MAC-aware Rate Control for Transport Protocol in Multihop Wireless Networks

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***Abstract*—Transport layer performance in IEEE 802.11 multihop wireless networks (MHWNs) has been greatly challenged by wireless medium characteristics and multihop nature which are the sources of several types of packet loss including collision, random channel errors and route failures. Rate control transport protocols, the candidates for multimedia streaming applications suffer from high loss rates and end-to-end delay in MHWNs. A common research direction is that the rate control mechanisms at transport layer should be aware of MAC layer contention to keep the network load at a reasonable level. In this paper, we introduce a new MAC metric which reflects the contention and congestion levels more accurately. The metric is then used to improve the rate control mechanism of a rate-based transport protocol in MHWNs. The simulation results show that the adapted mechanism introduces significant performance improvement in MHWNs.**

I. Introduction

In recent years, performance improvement for TCP in multihop wireless networks (MHWNs) has been one of the wireless research directions which receives major attention. Due to the shared wireless medium characteristics such as interference, error prone channels and the multihop nature of MHWNs, not only TCP but also other Internet predominant transport protocols face challenges to perform properly in MHWNs [1] [2]. Indeed, in MHWNs, nodes have to contend with each other to get access to the medium [3]. Transport protocols like TCP usually misbehaves in MHWNs by overloading the network, which in turn exacerbates the contention problem. As MAC contention becomes serious, queueing delay, backoff and transmission delays and collision losses increase while the throughput decreases. This in turn impacts the performance of VoIP or streaming applications which have strict requirements in terms of loss rate and latency. Hence, congestion control mechanisms at transport layer should be aware of MAC layer contention to keep the network load at a reasonable level.

Based on this idea, several solutions have been proposed to improve TCP operation in MHWNs by passing MAC layer information to transport level entity to perform loss differentiation and congestion control algorithms [4] [5]. In this work, our interest is focused on the improvement of rate-based transport protocols like TCP-Friendly Rate Control (TFRC) [8] over MHWNs. Although TFRC has several advantages when working in wired networks, such as smooth sending rate and fairness with TCP flows, it suffers from performance degradation in MHWNs, such as it exhibit conservative behavior and may experience higher packet loss rate than TCP [9]. Chen et al. [9] claim that the TFRC's loss rate estimation used in the throughput equation is highly inaccurate in MHWNs. In addition, delay measurement is unreliable in MHWNs and does not reflect the growth of the end-to-end hop distance [10]. Hence, the equation is not guaranteed to use in this kind of environment [10]. It creates the need of a new rate control mechanism based on MAC information which operates efficiently in MHWNs.

The aim of our work is to propose a new rate regulation method which adapts the source bit rate depending on the MAC layer contention level. In our opinion, to improve the transport service, the contention/congestion phase has to be detected as quickly as possible to reduce its duration. Thus, our work firstly proposes a new MAC metric called Medium Access Delay (MAD), which efficiently reflects the contention and collision levels around a node in IEEE 802.11 MHWNs. Based on this metric, we then introduce a rate adaptation mechanism to improve the transport service in terms of end-to-end (E2E) delay and packet loss.

The paper is organized as follows. The next section gives a brief review of related work. Section III provides the definition of the Medium Access Delay metric. The proposal of MAD-TP is described in detail in Section IV. Section V exhibits the simulation scenarios and results. Finally, we conclude the paper in Section VI.

II. Related work

Since end-to-end information such as Round Trip Time (RTT) and loss rate is not sufficient enough to solve the aforementioned problems in MHWNs, most of the proposed schemes have a common ground that they try to exploit the MAC layer information to have better knowledge about what happens at lower layers. The exploited MAC information form the MAC metrics, each of which is a collection of one or more parts taken from the DCF (Distributed Coordination Function) scheme of IEEE 802.11 standard [3]. They are then sent up to transport layer in a cross-layer manner and are used in various ways to improve the transport protocol.

Li et al. [11] proposed a mechanism which enables TFRC to estimate the optimal network load level by considering the MAC layer contention. An optimum round-trip time is computed from backoff delay and transmission delay at MAC layer collected from all hops from source to destination. The current RTT is then compared to this optimal value to estimate the contention level and to adjust the traffic rate accordingly. However, since the end-to-end delay measurement is unreliable in MHWNs as mentioned in the previous section [10], this scheme need to be proved in more complex topologies than the simple chain topology in the paper.

