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TCP-Jersey over High Speed Downlink Packet Access

Shupeng Li

Lucent Technologies
Whippany, NJ, 07981, USA
shupeng@lucent.com

Nirwan Ansari

Advanced Network Laboratory
New Jersey Institute of Technologies
Newark, NJ 07102-1982 USA
ang@njit.edu

Abstract - TCP performance over wireless networks is critical as the Internet gradually expands to the wireless territory. Traditional TCP schemes have been proven to be inefficient for the wireless network, as they are designed and optimized for the wired network. Recently, schemes that are aimed to improve TCP performance in wireless networks have been proposed. In particular, TCP-Jersey, a TCP scheme that was designed for heterogeneous network consisting of wired and wireless links, has shown significant improvement over other TCP variants. In this paper, performance of TCP-Jersey over High Speed Downlink Packet Access (HSDPA) in the Universal Mobile Telecommunication System (UMTS) is evaluated and compared with other TCP variants through detailed system modeling and extensive simulations. The result shows that TCP-Jersey produces significant goodput improvement over other TCP variants tested.

I. Introduction

The last few years have witnessed the explosion of portable and wireless devices. At the same time, the Internet has also experienced a dramatic growth, and is becoming a wired-cum-wireless network. This development is no doubt aided by the ideas and implementations of the TCP/IP protocol suites. However, traditional TCP algorithms are not optimized for the wireless part of the network, as they are developed for the wired network and are based on assumptions that are not necessarily held true in the wireless environment. For example, it is widely known that traditional TCP schemes cannot differentiate the packet losses caused by congestion from the packet losses caused by wireless link errors, thus leading to a certain degree of performance degradation. Since then, there have been a number of studies [1] [2] [3] aiming to improve its performance. Among them, TCP-Jersey [3] adopts an end-to-end modification approach. This way, it preserves the original TCP semantics, maintains the network layer structure and requires minimum modification at end hosts and routers. It was shown in [3] that TCP-Jersey demonstrates a promising performance improvement, and at the same time maintains fairness and friendliness. However, how the proposed TCP schemes perform in[1] the real wireless systems remains to be addressed, as past studies assume a simple Markov error model to represent the wireless part of the network. In reality, wireless systems, like the 3G cellular networks, are complicated systems that consist of multiple protocol layers. Among them, link layer recovery protocols operate below the TCP layer and have complex interactions with TCP. Also, TCP functionality is complicated by the fact that the radio resources allocated to each user is varying, sometimes quite fast. This is because base stations typically adopt a MAC layer scheduling algorithm to maximize air interface throughput as well as to keep a certain degree of fairness. This MAC scheduling has two immediate effects on TCP flow control: First, the TCP sender sees a varying queuing delay of its packets. Second, TCP experiences a time varying bandwidth more volatile than the bandwidth oscillation expected in wired networks. To assess TCP performance over High Speed Downlink Access (HSDPA) for UMTS, we have developed detailed modeling that includes models from the physical layer up to the application layer, and conducted extensive simulations.

The rest of the paper is organized as follows: Section II describes some of the TCP variants as well as the basic characteristics of TCP-Jersey. In Section III, we present a brief overview of the HSDPA system. In Section IV, we present the simulation model. In Section V, we present the simulation results. Concluding remarks are given in Section VI.

II. TCP Evolvement and TCP-Jersey

TCP is designed for connection-oriented, reliable, byte stream service with friendly congestion control and flow control mechanisms. TCP transport protocols have evolved gradually since its original incarnation.

A. *TCP Tahoe*

In the original TCP implementation, a go-back-n model using cumulative positive acknowledgements is employed. This implementation also needs a retransmit timer expiration to re-send data lost during the transportation. It is later found that this type of TCP did little to minimize network congestion. TCP Tahoe [4] divides the transmission into three phases, namely slow-start (SS), congestion avoidance (CA), and fast retransmit. These new features improve channel utilization and connection throughput.

B. *TCP Reno*

TCP Reno [5] introduced a mechanism which is called fast recovery, and it is based on the idea that the only way for a loss to be detected via a timeout and not via the receipt of a duplicated packet is when the flow of packets has completely stopped, thus indicating heavy congestion. By using Fast

[1] This work has been supported in part by the New Jersey Commission on Science and Technology via NJWINS.

