Improving TCP Performance in Buffer-bloated Cellular Networks via Dynamic Receive Window Adjustment

ABSTRACT

As smart phones and tablets become more and more popular, TCP performance over cellular networks is of growing importance for today's Internet. However, our extensive measurements over 3G/4G networks of four major U.S. carriers and the largest carrier in Korea reveal that TCP may suffer from extremely long delay and sub-optimal throughput. From the measurements, an important problem of TCP in cellular networks is observed: the current cellular networks are over-buffered and the huge buffers nullify loss-based TCP congestion control, resulting in excessive growth of congestion window. To mitigate the problem, smart phone vendors rely on an ad-hoc solution that sets the maximum receive buffer size to a relatively small constant value than its actual size. Although this simple scheme alleviates the problem, it is sub-optimal in a number of scenarios due to its *static* nature. In this paper, we propose *dynamic* receive window adjustment (DRWA) and implement it in Android phones. DRWA only requires modifications on smart phones and is immediately deployable. Our extensive real-world tests confirm that DRWA reduces the average delay of TCP by 24.09 ∼ 48.97% in general scenarios and achieves up to 51.06% throughput improvement in specific scenarios.

Categories and Subject Descriptors

C.2.2 [**Computer-Communication Networks**]: Network Protocols

General Terms

Design, Measurement, Performance

Keywords

TCP, Cellular networks, Bufferbloat, Receive window adjustment

1. INTRODUCTION

TCP is the dominant transport layer protocol of the current Internet, carrying around 90% of the total traffic [13,18]. Hence, the performance of TCP is of utmost importance to the well-being of the Internet and has direct impacts on user experience. Although TCP is well-studied in traditional networks, its performance over cellular networks has not been given adequate attention.

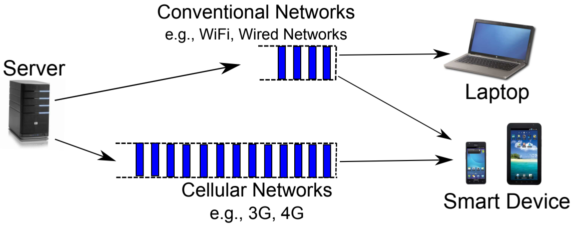


Figure 1: Over-buffering has been widely observed in the current Internet [9] but is especially severe in cellular networks, resulting in up to several seconds of end-to-end delay.

According to our measurements, TCP has a number of performance issues in this relatively new environment, including extremely long delay and sub-optimal throughput in certain scenarios. The reasons behind such performance degradations are two-fold. First, most of the widely deployed TCP implementations use loss-based congestion control where the sender will not slow down its sending rate until it sees packet loss. Second, most cellular networks are over-buffered to accommodate bursty traffic and channel variability [9, 15] as depicted in Figure 1. The huge buffer along with link layer retransmission conceals packet losses in cellular networks from TCP senders. The combination of these two facts leads to the following phenomenon: the TCP sender keeps increasing its sending rate to probe the available bandwidth along the path. Even if it has already reached the bottleneck link capacity, the congestion window will continue to grow since all the overshot packets are absorbed by the buffers and all the packets corrupted over the wireless links are recovered by link layer retransmission. This results in a long queue in the cellular network system and up to several seconds of end-to-end delay.

To solve this problem, smart phone vendors come up with a small trick: they set a relatively small value for TCP maximum receive buffer size although the physical buffer size of a smart phone is much larger. Since the advertised receive window cannot exceed the receive buffer size and the sender cannot send more than what is allowed by the advertised receive window, this limit effectively prevents TCP congestion window from excessive growth and controls the RTT (round trip time) experienced by the flow in a reasonable range. However, since the limit is statically configured, it is sub-optimal in many scenarios, especially considering the dynamic nature of the wireless mobile environments. In high speed long distance networks (e.g., downloading from an oversea server over 4G LTE (Long Term Evolution) network), the static value is too small to saturate the link and results in severe throughput degradation. In small bandwidth-delay product (BDP) networks, the static value is too large and the flows in a smart phone experience excessively long RTT.

In this paper, we propose a practical, receiver-based remedy to this issue called dynamic receive window adjustment (DRWA) . DRWA runs only on the receiver side (i.e., smart phone) and continuously adjusts the receiver window to be proper instead of reporting a static value. DRWA aims to keep the buffer at the bottleneck link (i.e., cellular link) non-empty while avoiding unnecessarily long queue. We achieve this goal by letting DRWA enforce RTT to stay around $\lambda * RTT_{min}$ where λ is our control parameter. Our extensive experiments over various cellular networks of different carriers reveal that $\lambda = 3$ keeps RTT 24.09 $\sim$ 48.97% lower than the current TCP implementations in smart phones while throughput is guaranteed to be the same in general cases and upto 51.06% higher in a large BDP network.

Our proposal is similar in spirit to delay-based congestion control algorithms but does not require modifications on large-scale servers (i.e., TCP senders). It is fully compatible with existing TCP protocol and can be easily deployed. Carriers or device manufacturers can simply issue an OTA (over the air) update to the smart phones' protocol stack so that these devices could immediately enjoy enhanced performance when interacting with existing servers.

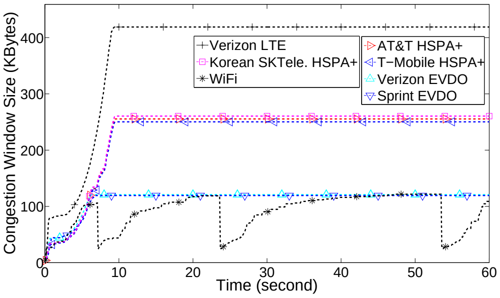
In summary, the key contributions of this paper include:

- We report extensive observations of TCP's behavior in a range of different cellular networks and point out its negative impacts on user experience.
- We anatomize the TCP implementation in state-of-the-art smart phones and locate the root cause of its performance issue in cellular networks.
- We propose a simple and backward-compatible remedy that is experimentally proven to be safe and effective. It provides substantial fixes to TCP performance issues and is immediately deployable.

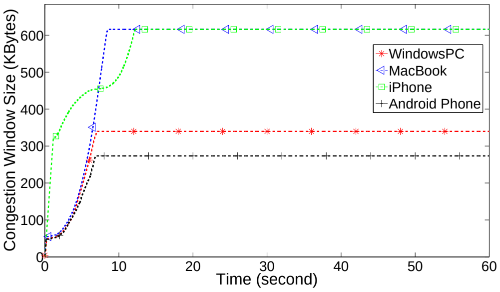
The rest of the paper is organized as follows. In Section 2, we provide detailed observations of TCP's abnormal behavior over various cellular networks. In Section 3, we anatomize the root cause of such behavior and identify potential limitations of current TCP implementation in smart phones. Then, we suggest a simple remedy DRWA in Section 4 and extensively show its experimental performance in comparison with the current implementation in Section 5. We conclude our work in Section 6.

2. OBSERVATIONS

In the past few decades, TCP has been well-studied in traditional networks, especially wired networks. With the exponential growth of hand-held devices like smart phones and tablet computers, TCP performance in cellular networks is becoming more and more important. Unfortunately, TCP performance of smart mobile devices over cellular networks is still lack of deep investigation though there exist a number of measurement work for different types of cellular networks [4, 12, 14, 15]. During our extensive real-world TCP performance measurements over various cellular networks, we found that the current TCP implementation shows abnormal behaviors in cellular networks and has a number of performance issues including extremely long delay and sub-optimal throughput.



(a) Abnormality exists across different cellular networks (tested with Android phones)



(b) Abnormality exists across different platforms (tested over AT&T HSPA+ network)

Figure 2: Abnormal TCP behavior in cellular networks: in this test, clients of different platforms download a large file from a university server over various cellular networks. The TCP congestion window ($cwnd$) is monitored on the server side using Web100 project [16].

Figure 2 depicts the TCP congestion window measured from a Linux server when clients of different platforms (Android phone, iPhone, Windows PC, Macbook) download a large file from it over various cellular networks. The server uses CUBIC [10] as the TCP congestion control algorithm. The cellular networks we have tested include the four major U.S. carriers' 3G networks, Verizon's 4G LTE network and Korean SKTelecom's HSPA+ network. As a reference, we also plot TCP's behavior in WiFi networks under similar conditions.