To obtain the channel utilization information, Zhai et al. [5] collected the channel busyness ratio computed at each node and then estimate the network available bandwidth. This information is then used to adjust the traffic rate pumped into the network. This scheme has some disadvantages such as it requires some transport information present at MAC layer such as the packet sending rate. Moreover, the available bandwidth is estimated based on an assumption that the collision probability $p \leq 0.1$ [12]. But in fact, it is hard to fulfil these requirements because of the hidden node problem, which is the source of high collision loss rate and is very common in MHWNs.

Navaratnam et al. [13] proposed to use both channel busyness ratio and effective throughput computed at each node to assess the current network capacity in terms of both channel utilization and collision levels. Their proposed link adaptive transport protocol, LATP, uses this estimated available bandwidth, named permissible throughput, to control the sending rate. Nevertheless, the load at each link is contributed by several flows passing across it. So the available bandwidth should be fairly shared by all. But in LATP, each flow increases its sending rate with an amount of available bandwidth reported by feedback packets. This increase is sometimes too large and the total traffic increase by all of the flows may quickly overload the network.

With regard to these work, we remark that the proposed metrics reflect the global network state which includes both contention and successful transmission periods. Since the congestion is closely coupled with the contention as demonstrated in [10], we propose to improve the performance of transport protocols in considering only the contention state of the network. Moreover, the contention is a critical situation which should be reacted rapidly enough by an appropriate rate adaptation mechanism. The next section will introduce our novel MAC metric.

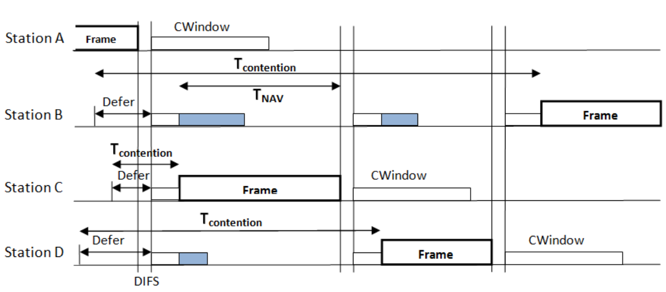

Fig. 1: IEEE 802.11 basic DCF

III. The Medium Access Delay

The metric Medium Access Delay is the delay information taken from IEEE 802.11 DCF model as showed in Figure 1. Due to the page limitation, we suppose that the reader is familiar with IEEE 802.11 DCF mechanism (for more details see [3]). We consider $T_{contention}$ in the DCF mechanism, which is the time a packet has to wait at MAC level before it is actually transmitted over the medium. By this definition we have:

$$T_{contention} = \sum^{n_{NAV}} T_{NAV} + \sum^{n_{busy}} T_{busy} + T_{backoff} \qquad (1)$$

where $\sum T_{NAV} + \sum T_{busy}$ represents the total channel busyness time due to the transmission of neighbor nodes that the packet has to defer during a backoff stage, in which T_{NAV} is the time indicated by received RTS/CTS packets (if used) and T_{busy} is the time indicated by physical Carrier Sensing (CS) mechanism. Note that the MAC protocol may freeze the backoff procedure as often as it receives RTS/CTS packets and busy channel indications from physical Carrier Sensing (CS) mechanism. Thus, the number of T_{NAV}, n_{NAV}, and the number of T_{busy}, n_{busy}, experienced by the node in a backoff stage duration depends on the number RTS/CTS packets received and the number of channel busyness indications from CS mechanism during that duration. $T_{backoff}$ is the backoff time which is calculated as follows:

$$T_{backoff} = N * aSlotTime \qquad (2)$$

where N is a random integer between [0,CW] and CW and $aSlotTime$ are respectively the Congestion Window used in DCF mechanism and the time unit defined in IEEE 802.11 PHY [3].

