A Queue Management Mechanism for Improving TCP fairness in Wireless Access Networks

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Abstract. *Due to the contention nature of Wi-Fi medium access control, several fairness issues exist. Moreover, the Access Point (AP) buffer space plays an important role in the provisioning of fairness to the flows. This paper presents a novel queue management technique designed to ameliorate fairness problems in Wi-Fi last mile networks. The proposed mechanism allocates the AP buffer space based on information available at the mobile nodes and sent to AP via link layer acknowledgement frames. The proposed mechanism is transparent to higher layers as well to the wired part of the network. Performance results show that the novel mechanism leads to considerable improvement when compared to traditional drop-tail queue management.*

1. Introduction

The IEEE 802.11 (Wi-Fi) *infrastructure mode* consists of two types of nodes: access point (AP) and mobile station (MS). The AP connects the wired and the wireless parts of the network, allowing the MSs to communicate with fixed stations and servers on the Internet. The IEEE 802.11 standard does not differentiate the operation of AP and MSs for the access to the medium and for packets buffering. As a result, the APs become the bottlenecks for end-to-end connections, and their buffer key points for enhancing the network performance.

Despite the emergence of real-time and peer-to-peer application, mostly based on User Datagram Protocol (UDP), the Transmission Control Protocol (TCP) is still responsible for the majority of the bytes transferred on the Internet [Williamson 2001]. The throughput of TCP connections is controlled by the congestion control mechanism, which evolution is driven by the pattern of acknowledgements generated by the TCP receiver. Flow and congestion control mechanisms control the amount of data with pending acknowledgement the TCP sender is allowed to send into the network. In steady state, TCP increases its transmission rate until a segment is lost. If the loss is due to the reception of three duplicate acknowledgements, the TCP sender reduces its sending window by half. This mechanism is intended to promote equal share of the available bandwidth among TCP connections using the same link. However, such fairness cannot be guaranteed when a part of the flows receive acknowledgements from the receivers transmitting at low rates and located far away, because the transmission windows of the connections with small Round-Trip Time (RTT) increase significantly faster than the rate of increase of connections with large RTT. This is the so called RTT-fairness problem reported in various studies [Marfia *et al.* 2005]. This problem is especially relevant to mobile stations in infrastructure Wi-Fi networks since most of the traffic comes from the TCP senders located on the Internet, limiting the receiving rate of mobile users located far away from their communicating peers. Solutions proposed so far to this problem demand changes in the TCP protocol, making difficult, if not infeasible, their deployment on a large scale.

A second important fairness issue in Wi-Fi network is known as anomaly of IEEE 802.11 MAC which arises in networks with multi-rate nodes. In such scenarios, cumulative throughput becomes limited by the throughput of the node transmitting at low rates, which degrades the performance of those stations that could transmit at high rates. This is due to the fact that the medium access protocol mechanism provides equal opportunities for channel access to all nodes in the network, but it does not guarantee the same utilization opportunities in time-domain. This means that, in the long term, all competing nodes with the same backlog can access the medium an equal number of times, but nodes with low data rate will use the medium for longer periods than those with high data rates since their packets demand longer times to be transmitted. Consequently, nodes with high data rates defer longer to access the medium. Moreover, resources are wasted, since performance of all flows is limited by the performance of the slower flow [Heusse *et al.* 2003]. Several investigations have analyzed and proposed adaptation mechanisms to circumvent this anomaly [Heusse *et al.* 2003, Kim *et al.* 2005, Branquinho *et al.* 2006, Yoo *et al.* 2005]. These solutions involve modifications of 802.11 parameter values, such as the frame size [Yoo *et al.* 2005] and the contention window size values [Kim *et al.* 2005, Branquinho *et al.* 2006]. These are non-standardized solutions and as a consequence their deployment is limited.

This paper proposes a novel active queue management for the buffer space at the APs called Receiver-Driven Queue Management (RDQM) for mitigating unfairness problems, and, consequently, enhancing performance. Actually, the RDQM framework can implement different algorithms for different goals. In this paper, the specific scheme aims at providing RTT-fairness as well as anomaly mitigation. To handle RTT-fairness problems in wired-cum-wireless scenario, the proposed algorithm limits the maximum buffer space that can be allocated to a flow proportionally to its RTT value. In networks with nodes running at different data rates, the buffer space is allocated proportionally to the bandwidth-delay product (BDP) of the flows, for mitigating the multi-rate anomaly. The advantages of this management scheme resides in the fact that it requires no changes in high-level protocols as well as in the wired network part and it can be easily implemented in legacy devices, given that it is based on virtual-queue management. Moreover, the RDQM mechanism is transparent to and consistent with any TCP version operating on top of it, demanding no change in TCP since it operates at the link layer.