2.1 TCP Congestion Control Collapse in Cellular Networks

	Samsung Galaxy S2 (AT&T)	HTC EVO Shift (Sprint)	Samsung Droid Charge (Verizon)	LG G2x (T-Mobile)
Wi-Fi	110208	110208	393216	393216
UMTS	110208	393216	196608	110208
EDGE	35040	393216	35040	35040
GPRS	11680	393216	11680	11680
HSPA+	262144	N/A	N/A	262144
WiMAX	N/A	524288	N/A	N/A
LTE	N/A	N/A	484848	N/A
Default	110208	110208	484848	110208

Table 1: Maximum TCP receive buffer size (*tcp_rmem_max*) in bytes on different Android phones for different carriers. Note that these values may vary on customized ROMs and can be looked up by looking for "setprop net.tcp.buffersize.*" in the init.rc file of Android phones. Also note that different values are set for different carriers even if the network types are the same. We guess that these values are experimentally determined based on each carrier's network conditions and configurations.

As shown in Figure 2, TCP congestion window over cellular networks does not show its conventional sawtooth behavior while WiFi networks show that behavior clearly. The TCP congestion window over cellular networks grows to a static value and stays there until the session ends. Through our extensive testing, this strange behavior, which we call "TCP congestion control collapse", turns out to be *universal* in cellular networks. It exists in the 3G networks of all four major cellular network carriers in the US. It also exists in 4G LTE networks. We further confirmed that the same abnormal TCP behavior is observed in the largest cellular network carrier in Korea. Moreover, this problem is observed across Android phones, iPhones as well as computers running Windows 7 or Mac OS X Lion. This abnormal phenomenon caught our attention and revealed an *untold story of TCP over cellular networks*. We became interested in the reasons behind this behavior and its impact on user experience.

2.2 Understanding the Problem

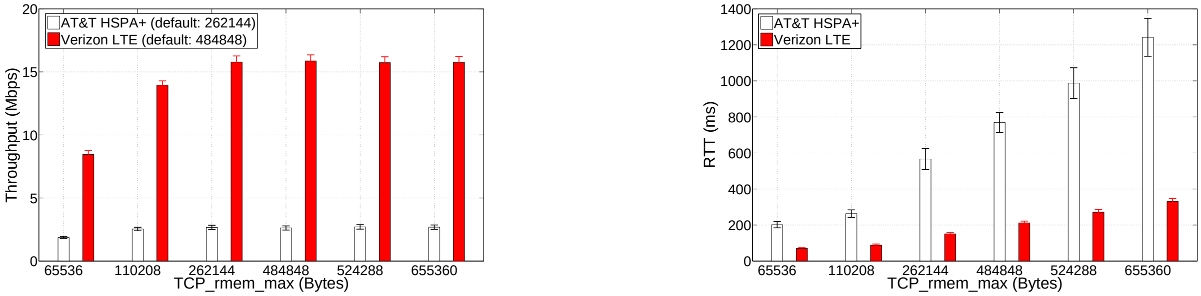
To understand the behavior of TCP over cellular networks, we focused on why the congestion window stays at a static value. The static congestion window first indicates that no packet loss is observed by the TCP sender (otherwise the congestion window should have decreased multiplicatively at any loss event). In a large scale real world measurement [12], it is shown that cellular networks typically experience packet loss rates close to zero. This no-packet-loss phenomenon can be explained by the large buffers existent in cellular base stations or middleboxes as well as the link layer retransmission mechanisms prevalent in cellular networks: most of the current 3G networks as well as the forthcoming 4G networks are over-buffered (or buffer-bloated as termed by [9]). These excessive buffers were originally introduced into cellular networks due to a number of reasons. First, the channel status of cellular links fluctuates quickly and the corresponding channel rate varies from dozens of Kbps to tens of Mbps. Second, the data traffic over such links is highly bursty. To absorb such bursty traffic over such a variable channel, the simple yet effective approach adopted by current cellular networks is to provide large buffers. These buffers smooth the bursty traffic and reduce the packet loss rate in cellular networks. Further, due to the relatively high bit error rate over the wireless channel, link layer retransmission is typically performed in cellular networks, which also requires large buffers in the routers or base stations to store the unacknowledged packets.

Providing large buffers seems to be a viable solution at Layer 2, but it has an undesirable interaction with the TCP congestion control at Layer 4. TCP mostly relies on packet loss to detect network congestion. Although other variants such as delay-based congestion control exist, most of the widely deployed TCP implementations (e.g., Newreno [7], BIC [21], CUBIC [10]) still use loss-based congestion control [22]. Excessive buffers in cellular networks prevent packet losses from happening even if TCP's sending rate far exceeds the bottleneck link capacity. This "hides" the network congestion from the TCP sender and makes its congestion control algorithm malfunction.

If packet losses are perfectly concealed, the congestion window may not drop but it should persistently grow up. However, it strangely stops at a certain value and this static value is different for each cellular network or client platform. Our deep inspection to the TCP implementation in Android phones (since it is open-source) reveals that the value is determined by a parameter *tcp_rmem_max* that specifies the maximum receive window advertised by an Android phone. This gives an intuitive answer why the congestion window shows the flat behavior: the receive window ($rwnd$) advertised by the receiver crops the congestion windows ($cwnd$) in the sender. By inspecting various Android phone models, we found that *tcp_rmem_max* has diverse values for different types of networks as shown in Table 1. Generally speaking, larger values are assigned to faster communication standards (e.g., LTE).

To understand the impact of *tcp_rmem_max*, we show the TCP performance under various *tcp_rmem_max* settings for Verizon's LTE and AT&T's HSPA+ networks in Figure 3. Obviously, a larger *tcp_rmem_max* value allows the congestion window of the TCP sender to grow to a larger size and hence leads to higher throughput. But this throughput improvement will flatten out once the link capacity is saturated. Further increase of *tcp_rmem_max* brings nothing but longer queuing delay.

As the figure shows, the current *tcp_rmem_max* values for LTE and HSPA+ networks are carefully chosen to balance



(a) Throughput Performance

(b) RTT Performance

Figure 3: Throughput and RTT performance measured for a whole day when downloading from a local server over LTE and HSPA+ networks with various *tcp_rmem_max* values. For this test environment, 110208 may work better than the default 262144 in AT&T HSPA+ network. Similarly, 262144 may work better than the default 484848 in Verizon LTE network. However, the optimal value depends on the environment and is hard to set statically in advance.

between throughput and RTT in typical scenarios. However, despite the careful choice of the parameter values, the static nature of such parameter setting is doomed to be sub-optimal in certain cases. For instance, in the testing of Figure 3, the default value set by Verizon for its LTE network and AT&T for its HSPA+ network do not provide the best throughput and end-to-end latency. The downlink traffic is from a local university server in U.S. In this case, the end-to-end latency is relatively small and the resulted pipe size (i.e., BDP) is small. This default values for both LTE and HSPA+ are large enough to achieve full bandwidth utilization as shown in Figure 3(a). However, the default value triggers excessive packets in network and thus resulting in unnecessarily larger RTT as shown in Figure 3(b). This demonstrates the fundamental limitations of the static parameter setting: it mandates one specific trade-off point in the system which may be sub-optimal for other applications. Two realistic scenarios where the current implementation may hurt user experiences are discussed below.

2.3 Impact on User Experience

Web Browsing with Background Downloads: Top lines of smart phones scheduled to be launched during 2012 are mostly with a quad core CPU of about 1.5GHz per core, more than 1GB RAM, and a high resolution screen (e.g., 1280×720 pixels). Due to their drastically improved capability, the phones are expected to perform multi-tasking more often. For instances, people will enjoy web browsing or online gaming while downloading files such as books, musics, movies or applications from on-line markets in the background. In such cases, we found that the current TCP implementation incurs long delays for the interactive flow (Web browsing or online gaming) since the buffer is filled with packets belonging to the background download flow.

Figure 4 shows that the Web object fetching time are severely degraded when background downloads are under way. Since Web objects are typically small in size (for instance, we use 8KB, 16KB, 32KB and 64KB in this test), their fetching time mainly depends on RTT rather than throughput. When a background flow causes long queues to be built up at the base station, the Web objects will be severely delayed. As the figure shows, average Web object fetching time is 2.6 times longer with background download[1].