MAD metric is then simply defined as the average total contention delay for a packet at MAC layer before it is successfully transmitted or dropped after several failed retransmissions in an interval.

$$MAD = \frac{\sum^{N_{ap}} \sum T^{i}_{contention}}{N_{ap}} \qquad (3)$$

where N_{ap} is the number of arrival packets in the interval and $T^{i}_{contention}$ is the contention time at the i^{th} transmission attempt. Note that maximum retransmission number is limited by the parameter RetryLimit defined in the standard. The MAD metric is simple to implement with available functions provided by IEEE 802.11 standard [3]. The node's MAC can take T_{NAV} from the header of RTS/CTS packets, $T_{backoff}$ from its intrinsic variables, and physical and virtual carrier sensing mechanisms provide function to determine whether the channel is busy or not.

If the value of MAD increases, either or both possibilities may arise. Firstly, the channel is mostly used by other nodes' transmission so that the node has to defer longer to have a transmission opportunity. Secondly, the number of retransmissions increases due to higher level of collision with a note that the node returns to backoff stage after each failed transmission.

Obviously, MAD takes into account the medium busyness and hidden node problem.

We also evaluated the effectiveness of MAD by simulation in comparison with the metric Channel Busyness Ratio proposed in [12]. However, due to the page limitation, please refer to [14] for more details. The results show that the MAD metric is a good early signal of network congestion in both non saturated and saturated states since its behavior is representative to contention/congestion level in the network.

The next section will present in detail a rate control mechanism which employs MAD to adapt appropriately its sending rate in MHWNs.

IV. MAD-TP: NEW RATE ADAPTATION TRANSPORT PROTOCOL

The MAD-TP protocol is aimed at providing an efficient rate control mechanism at transport layer which can reduce the contention effect of MHWNs. This mechanism uses the MAD metric as an early indication of network contention level in order to adjust appropriately the pace of sending packets over the network. To do that, every node on the network measures the MAD value periodically. For every packet passing the node, it adds its MAD value to the existing value stored in an option field in the IP header, called Contention Delay (CD). With this rule, when the packet reaches the destination, the CD field will contain the cumulative contention delay along the path it has travelled. After process the cumulative contention delay from the arrival packet, the MAD-TP receiver feeds the network contention information back to the sender together with the receiving rate by using appropriate acknowledgement mechanism. The MAD-TP sender then uses this information to control the sending rate. The proposal is explained for each actor of the MAD-TP protocol.

A. Intermediate nodes

The role of intermediate nodes is to provide estimation of contention level experienced by each node along the connection path. Each node maintains the measurement of MAD in every interval. If the interval duration is short, the value of MAD may vary largely due to the change of contention level and therefore the sending rate which is based on MAD may fluctuate as well. In contrast, if the interval is too long, the value of MAD can not react quick enough to the change of the network status, thus reduce the effectiveness of MAD-TP. In our implementation, we chose the interval duration to 0.1 second as the trade-off between the smoothness and the effectiveness.

For i^{th} transmission of every arrived packet in the interval, the node's MAC records the time instant the packet starts to contend for medium access t_s^i and the time instant the packet starts to be actually transmitted over the medium t_t^i. Then the contention time $T^i_{contention}$ is simply calculated as $t_t^i - t_s^i$. During the interval, the contention time is aggregated over all transmission attempts of all arrived packets and the final value is divided by the number of arrival packets in that interval to form the MAD metric.

For all outgoing packets, the node updates the aggregated contention level in the field CD in IP header by adding its MAD value to the value existing in that field. When the packet reaches its destination, the receiver will obtain the cumulative value of MAD from all nodes along the path. Thus, the change of MAD of the critical nodes will also change the total MAD along the path, which in turn reflects the change of contention level of the connection.

B. MAD-TP receiver

The function of MAD-TP receiver is relatively simple. Every time receiving a packet, it takes the MAD value from CD field, noted MAD_{CD}, and number of hops, noted N_h, from TTL field in the IP header or from the routing table of source routing protocols and compute the $MAD_{sample} = MAD_{CD}/N_h$. The MAD-TP receiver then derives the mean contention delay per hop by using the Exponentially Weighted Moving Average (EWMA) function as follows:

