유무선 혼합 네트워크 환경에서 TCP 스킴의 성취도측정 평가

Measuring Achievement of TCP Schemes over Heterogeneous Wireless Networks

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요 약

본 논문에서는 유무선 혼합 네트워크상에서 TCP 스킴을 평가하기 위해 굿풋, 굿풋 인덱스, 정규화된 굿풋 인덱스 등을 정의하였다. 이러한 메트릭스들은 주로 TCP 스킴을 측정하기 위해 사용된 성능 메트릭스와는 달리 주어진 네트워크 상황에서 목적을 달성할 수 있는 능력인 성취도 개념을 포함하고 있다.

Abstract

In this paper, we define Achieved Goodput(AG), Achieved Goodput Index(AGI) and Normalized Achieved Goodput Index(NAGI) in order to evaluate TCP schemes over heterogeneous wireless networks. These metrics contain a concept of achievement, the ability of accomplishing an objective in a given network situation unlike performance metrics commonly used to evaluate TCP schemes.

Keyword : Transmission control protocol, achievement, metric, heterogeneous wireless network

1. Introduction

The transmission control protocol (TCP) is one of the core communication protocols of the Internet protocol suite which provides end-to-end reliable, in-order delivery of a stream of bytes, making it suitable for streaming media applications like world wide web, e-mail, and file transfer [1].

For this reason, significant enhancements on TCP have been made and proposed over the years through several experiments and analyses.

The main performance metrics used to evaluate TCP schemes are the throughput of a single TCP connection, the throughput of a set of TCP connections, and the fairness between a set of TCP connections. More specifically, the throughput of a single TCP connection is defined as the number of segments sent and acknowledged in a given time interval. The throughput of a set of n-TCP connections is defined as the sum of sent and acknowledged segments of n-TCP connections divided by the overall duration of these TCP connections. Meanwhile, the Jain's fairness index [2] is normally used to show the fairness between a set of n-TCP connections.

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Furthermore, the goodput performance for evaluating TCP schemes is commonly used because it is a direct indicator of network performance [3]. The goodput of a flow is the bandwidth delivered to the receiver from the sender, excluding duplicate packets. Since goodput means the effective amount of data delivered through the network, it is more realistic and suitable than throughput in showing the performance of TCP schemes. In addition, the dynamics of congestion window, slow-start- threshold (ssthresh), round-trip-time (RTT), and packet loss in the network path are considered. Most these metrics mainly focus on identifying the characteristics and behavior of TCP itself in details, whereas it is important to note that there isn't any metric containing a concept of achievement on TCP.

This paper introduces achievement metrics and measurement methodology for evaluating TCP schemes in terms of achievement. Meanwhile, recent advances in wireless communication technology have led to significant innovations that enable wireless networks to provide much more bandwidth. However, wireless networks still have link errors in spite of the evolution of error detection and correction. Therefore, at this point, we may be interested in the relationship between high bandwidth wireless networks with link error and TCP schemes over them. Hence, the simulation environment we presented in this paper reflects these needs.

2. Related Works

In this section, we summarize commonly used metrics that have been proposed to evaluate the performance of TCP congestion control schemes in wired, wireless and heterogeneous networks

2.1 Throughput, Delay, and Loss Rates Metrics

Throughput can be measured as a router-based metric of aggregate link utilization. It is sometimes distinguished from goodput. Note that maximizing throughput is of concern in a wide range of environments, from highly-congested networks to under-utilized ones, and from long-lived flows to very short ones. Throughput has been used as one of the metrics for evaluating Quick-Start, a proposal to allow flows to start-up faster than slow-start, where throughput has been evaluated in terms of the transfer times for connections with a range of transfer sizes [4] [5] [6].

Some researchers evaluate transport protocols in terms of maximizing the aggregate user utility, where a user's utility is generally defined as a function of the user's throughput [7].

Like throughput, delay can be measured as a router-based metric of queuing delay over time, or as a flow-based metric in terms of per-packet transfer times. Per-packet delay can also include delay at the sender waiting for the transport protocol to send the packet. For reliable transfer, the per-packet transfer time seen by the application includes the possible delay of retransmitting a lost packet.

Packet loss rates can be measured as a network-based or as a flow-based metric. In RFC 3611, a burst is defined as the maximal sequence starting and ending with a lost packet [8]. In some cases, it is useful to distinguish between packets dropped at routers due to congestion, and packets lost in the network due to corruption. Note that in some cases the retransmit rate can be high, and the goodput correspondingly low, even with a low packet drop rate [9].