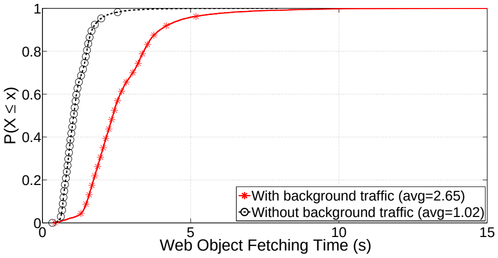


Figure 4: Web object fetching performance with and without background traffic. The time taken to fetch the same web objects becomes 2.6 times longer if a file download coexists. If TCP maintains a smaller queue in the network, the drastic performance degradation can be mitigated.

Throughput from Servers with Long Latency: The contents that smart phone users visit are diverse. Some contents are well maintained and CDNs (content delivery networks) are assisting them to get "closer" to their customers via replication. In such cases, the throughput performance can be well-supported by the static setting of *tcp_rmem_max*. However, there are still many websites or files showing long latencies due to their remote locations, such as the most common usage of smart phones: web browsing, market application download and streaming. In such cases, the static setting of *tcp_rmem_max* (which is tuned for moderate latency case) fails to provide the maximum possible throughput since it cannot fill the long fat pipe. Figure 5 shows that when downloading some contents from a server abroad, the client suffers from sub-optimal throughput performance under the default setting. A larger *tcp_rmem_max* can achieve

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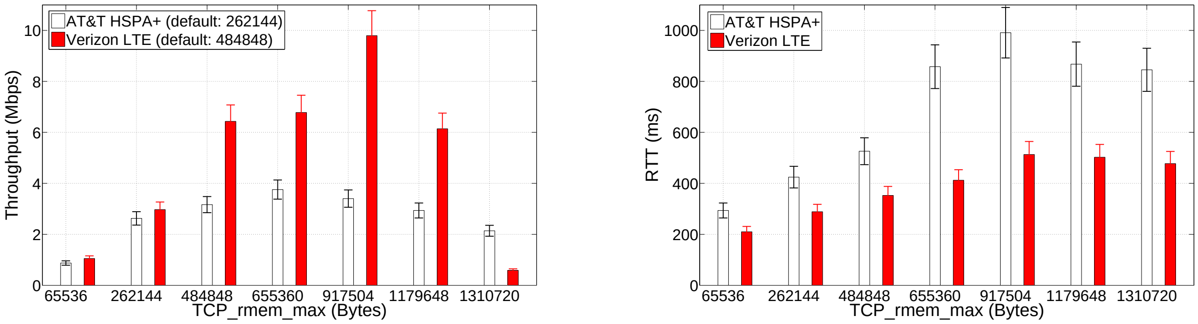
[1] Note that the fetching times of multiple small objects in parallel do not affect each other since the bandwidth is not saturated.



(a) Throughput Performance

(b) RTT Performance

Figure 5: Throughput and RTT performance measured for a whole day when downloading from a remote server in Korea via LTE and HSPA+ networks with various *tcp_rmem_max* values. The static setting results in sub-optimal throughput performance since it fails to probe maximum available bandwidth of the long fat pipe. 655360 for AT&T and 917504 for Verizon provided much higher throughput than their default values.

higher throughput, but if it is too large, packet loss will eventually happen in which case throughput degradation will occur as well.

3. ANATOMY OF TCP BEHAVIOR IN CELLULAR NETWORKS

In this section, we elaborate the details of the current TCP implementation in Android phones, especially the receive window adjustment algorithm and investigate the potential limitations of the current implementation. Inspired by the limitations, we suggest several possible solutions and discuss their advantages and disadvantages.

3.1 Details of Current Implementation

We use Android phones to look into the implementation detail of the receive window adjustment algorithm due to the closed nature of other platforms (e.g., iPhone). We are unable to pinpoint the exact implementations of such closed platforms, but testing results shown earlier suggest that they exhibit similar behaviors. This observation imply that the underlying implementation may be very similar even in the closed platforms.

TCP receive window was originally designed to prevent a fast sender from overwhelming a slow receiver with limited buffer space. It reflects the available buffer size on the receiver side so that the sender will not send more packets than the receiver can accommodate. The combination of this flow control and TCP congestion control ensures that neither the receiver nor any intermediate router along the path will be overloaded.

With the advancement in storage technology, memories are becoming cheaper and cheaper. Nowadays, it is not uncommon to find a computer equipped with several gigabytes of memory and even smart phones are now equipped with 1GB of RAM (e.g. Motorola Droid Razr, Samsung Galaxy S2). Hence, buffer space on the receiver side is hardly the bottleneck in the current Internet. To improve TCP throughput, a receive buffer auto-tuning technique called Dynamic Right-Sizing (DRS [6]) was proposed and many operating systems (including Linux and hence Android) have adopted the same or similar scheme in their kernels. In DRS, instead of determining the receive window based on the available buffer size, the receive buffer is dynamically adjusted so as to suit the connection's demand. The fundamental goal of DRS is to allocate enough buffer (as long as we can afford it) so that the throughput of the TCP connection is never limited by the receive window size but only constrained by network congestion. Meanwhile, DRS tries to avoid allocating more buffers than necessary.

Algorithm 1 DRS

```
Initialization:
$rwnd \leftarrow 0$;

RTT Estimation:
$RTT_{est} \leftarrow$ the time between when a byte is first acknowledged and the receipt of data that is at least one window beyond the sequence number that was acknowledged;

Dynamic Right-Sizing:
if data is copied to user space then
    if $elapsed\_time < RTT_{est}$ then
        return;
    end if

    $cwnd_{est} \leftarrow data\_rcvd$;
    $rwnd \leftarrow max\{2 * cwnd_{est}, rwnd\}$;
    Advertise $rwnd$ as the receive window size;
end if
```

Algorithm 2 DRS with *tcp_rmem_max* clamping

```
Same as Algorithm 1 except adding:
$rwnd \leftarrow min\{tcp\_rmem\_max, rwnd\}$;
```

Algorithm 1 gives the details of DRS. The first step in DRS is to measure the RTT of the TCP connection at the receiver. By definition, RTT is the time between when data is sent and the corresponding acknowledgement is received. However, since the receiver typically has no data to send, it is hard to measure RTT on the receiver side. DRS emulates RTT by monitoring the time between when a packet is first acknowledged and the receipt of data that is at least one window beyond the sequence number that was acknowledged. Although there is some deviation, this RTT measurement is proved to be good enough for the purpose as demonstrated by its wide deployment in the Internet.

Once the RTT is known, the current congestion window of the sender can be trivially estimated on the receiver side by counting the amount of data received within one RTT. Since TCP congestion window can at most double within an RTT (e.g., during slow start), DRS set the advertised receive window to be twice of the estimated congestion window so that the TCP sender is always congestion-window-limited rather than receive-window-limited. Further, since the TCP sender may be application-limited and have not fully used the congestion window, the data received in an RTT may be smaller than the actual window size. DRS therefore uses the largest received window advertised during any RTT by the max operation in Algorithm 1. Note that this makes adjustment of the receive window in DRS *non-decreasing*. It is acceptable because the sole goal of DRS is to set the congestion window "free" from the constraints of the receive window. It does not matter if the advertised receive window is a bit too large. Finally, the receive window is of course bound by the maximum receive buffer size.

Linux adopted a receive buffer auto-tuning scheme similar to DRS since kernel 2.4.27. Since Android is based on Linux, it inherits the same receive window adjustment algorithm. Other major operating systems also implemented certain kind of TCP buffer auto-tuning (Windows since Vista, Mac OS X since 10.5, FreeBSD since 7.0). This implies a significant role change of TCP receive window. Although the functionality of flow control is still preserved, most of the time the receive window is set to a value that lets TCP congestion control fully explore the available bandwidth while preventing the receiver from allocating unnecessarily large buffers to a connection.

Although DRS works fairly well in traditional networks and improves TCP throughput to a great extent, it actually incurs a *vicious cycle* in cellular networks. When the congestion window of the TCP sender increases, more packets are clocked out within an RTT and the receiver's estimation of the congestion window also increases. Since DRS requires the receive window to keep in pace with the growth of the congestion window, the advertised window will be increased, leaving more space for the congestion window to grow. This ever increasing trend will lead to long queues being built up at the routers or base stations and result in extremely long RTT. The current solution adopted by many Android smart phone vendors to break this vicious cycle is to set a relatively small maximum receive buffer size via *tcp_rmem_max* in Algorithm 2. This static size is strategically configured for each carrier's different cellular networks (GPRS, HSPA+, EVDO, LTE, etc.) so that the performance is tolerable in ordinary scenarios. As we know, the cellular networks are highly dynamic due to the nature of wireless channel as well as user mobility and diversity. The static setting will lead to sub-optimal performance in numerous scenarios which we will detail in the following section.