Recovery, fewer packets are sent out during fast retransmit until TCP knows that there is no imminent congestion.

C. *TCP New Reno*

It was found that TCP Reno encounters several problems with multiple packet losses in a window of data (usually in the order of half of a window). Studies in [6] have shown that when more than one loss occurs within one window of data, fast retransmit and fast recovery will be triggered several times in one RTT. In order to fix the above problems, TCP New Reno was proposed. It acts the same way as Reno in the case of one segment loss; however, when there are multiple packet losses, the ACKs for the retransmitted packets will acknowledge some, but not all the segments sent before the Fast Retransmit. Unlike Reno, New Reno retransmits one packet per RTT until all the lost packets are retransmitted. By doing this, it avoids triggering multiple fast retransmits from a single window of data.

D. *TCP SACK*

TCP "Selective Acknowledgement" (SACK) [7] is another effort to solve the problem of TCP Reno, but this approach requires changes to both the sender and receiver side. SACK requires that segments should not be acknowledged cumulatively but selectively, since a TCP sender can only learn about a single lost packet per RTT with the limited information available from cumulative acknowledgements. So, the SACK option field is added in the ACK packet, which reports non-contiguous blocks of data that have been received correctly. In this way, the sender can retransmit only the missing part.

E. *TCP-Jersey*

The aforementioned TCP schemes work fairly well in the wired network where packet losses are mainly due to link congestions, and those caused by link errors are usually trivial. However, in the wireless environment, traditional TCP exhibits throughput degradations [8], because packet losses, now most likely due to radio channel (slow and multi-path fading), handoff and mobility, are no longer negligible.

In [3], an end-to-end approach modification, TCP-Jersey was proposed. TCP-Jersey adds two improvements, the available bandwidth estimation (ABE) algorithm and the congestion warning (CW) router configuration. ABE is meant to more accurately estimate the time-varying network bandwidth. It modifies at the TCP sender side to monitor the rate of receiving ACKs, and then calculates the optimum congestion window size based on this estimation and guides the sender to adjust its transmission rate when the network becomes congested. TCP-Jersey computes the optimum congestion window once every RTT using a modified time-sliding window algorithm. The ABE estimator works as follows [3]:

$$R_n = \frac{RTT * R_{n-1} + L_n}{(t_n - t_{n-1}) + RTT} \qquad (1)$$

where R_n is the estimated bandwidth when the n^{th} ACK arrives. t_n and t_{n-1} are the n^{th} and $(n-1)^{th}$ ACK arrival time, respectively. L_n is the amount of data that the n^{th} ACK acknowledges, and RTT is the TCP's estimation of the end-to-end RTT delay at time t_n . With the estimated network bandwidth, the optimum congestion window cwnd in units of segments can be calculated as:

$$cwnd_n = \frac{RTT * R_n}{segment\_size} \qquad (2)$$

CW is an ECN [9] like configuration of network routers so that routers can alert end stations by marking all the packets when there is a sign of network congestion. The differences between ECN and CW lie mainly in the marking scheme. ECN marks packets probabilistically when the average queue length lies between $\min_{th}$ and $\max_{th}$, while CW marks all the packets when the average queue length exceeds a threshold. The advantage of non-probabilistic marking is that the TCP sender who receives these marks can decide its window adjustment strategy rather than being influenced by the probabilistic marking of the packet in ECN. The purpose of CW is to convey a simple image of the bottlenecked queue to the sender. Also, the marking of packets by the CW configured routers helps the sender of the TCP connection to effectively differentiate packet losses caused by network congestion from those caused by wireless link errors.

TCP-Jersey works on the combination of ABE and CW. The detailed operation and flowchart can be found in [3].

III. High Speed Data Packet Access (HSDPA)

To provision the enhanced support for wireless Internet services, HSDPA was standardized in 3GPP R5 and R6; it offers peak air interface data rates up to 14.4Mbps, thus dramatically improving the existing packet service capability of UMTS R99. The support of High-Speed Downlink is provided by means of a shared channel called High Speed Downlink Shared Channel (HS-DSCH). HS-DSCH channel can accommodate a maximum of 15 codes, and the sharing of HS-DSCH is facilitated in both the time and code domain.