2.2 Fairness and Convergence

There are fairness and convergence times as another set of metrics. Fairness can be considered between flows of the same protocol, and between flows using different protocols as well as between sessions, between users, or between other entities. A number of different fairness measures is the max-min fairness [10], proportional fairness [11], the fairness index proposed in [12], and the product measure, a variant of network power [13]. Metrics for fairness between flows not consider different characteristics of flows, such as the number of links in the path, or the round-trip time. In order to fairness metrics, x_i is to the throughput for the i-th connection. Jain's fairness index: The fairness index in [12] is where there are n users.

$$\frac{(\sum_{i}^{n} x_i)^2}{n\sum_{i}^{n}(x_i)^2}$$

The ranges of fairness index is from 0 to 1 and is maximum when all users receive the same allocation. When k users equally share the resource, this index is k/n. and the other n-k users receive zero allocation. The product measure: The product measure $product_i$ x_i, the product of the throughput of the individual connections, is also used as a measure of fairness. It is shown that for a network with many connections and one shared gateway, the product measure is maximized when all connections receive the same throughput in [14].

The metrics for fairness between flows with different resource requirements are with different utility functions, round-trip times, or numbers of links on the path. Fairness can be fulfilled in 3 ways, max-min fairness, proportional fairness and minimum potential delay fairness. In order to satisfy the max-min fairness criteria, the smallest throughput rate must be as large as possible. Given this condition, the next-smallest throughput rate must be as large as possible, and so on. Thus, the max-min fairness gives absolute priority to the smallest flows. In contrast, with proportional fairness, a feasible allocation x is defined as proportionally fair if for any other feasible allocation x*, the aggregate of proportional changes is zero or negative:

$$\frac{\sum_{i}(x_i^* - x_i)}{\sum_{i} x_i} \leq 0$$

On the other hand, minimum potential delay fairness has been shown to model TCP [15], and is a compromise between max-min fairness and proportional fairness.

When we look into the fairness, there're several factors to be considered. They're throughput, the number of congested links, round-trip times, packet size and convergence times.

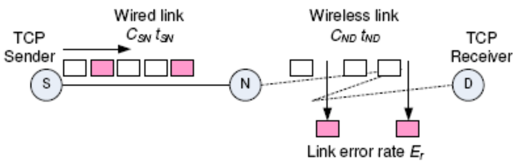
There're tradeoffs between fairness and throughput. The fairness measures in the section above generally measure both fairness and throughput, giving different weights to each. Regarding the relation between fairness and the number of congested links, there is not a clear consensus for the fairness goals, in particular for fairness between flows that traverse different numbers of congested links [16]. Fairness between flows with different round-trip times [17] has been one of the goals cited in a number of new transport protocols. However, there's not a consensus in the networking community about the desirability of this goal, or about the implications and interactions between this goal and other metrics [18]. When it comes to packet size, one fairness issue is that of the relative fairness for flows with different packet sizes like between file transfer applications and low-bandwidth VoIP flows. Lastly, convergence times concern the time for convergence to fairness between an existing flow and a newly-starting one, and are a special concern for environments with high-bandwidth long-delay flows.

3. Achievement Metrics and Measurements

In this session, we introduce achievement metrics and measurement methodology for evaluating TCP schemes in terms of achievement.

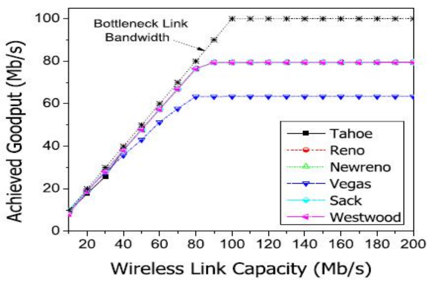
3.1 Achieved Goodput

Achievement is the ability of accomplishing an objective in a given network situation. For instance, we assume that the sender connects to a wireless base station with a 10 Mb/s error free link, the base station is linked to the receiver with a 2 Mb/s lossy channel and a single TCP connection running a FTP application delivers data from the sender to the receiver. In this case, we assume that the objective of the TCP scheme is to transmit as many data as end-to-end bottleneck link bandwidth of 2 Mb/s. For this reason, we define Achieved Goodput (AG) as a pair of (g, b) where g is goodput performance of a TCP scheme and b is end-to-end bottleneck link bandwidth in a given network topology. In the topology, N is edge router at wired networks that is, there is one of node and the D is nodes of destination

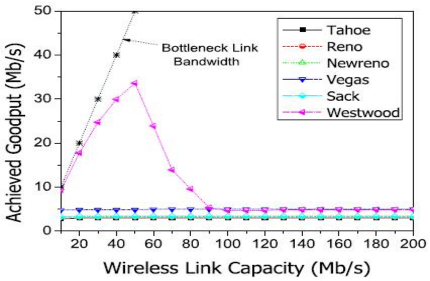


(Fig. 1) Network topology for obtaining achievement measurements.