3.2 Limitations of Current Implementation

Long End-to-End Latency: On the other hand, in many cases where the static value of *tcp_rmem_max* may overestimate the pipe size, the RTT will be unnecessarily long though the bandwidth can be fully utilized. The excessive packets make the network more congested and can severely degrade the performance of short-lived flows such as web browsing or online gaming because their performance are more dependent on RTT than throughput.

Throughput Degradation: RTT in cellular networks is relatively long (up to several seconds) because of long distance transmission (remote server), link layer latency (retransmission and contention at L2) and congestion within the network (queuing delay along paths). Since the maximum throughput of a TCP flow using Algorithm 2 becomes roughly *tcp_rmem_max* divided by RTT, the static setting of *tcp_rmem_max* may result in bandwidth under-utilization if RTT become larger than certain limit.

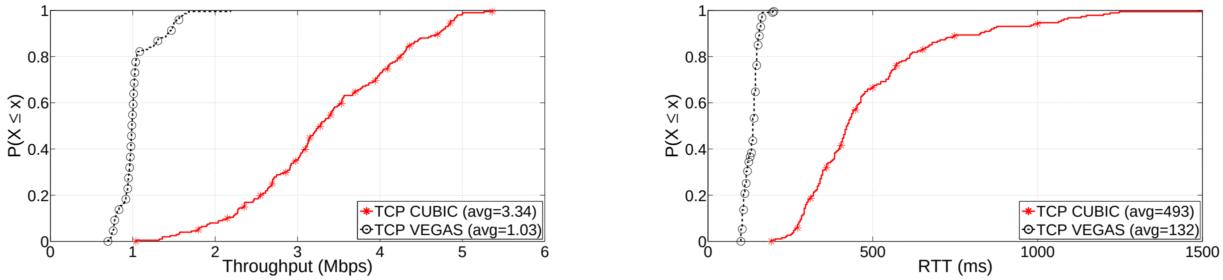
Expensive Recovery from Packet Loss: Although packet loss is uncommon in cellular networks, once it happens the recovery may be very expensive. In particular, we found that in AT&T's HSPA+ network the sender almost always resort to the expensive retransmission time-out (RTO) rather than *fast retransmit* to recover from packet losses. According to [19], the reason may be the middleboxes hold all out-of-order packets until everything is in order so that it can conduct some deep packet inspection and then forward them to the downstream.

3.3 Candidate Solutions and Our Approach

To address TCP's problem in buffer-bloated cellular networks, there are a few possible solutions. One obvious solution is to reduce the buffer size in cellular networks so that TCP can function the same way as it does in wired networks. However, as explained earlier these extra buffers are essential to ensure the performance of cellular links under dynamic conditions and cannot be easily removed.

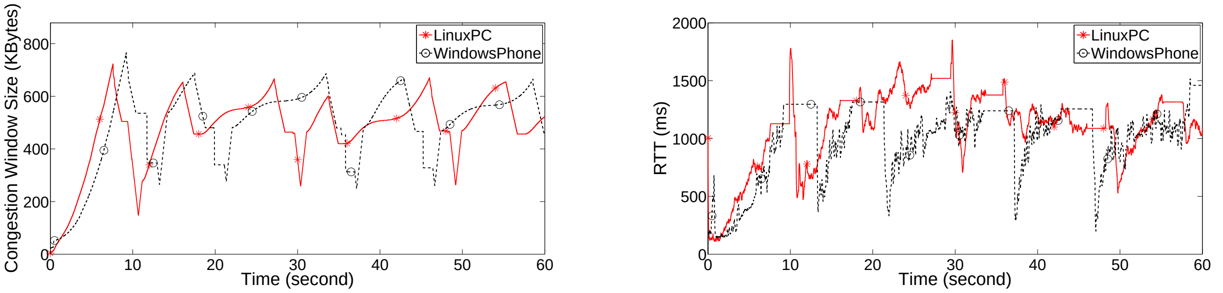
An alternative to this solution is to employ certain Active Queue Management (AQM) schemes like RED [8] or REM [2]. By randomly dropping or marking certain packets before the buffer is full, we can notify TCP sender in advance and avoid the excessively long delay. However, despite being studied extensively in the literature, few AQM schemes are actually deployed in the Internet due to the complexity of their parameter tuning, the extra packet losses introduced by them and the limited performance gains provided by them.

Another possible solution to this problem is the modification of the TCP congestion control algorithm at the sender. Instead of a loss-based approach, delay-based congestion control such as TCP Vegas [3] or FAST TCP [20] can be used. Since delay-based congestion control backs off when



(a) Throughput (b) RTT

Figure 6: Throughput and RTT performance of TCP Vegas in cellular networks: although delay-based congestion control reduces the RTT, it suffers from throughput degradation.



(a) CWND (b) RTT

Figure 7: TCP behaviors of a Linux PC and a Windows phone in cellular networks: They set *tcp_rmem_max* to a very large value. Their TCP exhibits the usual saw-tooth behavior, but this results in excessively long RTT in bufferbloated cellular networks.

RTT starts to increase rather than waiting until packet loss happens, they may serve the over-buffered cellular networks better than loss-based congestion control. However, as Figure 6 shows, although delay-based TCP congestion control decreases RTT to a great extent, they suffer from throughput degradation. This agrees with the observation over a cellular network in [15]. Further, adopting delay-based congestion control requires modifications on the sender side (typically large-scale servers) which may incur considerable deployment cost and affect both wired and wireless users.

In light of the problems with the above-mentioned solutions, we suggest to handle the problem on the receiver side by changing the static setting of *tcp_rmem_max*. That is because receiver (mobile device) side modification has minimum deployment cost. Vendors may simply issue an OTA update to the protocol stack of the mobile devices so that they can enjoy a better TCP performance without affecting other wired users. It is a light-weight, effective and immediately deployable solution to the problem.

To change the static limit imposed on the sender's congestion window, one may simply remove the *tcp_rmem_max* parameter or set it to a very large value so that congestion window is free to grow (and drop). This will bring back the sawtooth behavior of TCP but will incur extremely long RTT due to the bufferbloat in cellular networks. As shown in Figure 7, Windows Phone (Samsung Focus) and Linux PC work in this way. Unfortunately, this type of solution leads to even longer RTT and potential throughput degradation. Therefore, we need a dynamic receive window adjustment (DRWA) algorithm to ensure full utilization of the available bandwidth while maintaining RTT small.

4. DYNAMIC RECEIVE WINDOW ADJUSTMENT (DRWA)

4.1 Algorithm

The aim of DRWA is to adaptively set the receive window to a proper size in different environment. Sometimes, it should be larger than the current static limit to get more throughput and sometimes it should become smaller than the current value to avoid unnecessary queues in the link. The challenges in this work lie in three parts. First, DRWA should remove the static setting of a relatively small maximum receive buffer size. Second, DRWA should bear the capability to estimate the proper pipe size of a link via a new window adjustment algorithm. Finally, DRWA should be compatible with the current TCP protocol and easy to deploy.

DRWA is built on top of DRS. Instead of an unidirectional adjustment where the advertised window is non-decreasing, we need a bidirectional adjustment algorithm to rein TCP in the buffer-bloated cellular networks but at the same time ensure full utilization of the link. To accomplish that, DRWA

Algorithm 3 DRWA

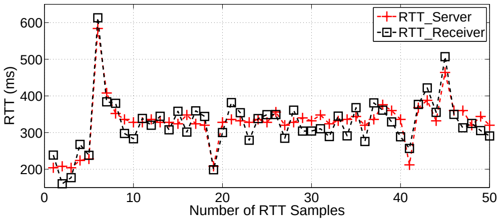
```
Initialization:
$RTT_{min} \leftarrow \infty$;
$cwnd_{est} \leftarrow data\_rcvd$ in the first $RTT_{est}$;
$rwnd \leftarrow 0$;

RTT and Minimum RTT Estimation:
$RTT_{est} \leftarrow$ the time between when a byte is first acknowledged and the receipt of data that is at least one window beyond the sequence number that was acknowledged;

if TCP timestamp option is available then
    $RTT_{est} \leftarrow$ averaging the RTT samples obtained from the timestamps within the last RTT;
end if

if $RTT_{est} < RTT_{min}$ then
    $RTT_{min} \leftarrow RTT_{est}$;
end if

DRWA:
if data is copied to user space then
    if $elapsed\_time < RTT_{est}$ then
        return;
    end if

    $cwnd_{est} \leftarrow \alpha * cwnd_{est} + (1 - \alpha) * data\_rcvd$;
    $rwnd \leftarrow \lambda * \frac{RTT_{min}}{RTT_{est}} * cwnd_{est}$;
    Advertise $rwnd$ as the receive window size;
end if
```

needs to keep the queue size proper as necessary but non-empty always. Algorithm 3 gives the details.