A fast MAC layer scheduling algorithm determines to which User Equipment (UE) the shared channel transmission should be directed at each Transmission Time Interval (TTI). The key is a rapid adaptation to changes in the radio environment. Also, HSDPA employs a more effective rate control strategy called adaptive modulation and coding (AMC). The principle of AMC is to change the modulation and coding format in

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accordance with variations in the channel conditions. The channel conditions can be estimated based on feedback (channel quality indicator) from the receiver. In a system with AMC, users with higher geometry, e.g., users close to the cell site, are typically assigned higher order modulation with higher code rates, while users in low geometry positions are assigned lower order modulation with lower code rates. Also, instead of sequentially allocating radio resources among users (Round-Robin scheduling), the aggregated cell capacity can be increased significantly by using a channel-dependent scheduler. At the same time, each user can be guaranteed a minimum throughput by applying a certain fairness constraint.

Another important technique used by HSDPA is fast hybrid ARQ. Rather than discarding erroneous packets, UE can rapidly request the retransmission of missing data and combine information from the original transmission with that of the later transmission before attempting to decode the packet. The ARQ combining scheme is based on Incremental Redundancy (IR), wherein instead of sending simple repeats of the entire coded packet, additional redundant information is incrementally transmitted if the decoding fails on the first attempt. The decoder at the receiver combines these multiple copies of the transmitted packet weighted by the received SNR. Time Diversity gain is thus obtained. H-ARQ autonomously adapts to the instantaneous channel conditions and is insensitive to the measurement error and delay. Combining AMC with H-ARQ leads to the best of both worlds - AMC provides the coarse data rate selection, while H-ARQ facilitates for fine data rate adjustment based on channel conditions. In short, the primary benefit of HSDPA is improved end-user experience for wireless broadband applications: it brings higher bit rate, shorter round trip time, and much improved system capacity.

IV. Simulation Models

In the simulation, we consider the scenario where UE is downloading files from a fixed host as shown in Figure 1.

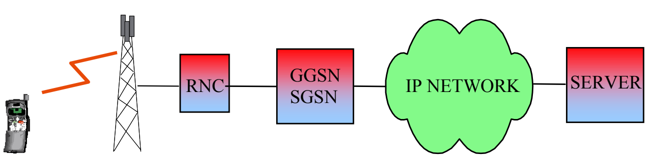

Fig. 1. Simulated link between the UE and the Server

The UE connects with the server through the Radio Network Controller (RNC), Gateway GPRS (General Packet Radio Service) Support Node (GGSN), and Serving GPRS Support Node (SGSN). The wired delay taken in the simulation is 50ms, and the wired network bandwidth is set to be 100Mb/s so that it will not become the bottleneck in the simulation.

A. *System & Link Level Modeling*

The cellular system consists of 19 wrap-around hexagonal cells as shown in Figure 4. Six cells of the first tier and 12 cells of the second tier surround the central cell. A Node-B (UMTS term referring to a base station) is located at the center of each cell. The adopted propagation model is a modified Hata model specified in [10].

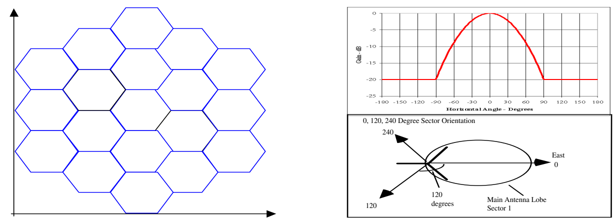

Fig. 4. Cell layout and adopted antenna pattern

The shadowing modeling adopted from [10] is a time correlated process with mean value of 0 dB and standard deviation of 8.9dB. There is also correlation between cells and base stations. Each multi-path component modeled in the simulator has its own fast fading process based on the Jakes model. Channel models and assignment probability used in the simulations are given in Table 1.

