm 3: On LL-ACK packet arrival at BS

```
Get Flow ID
If Flow RTT is greater that zero Then
    Set RTT Sum equal to zero
    For Each Flow Do
        Increase RTT Sum by Flow RTT
    Endfor
    For Each Flow Do
        Set Flow Weight to Flow RTT / RTT Sum
        Set Flow Packet Limit to max between Flow Weight and buffer size
    Endfor
Endif
```

III. PEFORMANCE EVALUATION

This section assesses the performance of the proposed RDQM scheme. The aim is to confirm that the proposed scheme is capable of ammeliorating the RTT-fairness problem.

A. *Scenario description*

RDQM is implemented as an extension of ns-2 [8] simulator. Fig. 3 shows the details of the simulated topology which includes four Fixed Hosts (FH) generating TCP traffic and four wireless receivers. TCP transmissions are initiated at the FHs. The BS interconnects fixed and wireless segments of the network. FHs are connected to the router with links experiencing 5ms, 20ms, 50ms, and 100ms one-way propagation delays. The bandwidth of all wired links is set to 100Mb/s which corresponds to the transmission rate of the most commonly used Ethernet standard while the IEEE 802.11b wireless link operates at 11 Mb/s.

Table I summarizes the setup of simulation parameters. All simulation results were averaged from 10 runs. Confidence intervals have 95%confidence level.

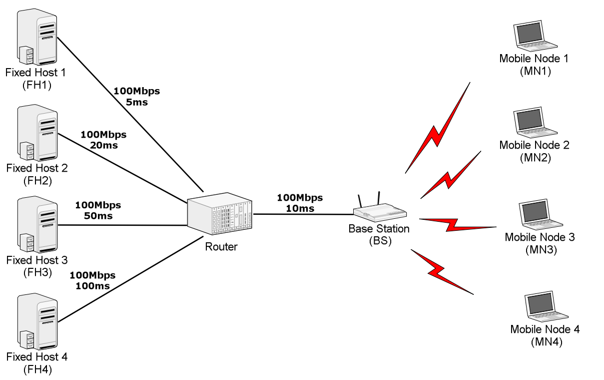

Figure 3. Simulation scenario.

B. *Experimental results*

The experiments aim at demonstrating the improvement of the throughput fairness between flows with different RTT values. In order to do so, we confirm by simulations the fact

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that flows with large RTT cannot fully utilize available resources when the BS has a small buffer size which does not happen to flows with small RTTs (Fig. 4). Then, we measure the throughput and fairness level between flows with different RTTs for a standard drop tail queue (Fig. 5). Finally, we analyze the throughput and the fairness when the RDQM scheme is used (Fig. 6).

With RDQM all flows achieve equal throughput level regardless of their RTTs and of the allocated buffer space at the BS (with the exception of l buffers smaller than 200 packets). Finally, the level of the fairness achieved is analyzed using Jain's index [9] (Fig. 7) and the coefficient of variation of the obtained throughputs (Fig. 8).

Fig. 4 presents the throughput as a function of the BS buffer size achieved by single TCP connections originated at the FH towards the wireless part of the network. The propagation delays connecting FHs to the router ranged from 5ms to 150ms.

TABLE I
SIMULATION PARAMETERS

ns-2	
Simulation Time	1000 s
IEEE 802.11b	
Propagation Model	TwoRayGround
PHY data rate	11 Mbps
PHY basic rate	1 Mbps
RTSThreshold	3000 bytes
TCP NewReno	
Packet size	1500 bytes
Max Congestion Window Size	4096 packets
Max Receive Window	1024 packets
Slow Start Threshold	64 packets

The results obtained indicate that small buffer sizes prevent TCP flows of reaching full utilization of the wireless link whereas large buffer sizes allow individual flow to grow their sending rate fully utilizing the available bandwidth. It should also be noted that no further improvement in the throughput can be achieved by increasing the buffer size beyond the available bandwidth. In the example in Figure 4, the available wireless link capacity is equal to 4.5 Mb/s. This value will be used as a reference for the calculation of BDP during the execution of the RDQM algorithm.