$$MAD = \alpha MAD + (1-\alpha) MAD_{sample} \tag{4}$$

Since the sending rate depends on the value of MAD, we set $\alpha = 0.5$ as the trade off between the contention sensitiveness and the rate smoothness. This mean value is calculated as in equation 4 for every received packets. Whenever the receiver detects a loss, it immediately sends a feedback packet with MAD value to the sender so that the sender can update the sending rate according to the change of contention delay of the connection. In addition, the receiver should send at least a feedback every round trip time RTT if no loss is detected. This will help the sender keep in mind the updated knowledge about the connection. Note that the RTT is introduced to the receiver by a field in the data packet as in TFRC [8]. Beside the mean contention delay, the receiver also estimates the average receiving rate R_{rcv} from the last report until the current feedback packet is generated. These two values will be used in the rate control mechanism at the sender side MAD-TP.

C. MAD-TP sender

When the sender starts a new connection, the slow-start is invoked as following. The initial rate is set $1pkt/s$ when the sender has no sample of RTT. Every time receiving a feedback packet, the sending rate R is updated by the rule

$$R = max(2 * R_{rcv}, S/RTT) \tag{5}$$

where S is the packet size. The slow-start is terminated whenever the feedback MAD is greater than its predefined threshold MAD_{TH} or the sender does not receive any feedback packet after RTO as in TFRC [8]. In our implementation, we set $RTO = 4 * RTT$.

If the received $MAD \leq MAD_{TH}$, a conservative amount of traffic ΔR may still be pumped into the network. ΔR may be coarsely estimated as follows

$$\Delta R = (\frac{MAD_{TH}}{MAD} - 1) * R \tag{6}$$

The new expected sending rate is then $R + \Delta R$. However, to avoid the sudden change in the sending rate, we employed the rule deployed in the work of LATP [13] by which the new sending rate is chosen as follows :

$$R = max(min(2 * R_{rcv}, R + \Delta R, R + N * S/RTT), S/RTT) \quad (7)$$

where N is the number of RTTs from the last rate change. The equation 7 controls the update of sending rate such that it ensures that the MAD-TP rate is at least one packet per RTT and should not increase more than one packet per RTT.

If $MAD \geq MAD_{TH}$, the MAD-TP sender assumes that the connection experiences a severe contention along the path and will decrease the sending rate. We also use the decrease rule proposed by LATP [13] by which, the sending rate is reduced by 1/8 its current sending rate after each RTT but never smaller than one packet per RTT. The sender also halves the sending rate when the "NoFeedbackTimer" expires as in TFRC.

V. Simulation and results

The performance evaluation for our rate control proposal MAD-TP is carried out in comparison with TFRC and LATP. We use NS-2 simulator version 2.34 [15] to conduct the evaluation with the general configuration as in Table I. In all topologies, the nodes in the MHWNs are static to reduce the effect of mobility and the channel is set to be perfect to eliminate the effect of channel error loss. In the simulation, MAD-TP, TFRC and LATP operate as they always have packets to send for the scheduled sending time instants, thus their operation does not depend on the application rate. The performance metrics are Throughput, End-to-End (E2E) Delay and Packet Loss Ratio (PLR). They are averaged from 16 simulation runs in each scenario, each run is performed in 400s. Note that we do not consider the fairness in our simulation scenarios.

TABLE I: General configuration for simulation

Parameters	Value
Propagation Model	TwoRayGround
MAC protocol	802.11a DCF
Channel Capacity	6Mbps
Interface queue size	50
Carrier Sensing Range	$\simeq$ 500m
Transmission Range	$\simeq$ 250m
Data packet size	1000 bytes
Routing protocol	AODV

A. Scenarios

The simulations take place in three types of topology chain, grid and random because of the variety of interference schemes they represent.

In chain topology, a pair of nodes is 200m apart, this makes the two adjacent nodes are in the transmission range of each other and two nodes 2 hops away from each other are in their interference range. The first scenario in chain topology is to evaluate the performance of MAD-TP, TFRC and LATP with different number of hops. In this scenario, we set the number of hops ranging from 4 to 13 in order to observe more clearly the difference between the three protocols' operation. A connection is established from two end nodes in each network during the simulation. The second scenario is to evaluate MAD-TP performance in case of competing flows of 4 connections coexisting in the network with 8 hops. All connections have the same pair of source and destination as in the first scenario. Each one starts randomly in the first 3 seconds of each simulation.