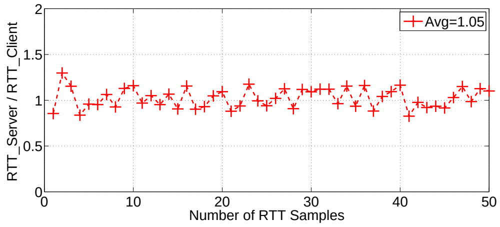
DRWA uses the same technique as DRS to measure RTT on the receiver side if TCP timestamp option is unavailable. However if TCP timestamp option is available, DRWA uses it to obtain a more accurate estimation of the RTT (Timestamps can provide multiple RTT samples within an RTT whereas the traditional DRS way provides only one sample per RTT). We surveyed the support for TCP timestamp option in Windows Server and Linux (Table 2) and found that when DRWA runs on Android phones, it could turn on timestamp no matter it talks to a Linux server or a Windows server. With assistance of timestamps, DRWA is able to achieve robust RTT measurement on receiver side and thus conquering the well-known battle of accurately measuring RTT in dynamic networks, as shown in Figure 8. In addition to RTT measurement, DRWA also records the minimum RTT ever seen in this connection and use it later to determine the receive window size. Since the minimum RTT approximates the round-trip propagation delay between the two hosts when no queue is built up in the intermediate routers especially in the cellular base station, we use it as an indication on what the network and channel conditions are.

Server	Client	Timestamp Option
Linux	Linux (Android)	Enabled
Windows	Linux (Android)	Enabled
Linux	Windows	Disabled
Windows	Windows	Disabled

Table 2: Support for TCP timestamp option on Linux and Windows: both Linux and Windows support TCP timestamp and Linux clients turn it on by default when initiating a connection while Windows clients do not. Therefore, Android phones can use timestamp no matter they talk to Linux or Windows servers.



(a) RTT Measurement

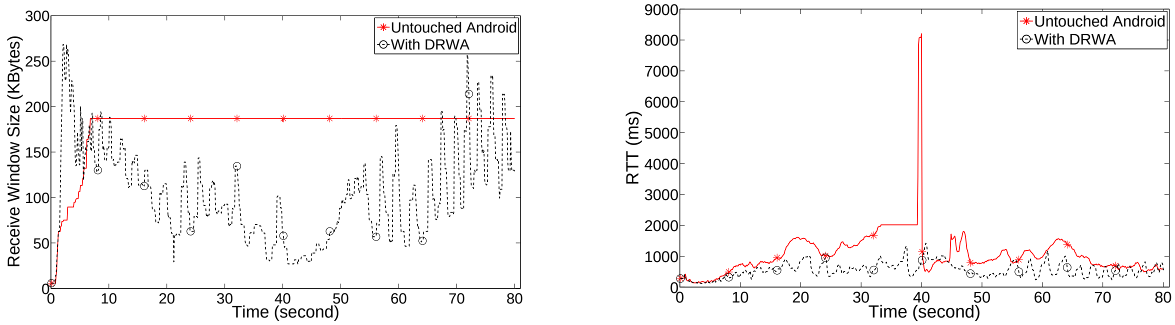


(b) Ratio between RTT measurements

Figure 8: With timestamp option, DRWA is able to achieve robust RTT measurement on the client side. The testing was conducted over AT&T HSPA+ network by using Samsung Galaxy S2 phone. We print the RTT values to kernel message on the client side and use Web100 to monitor RTT value on server side. The two RTT measurements are consistent though there exists minor deviation.

After knowing the RTT, DRWA counts the amount of data received within one RTT in the same way as DRS. However, DRWA further smooths the estimated congestion window by using a moving average with a low-pass filter (α is set to 7/8 in our current implementation). This smoothed value is used to determine the receive window we advertise. In contrast to DRS who always sets $rwnd$ to $2 * cwnd_{est}$, DRWA sets it to $\lambda * \frac{RTT_{min}}{RTT_{est}} * cwnd_{est}$. When RTT_{est} is close to RTT_{min}, implying the network is not congested, $rwnd$ will increase quickly to give the sender enough space to probe the available bandwidth. As RTT_{est} increase, we gradually slow down the increment rate of $rwnd$ to stop TCP from overshooting. The operation of taking the maximum of the newly calculated $rwnd$ and the previous $rwnd$ in DRS is also removed so that DRWA makes bidirectional adjustment of the advertised window and controls the RTT_{est} to stay around $\lambda * RTT_{min}$. More detailed explanation of λ will be given in the following section.

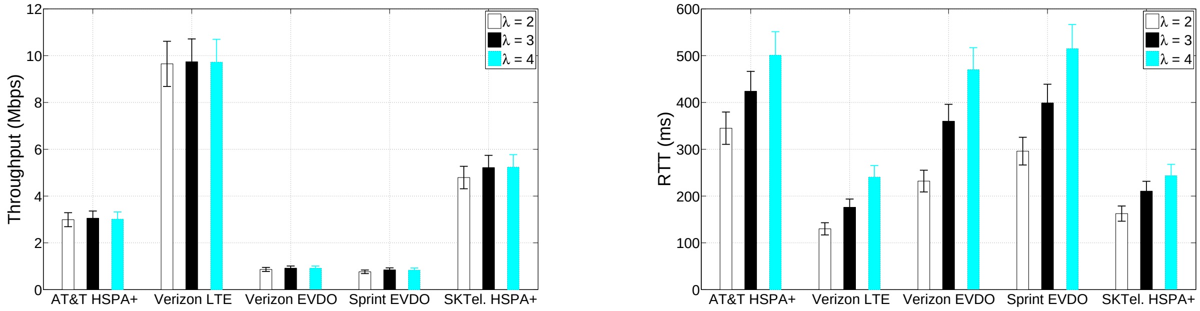
This algorithm is simple yet effective. Its ideas stem from delay-based congestion control algorithms but work better than them for two reasons. First, since DRWA only *guides*



(a) Receive Window Size

(b) RTT Performance

Figure 9: TCP behavior comparison between untouched Android phones and phones with DRWA: in this test the phones are moved from an area with good signal to an area with weak signal and then moved back again. In contrast to the static setting of the receive window in untouched Android phones, DRWA nicely tracks the variation of the channel conditions and dynamically adjusts the receive window. Due to the dynamic adjustment, DRWA is able to keep the RTT constantly low while the untouched Android phone experiences drastic increase in RTT under weak signal.



(a) Throughput Performance

(b) RTT Performance

Figure 10: Impact of λ on the throughput and RTT performance of TCP with DRWA in different cellular networks. $\lambda = 3$ gives a good balance between throughput and RTT in four major U.S carriers as well as the largest Korean carrier.

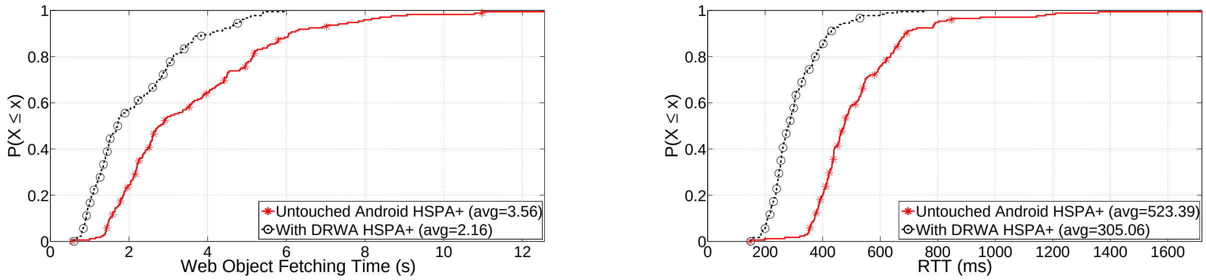
TCP congestion window by advertising an adaptive $rwnd$, the bandwidth probing responsibility still lies with the TCP congestion control algorithm at the sender. Therefore, typical throughput loss seen from using delay-based TCP will not appear. Also, due to some unique characteristics of cellular networks, RTT based control can work more effectively. In wired networks, a router may handle hundreds of TCP flows at the same time and they may share the same output buffer. That makes RTT measurement more noisy and delay-based congestion control less reliable. However, in cellular networks, a base station typically has separate buffer space for each user and a mobile user is unlikely to have many simultaneous TCP connections. This makes RTT measurement a more reliable signal for network congestion.