TABLE 1 CHANNEL MODEL

Channel Model	**Multi-path Model**	**#of Fingers**	**Speed (km/h)**	**Assignment Probability**
Model A	Pedestrian A	1	3	0.30
Model B	Pedestrian B	5	10	0.30
Model C	Vehicular A	4	30	0.20
Model D	Pedestrian A	1	120	0.10
Model E	Single path	1	f_D=1.5 Hz	0.10

The combined signal-to-interference ratio of each UE at the output of the pilot-weighted combiner at each time slot is given by (3)

$$(C/I)_{combined} = \frac{\left(\sum_{i=1}^{J}\|\gamma_i\|^2\right)^2}{\sum_{j=1}^{J}\|\gamma_j\|^2\left(G^{-1}+\|\lambda\|^2+\sum_{1\le k\le J,k\ne j}\|\gamma_k\|^2\right)} \quad (3)$$

In the equation, γ_i denotes the sample of the fading process of the i^{th} recovered ray, λ the fading process of the un-recovered power, and G the geometry, which is expressed as:

$$G=\frac{\hat{I}_{or}}{N_0+\sum_{n=1}^{N} I_{oc}(n)\|\rho_n\|^2} \quad (4)$$

Where N is the number of interfering cells, ρ_n the fading process, N_0 the variance of the thermal noise, $\hat{I}_{or}$ the total energy per chip averaged over fading and received from the

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serving cell, and $I_{oc}(n)$ the total energy per chip averaged over fading and received from the n^{th} interfering sector.

An Es/Nt vs. block error rate (BLER) curve for the AWGN channel was created *a priori* using the separate link level simulation for each Modulation and Coding Set (MCS). To map Es/Nt to BLER, a quasi-static approach with fudge factors [11] is employed.

In each TTI, the MAC scheduler determines the UEs to which packets are sent and what MCS and power level are allocated. The packet size is determined by the selected MCS. For MAC scheduling, a proportional fairness (PF) scheduling algorithm is used. The PF scheduler maintains an average rate R_{ave} transmitted to each UE over a fixed number of TTI and calculates the ratio of the instantaneous to the average channel condition (R(t)) experienced by each UE. It then sets the priority based on the ratio $R(t)/R_{ave}$. Fairness among UEs is maintained since a UE that has not been scheduled for a period of time would have a lower throughput than a UE that has just been recently scheduled for transmission. So, the first UE would be assigned a higher priority for scheduling. The detailed MAC layer protocol model is based on the 3GPP standard specification.

Other system level parameters used in the simulation are given in Table 2.

TABLE 2. SYSTEM LEVEL PARAMETERS

Parameter	Explanation/Assumption
Cellular layout	Hexagonal grid, 3-cell/Node-B, total 19 Node-B, clover leaf
Site to Site distance	2800 m
Antenna pattern	As proposed in [10]
Propagation model	L = 128.1 + 37.6 Log10R
CPICH power	-10 dB
Other common channels	-10 dB
Std. Deviation of slow fading	8.0 dB
Correlation between sectors	1.0
Correlation between sites	0.5
Carrier frequency	2000 MHz
BS antenna gain	14 dB
UE antenna gain	0 dBi
UE noise figure	9 dB
BS Total Power	20w
Fast HARQ scheme	IR combining
BS total Tx power	42.3 dBm
UE receiver type	Rake
PF scheduler Tc	500

B. *Upper Layer Modeling*

a) RLC Layer:

Radio Link Control (RLC) [13] is a layer 2 protocol in UMTS to provide segmentation and retransmission services for both user and control data. There are three RLC modes: Transparent Mode (TM), Unacknowledged Mode (UM), and Acknowledged Mode (AM). TM is mainly used for streaming type of service where no segmentation is needed. In the UM mode, there is no retransmission, and data delivery is not guaranteed, but segmentation and concatenation are provided by means of the length indicator in RLC PDU. In the simulation, we use UM for VoIP traffic. The ARQ mechanism is used only in the AM mode to provision the in-sequence guaranteed packet delivery to the peer entity. We will use the AM mode for the TCP protocol studied in the simulation. Details of RLC parameters are given in Table 3 and Table 4.

TABLE 3. PARAMETER FOR FTP TRAFFIC

Parameter	Explanation/Assumption
RLC Mode	AM
RLC PDU Size	40 bytes
SDU Discard function	Timer based discard
TCP Segment Size	1500 bytes

b) Packet Data Convergence Protocol (PDCP) Layer

The function of the PDCP [14] layer is to receive PDCP SDU from the Non-Access Stratum (NAS) and forwards it to the RLC layer and vice versa. During this process, header compression/decompression (TCP/IP and RTP/UDP/IP headers) may be configured. PDCP parameters are listed in Table 4.

