Enhanced Packet Scheduling Algorithm for Multihop Wireless LANs

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Abstract

*The use of wireless devices are flooding the network, thus giving rise to extensive data transmission. One of the main concerns is how to dictate the buffering of data so that throughput fairness can be achieved, specifically in the multihop wireless networks. The technique of buffering has to be based on the scheduling algorithms. Several studies on scheduling algorithms as a part of congestion control have been carried out, such as by Shagdar, Nakagawa and Zhang [1]****,*** *Izumikawa, Ishikawa, and Sugiyama [2], and Giang and Nakagawa [3]. The Round Robin (RR) scheme is considerably simpler than other known schemes for achieving throughput fairness in multihop wireless networks. As is RR scheme, the main cause of concern is the delay that is been caused by high volumes of workloads at the wireless source nodes. It is therefore the purpose of this paper to enhance the network performance in wireless LANs by improving the fairness index and average end-to-end delay the through the scheme identified as Proposed Packet Reverse Function.*

Keywords: *Network; congestion; scheduling; round-robin; fairness*

1. Introduction

Wireless network attracts tremendous attention in network configuration because it is free from any involvement of infrastructure such as routers and base stations. It is foresee that the use of wireless LAN devices of IEEE 802.11 standards is expected to be common in various kinds of networks including ad hoc networks, sensor networks, personal area networks, and networks that interconnect the base stations of the fourth-generation cellular system [1].

While there existed considerable literatures that discuss issues with regard to 802.11 wireless multi-hop networks, but issues that are associated with interference, routing and congestion controls remain as the main focus [4].

New mechanisms have been proposed in the field of scheduling algorithm in trying to overcome the problem related to interference and routing during the scheduling of packets. Scheduling algorithm will determine among the packets in the buffer which one will be served. Round-robin (RR) scheduling is one of the examples of scheduling algorithm. This algorithm can be used to assist in creating process or job schedules to ensure that each process required to complete a job gets a fair amount of run time [5].

Fairness in network scheduling can be achieved by using RR mechanism. Fairness is one of the most important properties of a computer network: when network resources are unable to satisfy demand, they should able to divide the resources fairly between the clients of the network. Most often, max-min fairness [6] is the desired fairness scheme. Under this scheme, the clients are split into two groups: the first group consists of the clients that cannot be completely satisfied by network resources; they all receive the same share of bandwidth. The second group is made up of the clients that need less bandwidth than their fair share; they receive exactly the amount of bandwidth that they ask for. Neither of the two groups can be empty [7].

By definition, congestion control can be defined as a set of mechanism that prevents or reduces such a drop of networks service or throughput due to the increase in the network load [8]. Matters that are been studied currently with regards to congestion issue includes avoidance, congestion prediction and congestion routing [9, 10, 11].

D. Gaiti and G. Pujolle explained that congestion occurred because of unpredictable statistical fluctuations of traffic flows and fault conditions within the network [12]. Earlier in 1989, D. M. Chiu and R. Jain stated that throughput may suddenly drop when load increased beyond the network capacity and therefore the network was said to be congested [13]. The emergence of the congestion control algorithm had brought to several proposals and implementations which had made the internet so useful and convenience until today.

2. Preliminaries

The aim of the buffer management mechanism is to maximize the numbers of transmitted packets. The buffer management mechanism has to decide which packets have to be dropped and which packets have to be transmitted because of the buffer capacity constraints. Buffer management is able to support precise and efficient per-flow fairness and achieve very good performance.

2.1. Buffer Management Issue for Multi-hop Wireless LANs

In multi-hop environment, each node may transmit new packets and those forwarded from the other nodes through a buffer. This will induce unfairness in per-flow end-to-end throughput. A new packet will be generated by each node for every 3T period and that the transmission will alternate between Wireless Node 1 (WN1) and Wireless Node 2 (WN2) in 2T period, as shown in Figure 1(a). In Figure 1(a), with the assumption of the buffer size to be four (packets), the offered load G was small and hence no packet loss occurs.

