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Evaluating TCP Flows Behavior by a Mathematical Model in WLAN

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ABSTRACT

This paper presents a mathematical model for evaluating TCP flows behaviors in WLAN. Now a days Downstream flows play an important role in traffic of a wireless networks. In many networks TCP downstream flows have less throughput than upstream flows. Simulated results of our proposed scenario show base station buffer size cause unfairness between flows. This unfairness specially injures downstream flows throughput. Our main goal is demonstrating this tow flow behaviors by simulating a proposed scenario and presenting an analytical and mathematical model for evaluating these flows behavior. You will see if the base station buffer size be larger than 1.5 time of TCP receiver window size, the throughput of tow flows is nearly equal hence accessing the sharing bandwidth remains fair

INDEX TERMS: WLAN,TCP, throughput, downstream flow, upstream flow, glomosim simulator.

I. INTRODUCTION

Over the past few years, Wireless Local Area Networks (WLANs) have gained an increased attention and a large number of WLANs are being deployed in universities, companies, airports etc.

now a days, wireless networks application is under the territory of WiFa and this standard decrease the limitation of transmission media of bandwidth , efficiency management and offering upmost quality of service to users.

Transmission Control Protocol(TCP) is used as a popular protocol of transport layer. TCP is a connection oriented protocol that that transmit reliably data packet(segment) , among end users .

Improvement of efficiency and throughput of TCP protocol in Wireless network is as important as improvement of usage of these kind of networks. On the other hand a lot of application use TCP as a protocol of transport layer and whereas many users of Internet employ Web based service, hence most significant capacity of Internet traffic is base on the TCP packet. Some observation on Internet traffic demonstrate TCP packets make more that 90% network traffic.

Fundamental problem of this protocol is the Unfairness problem. It means that allocating media is almost equal and fairness by all of the users. since creation of suitable situation for transmission TCP packet in wireless networks plays a significant role in enhancement of these network capability, hence unfairness problem is investigated and studied[1,2,3,4]. in this paper , we decide to investigate TCP flows behavior , in proposed scenario. We will prove that throughput of downstream and upstream flows have a vast dependency to buffer size of base station and window size of receiver. we evaluate simulated results with by a mathematical model and the results of this model demonstrate ,our model is similar to simulated results. And finally it is clear than for having fairness between flows the base station buffer sixe must be at last 1.5 time of receiver window size.

II. PROBLEM DESCRIPTION

In this section , we demonstrate a proposed solution to overcoming of TCP unfairness problem . Unfairness is researched as many standpoints [5,6,7,8,9] but our interest is “investigating and overcoming unfairness problem between throughput of upstream against downstream TCP flows”.

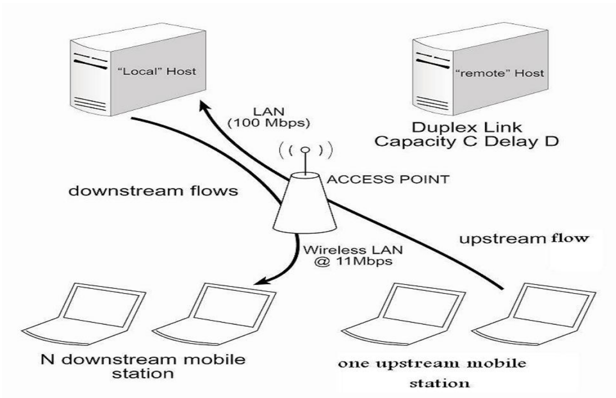

Figure 1: N upstream flows against N downstream flows

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Figure(1) shows a an common structure of on wireless lan where mobiles nodes access network by on work station(access point).

Majority of the IEEE 802.11 based WLANs employ Distributed Coordination Function (DCF) in Wireless Access Points (AP) to arbitrate the wireless channel among mobile nodes.

because of DCF algorithm of MAC sub layer , all of mobile nodes access to network(chanel) equally. It means that access point and mobile nodes have equal access to bandwidth .if all of the nodes is being in sending mode(upstream flow) or receiving mode (downstream flow) , they can divide the bandwidth equally and fairness is established between nodes. But imagine the case that one mobile node is in Sending mode and N mobile nodes are in Receiving mode(figure 1). In this case half of bandwidth is accessed by one sender and another half is divided between the N nodes which are in receiving mode (downstream flow).1/N of half of the bandwidth is allocated to each node which in receiving mode. therefore , whatever the number of receiver(node) is increased ,then a sender can access bandwidth manifold of one receiver and this ratio is increased linearly (base on increasing N). this unequal access of every node to bandwidth posed "UNFAIRNESS" between upstream and down stream flows.As you see , DCF protocol that is devised for equal accessing of bandwidth , poses unfairness.