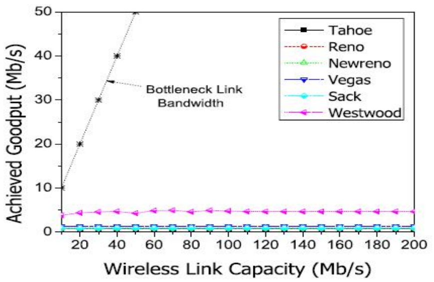
In order to measure the AG, we consider the well-known TCP schemes, e.g., TCP Tahoe, -Reno, -Newreno, -Vegas, -Sack and -Westwood, in mixed wired and wireless networks. As shown in Fig. 1, a single TCP source performs a file transfer. The wired link capacity C_{SN} is set to the fixed value 100 Mb/s and the wireless link capacity C_{ND} varies from 10 Mb/s to 200 Mb/s. Propagation delay t_{SN} and t_{ND} are set to 45 ms and 5 ms, respectively so that the RTT can be equal to 100 ms. The link error rate Er at the wireless link is the packet loss rate where the link drops packets according to a Poisson process. Errors can be generated from a simple model such as the packet error rate, or from more complicated statistical and empirical models. In other words, we can use these error models properly to implement wireless network environments suffering from errors or losses. Therefore, we implemented a function of random simple error model in NS-2 network simulator. The packet size of the TCP source is equal to 1,000 bytes and all queues can store a number of packets equal to the bandwidth-delay product. We run the simulation using the NS-2 network simulator [19]. There is no special reason to select the TCP schemes mentioned above. The simulation time is 200s. The AG result is shown in Fig. 2. In case of Er = 0%, for the wireless link capacity smaller than 80 Mb, the AG of all TCP schemes closely approaches to the curve of end-to-end bottleneck link bandwidth. Beyond that point, most TCP schemes perform the AG of less than 80 Mb/s without more increase even if the end-to-end bottleneck link bandwidth is 100 Mb/s. It is important to note that the wireless link capacity is higher than the wired link capacity and its error rate is 0%, which means this environment is matched to



(a) *Er* = 0%

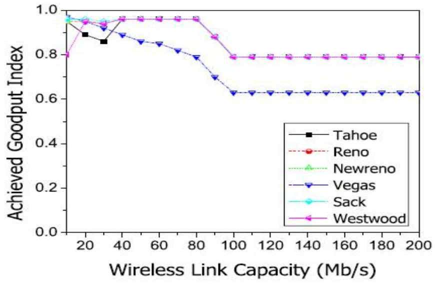


(b) *Er* = 0.1%

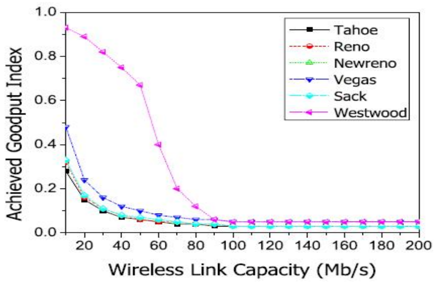


(c) *Er* = 1%

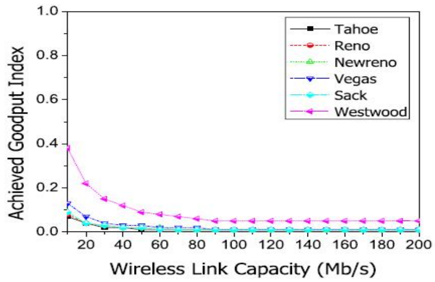
(Fig. 2) AG of TCP schemes with the random link error rate of (a) 0%, (b) 0.1% and (c) 1%.



(a) *Er* = 0%



(b) *Er* = 0.1%



(c) *Er* = 1%

(Fig. 3) AG I of TCP schemes with the random link error rate of (a) 0%, (b) 0.1% and (c) 1%.

broadband optical access networks, e.g., fiber to the x (FTTx) and passive optical network (PON).

In case of Er = 0.01%, for the wireless link capacity smaller than 50 Mb, only TCP Westwood closely approaches to the curve of end-to-end bottleneck link bandwidth whereas the other TCP variants experience a severe degradation in performance. However, the performance of TCP Westwood decreases significantly beyond the wireless link capacity of 50 Mb. In the case of Er = 1%, all TCP schemes experience a severe degradation in performance for all the wireless link capacities. The case that the link errors on the network exist and the wireless link capacity is higher than the wired link capacity is matched to broadband wireless access networks, e.g., WiMAX. Consequently, it shows that TCP schemes presented in this paper have limitation of scale when they are used in broadband optical or wireless access networks.

3.2 Achieved Goodput Index

Next we define Achieved Goodput Index(AGI) as the ratio of AG to end-to-end bottleneck link bandwidth. Given a network topology i, the AGI of network topology i is defined as

$$a_i = \frac{g_i}{b_i} \tag{1}$$

where gi and bi are AG and end-to-end bottleneck link bandwidth in network topology i, respectively. The result of AGI of TCP schemes is shown in Fig. 3.