This method of BDP calculation tends to overestimate the end-to-end bandwidth between the sender and the receiver nodes especially when the wireless link is shared by multiple flows or the connection bottleneck is somewhere in the fixed part of the network. Such, overestimated BDP value leads to overallocation of the BS buffer space. However, this small increase in memory usage does not imply on significant disadvantage since the utilization remains at 100% for any buffer size greater than the actual BDP of the flow.

Figures 5 and 6 show the throughput as a function of the buffer size when bottleneck is shared by four connections ordinate at differed FH nodes. The links connecting the FHs to the network have different propagation delays ranging from 5 ms to 100 ms. Figure 5 illustrates the results obtained for a drop tail queue management while Figure 6 provides results when RDQM is used.

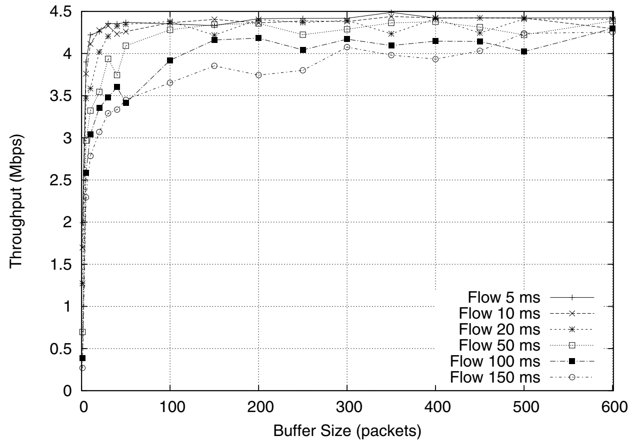

Figure 4. Single-flow throughput as a function of buffer size.

The effectiveness of RDQM can be assessed by measuring the distance between the curves produced by connections flowing through links with different propagation delays. For buffer sizes smaller than 100 packets the difference in throughput produced by drop tail is 400 kbps higher than that given by RDQM. The throughput difference decreases significantly for buffer sizes larger than 300 packets when RDQM is employed. Moreover, the difference is smaller than 100 kbps under RDQM. However, such decrease is not observed when drop tail is used. Under drop tail, the difference between flows does not decrease significantly and it can be of the order of 500 kbps.

The comparison of these scenarios unveils the benefits of RDQM which considerably outperform the traditional drop tail approach when comparing the fairness among flows with different RTTs. Even for buffer sizes smaller than the BDP of the flow, the RDQM scheme is able achieve better fairness level than do drop tail.

Figure 7 plots the Jain index of RDQM flows, achieving values close to 1.0 for buffer sizes larger than 200 packets. Moreover, Figure 8 shows the coefficient of variation of the obtained throughput for each buffer size which is 0.1 smaller than that achieved by the drop tail.

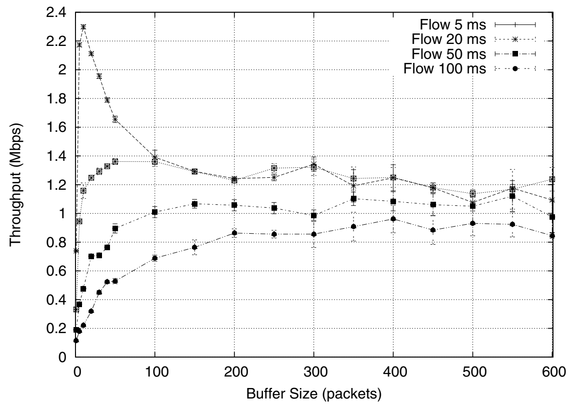

Figure 5. Multiple-flows and no queue management used.