OUR main goals are:

1- We assess relationship between TCP and MAC by simulation and analytical models.
2- We ILLUSTRATE 2 regions of TCP unfairness that depend on the buffer availability at the base station.

We want to develop a novel method to overcome Unfairness in TCP .

III. DESCRIPTION OF SCENARIO(ONE UPSTREAM AND ONE DOWNSTREAM FLOW)

We need to identify some performance parameters that use them in the rest of the paper. R_u : average TCP uplink throughput

R_d : average TCP downlink throughput

$\overline{R}$:ratio between R_u to R_d .

B: Base station buffer size

W: TCP receiver window size

(α) : The number of ACK packets per data packet

In order to analysis $\overline{R}$ ratio and identifying relevant parameters ,we simulate different scenario by glomosim simulator.

In order to describe relation between TCP and MAC sub layer which caused unfairness , assume the simple case of one mobile TCP sender and one mobile TCP receiver interacting with a server through a base station.

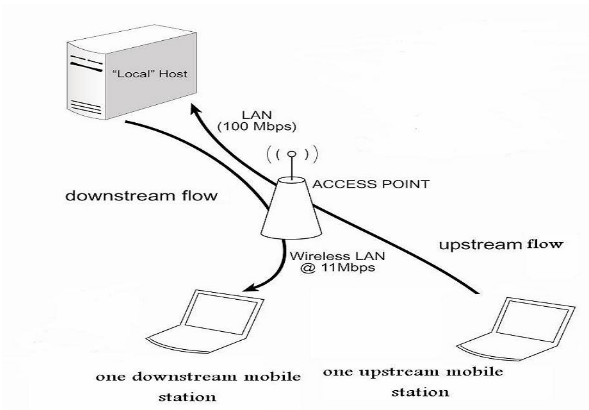

figure2: one upstream flow against one downstream flow[10]

Here, we are going to describe influence of Buffer size(B) of Base station and TCP receiver window size (W) on the average throughput ratio ($\overline{R}$).

Simulation of scenario

We start with the basic case of one mobile sender and one mobile receiver scenario. We set TCP receiver window to 39 . We vary the base station buffer size from 1 to 92. The results are shown in Figure 3. We also plot the total throughput in order to verify that it remains stable. For each buffer size. The number of ACK packets per data packet was set to 1.

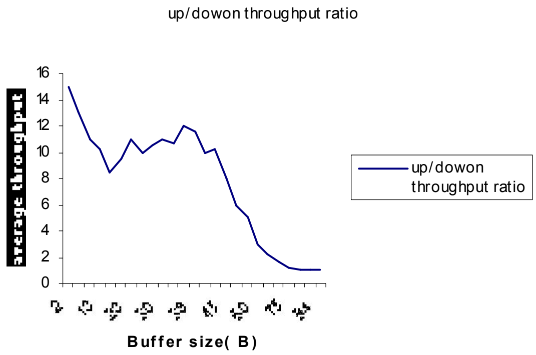

Figure 3:simulated results of $\overline{R}$

Figure 3:average throughput ratio for mentioned scenario

It is clear from above figure the base station Buffer size play a vital role in determining $\overline{R}$ ratio. as it can be observable