In Figure 1(b), each node will generate a new packet at every T period and the transmission alternates between the two nodes in 2T. BS will receive more packets that originated from WN1 compared to packets that are forwarded from WN2. Packets that originated from WN2 also suffered from packet loss at WN2’s transmission buffer. This situation is caused by buffer saturation due to the small size of buffer, which is four. Therefore, the end-to-end throughput of WN2 packets has degraded significantly in comparison with WN1 packets. WN1 discards packets from WN1 and WN2 at 4T and 6T.

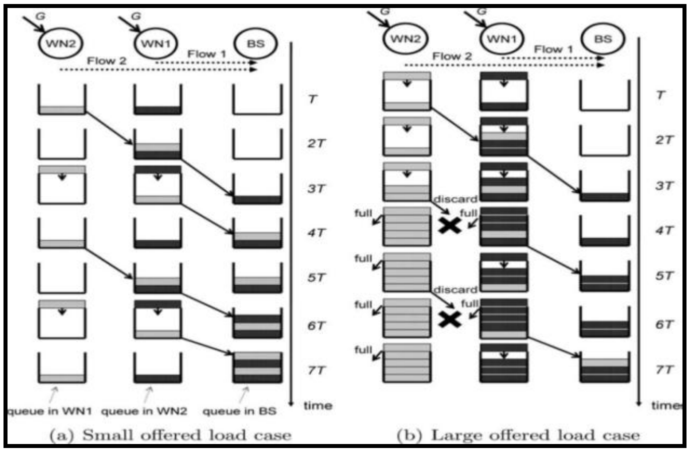

Figure 1(a & b). Queuing Dynamics

2.2. Fairness and Max-min Fairness in Multi-hop Wireless Networks

A data communication network is made of store-and-forward switches also identified as nodes which are joined by point-to-point communication channels (links). Each network user (session) is assigned a fixed path (virtual circuit) through the network, and data for the session are sent in manageable parcels (packets) along this path. In such a network, occasional surges in user demand can overload network links, causing packet buffers to build up in network nodes. These buffers may eventually overflow the nodes' storage space, or the delay of acknowledgments may cause transmitters to assume that data were lost. Flow control procedures attempt to prevent or alleviate this degradation by regulating the appropriate traffic sources. It would be desirable for flow control procedures to regulate network inputs so as to grant each session a fair throughput rate [14].

A fairness approach called max-min flow controls (or bottlenecks flow control) can be used in various parts of networking such as packet-switched network and data network. In multi-hop wireless networks environment, max-min fairness can be achieved by ensuring that the smallest session rate in the network must to be as large as possible and secondly the second-smallest session rate must to be as large as possible [15].

2.3. Round Robin Scheduling

Scheduling can be recognized as a critical mechanism in resource allocation for multihop wireless network. [16] The Round robin (RR) scheduling algorithm is considerably simpler than most algorithms. The function of RR scheduling algorithm is to maintain the per-flow buffers. Each buffer offers its packet transmission slots to its users by polling them in round-robin order. If a session is offered a chance to use a link slot but has no packets ready, then that same slot is offered to the next session, and perhaps the next, etc., until a ready session is found. In order to prevent excessive buffers at the network nodes, window flow control is also employed with RR scheduling.

A lot of researches have been done on enhancing the scheduling algorithms by using RR scheduling. The Distributed Coordination Function (DCF) mechanism [1] focused on the contention of direct and forwarding flows, and proposed scheduling algorithms by using RR buffer. This mechanism was modified to achieve the bandwidth utilization by sending all the packets at the head of RR buffers continuously without any delay due to back-off algorithm. This mechanism also enabled a node to send several packets in a single channel access in order improve per-flow fairness in wireless networks. Even though the proposed scheme will require minor modification to the original IEEE 802.11MAC protocol, but it has successfully improved the per-flow fairness, total performance and medium utility of the network. Although the method used RR buffer, but it worked ineffectively because the allocated bandwidths at the MAC layer were not suitable for forwarding and direct flows in the link layer.