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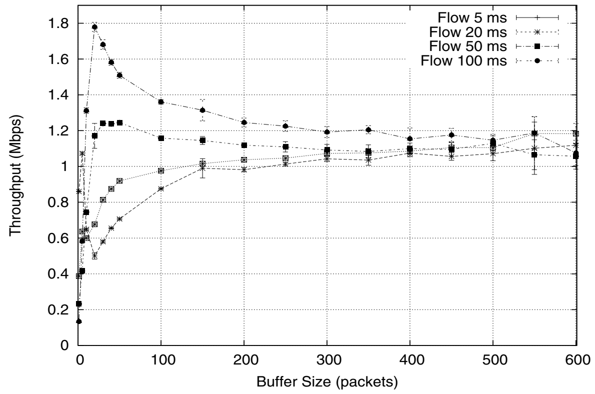

Figure 6. Multiple-flows and RDQM scheme used.

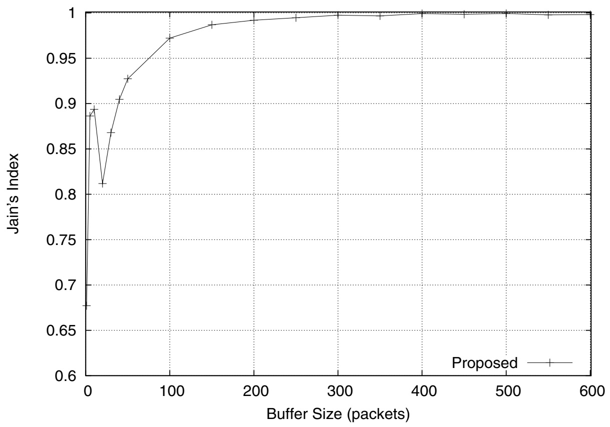

Figure 7. Jain's fairness index

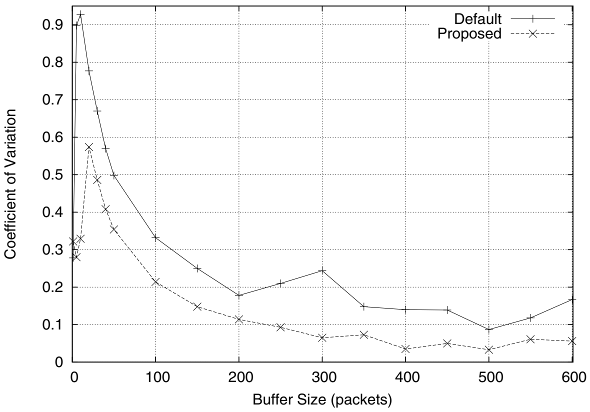

Figure 8. Coefficient of variation

IV. CONCLUSIONS

As portable devices become popular, it is common to have mobile users accessing data and services available on fixed host nodes on the Internet. In this scenario, mobile users tend to be consumers of information available on the Internet.

The distances between fixed nodes and the base station, which bridges fixed and mobile network parts, vary and so does the propagation delays of their links. Such variety contributes to the problem of RTT unfairness making flows over links with long propagation delays receive smaller portions of the available bandwidth and buffer resources than do flows over links with shorter propagation delays.

This paper introduced a novel management scheme for the buffers at the base station which takes into account information available at the mobile receivers to achieve specific goals. The employment of such scheme to ameliorate the RTT unfairness problem was illustrated using different scenarios.

Results indicate that the proposed scheme outperforms traditional drop tail achieving high levels of fairness of throughput among flows with different RTTs.

Future work will explore other design goals.

REFERENCES

[1] G. Lu and X. Li, "On the correspondency between TCP acknowledgment packet and data packet," 3rd ACM SIGCOMM conference on Internet measurement, Miami Beach, FL, USA, 2003.

[2] R. N. Shorten and D. J. Leith, "On Queue Provisioning, Network Efficiency and the Transmission Control Protocol," IEEE/ACM Transactions on Networking, vol. 15, no. 4, pp. 866 – 877, Aug. 2007.

[3] Hao Jiang and C. Dovrolis, "Passive estimation of TCP round-trip times," ACM SIGCOMM Computer Communication Review, vol. 32, no. 3, pp. 75 – 88, 2002.

[4] IEEE Std. 802.11-1999, Part 11, "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications," ISO/IEC 8802-11:1999, 1999.

[5] S. Floyd and T. Henderson. "The NewReno Modification to TCP's Fast Recovery Algorithm", RFC2582, Apr. 1999.