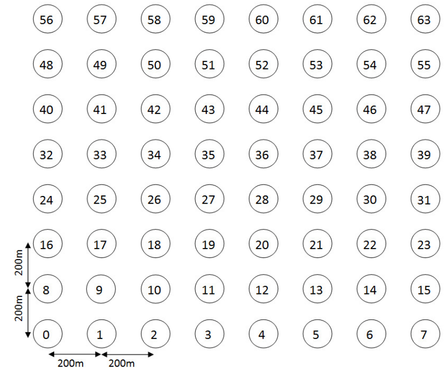

Fig. 2: Grid 8x8 topology

We use the grid topology as showed in Figure 2. This topology provides more intricate and adjustable node contention patterns. We set up 4 connection patterns such that they provide different contention levels in the network. Therefore, the performance of the MAD-TP can be evaluated thoroughly. In the pattern 1, we set up two parallel flows from node 16 to node 23 and from node 39 to node 32. These two flows are 400m apart so that each pair of nodes of the two connection lying on the same column of the grid are out of transmission range but on the carrier sensing range of each other. In the pattern 2, 4 parallel flows are established from nodes 8, 31, 40, 63 to nodes 15, 24, 47 and 56 respectively. Each pair of flows is also 400m apart. 8 parallel flows are initialized from nodes 0, 8, 23, 31, 32, 40, 55, 63 to nodes 7, 15, 16, 24, 39, 47, 48 and 56 respectively in pattern 3. Pattern 4 has 5 cross flows established from nodes 2, 8, 31, 40, 61 to nodes 58, 15, 24, 47 and 5 respectively where two parallel flows (2-58 and 61-5) cross three parallel flows (8-15, 31-24 and 40-47). Each connection corresponding to a flow starts randomly in the first 3 seconds of the simulation.

The random topology used in this simulation has 60 nodes placed randomly in 1500mx1500m area. The number of connections running simultaneously in the network is 5, 10, 15 and 20. Each pair of source and destination of a connection is chosen randomly with its hop distance of at least 3 hops.

The simulation results are explained in the following sections.

B. Results and discussion

The MAD threshold for MAD-TP

MAD_{TH} is an important parameter in the operation of MAD-TP. From our previous work [14], we observed that the network may work in two states: saturated and not saturated.