[2] has proposed a scheduling algorithm to solve the problems in a multihop wireless network. The authors stated that the differences arose in the throughput is related to the number of hops from the base station connected to wired networks. In the proposed algorithm, the uplink packets and downlink packets were buffered separately, and the uplink packets were classified into packets originated within the node and forwarded packets from adjacent wireless nodes and buffered. When a node transmitted a packet, the fairness among the nodes could be improved by scheduling based on the source identifiers of the buffered packets. This scheme is designed for achieving the same throughput among data flows regardless of the number of hops. Therefore, it works well only when all sender nodes transmit packets at the same rate and the link capacities are the same.

Giang and Nakagawa [3] used RR buffer with three algorithms to control input/output packets. The proposed algorithms have achieved improved result in per-flow fairness. In Probabilistic Control on Round robin Buffer (PCRQ) scheduling, the MAC layer fairness was improved indirectly because PCRQ scheduling helped flows of small loads to get more chance in using channel bandwidth by giving delay to the flow of heavy loads to send a packet. However, the total throughput performance was slightly degraded. In PCRQ scheduling, RR buffers were used with three algorithms: The first algorithm controlled the number of input packet to buffers, while the second algorithm controlled the turn of reading buffers, and the third controlled the number of output packets from buffers.

3. Simulation Design

3.1. Base Work

This study will involve two main components, the base station (BS) and the wireless node (WN). The BS is made of scheduling algorithms module which is using the RR scheme. The scheduling algorithms module contains the detailed information about how the packets in the buffers are being identified for departure. Packet scheduler module selects the appropriate packets from different buffers after the packets have been classified according to traffic type.

WN module consists of one BS and three WNs. Each WN has a per-flow buffers which is functioning in a round-robin fashion. The buffer in each node can be classified into two types: one is an origin buffer for packet originating from the node, and the other one is forwarding buffers for packets sent from the other nodes. This simulation uses the reverse function to ensure no packet loss as will be discussed in the later section of this paper. However, packets will also eliminated in the cases of en all previous nodes are already full.

3.2. Simulation Model

A discrete event simulator has been developed to simulate the proposed algorithm and it is based on previous research from [5]. However, there are few changes that have been made in the parameters of the algorithm which did not affect the original results. The parameters that are been used in the experiment are listed in Table 1.

Data are transmitted through the network in packets of equal length. A time interval during which one packet is transmitted over a link is called inter-arrival time for that link. All links have the same buffer size.

Table 1. Parameters for Research Experiment

Number of Nodes	**3**
Maximum buffer size	50
Total packet generation rate	Variations between 100-2000
Length of simulation run	30s
Packet size	1500 bytes
Bandwidth	11Mbps

In this simulation, RR scheduling algorithm is used to serve both buffer in each node as in Figure 2. The Q1 and Q2 are referring to the direct and forwarding buffer. The original packets from WN1 are buffered at Q1 and packets from WN2 are forwarded to Q2. Each buffer is served if it has a packet to depart before moving to the next buffer. If the current buffer for example Q1 has no packet, the transmission is immediately assigned to Q2.

The packets originated within the node and forwarded packets from another WN are buffered separately at Q1 and Q2. When the node transmits a packet based on the source identifiers of the buffered packets by using RR scheduling algorithm, the fairness among the nodes is improved. Furthermore by using RR scheduling algorithm, a packet experience no processing delay at a node, no propagation delay on a link, but has a possible queuing delay as its waits for transmission.

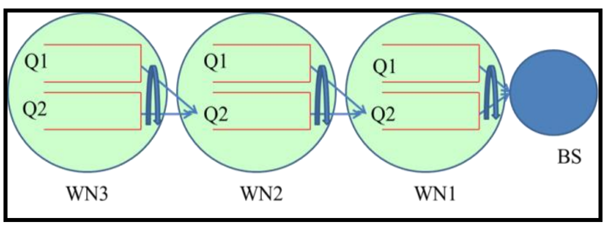

Figure 2. Simulation Model Diagram

3.2.1. Packet Generation: Our proposed scheduling algorithm defines the packet arrival time and inters arrival time of the packets for transmission based on random packet generation. The packet arrival time need to be reserved in a buffer in order to wait for its turn to be selected for the minimum time of the packets to be departed.