4.2 Adaptive Nature of DRWA

DRWA allows a TCP receiver to report a proper receive window size to its sender in every RTT rather than advertising a static limit. Due to its adaptive nature, DRWA is able to track the variability of channel conditions. Figure 9 shows the evolution of the receive window size and the corresponding RTT performance. During this test, we moved the Android phone from a good signal area to a weak signal area (from 0 second to 40 second) and then returned back to the good signal area (from 40 second to 80 second). As shown in Figure 9(a), the receive window size dynamically adjusted by DRWA well demonstrates the signal change incurred by the movement. This leads to a steadily low RTT while the static setting of untouched Android results in an ever increasing RTT as the signal strength decreases and the RTT blows up in the area of the weakest signal strength.

4.3 Impact of λ on TCP Performance

λ is a key parameter in DRWA. It tunes the operation region of the algorithm and reflects the trade-off between throughput and delay. Note that when RTT_{est}/RTT_{min} equals to λ, the advertised receive window will be equal to its previous value, leading to a steady state. Therefore, λ reflects the target RTT of DRWA. If we set λ to 1, that means we want RTT to be around RTT_{min} so that almost



(a) Web page fetching time

(b) RTT experienced by the Web browsing TCP flow

Figure 11: Web browsing with background download over AT&T HSPA+: DRWA reduces the RTT experienced by the TCP flows and hence improves Web browsing performance.

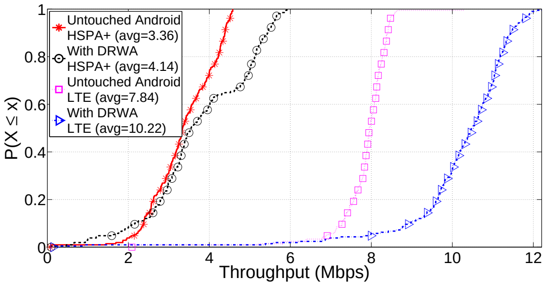
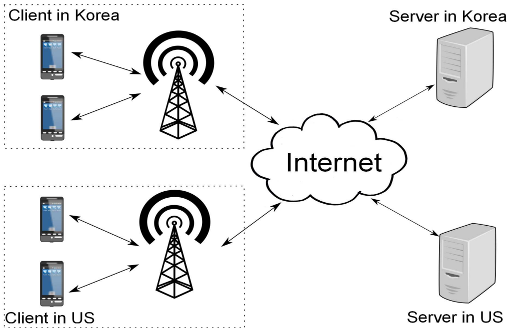


Figure 12: Throughput improvement brought by DRWA when clients in U.S. download from a server in Korea: Each test lasts for 24 hours. The improvement ratios are 23.21% in AT&T HSPA+ network and 30.36% in Verizon LTE network.

no queue is built up. This ideal case only guarantees high throughput if 1) the traffic has constant bit rate, 2) the available bandwidth is also constant and 3) the constant bit rate equals to the constant bandwidth. In practice, Internet traffic is bursty and the channel condition varies over time. Both necessitate the existence of some buffers to absorb the temporarily excessive traffic and drain the queue later on when the load becomes lighter or the channel condition becomes better. Otherwise, we cannot fully utilize the link. λ determines how aggressive we want to be to keep the link busy and how much delay penalty we can tolerate. The larger λ is, the more aggressive the algorithm is. It will guarantee the throughput of TCP to be saturated all the time but at the same time introduce extra delay. Figure 10 gives the comparison of performance among different values of λ. This test combines multiple scenarios ranging from local to remote access, good to weak signal. Each has been repeated for over 400 times over the span of 24 hours so as to find the optimal parameter setting. In our current implementation, we set λ to 3 which works very well for most cellular networks. However, a better approach may exist, which may make this parameter adaptive. We leave this as our future work.

4.4 Improvement in User Experiences



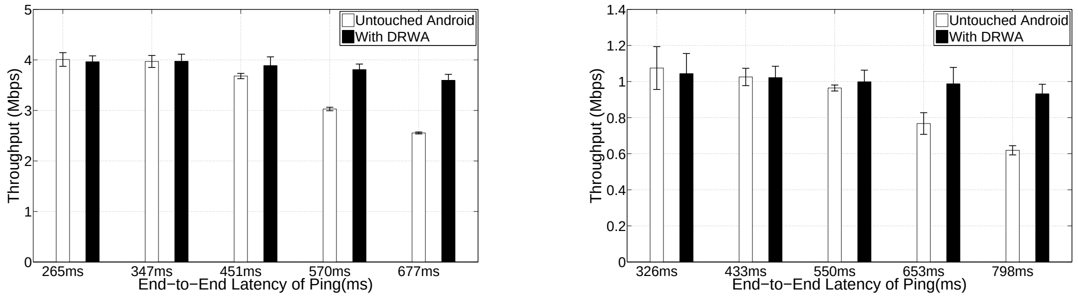
(a) Experiment Architecture

(b) Experiment Phones

Figure 13: Our test environment: We have TCP servers in U.S. and Korea and pairs of smart phones for major cellular carriers.

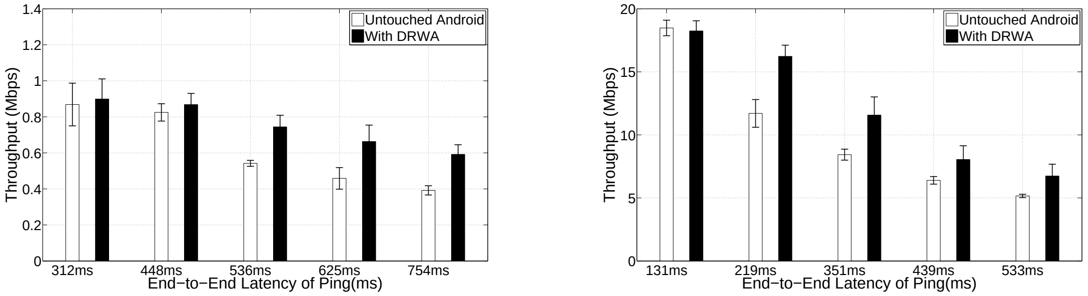
Section 2.3 lists two scenarios where existing TCP implementation may have a negative impact on user experience. In this section, we demonstrate that, by applying DRWA we can drastically improve user experience in such scenarios. More comprehensive experiment results are provided in Section 5.

Figure 11 shows Web object fetching performance with background download traffic. Since DRWA reduces the length of the queue built up in the cellular networks, it brings on average 41.72% reduction in the RTT experienced by all the TCP flows coming down to the receiver. This translates into 39.46% speed-up in Web object fetching since the download completion time of (typically small) Web pages and the embedded objects (e.g., images, flash clips) are mainly deter-



(a) Comparison in AT&T HSPA+

(b) Comparison in Verizon EVDO



(c) Comparison in Sprint EVDO

(d) Comparison in Verizon LTE

Figure 14: Throughput improvement provided by DRWA for various cellular networks under different network latencies: we see significant throughput improvement when the end-to-end latency is long.

	265 ms	347 ms	451 ms	570 ms	677 ms
AT&T HSPA+	-1.09%	0.12%	5.65%	25.78%	40.77%
	326 ms	433 ms	550 ms	653 ms	798 ms
Verizon EVDO	-2.85%	-0.31%	3.58%	28.65%	50.51%
	312 ms	448 ms	536 ms	625 ms	754 ms
Sprint EVDO	3.54%	5.19%	37.39%	44.62%	51.06%
	131 ms	219 ms	351 ms	439 ms	533 ms
Verizon LTE	-1.27%	38.67%	37.08%	25.64%	30.62%

Table 3: Throughput improvement shown in Figure 14 for difference latency values over various cellular network: as the end-to-end latency increases, the throughput improvement ratio becomes higher.

mined by the RTT.