[6] G. Marfia, C. Palazzi, G. Pau, M. Gerla, M. Y. Sanadidi and M. Roccetti. "TCP Libra: Exploring RTT-fairness for TCP", Technical Report UCLA-CSD TR-050037, UCLA, Computer Science Dept., Los Angeles, CA, 2005.

[7] C. Caini and R. Firrincieli, "TCP Hybla: a TCP Enhancement for Heterogeneous Networks", International Journal of Satellite Communications and Networking, John Wiley & Sons, vol. 22 , no. 5, pp 547 – 566, Sep. 2004.

[8] ns2, The Network Simulator. Available at http://www.isi.edu/nsnam/ns/

[9] R. Jain, D. Chiu, and W. Hawe, "A Quantitative Measure Of Fairness And Discrimination For Resource Allocation In Shared Computer Systems", DEC Research Report TR-301, September 1984.

[10] B. Veal, K. Li and D. K. Lowenthal, "New Methods for Passive Estimation of Round-Trip Times Using TCP Timestamps," 6th Workshop on Passive and Active Measurement (PAM), Mar. 2005.

[11] F. Kamoun and L. Kleinrock, "Analysis of shared finite storage in computer network node environment under general traffic conditions", IEEE Transactions on Communications, vol. 28, pp. 992-1003, 1980.

[12] S. Yi; M. Kappes; S. Garg; X. Deng; G. Kesidis; C. R. Das "Proxy-RED: An AQM Scheme for Wireless Local Area Networks", Wireless Communications & Mobile Computing, vol. 8, no. 4, pp. 421-434, May 2008.

[13] X. Lin, X. Chang and J. Muppala, "VQ-RED: an efficient virtual queue management approach to improve fairness in infrastructure WLAN", Proceedings of IEEE Workshop on Wireless Local Networks (WLN), Sydney, Australia, pp. 632–638, Nov. 2005.

[14] M. Gong, Q. Wu and C. Williamson, "Queue management strategies to improve TCP fairness in IEEE 802.11 wireless LANs", Proceedings of RAWNET, Boston, MA, 2006.

[15] S. Pilosof, R. Ramjee, D. Raz, Y. Shavitt and P. Sinha, "Understanding TCP fairness over wireless LAN", Proceedings of IEEE INFOCOM, vol. 2, pp. 863–872, Apr. 2003.

[16] Eun-Jong Lee, Hyung-Taig Lim, Seung-Joon Seok, and Chul-Hee Kang1, "A Scheme for Enhancing TCP Fairness and Throughput in IEEE 802.11 WLANs", NETWORKING 2007. Ad Hoc and Sensor Networks, Wireless Networks, Next Generation Internet, Lecture Notes in Computer Science, vol. 4479, Nov. 2007.

[17] T. Li and D. J. Leith, “Adaptive Buffer Sizing for TCP Flows in 802.11e WLANs”, International Conference on Communications and Networking in China, 2008. ChinaCom 2008, pp. 7-11, Aug. 2008.

[18] Eun-Chan Park, Dong-Young Kim and Hwangnam Kim “A Cross-Layer Approach for Per-Station Fairness in TCP over WLANs”, IEEE Transactions on Mobile Computing, vol. 7, no. 7, pp. 898-911, Jul. 2008.

[19] N. Blefari-Melazzi, A. Detti, A. Ordine and S. Salsano, “Controlling TCP Fairness in WLAN access networks using a Rate Limiter approach”, 2nd International Symposium on Wireless Communication Systems, pp.375-379, Sept. 2005.

[20] H. Ma, H. Li, P. Zhang, S. Luo, C. Yuan, X. Li, “Dynamic optimization of IEEE 802.11 CSMA/CA based on the number of competing stations”, Proceedings of IEEE International Conference on Communications (ICC), vol. 1, pp. 191–195, June 2004.

[21] F. Cali, M. Conti, E. Gregori, “Dynamic tuning of the IEEE 802.11 protocol to achieve a theoretical throughput limit”, IEEE/ACM Transactions on Networking, vol. 8, no. 6, pp. 785– 799, Dec. 2000.

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