It is shown that the RDQM ameliorates RTT-fairness problems as a consequence of the distribution of the buffer space among flows according to the connections RTT values, as well as it ameliorates the anomaly problem, since it reserves less resources to flows with low data rate, back-pressuring their transmission rates.

This paper is organized as follows: Section 2 reviews related works on Wi-Fi unfairness. Section 3 presents a description of the proposed approach. Section 4 evaluates the performance of the proposed mechanisms. Finally, Section 5 concludes the paper.

2. Related Works

Fairness in wireless networks has been extensively studied. One type of unfairness results from performance difference between the upstream and the downstream [Freitag 2006]. Such unfairness is typically ameliorated by advertising the receiver window size to ensure equal resource allocation between all flows in the network [Lee *et al.* 2007]. In [Li and Leith 2008], buffer resizing and flow prioritization for TCP-ACK packets of downlink flows to avoid the unfairness problem was proposed, whereas in [Park *et al.* 2008] a per-station fairness algorithm based on network cost for setting the ECN bits was introduced. In [Blefari-Melazzi *et al.* 2005], it was presented a fair scheduling mechanism that employs a token-bucket filter for achieving fairness between the downlink and the uplink. Most of these solutions require modification of the TCP receiver making them difficult for large scale deployment.

RTT-fairness is specifically addressed by some TCP variants for wired networks such as TCP Libra [Pilosof *et al.* 2003] and TCP Hybla [Caini and Firrincieli 2004]. These protocols set their sending rate as a function of RTT values to ensure balanced resource sharing between all nodes in the network. However, such solutions require modification of the TCP sender code and, thus, their dissemination on a large scale can be infeasible.

Although traditional active queue management algorithms leads to good performance in wired networks, their impact is not so successful in wireless networks. For instance, the RED mechanism needs to be adopted jointly with ECN in order to achieve good performance [Yi *et al.* 2008], making the overhead impractical for most APs with light-weight software. Consequently, other queue management mechanisms have been proposed to ameliorate the fairness issues in Wi-Fi networks. In [Lin *et al.* 2005], the authors proposed a management scheme which admits flows into round-robin virtual queues. This mechanism guarantees fairness between uplink and downlink flows. The work in [Gong *et al.* 2006] proposed two alternative approaches: the first one, called *Selective Packet Marking with ACK filtering* (PMS-AF), is based on packet priority marking for potential discard at the AP; the second, called *Least Attained Service* (LAS), guarantees fairness among flows by giving higher priority to the least served flows.

All the proposed approaches overviewed above either depend on a specific network infrastructure [Yi *et al.* 2008] or require modifications of the TCP sender [Gong *et al.* 2006]. The closest approach to the one presented in this paper is [Lin *et al.* 2005]. However, as many other proposals, it operates as a standalone module and it does not use any parameter provided by other network nodes.

The MAC fairness anomaly problem has also been studied. In [Heusse *et al.* 2003], the existence of such anomaly was demonstrated by analytical derivation as well as by experimental tests in real networks. The work in [Yoo *et al.* 2005] proposed a solution for the anomaly problem by sizing the frames proportionally to the data rate of the nodes. In [Branquinho *et al.* 2006], the authors proposed to adjust the contention window thresholds, CW_{min} e CW_{max}, according to the nodes SNR. In [Kim *et al.* 2005], an initial contention window size inversely proportional to the nodes data rate is proposed. The adjustment of 802.11 parameters to guarantee fairness among multi-rate nodes has been the focus of several investigations. These, however, require that the entire network be changed and that nodes become non-standardized ones. Our approach is centralized at the AP and it is totally standard-compliant, allowing an incremental deployment of it.

3. Proposed mechanism

The proposed Receiver-Driven Queue Management (RDQM) scheme aims at optimizing the performance of TCP flows and resource sharing over Wi-Fi infrastructure networks. RDQM is essentially an algorithm implemented at the Access Point (AP) that manages the outgoing queue of the wireless interface, enqueuing or dropping packets to improve the use of the AP buffer in wired-cum-wireless networks. The queue policy is assisted by the mobile receivers through a feedback channel created at the link layer. Receivers encapsulate useful information into the acknowledgment frames (LL-ACK) to help the AP to manage the shared buffer space. Although RDQM is a generic framework, in this article, it will be used to address the RTT fairness problem and the MAC anomaly problem. In the following subsection, the mechanism will be explained.

3.1. Receiver-Driven Queue Management

RDQM is a buffer management architecture proposed for wired-cum-wireless networks, in which mobile users establish TCP connections with fixed peers on the Internet. Scenarios are composed by mobile and fixed nodes connected through heterogeneous channels and a last-mile wireless hop (see Fig. 1). In RDQM, the operation of the buffer management is driven by information sent by the mobile user, such as RTT, bandwidth-delay product, or any other metrics easily measurable at the Mobile Node (MN) end, which are difficult, and in some cases impossible, to be inferred by the AP.