Figure 12 shows the scenario where a mobile user in U.S. downloads from a remote server in Korea. Since the RTT is very long in this scenario, the BDP of the underlying network is fairly large. The static setting of *tcp_rmem_max* is too small to fill the long, fat pipe and results in throughput degradation. With DRWA, we are able to fully utilize the available bandwidth and achieve 23-30% improvement in throughput.

5. EXTENSIVE EXPERIMENTS

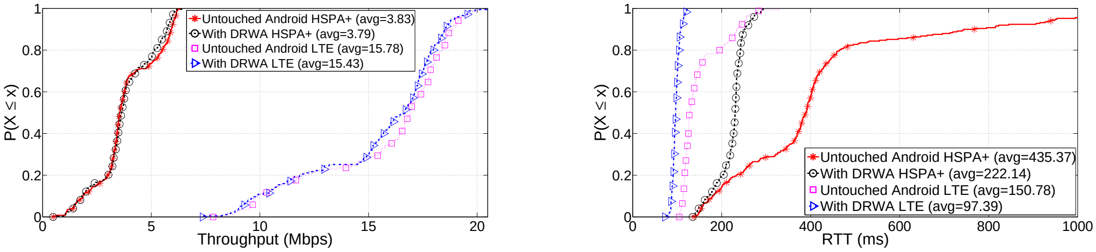
5.1 Test Environment

We implemented DRWA in Android phones by patching their kernels. It turned out to be fairly simple to implement DRWA in the Linux/Android kernel. It takes merely around 100 lines of code. We downloaded the original kernel source codes of different Android models from their manufacturers' website, patched the kernels with DRWA and recompiled them. Finally, the phones were flashed with our customized kernel images. We provided a procfs entry for users to easily turn on or off DRWA.

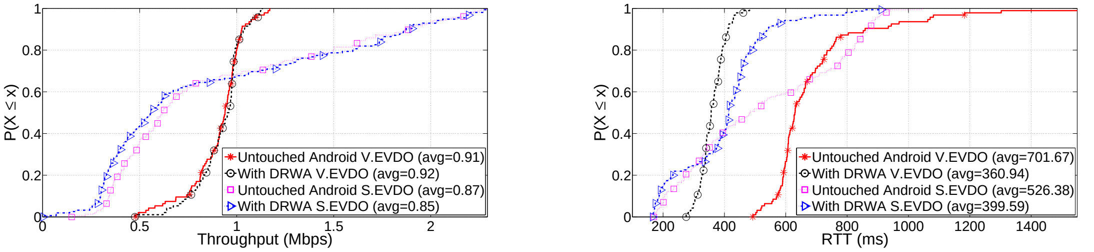
We did head-to-head comparisons between untouched Android phones and Android phones with DRWA. Figure 13 gives an overview of our test environment. We have both clients and servers in two places: a university in U.S. and a university in Korea. We evaluate different scenarios where clients download files from nearby servers or remote servers over various cellular networks operated by different carriers. All our servers run Ubuntu 10.04 (with 2.6.35.13 kernel) and use the default TCP congestion control algorithm, CUBIC [10]. The settings of the smart phones used as clients vary depending on the carriers and their networks (see Table 1). The signal strength during out tests is between -75dBm and -87dBm (typically considered as good signal condition in daily life) unless otherwise noted. We developed a simple Android application that downloads files from the designated servers with different traffic patterns. Traces were collected on the server side using tcpdump [1] and analyzed using tcptrace [17]. Internal states of TCP (e.g., $cwnd$) are probed with the help of Web100 project.

5.2 Throughput Improvement

Figure 14 shows the throughput improvement of Android phones with DRWA over untouched Android phones. Detailed percentage of improvement can be found in Table 3. The test involved file downloading (file size is 100MB) via

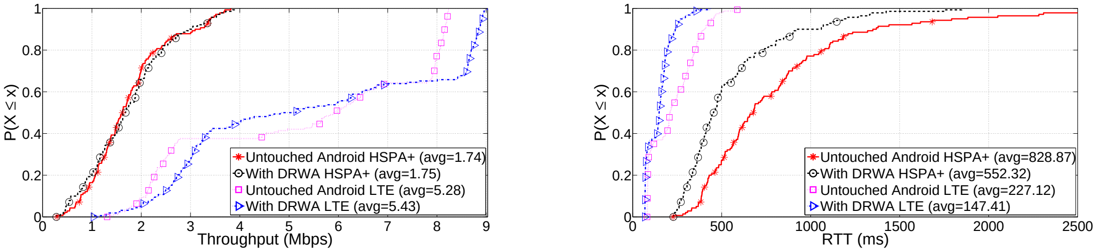


(a) Throughput: Verizon LTE and AT&T HSPA+ (b) RTT: Verizon LTE and AT&T HSPA+



(c) Throughput: Verizon EVDO and Sprint EVDO (d) RTT: Verizon EVDO and Sprint EVDO

Figure 15: RTT improvement in networks with small BDP: DRWA provides huge RTT reduction without throughput loss across different cellular networks. The RTT reduction ratios are 48.97%, 35.41%, 48.56% and 24.09% for AT&T HSPA+, Verizon LTE, Verizon EVDO and Sprint EVDO networks respectively.



(a) Throughput: Verizon LTE and AT&T HSPA+ (b) RTT: Verizon LTE and AT&T HSPA+

Figure 16: We repeat the same tests shown in Figures 15(a) and 15(b) under weak signal strength ranging between -95dBm and -105dBm. In such a condition, throughput and RTT performance are significantly degraded but the performance gain from DRWA is still clearly visible.

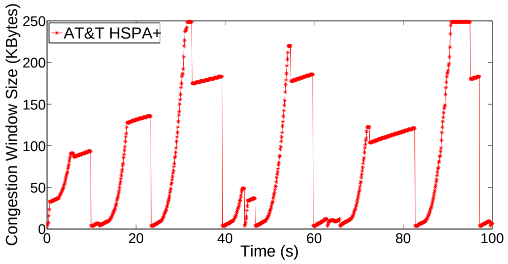
different cellular networks operated by various carriers. For each network we ran the test for 24 hours. During the test, we applied *netem*, the built-in network emulator in Linux [11] on the server side to emulate the scenarios of different end-to-end latencies. From Table 3, we see that Android phones with DRWA significantly improve the throughput in all cellular networks as the end-to-end latency increases. The scenario over the Sprint EVDO network with the end-to-end latency of 754 ms shows the largest improvement (as high as 51.06%). In LTE networks, the phones with DRWA show throughput improvement up to 38.67% under the latency of 219 ms.

The reason behind the improvement is obvious. When the latency increases, the static values set by the vendors fail to saturate the pipe, resulting in throughput degradation. In contrast, networks with small latencies do not show such degradation. According to our experiences, RTTs between 400 ms and 700 ms are easily observable in cellular networks, especially when using services from foreign servers.

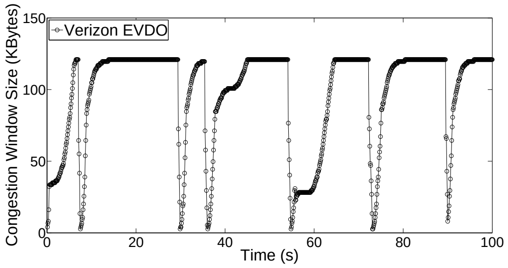
In the LTE networks, TCP throughput is even more sensitive to *tcp_rmem_max* setting. The BDP can be dramatically increased by a slight RTT increase. Therefore, the static configuration easily becomes far from optimal. However, DRWA is able to keep pace with the increasing BDP without any problem.

5.3 End-to-End Latency Reduction

In networks with small BDP, the static *tcp_rmem_max* setting is sufficient to fully utilize the bandwidth of the network. However, it has a side effect of long RTT. In such networks, the static receive window reported by current implementations misleads a TCP sender to put excessive packets in the network, resulting in unnecessarily long RTT. However, DRWA manages the RTT to be λ times of the RTT_{min}, which is substantially smaller than that of current implementations in networks with small BDP. Figure 15 shows the



(a) $cwnd$ in AT&T HSPA+



(b) $cwnd$ in Verizon EVDO

Figure 17: For 0.5% packet losses applied by *netem*, AT&T's HSPA+ network shows frequent RTO events while other cellular networks (e.g., Verizon's EVDO network) do not. A drop of $cwnd$ in the left figure indicates RTO but that in the right figure indicates fast retransmission although they look similar. Intermediate data points during the drop explains the difference.

improvement in RTT brought by DRWA while throughput is preserved.