TABLE 4. PARAMETERS FOR VOIP TRAFFIC

Parameter	Explanation/Assumption
RLC Mode	UM
RLC PDU Size	25 bytes
RLC UM header	1 byte
AMR rate	7.95kbps
ROHC compression	UDP/IP to 3 bytes

c) TCP/UDP Layer & Application Layer

Two types of traffic model are used in the simulation. The first one is FTP with infinite file size, which is running on top of the TCP protocol evaluated. The second one, which is used to model contending traffic for TCP, is VoIP. A two state Markov model with 50% activity is used to model the Adaptive Multi-Rate (AMR) speech coder VoIP service. TCP and UDP modules in the simulation are derived from the NS-2 [15] network simulator.

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V. Simulation Results

The simulation is conducted in the following way: In each trial, one TCP UE, together with *n* (*n*=9,19,29) UDP UE, are uniformly dropped into the 19-hexagon coverage area. A large number of trials are conducted, and the collected statistics are averaged. We focus on the goodput, which is usually the most important indicator of network performance and defines the effective amount of data bit delivered through the network. The goodput results from the simulations are showed in Figures 5 and 6.

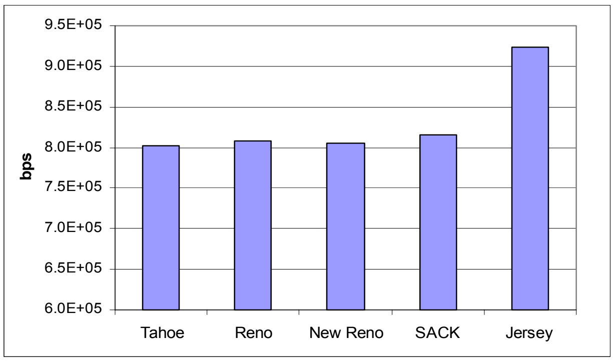

Fig. 5. Average TCP Goodput (with 9 UDP user/ cell)

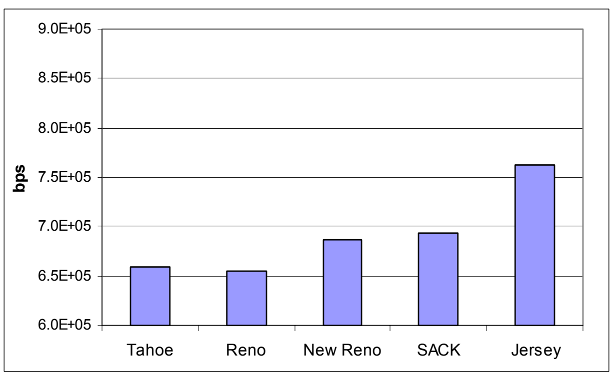

Fig. 6. Average TCP Goodput (with 19 UDP user/ cell)

From the simulation results, we can see that TCP Jersey has a significant improvement over other TCP variants. Two salient features of TCP-Jersey contribute to this improvement. First, CW enables TCP-Jersey to differentiate wireless packet errors from network congestion. When TCP sender receives a DUPACK without the CW mark, it assumes with high confidence that a link error occurred. In the simulation, it is found false detection rate is less than 1%. Secondly, it is well known that a cellular system's throughput is maximized when the radio resources are allocated according to the mobile's changing channel condition. Simulations show TCP-jersey works well with the MAC protocol. In fact, ABE helps the TCP sender more accurately estimate the changing bandwidth, thus better coordinating with the MAC layer resource scheduler.

However, the improvement shown is not as high as shown in [3], possibly for two reasons. Firstly, with the RLC AM and MAC layer's ARQ mechanisms, link errors encountered are partly hidden from the upper layer. Secondly, UE goodput is influenced by a variety of parameters, such as geometry distribution, total number of UEs being served, MAC scheduling algorithms, and the air interface capabilities.

Note that TCP-SACK showed limited improvement (excluding TCP-Jersey) even in the scenario when there are not many serving UEs (10).

VI. Conclusions

In this paper, we have evaluated the performance of TCP-Jersey over High Speed Downlink Packet Access (HSDPA) through intensive simulations. The simulation results have demonstrated that TCP-Jersey exhibits a consistent improvement over traditional TCP variants.

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