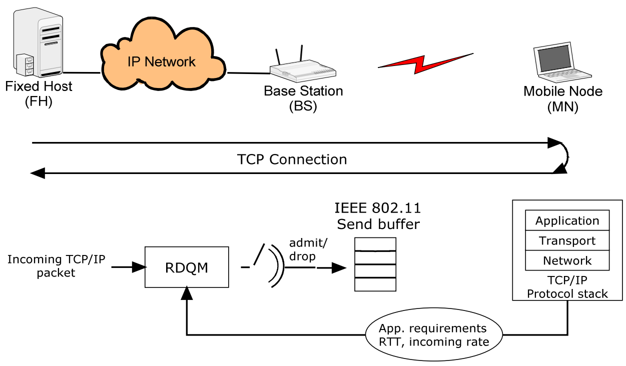

Figure 1. RDQM architecture.

The IEEE 802.11 [IEEE Std. 2007] standard uses an ARQ scheme, which requires the receiver to acknowledge positively every successfully frame received. In the LL-ACK frames, there are some reserved bits that can be used by the RDQM to create a feedback channel between mobile receivers and the AP, requiring no modification in the standard and being totally transparent to high-level protocols. More specifically, 14 reserved bits from the 16-bits duration field of the 802.11 link layer header is used for that purpose (see Fig. 2), allowing the MNs to send any signaling information to the AP. It is important to notice that the use of reserved portion of bits does not modify NAV mechanism during virtual carrier sensing. In this way, the coexistence of the standard implementation and that of RDQM is feasible.

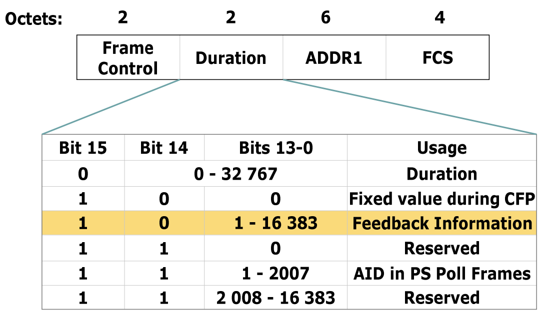

Figure 2. Reserved portion of IEEE 802.11 link layer ACK frame.

The common drop-tail buffer management policy can yield TCP connections with large RTT to starvation since packets belonging to connections with small RTT can monopolize the buffer space. However, the information sent to the AP can be used to better manage its buffer in order to specific goals, so that to promote RTT fairness among TCP connections, the RDQM can be used to divide the buffer space among the flows proportionally to their RTT measurements in a dynamic way. MAC anomaly degrades drastically overall network performance not because of buffer monopolization, but because of medium monopolization, since nodes running at lower rate use the wireless medium for a longer period of time. In this situation, the RDQM scheme can be deployed to backpressures the sending rate of mobile nodes. It is possible to propose a single solution for both problems by simply dividing the buffer space proportionally to the bandwidth-delay-product (BDP) of the flows, which can be estimated at the mobile receiver. Initially, the RTT is taken as that experienced by the SYN TCP segment [Jiang and Dovrolis 2002]. After that, the RTT value is measured during the connection lifetime [Lu and Li 2003, Veal *et al.* 2005]. The BDP is more difficult to estimate in heterogeneous networks, since bandwidth estimation is a hard task and usually requires specific tools [Strauss *et al.* 2003]. For the purpose of RDQM algorithm the exact value of BDP is not needed, but rather the ratio between the BDP of the established flows. For simplicity of analysis and fast computation of the BDP values, we can consider that the bottleneck of the end-to-end path is the wireless link and that the bandwidth of each flow is equal to the physical data rate of each node. It is well known that the bandwidth of Wi-Fi flows is a small percentage of the physical data rate, but in our analysis the most important is the ratio between the bandwidth of the flows, since the buffer space is allocated proportionally to their BDP values. Such simplification is valid and brings good results, as it will be shown later.

The proposed algorithm for RDQM operation can be deployed in any Wi-Fi network variant (IEEE 802.11a, 802.11b, 802.11g, 802.11n) with nodes running at different data rates.

3.2. Algorithm description

The RDQM algorithm runs at the AP with the support of agents running at the MNs. The queue management uses virtual queues. Each flow has its own virtual queue and associated parameters such as maximum virtual queue size, current number of packets and the weight values. All virtual queues are physically located at the AP´s outgoing buffer and their parameters can be adjusted dynamically (see Fig. 3). The virtual queue implementation requires no modifications to the memory allocation scheme of the APs that implement legacy IEEE 802.11 standard.