Basically, this algorithm acts as buffer management mechanism when the server status is busy in serving the packets. Even when the server status is busy; all sources will keep sending packets for transmissions. The flood of packets to be served need to be handled carefully so that any packets will not be missing on transmission. In this situation, a buffer is needed as a transit for packets while waiting for their turns to be served.

When the packet arrives at Q1 in WN1, the scheduler will check whether the Base Station (BS) is idle or busy. If the BS is idle, the packet will be sent to BS and service time is calculated. If the BS is busy, the scheduler inserts packets into the buffer. After that, the scheduler will check for the availability of packets at Q2. If Q2 has a packet, then the algorithm will check the status of BS. If the BS is idle, the packet will be sent to BS and the service time will be calculated. When another packet arrives, the scheduler will also check the availability of the BS status. All the other packets that come earlier than time the server is ready to serve (server is idle) will be sent to buffer. However, if the selected buffer has no packet, the transmission right is immediately assigned to Q1 and if the current buffer is Q1 has no packet, the transmission right is immediately assigned to Q2.

However, there exist differences in buffer management in WN2 and WN3. For example, when the original packet from WN2 arrives at Q1, the scheduler will check the availability of Q1. If the Q1 is less than maximum buffer size it will automatically insert packet into Q1. Then, it will check the availability of Q2 at WN1. If Q2 is less than maximum buffer size, first packet at Q1 in WN2 will be forwarded to the Q2 buffer at WN1. After that, it will check the availability of Q2 at WN2 whether it has a packet or not. If Q2 has a packet, the first packet is forwarded to the Q2 at WN1. When another packet arrives, the scheduler will also check the availability of Q1 size.

3.2.2. Proposed Packet Reverse Function: The special characteristic of proposed scheduler is on how it managed the packets to be transmitted. It uses the reverse function to ensure no packet loss as in Figure 3 – Figure 5.

i. Packet from Q1 at WN2 is forwarded to Q2 at WN1. The scheduler checks if Q2 at WN1 is equal to maximum buffer size. If buffer size at Q2 at WN1 is already full, then the packet is sent Q2 at WN2. The scheduler checks if Q2 at WN2 is less than maximum buffer size. If true, a packet is inserted into buffer (Figure 3).

ii. If buffer size at Q2 in WN2 is already full, then the packet is send to Q2 at WN3. The scheduler checks the Q2 if it less than maximum buffer size. If true, the packet is inserted into the buffer (Figure 4).

iii. If Q2 at WN3 is also full, then packet is loss (Figure 5).