In case of Er = 0%, for the wireless link capacity smaller than 80 Mb, the AGI of all TCP schemes has more than 0.95, which can be evaluated to outstanding performance. However, for the wireless link capacity greater than 80 Mb, the index decreases by about 20 % and there is no performance gain even if the wireless link capacity is much wider than the wired link capacity.

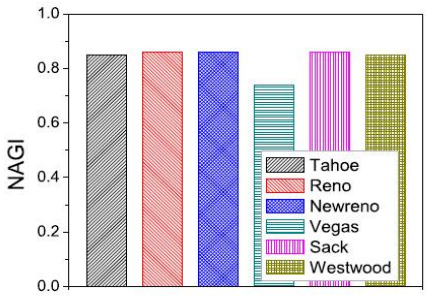
In case of Er = 0.1%, for the wireless link capacity smaller than 50 Mb, the indexes of most TCP schemes decrease significantly whereas the index of TCP Westwood has more than 0.6 which is satisfactory performance. However, TCP Westwood's index decreases significantly beyond the wireless link capacity of 50 Mb even if TCP Westwood is relatively robust to link error rate. In addition, all TCP schemes show a severe degradation having the index of less than 0.2 beyond the wireless link capacity of 70 Mb. In case of Er = 1%, all TCP schemes experience a severe degradation having the index of less than 0.4 for all the wireless link capacities.

3.3 Normalized Achieved Goodput Index

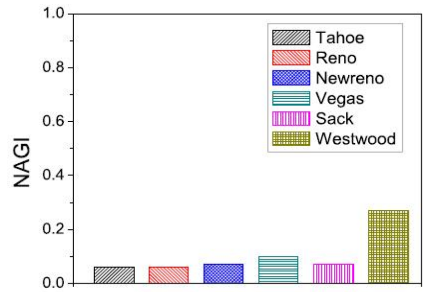
We also define Normalized Achieved Goodput Index (NAGI) as the average of a set of AGI. Given a set of AGI (a1, a2, , , , an), the NAGI of the set is defined as

$$NAGI = \frac{\sum_{i=1}^{n} a_i}{n} \tag{2}$$

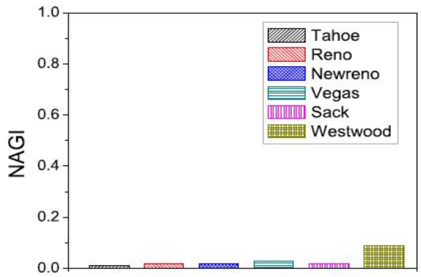
The value of NAGI is between 0 and 1. AG and AGI focus on the individual performance by independent network topology whereas NAGI focuses on the collective performance of the overall network topologies. The result of NAGI of TCP schemes is shown in Fig. 4.



(a) Er = 0%



(b) Er = 0.1%



(c) Er = 1%

Fig. 4. NAGI of TCP schemes with the random link error rate of (a) 0%, (b) 0.1% and (c) 1%.

In case of Er = 0%, the NAGIs of most TCP schemes except TCP Vegas have more than 0.8, which can be evaluated to satisfactory performance.

In case of Er = 0.1%, TCP Westwood has the NAGI of more than 0.2 whereas the others have the NAGI of around 0.1. In other words, TCP Westwood outperforms the other TCP variants by more than 50 % in the case of Er = 0.1%. Meanwhile, in the case of Er = 1%, TCP Westwood has the NAGI of around 0.1. Even if TCP Westwood has relatively the best performance of them as the link error rate is becoming high, the NAGI of around 0.1 means "not satisfactory".

4. Conclusion

We have considered "achievement," the ability of accomplishing an objective in a given situation. There are many performance metrics such as throughput, goodput, fairness and friendliness, which are used to evaluate TCP schemes.

However, these metrics have no concept of achievement because they don't have any objective.

In other words, they are just tools for comparing performance relatively among TCP schemes, not absolutely. Therefore, the paper focuses on the achievement measurements of TCP schemes over heterogeneous wireless networks. The network topology presented in this paper is commonly used one with its bandwidth and propagation delay. Under the condition, we have defined and measured AG, AGI and NAGI. AG and AGI are individual performance metrics, whereas NAGI is the collective performance metric. One of the benefits we can get using these achievement measurements is to index the achievement of TCP schemes according to a given objective.

Indexed achievement gives us "look and feel" performance comparisons between TCP schemes. Moreover, we have shown that it is possible to compare TCP schemes relatively as well as numerically.

To obtain the achievement measurements of well-known TCP schemes, we have considered the heterogeneous wireless network with a single TCP connection traversing a wired link as well as a wireless link affected by independent random link errors. For this network scenario, we found that TCP schemes mentioned in this letter would have limitation of scale if used in broadband access networks.

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