For every packet received from the wired network, the AP verifies whether the number packets enqueued for a given flow exceeds the maximum allowed value. If this is the case, the packet is dropped, otherwise it is enqueued. The buffer space is organized in a partial sharing fashion with thresholds defined dynamically [Kamoun and Kleinrock 1980]. For each virtual queue, the RDQM mechanism also tries to allocate a minimum amount of space equal to the bandwidth-delay product of the flow estimated by the product of the measured RTT at the MN by the physical data rate. Such policy is based on the well-knows principle for TCP New-Reno [Floyd and Henderson 1999] that states that the buffer space at a bottleneck link should be at least equal to the bandwidth-delay product between the end points for full utilization of the available bandwidth by the two TCP endpoints.

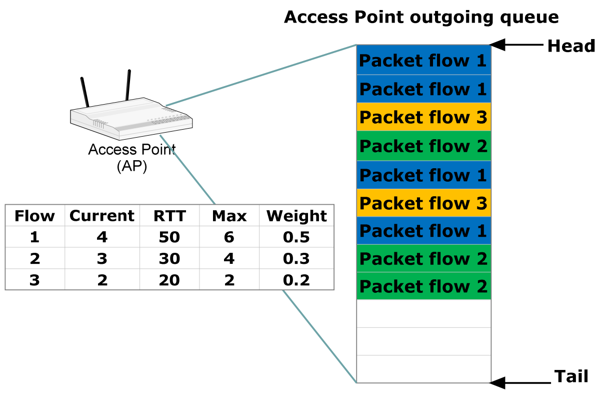

Figure 3. Virtual queue implementation

Algorithms 1 to 3 present the implementation details of the proposed approach. Algorithm 1 describes the procedures implemented at the mobile node, which essentially inserts the measured BDP values into the header of outgoing LL-ACK packets.

Algorithm 2 corresponds to the procedure implemented at the APs. It decides whether or not a packet should be discarded. Moreover, it updates the number of packets currently stored at all virtual queues. The algorithm is executed at the arrival of TCP data packet sent by the fixed sender. Parameters MAX_{def} and BDP_{def} correspond to pre-set default values used in the initialization of data structures for new flows, since measurements such as RTT are only available after three-way handshake.

Finally, Algorithm 3 updates the queue thresholds, i.e. *Flow BDP*, *Flow Weight* and *Flow Packet Limit*, which, respectively, correspond to the current BDP value of the flow, its weight in the buffer space allocation and the maximum number of packets that can be stored in the buffer. It is triggered by the arrival of every incoming link layer ACK from the MNs.

Algorithm 1: LL-ACK packet transmission

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

Algorithm 2: 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 BDP equal to BDP_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: LL-ACK packet arrival at BS

```
Get Flow ID
If Flow BDP is greater that zero Then
    Set BDP_Sum equal to zero
    For Each Flow Do
        Increase BDP_Sum by Flow BDP
    Endfor'
    For Each Flow Do
        Set Flow Weight to Flow BDP / BDP_Sum
        Set Flow Packet Limit to product between Flow Weight and Buffer Size
    Endfor
Endif
```

4. Performance evaluation

4.1. Simulation Setup

To assess the effectiveness of the proposed scheme, an ns-2 [ns-2] module was implemented. The simulated scenario consists of wireless nodes and Fixed Nodes (FNs) connected to the router by links of 100 Mb/s with delay ranging from 5 to 100 ms. The Mobile Nodes (MNs) are connected to the fixed network using IEEE 802.11 AP. The topology can be seen in the Figure 4. The wired part of the network represents the Internet and connections have different RTT values. The MNs use IEEE 802.11b standard, operating at 1, 2, 5.5, and 11 Mb/s data rates.

The packet size of TCP NewReno used in the simulation was 1500 bytes. The duration of all simulation runs were 600 seconds, which is sufficient to reach the steady-state phase of TCP flows. RTS/CTS threshold of 802.11 was set to 2000 bytes, avoiding the unnecessary exchange of RTS and CTS frames and improving the performance of wireless link. Results were averaged from 10 runs and 95% confidence intervals were derived.

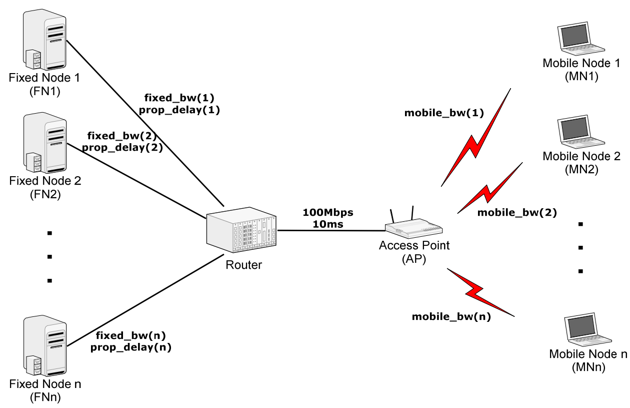

Figure 4. Wired-cum-wireless scenario

4.2. Scenario with fixed transmission rate

The experiments aim at evaluating the improvement of the throughput fairness among flows with different RTT values and physical data rates.