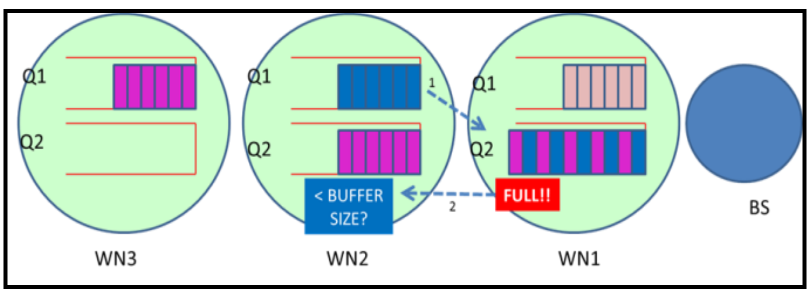

Figure 3. Case 1

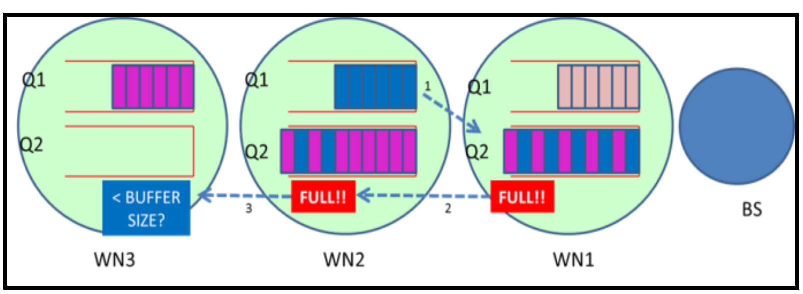

Figure 4. Case 2

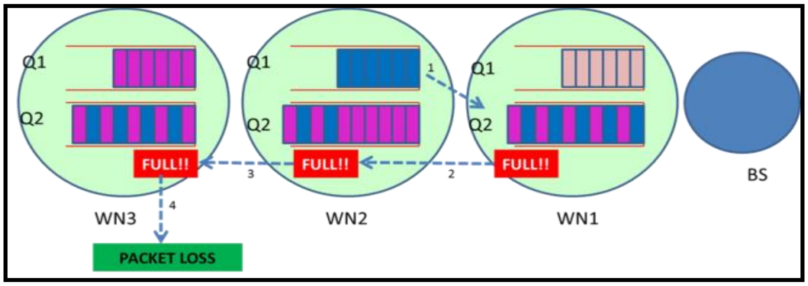

Figure 5. Case 3

4. Result and Analysis

The following figures (Figure 6 – Figure 8) represent the performance from the two different protocols called RR scheme and enhanced packet scheduling algorithm scheme for RR. Then the comparison between tha fairness index and RR schemes via simulation. The simulation of the new proposed scheme is being run within the range of offered load from 100 and 2000 [kBps]. We then proceed with the examining how the offered load affects per-flow fairness. We also simulated the average end-to-end delay comparison of RR scheme and proposed scheme and between direct (Q1) buffer and forwarding (Q2) buffer.

Figure 6 illustrates the fairness index for both schemes. We have discovered that the proposed scheme has always maintained high per-flow fairness, while per-flow fairness in RR is not consistent. This result shows that the proposed scheme has successfully maintained high per-flow fairness in comparison to RR scheme.

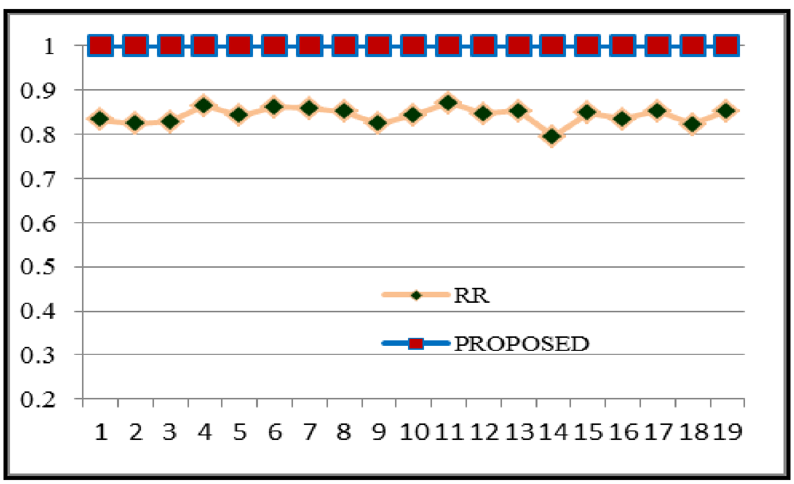

Figure 6. Fairness Index vs. Offered Load for each Scheme

In Figure 7, the results of average delay are been plotted in relation to the offered loads. Figure 7 depicts the average delay of proposed scheme is relatively lower than in the RR scheme. However, at offered load of 6, 13 and 14 [kBps], the proposed scheme has higher delay than RR. The higher delay is caused by the random generation of packet. Nevertheless, this result shows that the proposed scheme still performs well compare to RR scheme because it is taking less waiting time in terms of serving the packets.