- when buffer size is up to 78, the $\overline{R}$ ratio is one .this ratio demonstrates , fairness between upstream and downstream flow. this region reflect when the buffer size is as large as twofold of maximum receiver window size (46 packets) of both flows, hence there is loss-free transmission in both upstream and downstream directions.
- When buffer size is between 39 to 78, $\overline{R} = \frac{R_u}{R_d}$ ratio is decreased. This phenomenon is normal too. When the base station buffer size is increased from 39 to 78, downstream packet drop rate is decreased gently. Hence the R_d is increased , therefore $\overline{R} = \frac{R_u}{R_d}$ is decreased. Why is downstream packet drop rate decreased? As you know, the buffer of base station accommodates data packet of downstream flow and Ack packet of upstream flow. As you will observe in later sections , we will prove that nearly in all cases the upstream TCP window size reaches its maximum size, this results demonstrate that R_u ratio is in maximum window size (W) (R_u = W) but the downstream window size changes. When base station buffer size is increased , data packet drop rate is decreases and THEREUPON throughput of downstream flow is increased and $\overline{R} = \frac{R_u}{R_d}$ declines to 1 ratio.
- It can observable from figure 3 ,When buffer size is between 2 to39 , the result is very noisy and $\overline{R} = \frac{R_u}{R_d}$ ratio is vary between 8.5 to 15. the simulating results in this region is not compatible to our Ratiocination . because it is clear that $\overline{R}$ is increased according to increasing W. but in this region, when W is increased from 2 to 39 , the $\overline{R}$ is not increased expectable ($\overline{R}$ ratio is noisy).we try to justify this phenomenon.

IV. Evaluating simulating result by An Analytical model

We are going to model TCP flows behavior that is plotted in figure 3.this curve represented the simulated results of one TCP upstream and one downstream flows Scenario. As, it is mentioned in last section TCP behavior has drastic dependency to TCP buffer size (W) Base station buffer size(B),We assume that los of packet is due to overflow in Base station buffer. when B be large enough, the loss of TCP upstream ACK packets have not actual influence on W. this is due to cumulative acknowledgement nature of TCP. It means that when a TCP ACK packet is lost, this loss of ACK can not cause of decreasing TCP sender buffer size(B). Because of next ACK packet is inclusive sequence number of former ACK packet (that is lost or doped) , thus TCP upstream flows increase. Until it reaches to full window size situation(W) and will remain in this size during the connection. Now we understand the reason of our claim in last section “nearly in all cases the upstream TCP window size reaches its maximum size” is cumulative acknowledgement of ACK packet.

But downstream TCP window size is severely varied base on the B and W and this impressionable causes that TCP window size (receiver window) be half. Precisely , if the Base station buffer size is larger than twice the TCP receiver window size , NO packet will be dropped. in this case the bandwidth is shared fairness between all stations.

Figure 3 shows that when B is equal or larger than 78 (twice of TCP window size of receiver (39))., in this case R_u is equal to R_d and ($\overline{R}$) is one, because of no packet is dropped.

In the case that buffer size(B) is small ,we can see TCP upstream flow owns vast share of available bandwidth. there is a explanation for this case:

Consider a base station buffer that it has (α W) ACK packet. hence there are (B- αW) downstream data packet in it. When a data packet is lost, this is detected by sender stations . when it sent 3 duplicate ACK and they are received by the receiver station the window size decreases to half of its value. Thus the window size will varry between B- αW and (B- αW)/2 and average of window size is 3 (B- αW)/4.

Now $\overline{R}$ is countable:

$$\frac{4w}{3(B-aw)} = \frac{R_u}{R_d} = \overline{R} \tag{1}$$

This equation nearly provide a correct explanation of figure 3 behavior, when B is large enough. But for small value of B, this equation is not correct . main mistake of this equation is that it is assumed , the base station buffer size full with (αW) ACK packet . but in a real case it is observable ,most of the time, there are a fewer than αW ACK packet (of upstream flow) in this buffer. Thus there are more place in the buffer for data packet of downstream flow(growth of R_d and hence decline of $\overline{R}$).

The problem of former equation causes that we devise another solution for analyzing

The behavior of TCP flows(shown in figure 3).