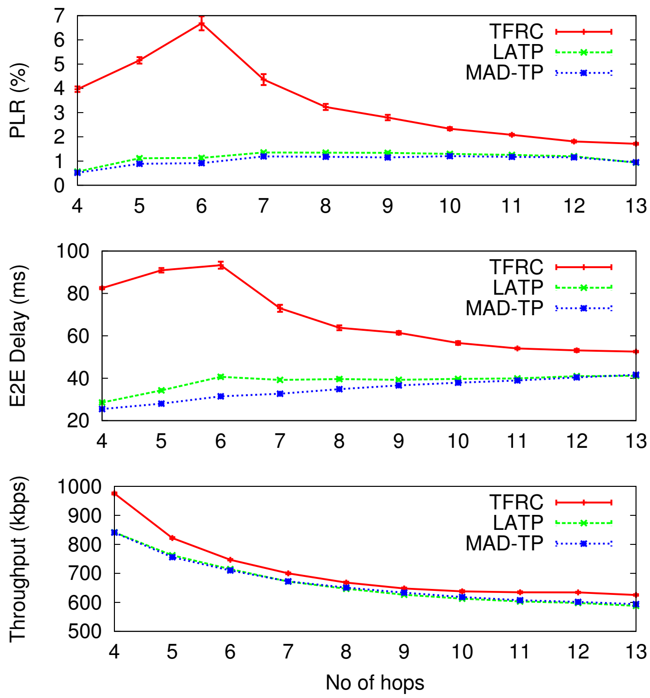

Fig. 3: Chain topology with 1 connection

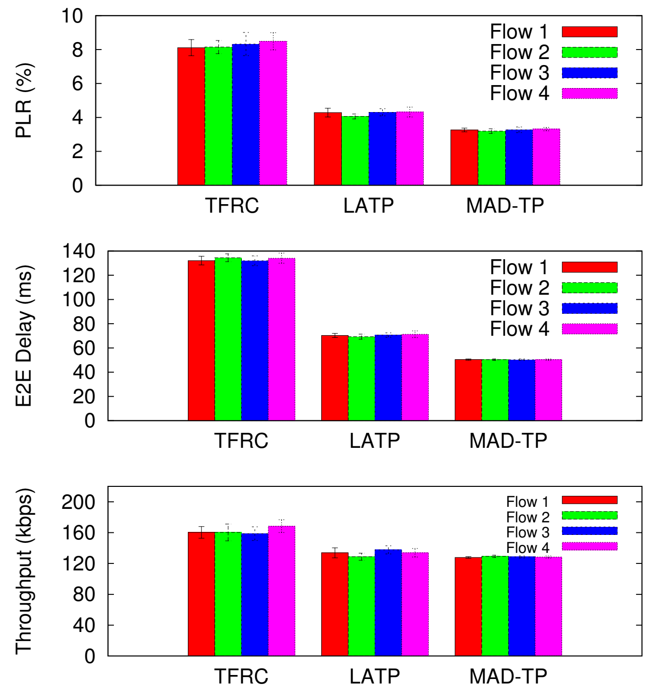

Fig. 4: Chain topology with 8 hops and 4 connections

In non-saturated state, the measured value of MAD is about 0.11 ms while in saturated state, that value is more than 1 ms. However, if the first value is used as the threshold for MAD-TP, it is too small to allow a reasonable throughput of MAD-TP. Thus, we set MAD_{TH} to 0.7 ms as the trade-off between Throughput and E2E Delay and PLR.

Chain topology

Figure 3 shows the result for the first scenario which varies the number of hops. We can observe that MAD-TP outperforms TFRC in terms of Packet Loss Ratio (PLR) and End-to-End delay. The PLR of TFRC is higher than that of MAD-TP from 0.8% (in case of 13 hops) to 6% (for network with 6 hops) and the time scale is from 10ms (for network with 13 hops) to 60ms (for network with 6 hops) for E2E delay. Particularly in the common MHWNs whose size is smaller than 10 hops, the difference is at least 1% for PLR and 20ms for E2E delay. The reason is that TFRC's rate control wrongly estimates the network capacity and tends to overload the MHWN which has limited resources. This problem is caused by TCP throughput equation used in TFRC which depends on inaccurate packet loss rate measurement in MHWNs [9], where losses are mostly due to channel contention. Thus, TFRC increases the rate inappropriately when the network contention is rather high and does not decrease the rate efficiently enough when the network contention becomes severe. As a consequence, the packets travelling along the path will suffer from high loss rate and delay caused by collision among contending nodes, multiple retransmission attempts at MAC layer as well as high level of channel busyness. In contrast, MAD-TP introduces small loss ratio and delay for all the number of hops. In addition, the E2E delay of a MAD-TP flow is getting longer with the increase of the hop number but is always smaller than the delay introduced by TFRC. This increase seems to be more "reasonable" than that of TFRC. These improved results come from the appropriate rate control of MAD-TP since it depends on the contention level in the network. Thus, MAD-TP always tries to keep the network operating in a low contention level status which in turn reduces the transmission attempts to successfully transmit a packet as well as the delay a packet experiences.

Figure 3 also shows that the average throughput of MAD-TP connection is smaller than that of TFRC but the difference is quite small. This is the price MAD-TP has to pay to achieve much better PLR and E2E delay. However, for applications which have strict packet drop rate and latency, we believe that this tradeoff is acceptable.

MAD-TP's performance is also better than that of LATP in terms of PLR and E2E Delay in chain network while it achieves almost the same throughput. The reason is that MAD detects heavy contention better than the metric Permissible Throughput used by LATP [13], which then makes MAD-TP control its sending rate more efficiently than LATP.