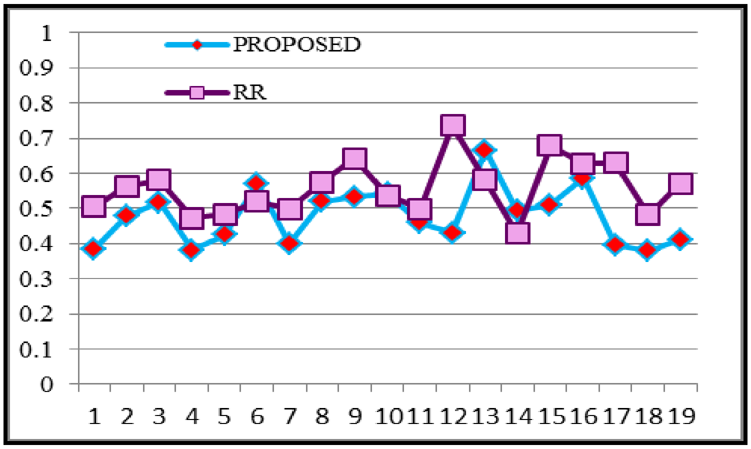

Figure 7. Average Delay vs. Offered Load for Each Scheme

Figure 8 shows the average delay against the offered load for each buffer in proposed scheme. The average delay of forwarding buffer (Q2) is almost higher than the direct buffer (Q1). The main reason behind the huge gap of average delay is that the direct buffer gets more advantages than forwarding buffer. This is because the direct buffer only receives original packet from each node. For example, Q1 at WN2 only receives original packet from WN2. However, forwarding buffer (Q2) at WN2 receive forward packet from WN3 and also reverse packet from WN1. Therefore there are more packets are waiting in the Q2 at WN2 to be served.

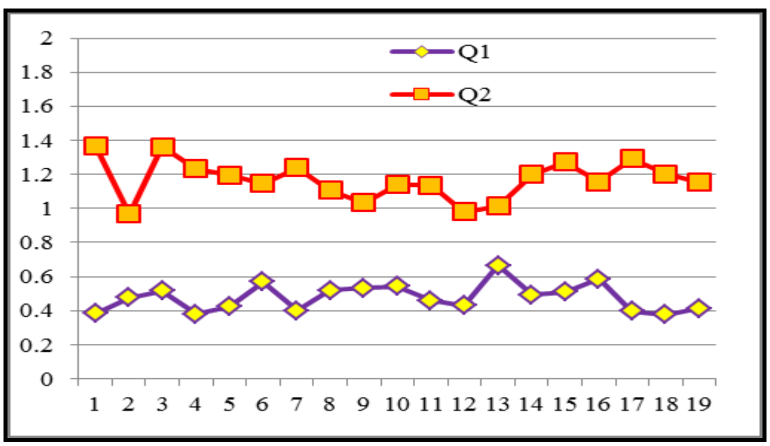

Figure 8. Average Delay vs. Offered Load for Each Buffer in Proposed Scheme

5. Conclusion and Future Works

In our research, we have dwelled on the issues of unfairness for the per-flow throughput in IEEE 802.11 multihop wireless LANs. As part of our study, we did some research on the performance of RR scheme and followed by a proposal of a new scheme. The proposal is done in line with our objective that is to enhance the network performance in terms of fairness index and average end-to-end delay.

We have compared RR scheme with our proposed scheme with different offered loads. The comparison is done and evaluated through the DES simulator. The main difference between our proposed scheme and RR scheme is that the proposed scheme is able to reduce the number packet loss during the transmission. The proposed scheme also focuses on reliable delivery of data from sources to its destinations. The simulation process consisted of several numbers of different scenarios and the results indicate that new proposed scheme has always outperformed the performance of RR scheme in terms of fairness index and average delay. The new proposed scheme also offers significant reduction in the average end-to-end delay when compared with the RR scheme.