In the first set of experiments the physical data rate was 11 Mbps. Each pair of nodes establishes only one connection and the number of flows varied from 1 to 10. The buffer space varied from 5 to 1000 packets. Each pair of nodes was configured to have different propagation delays in the wired part, varying randomly from 5 to 150 ms.

Figure 5 shows the throughput results obtained by a drop-tail queue (Fig. 5a) and by an RDQM queue (Fig. 5b) in a scenario with 6 flows and propagation delays ranging from 5 to 120 ms. It can be seen that flows with lower propagation delay are prioritized in drop-tail scheme, which does not happen under the RDQM scheme. For instance, the flows with 100 and 120 ms receive only a little percentage of the buffer space under drop-tail scheme and, consequently, achieve throughputs smaller than 700 Kbps even for buffer sizes larger than 400 packets. On the other hand, these two flows are prioritized under the RDQM scheme and the achieved throughput is larger than 800 Kbps for all buffer sizes. The effectiveness of the RDQM scheme can also be assessed by measuring the distance between the curves produced by the connections using links with different propagation delays.

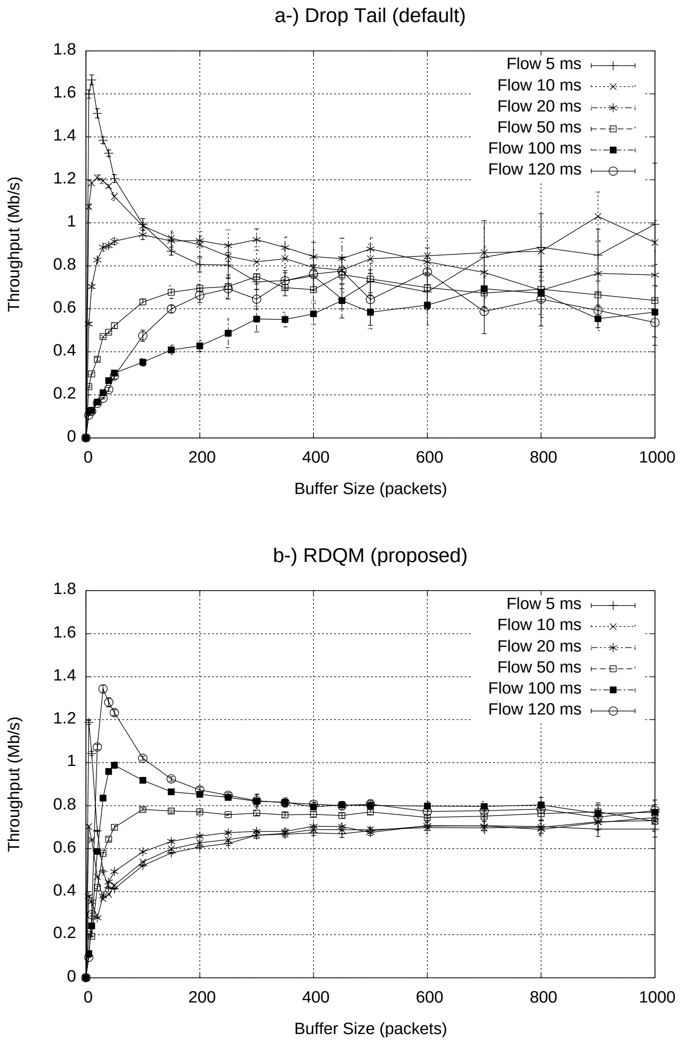

Figure 5. The throughput performance of 6 TCP flows with different RTTs for a) drop-tail and b) RDQM queues.

Figure 6 reports the distance between the curves, which is calculate as the maximum difference between the throughput values of the analyzed flows. For all buffer sizes, the throughput given by RDQM outperforms that given by drop-tail, which proves the efficiency of the proposed algorithm. For buffer sizes smaller than 100 packets the throughput difference is large for both schemes; it is higher than 1.5 Mbps under drop-tail scheme and it is bounded to 1 Mbps under RDQM. For buffers larger than 100 packets, the throughput difference tends to remain low and decreasing for RDQM, while it is high and increasing for the drop-tail scheme. For buffer sizes larger than 400 packets, the throughput difference of RDQM scheme is limited to 200 Kbps and for drop-tail scheme it is more than twice this value. It is clear that for buffer spaces larger than 200 packets, RDQM achieves its maximum fairness improvement; for limited buffer spaces (smaller than 200 packets), the throughput fairness is not significantly improved, but the RDQM scheme still provides better performance than does the default drop-tail scheme. For the sake of comparison, considering an MTU of 1500 bytes, typically used for wired-cum-wireless networks, a buffer space of 200 packets would require 300 KB of storage memory. This shows that the RDQM scheme can be successfully implemented and it can achieve its maximum fairness improvement in most of the available commercial equipments.