The service rate is considered equal to number of upstream packet which is sent to destination(R_u). The arrival rate is shown by λ is equal to $R_u + R_d$. utilization rate is equal to:

$$U = \frac{\lambda}{E} = \frac{R_u + R_d}{R_u} = 1 + \frac{1}{\overline{R}} \tag{2}$$

The probability that there are B packets in the buffer of base station is:

$$P = \frac{1-U}{1-U^{B+1}} U^B \tag{3}$$

$$P = \frac{1-(1+\frac{1}{\overline{R}})}{1-(1+\frac{1}{\overline{R}})^{B+1}} (1+\frac{1}{\overline{R}})^B \tag{4}$$

For small value of $\overline{R}$, $(1+\frac{1}{\overline{R}})^B$ is equal to $(1+B\frac{1}{\overline{R}})$ .thus the equation (3) converts to following equation:

$$P = \frac{1+B\frac{1}{\overline{R}}}{B+1} \tag{5}$$

We use the results of [11] for obtaining new value of R_u and R_d . it is assumed that no time out happen. R_u is equal to $R_u = \frac{W}{Rtt_u}$ and R_d is equal to $\frac{1}{RTT_d}\sqrt{\frac{3\alpha}{2P}}$. Thus $\overline{R}$ is equal to:

$$\overline{R} = \frac{RTTd}{RTTu} * \frac{w}{\sqrt{3\alpha/2p}} \tag{6}$$

Whereas delay of both flows is due to waiting in base station buffer and are equal , hence it is assumed $\frac{RTTd}{RTTu} = 1$ .and

$\bar{R}$ ratio is $\bar{R}=\sqrt{\frac{2PW^2}{3\alpha}}$. according equation (5) and (7) :

$$\alpha=1 \Rightarrow \bar{R}=\sqrt{\frac{2pw^2}{3\alpha}} \Rightarrow P=\frac{3\bar{R}^2}{2w^2} \tag{7}$$

$$\frac{1+B\frac{1}{\bar{R}}}{B+1}=\frac{3\alpha\,\bar{R}^2}{2W^2} \tag{8}$$

We assume that $(1+B\frac{1}{\bar{R}}) \approx B\frac{1}{\bar{R}}$ and. Thus equation (8) is changed to:

$$B\frac{1}{\bar{R}}=\frac{3\alpha\bar{R}^2}{2w^2}(B+1) \tag{9}$$

Now we success to compute $\bar{R}$ that is equal to:

$$\bar{R}=\sqrt[3]{\frac{2w^2B}{3\alpha(B+1)}} \tag{10}$$

For W=39 and $\alpha=1$, we can easily observe that the new curve of figure (4) which is computed by equation (10) matches the simulation value that is plotted in figure(3).as it is mentioned for B value between 39 to 78 ,the equation (1) matches to simulated result and for B value less than 39 , the equation (10) is nearly match to simulated value in figure(3).

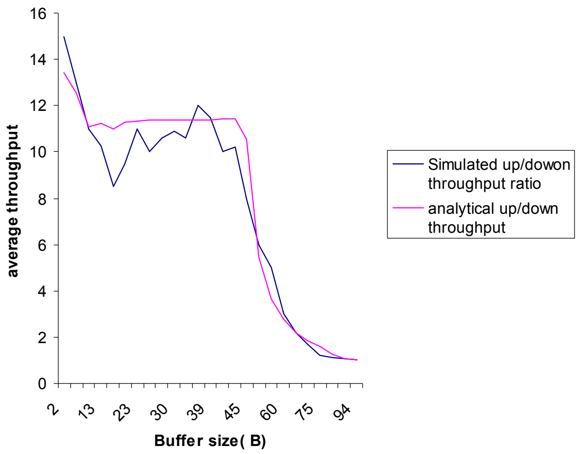

Figure 4: comparing computed and simulated results of $\bar{R}$

V. CONCLUSION

In this paper we try to show unfairness between TCP flows. We divided TCP flows to 2 flows called upstream and downstream. We run TCP flows behavior by glomosim simulator. In our scenario, there are one up stream and one down stream flow. We divides simulating result to 2 regions. First region is the buffer size between 39(equal to TCP sender window size) to 78. as , it is observed in figure (3), when the base station buffer size is enhanced from 39 to 78, downstream data packet can be located in buffer and the loss rate of data packet is decreased ,hence R_d is increased and there is fairness between 2 flows. But in region 2 ,when buffer size is between 2 to 39 .the simulated results are noisy. we understand , because of ACK packet cumulative nature , and probability of more drop rate of downstream data packet R_u is increased and R_d is decreased hence $\bar{R}=\frac{R_u}{R_d}$ is increased and this probability causes the simulated curve have not have its common behavior and results be noisy.

At last we interpret our simulated results by an mathematical model and figure(4) shows us our model is nearly equal to simulated results.

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