Figure 4 shows the results for scenario with 4 connections. We have the same observation as the previous scenario since MAD-TP outperforms TFRC in terms of PLR and E2E delay with a price of small degradation of throughput. MAD-TP also presents smaller PLR and E2E delay than LATP with almost the same throughput. The improvement is about 1% for PLR and 20ms for E2E delay. Note that this scenario has four connections which send more packets into the network than the previous scenario. Since the network capacity is unchanged, the network contention level becomes higher than that in the previous scenario due to the increase of packet number

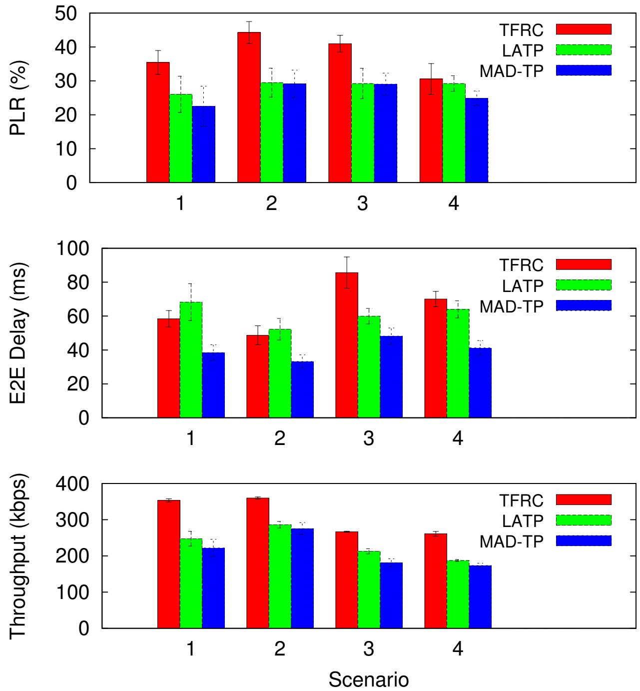

Fig. 5: Simulation results in grid topology

pumped into the network. The MAD metric signals the growth of network contention level sooner and more accurately than the metric used in LATP. The MAD-TP, therefore, controls the sending rate more efficiently than LATP and TFRC .

Grid topology

Figure 5 shows the simulation results for the 4 connection patterns. We can observe that MAD-TP operates efficiently in different levels of network contention. MAD-TP outperforms TFRC in terms of E2E Delay and PLR for all the scenarios while the Throughput is slightly smaller. MAD-TP also provides better performance compared to LATP in terms of E2E delay and PLR while the aggregated throughput is almost the same for both protocols. This prominence comes from the fact that the MAD-TP flows can soon realize that the network is becoming overloaded and thus reduce appropriately the sending rate. The increase of the MAD-TP flow is also not aggressive, hence the network avoids being quickly overloaded.

Random topology

Table II displays the simulation results. Random topology can be considered as a general case. We can observe that MAD-TP still provides better performance than TFRC and LATP in terms of PLR and E2E delay but with slightly smaller throughput as same as for other topologies. These results reinforce the efficiency of our approach.

TABLE II: Simulation results for random topology

Flows	PLR			Delay			Throughput		
	TFRC	LATP	MAD-TP	TFRC	LATP	MAD-TP	TFRC	LATP	MAD-TP
5	20.71	11.06	10.24	166.47	64.03	55.68	134.34	108.66	99.38
10	33.49	22.87	19.01	413.06	162.79	94.40	126.54	95.22	81.97
15	38.35	22.83	21.20	506.78	178.31	140.93	90.26	62.75	56.15
20	45.30	22.22	22.55	772.88	181.37	174.72	138.99	85.51	87.42

VI. Conclusion

In this paper, we have introduced a new MAC metric, named Medium Access Delay, which can describe accurately the network contention and network congestion states. We have also proposed a rate control mechanism using the MAD metric, called MAD-TP, that alleviates the main drawback of TFRC caused by the unreliable estimation of delay and loss rate. The simulation results show that MAD rate-based transport protocol outperforms TFRC and LATP in terms of End-to-End delay and Packet Loss Ratio which are the two critical criteria for streaming applications. This better performance comes from the fact that the MAD-TP can detect earlier the high contention state of the network, and provides a more efficient rate control.

Our future work is to model analytically the MAD metric and to prove the effectiveness of our control. In a second step, we plan to use the combination of several MAC metrics to reflect not only the contention but also the effect of mobility, and its respective rate adaptation policy which can meet QoS requirements of media streaming application in Mobile Ad hoc Networks.

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