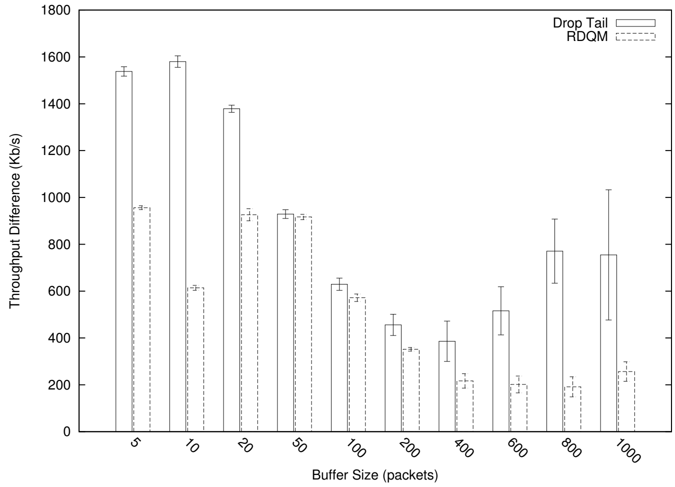

Figure 6. Maximum throughput difference – scenario with 6 flows

Figure 7 presents results for a network scenario with increasing number of flows. The number of network flows varied from 2 to 10, the increasing RTT values of the flows ranged from 5 to 150 ms and the buffer size was 200 packets. Figure 7 shows the maximum throughput difference, which correspond to the difference between the achieved throughput of the fastest and the slowest flows. The RDQM algorithm keeps the throughput difference lower than does the drop-tail scheme. For scenarios with more than 3 nodes, RDQM keeps the throughput difference bounded to 200 Kbps, while this difference is about 300 Kbps for drop-tail; an improvement of 50%. This set of experiments proves that RDQM scheme can be applied to scenarios with varying number of flows and yet provide great fairness improvement.

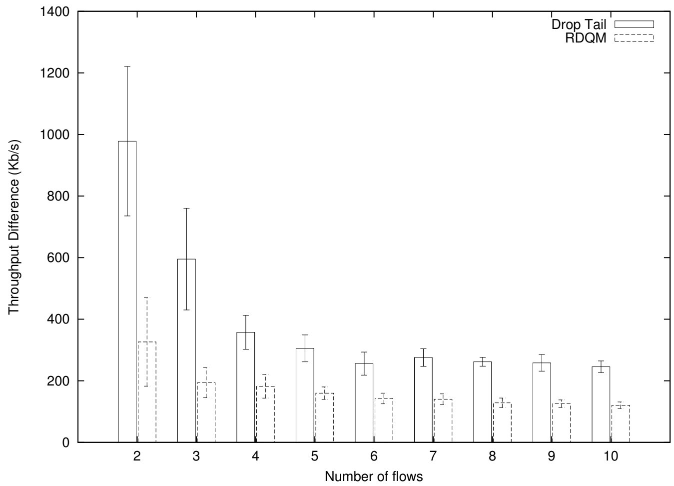

Figure 7. Maximum throughput difference – scenarios with many flows

4.3. Scenario with transmission rate adaptation

The second set of experiments consisted of a network with multi-rate nodes. Figure 8 shows the obtained throughput results for 6 TCP flows simultaneously obtained by the drop-tail (Fig. 8a) and the RDQM (Fig. 8b) queues. The throughput anomaly happened at the drop-tail queue can be clearly noticed. The mean throughput per flows is about 350 Kbps, which implies on an aggregated throughput of about 2.1 Mbps, much lower than the 4.5 Mbps available in 802.11b networks. It can be observed that the RDQM mitigates the MAC anomaly effect by keeping the throughput at 700 Kbps for nodes running at 11 Mbps, 500 Kbps for nodes running at 5.5 Mbps and 100 Kbps for nodes running at 1 Mbps. The aggregated throughput is almost 3.2 Mbps, which represents a gain of more than 50% when compared to drop-tail.

Figure 9 shows the aggregated throughput for this scenario. RDQM keeps the aggregated throughput higher than does drop-tail, which shows that it mitigates the anomaly, even for small buffers. For buffer sizes larger than 50 packets RDQM scheme outperforms drop-tail, achieving an aggregated throughput more than 50% higher.