Future researches in multihop wireless LANs can be done with regards to application demands instead of application types. Different applications have different sensitivity factors. Thus, different network designs have different constraints according to varying challenges issues. There are many opportunities for future work regarding to the enhancement of this model:

i. Applying other type of scheduling algorithms in the simulation model for comprehensive performance study regarding efficient, fair and robust scheduler.

ii. Adding real-time traffic such as video and audio in traffic generator to analyze the performance.

iii. Evaluating various performance analysis parameters such as average queuing delay, throughput, and end-to-end throughput.

References

[1] O. Shagdar, K. Nakagawa and B. Zhang, "Achieving Per-Flow Fairness in Wireless Ad Hoc Networks", Elec. Comm. in Japan, Part 1, vol. 89, no. 8, **(2006)**, pp. 37–49.

[2] H. Izumikawa, H. Ishikawa and K. Sugiyama, "Scheduling Algorithm for Fairness Improvement Among Subscribers in Multi-Hop Wireless Networks", Electronics and Communications in Japan (Part I: Communications), vol. 90, no. 4, **(2007)**, pp. 11–22.

[3] P. T. Giang and K. Nakagawa, "Achieving Fairness Over 802.11 Multihop Wireless Ad Hoc Networks", IEICE TRANS> COMMUN, vol. E92-B, no. 8, **(2009)**.

[4] V. Gambiroza, B. Sadeghi and E. W. Knightly, "End to End Performance and Fairness in Multihop Wireless Backhaul Networks", Proc. of ACM MOBICOM, (2004) September.

[5] K. Wakuda, S. Kasahara, Y. Tkahashi, Y. Kure and E. Itakaru, "A Packet Scheduling Algorithm For Max–Min Fairness in Multihop Wireless LANs", Computer Communications, vol. 32, **(2009)**, pp. 1437–1444.

[6] L. Kalampoukas, A. Varma and K. K. Ramakrishnan, "An Efficient Rate Allocation Algorithm for ATM Networks Providing Min-Max Fairness", UCSC-CRL-95-29 Computer Engineering Dept. University of California, Santa Cruz, Tech. Rep., **(1995)**.

[7] J. Jun and M. L. Sichitiu, "Fairness And Qos In Multihop Wireless Networks", Proc. IEEE 58th Vehicular Technology Conference, vol. 5, **(2003)**, pp. 2936–2940.

[8] S. Keshav, "Congestion Control in Computer Networks", PhD Thesis, University of California, Berkeley, **(1991)**.

[9] C. W. Sham and F. Y. Young, "Congestion Prediction in Early Stages", Proc. International Workshop on System Level Interconnect Prediction, **(2005)**.

[10] R. S. Shelar, P. Saxena, X. Wang and S. S. Sapatnekar, "An Efficient Technology Mapping Algorithm Targetting Routing Congestion Under Delay Constraints", Proc. of International Symposium on Physical Design, **(2005)**.

[11] J. Westra and P. Groeneveld, "Is Probabilistic Congestion Estimation Worthwhile?", Proc. International Workshop on System Level Interconnect Prediction, **(2005)**.

[12] D. Gaiti and G. Pujolle, "Performance Management Issues in ATM Networks: Traffic and Congestion Control", IEEE/ACM Transaction on Networking, vol. 4, **(1996)**.

[13] D. M. Chiu and R. Jain, "Analysis of the Increase and Decrease Algorithms for Congestion Avoidance in Computer Networks", Computer Networks and ISDN Systems, vol. 17, **(1989)**.

[14] M. Gerla and L. Kleinrock, "Flow Control: A Comparative Survey", IEEE Trans. Commun., vol. COM-28, no. 4, **(1980)**, pp. 553-574.

[15] E. L. Hahne, “Round-Robin Scheduling for Max-Min Fairness in Data Networks”, IEEE Journal on Selected Areas Communications, vol. 9, **(1999)**, pp. 1024-1039.
[16] B. Ji, C. Joo and N.B Shroff, “Delay-Based Back Pressure Scheduling in Multihop Wireless Networks”, IEEE/ACM Transaction on Networking, Issues 99, **(2012)**.

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