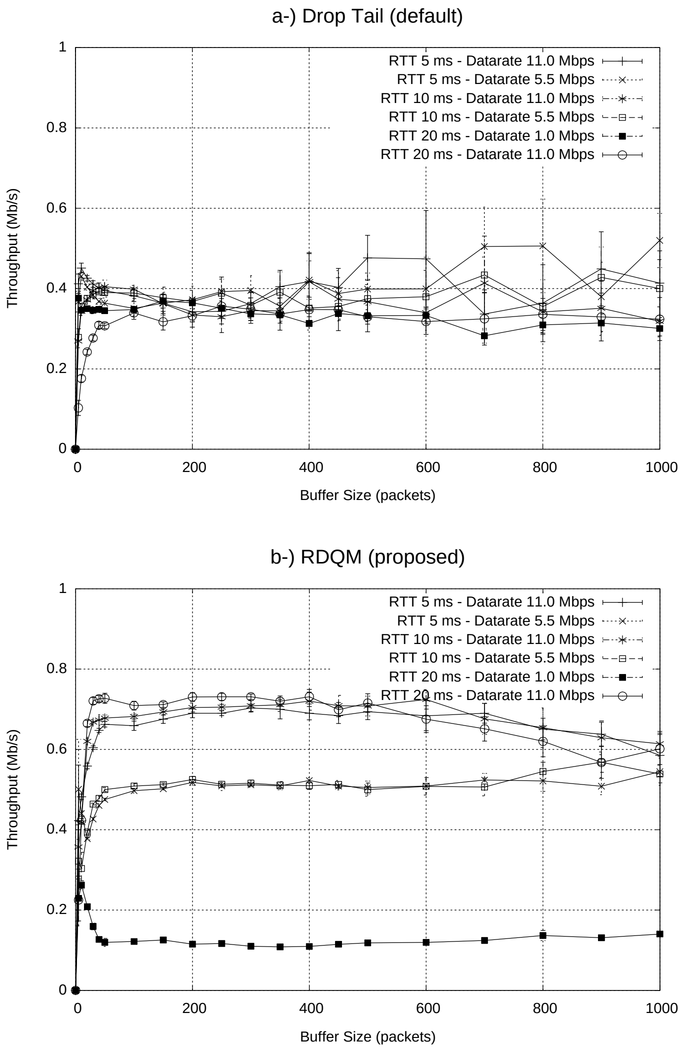

Figure 8. The throughput performance of 6 TCP flows with different RTTs for a) drop-tail and b) RDQM queues.

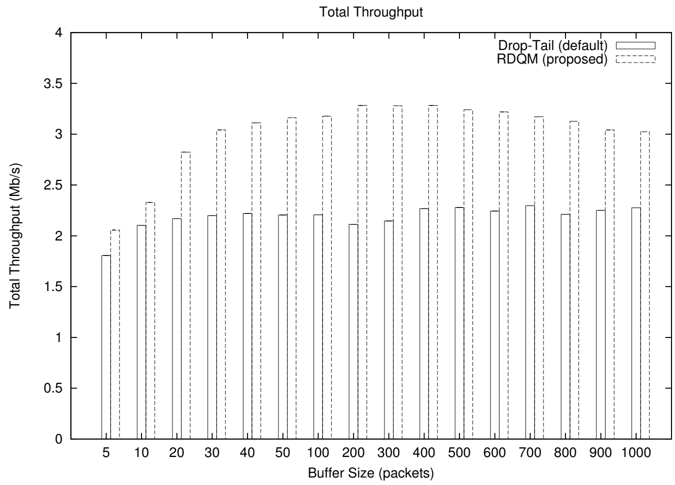

Figure 9. The aggregated throughput of 6 TCP flows

5. Conclusions and future works

The most common use of wireless network technologies consists of mobile users accessing data and services available on fixed nodes on the Internet using Wi-Fi. In such scenario, users can be affected by many different fairness issues.

The distances between fixed pairs on Internet and the AP, which bridges fixed and mobile network parts, vary and so does the propagation delays of their links. Such variation contributes to the problem of RTT unfairness making flows paths with long propagation delays receive smaller portions of the available bandwidth and buffer space than do flows paths with shorter propagation delays. Another fairness issue that affects the efficiency of the transmissions is the MAC anomaly which arises in Wi-Fi networks with nodes running at different physical data rates. In such scenario, the wireless medium is monopolized by flows with low data rates.

This paper introduced a novel management scheme for the buffers at the Access Point which takes into account information available at the mobile receivers to achieve specific goals. The employment of this scheme to ameliorate the RTT unfairness problem and the MAC anomaly was illustrated using different scenarios. In the proposed scheme, buffer allocation was proportional to the bandwidth-delay product of the connections, which is continuously computed by the mobile receivers and sent to the AP through a feedback channel overlayed on the link layer. Results indicate that the proposed scheme outperforms traditional drop-tail, achieving high levels of fairness among flows with different bandwidth-delay products.

References

Williamson, C. (2001) “Internet Traffic Measurement”, In: IEEE Internet Computing, vol. 5, no. 6, p. 70-74.

Marfia, G., Palazzi, C., Pau, G. Gerla, M., Sanadidi, M. Y. and Rocetti, M. (2005) “TCP Libra: Exploring RTT-fairness for TCP”, Technical Report UCLA-CSD TR-050037, UCLA, Computer Science Dept., Los Angeles, CA.

Pilosof, S., Ramjee, R., Raz, D., Shavitt, Y. and Sinha, P. (2003) "Understanding TCP fairness over wireless LAN", In: Proceedings of IEEE INFOCOM, vol. 2, p. 863–872.

Lee, E., Lim, H. Lim, Seok, S. and Kang, C. (2007) "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, p. 368-379.

Li, T. and Leith, D. J. (2008) "Adaptive Buffer Sizing for TCP Flows in 802.11e WLANs", In: International Conference on Communications and Networking in China, 2008. ChinaCom 2008, p. 7-11.

Park, E., Kim, D. and Kim, H. (2008) "A Cross-Layer Approach for Per-Station Fairness in TCP over WLANs", In: IEEE Transactions on Mobile Computing, vol. 7, no. 7, p. 898-911.

Blefari-Melazzi, N., Detti, A., Ordine, A. and Salsano, S. (2005) "Controlling TCP Fairness in WLAN access networks using a Rate Limiter approach", In: 2nd International Symposium on Wireless Communication Systems, p. 375-379.

Caini, C. and Firrincieli, R. (2004) "TCP Hybla: a TCP Enhancement for Heterogeneous Networks", In: International Journal of Satellite Communications and Networking, John Wiley & Sons, vol. 22, no. 5, p. 547-566.

Yi, S., Kappes, M., Garg, S., Deng, X., Kesidis, G. and Das, C. R. (2008) "Proxy-RED: An AQM Scheme for Wireless Local Area Networks", In: Wireless Communications & Mobile Computing, vol. 8, no. 4, p. 421-434.

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

Gong, M., Wu, Q. and Williamson, C. (2006) "Queue management strategies to improve TCP fairness in IEEE 802.11 wireless LANs", In: Proceeding of International Symposium on Modeling and Optimization in Mobile, Ad Hoc and Wireless Networks, p. 1-8.

IEEE Std. 802.11-2007 (Revision of IEEE Std. 802.11-1999), Part 11, (2007) "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specifications".

Jiang, H. and Dovrolis, C. (2002) "Passive estimation of TCP round-trip times", In: ACM SIGCOMM Computer Communication Review, vol. 32, no. 3, p. 75-88.

Lu, G. and Li, X. (2003) "On the correspondency between TCP acknowledgment packet and data packet", In: Proceedings of the 3rd ACM SIGCOMM conference on Internet measurement p. 259-272.

Veal, B., Li, K. and Lowenthal, D. K. (2005) "New Methods for Passive Estimation of Round-Trip Times Using TCP Timestamps", Passive and Active Network Measurement, Lecture Notes in Computer Science, vol. 3431, p. 121-134.

Strauss, J., Katabi, D. and Kaashoek, F. (2003) “A measurement study of available bandwidth estimation tools”. In: Proceedings of the 3rd ACM SIGCOMM conference on Internet measurement, p. 39-44.

Kamoun, F. and Kleinrock, L. (1980) "Analysis of shared finite storage in computer network node environment under general traffic conditions", In: IEEE Transactions on Communications, vol. 28, p. 992-1003.

Floyd, S. and Henderson, T. (1999) “The NewReno Modification to TCP’s Fast Recovery Algorithm”, RFC2582.

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

Heusse, M., Rousseau, F., Berger-Sabbatel, G. and Duda, A. (2003) “Performance Anomaly of 802.11b”, In: Proceedings of IEEE INFOCOM.

Kim, H., Yun, S., Kang, I. and Bahk, S. (2005) “Resolving 802.11 Performance Anomalies through QoS Differentiation”, In: IEEE Communications Letters, vol. 9, no. 7, p. 655-657.

Branquinho, O. C., Reggiani, N. and Ferreira, D. M. Ferreira (2006) “Mitigating 802.11 MAC Anomaly Using SNR To Control Backoff Contention Window”, In: ICWMC´06 IEEE International Conference on Wireless and Mobile Communications, p. 29-31.

Yoo, S., Choi, J, Hwang, J. and Yoo, C. (2005) “Eliminating the Performance Anomaly of 802.11b”, Networking - ICN 2005, Lecture Notes in Computer Science, p. 1055-1062.

Freitag, J., da Fonseca, N. L. S., de Rezende, J. F. (2006) “Tuning of 802.11e network parameters”, IEEE Communications Letters, vol. 10, no. 8, p. 611-613.