In Figure 15, we downloaded a file from a local server installed at a university campus in U.S. to Android phones in U.S. to explore the throughput and end-to-end delay performance in small BDP networks. We measured the performance for a whole day per each carrier and compared the performance between Android phones with and without DRWA. During the whole day run, each round of the file downloading took three minutes, resulting in over 400 runs within a day. From Figure 15, we can verify that remarkable reduction of RTT up to 48.97% is achieved while the throughput is guaranteed in a similar level (4% difference at maximum). The same tests were repeated under weak signal conditions and the results are shown in Figure 16. In this scenario, the average throughput of the TCP flows is much lower than that in Figure 15(a) while the RTT is much longer. But DRWA can still reduce the RTT with no throughput loss.

Another important observation from the experiments is that the current implementation with a static receive window experiences much larger RTT variation than DRWA. As Figures 15(a) and 15(c) show, the RTT values of untouched Android phones are distributed over a much wider range than that of phones with DRWA. The reason is clear because DRWA intentionally enforces the RTT to stay around the target value of $\lambda * RTT_{min}$. This property of DRWA will potentially benefit jitter sensitive applications such as live video communications and voice chats.

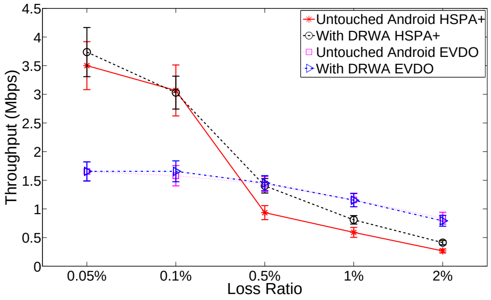


Figure 18: DRWA alleviates TCP throughput degradation in AT&T HSPA+ network: throughput is increased by 6.75%, 50.09% and 54.30% under loss ratio of 0.05%, 0.5% and 2% respectively.

5.4 Loss Recovery

Although buffer-bloated cellular networks rarely have packet losses, other part of the end-to-end TCP connection may still experience them. For instance, according to the large-scale measurement study in [5], the Internet experiences a non-negligible packet loss rate between 1% and 6%. If such packet losses result in RTO, TCP running over an excessively long queue will need to pay expensive recovery cost. For an RTO, the TCP sender has to retransmit all of the packets in flight over the network. When there are more queued packets, the amount of discarded packets due to the retransmissions becomes higher. In general, RTO rarely happens in cellular networks because SACK (selective acknowledgement) is enabled in Android phones by default and it typically recovers packet losses before time out happens. However, in some cellular networks, special configurations make RTO happens with high probability even with a single packet loss. The HSPA+ network operated by AT&T is one example of such networks. As shown in Figure 17(a), $cwnd$ of TCP over AT&T's HSPA+ network shows very frequent RTO operations when we emulate packet losses in the Internet using *netem*, while other cellular networks do not show such $cwnd$ behaviors. Figure 17(b) exemplifies that RTO does not happen in Verizon's EVDO network. Instead, fast retransmit recovers the lost packets. Since *netem* imposes bursty packet losses, the RTO and fast retransmit look similar, but $cwnd$ in Verizon's EVDO network shows intermediate data points while falling down, which tells that the drops are not from RTO.

It is hard to understand and explain why a specific network suffers this problem while others do not. Recent observations and corresponding analysis claimed in [19] give a hint to the problem. According to the work, some cellular network carriers manage middleboxes in their networks for deep packet inspection or other security purposes and the middelboxes buffer out-of-order packets and deliver them to the receiver when the packets become in-order. Therefore, in such configuration of a network, a single lost packet will block the middlebox from delivering the packets to the receiver, resulting in huge delays for a bunch of ACK packets from the receiver. If this happens, RTO may happen for packet loss.

To further explore this issue, we set up a test scenario over the AT&T's HSPA+ network and Verizon's EVDO network. We apply random bursty packet losses whose ratios range from 0.05% to 2% using *netem* on a router connected to our local server in U.S.. This range of packet loss rate is observable in the wild Internet [5]. Figure 18 shows the throughput results for different loss ratios with and without DRWA. Since DRWA keeps the queue size smaller, when RTO happens, the amount of packets to be recovered is less. Therefore, Android phones with DRWA experience less throughput degradation comparing to the untouched Android phones for the same loss ratio. Up to 54% of throughput improvement is observable when the loss ratio goes beyond 0.5%.

6. CONCLUSION

In this paper, we thoroughly investigated TCP's behavior and performance over cellular networks. We reveal that the excessive buffers available in existing cellular networks void the loss-based congestion control algorithm used by most TCP implementations and the naive solution adopted of setting a static *tcp_rmem_max* is sub-optimal. Built on top of our observations, a dynamic receive window adjustment algorithm is proposed. This solution requires modifications only on the receiver side and is backward-compatible as well as incrementally deployable. We ran extensive experiments over various cellular networks (EVDO, HSPA+, LTE, etc.) to evaluate the performance of our proposal and compare it with the current implementation. Experiment results show that our scheme makes RTT 24.09 $\sim$ 48.97% lower than the current implementation of TCP while throughput is guaranteed to be the same in general cases or up to 51.06% higher in a high speed network with long latency. The bufferbloat problem is becoming more and more prevalent in the Internet. It is not specific to cellular networks although it might be the most prominent in this environment. A more fundamental solution to this problem may be needed. Our work provides a good starting point and is an immediately deployable solution for smart phone users.

7. REFERENCES

[1] tcpdump. http://www.tcpdump.org/.

[2] S. Athuraliya, S. Low, V. Li, and Q. Yin. REM: Active Queue Management. *IEEE Network*, 15:48–53, May 2001.

[3] L. S. Brakmo, S. W. O'Malley, and L. L. Peterson. TCP Vegas: New Techniques for Congestion Detection and Avoidance. In *Proceedings of ACM SIGCOMM*, 1994.

[4] M. C. Chan and R. Ramjee. TCP/IP Performance over 3G Wireless Links with Rate and Delay Variation. In *Proceedings of ACM MobiCom*, 2002.

[5] N. Dukkipati, T. Refice, Y. Cheng, J. Chu, T. Herbert, A. Agarwal, A. Jain, and N. Sutin. An Argument for Increasing TCP's Initial Congestion Window. *SIGCOMM Comput. Commun. Rev.*, 40:26–33, June 2010.

[6] W.-c. Feng, M. Fisk, M. K. Gardner, and E. Weigle. Dynamic Right-Sizing: An Automated, Lightweight, and Scalable Technique for Enhancing Grid Performance. In *Proceedings of the 7th IFIP/IEEE International Workshop on Protocols for High Speed Networks (PIHSN)*, pages 69–83, 2002.

[7] S. Floyd and T. Henderson. The NewReno Modification to TCP's Fast Recovery Algorithm. IETF RFC 2582, April 1999.

[8] S. Floyd and V. Jacobson. Random Early Detection Gateways for Congestion Avoidance. *IEEE/ACM Transactions on Networking*, 1:397–413, August 1993.

[9] J. Gettys. Bufferbloat: Dark Buffers in the Internet. *IEEE Internet Computing*, 15(3):96, May-June 2011.

[10] S. Ha, I. Rhee, and L. Xu. CUBIC: a New TCP-friendly High-speed TCP Variant. *ACM SIGOPS Operating Systems Review*, 42:64–74, July 2008.

[11] S. Hemminger. Netem - emulating real networks in the lab. In *Proceedings of the Linux Conference*, 2005.

[12] J. Huang, Q. Xu, B. Tiwana, Z. M. Mao, M. Zhang, and P. Bahl. Anatomizing Application Performance Differences on Smartphones. In *Proceedings of ACM MobiSys*, 2010.

[13] K.-c. Lan and J. Heidemann. A Measurement Study of Correlations of Internet Flow Characteristics. *Computer Networks*, 50:46–62, January 2006.

[14] Y. Lee. Measured TCP Performance in CDMA 1x EV-DO Networks. In *Proceedings of the Passive and Active Measurement Conference (PAM)*, 2006.

[15] X. Liu, A. Sridharan, S. Machiraju, M. Seshadri, and H. Zang. Experiences in a 3G Network: Interplay between the Wireless Channel and Applications. In *Proceedings of ACM MobiCom*, pages 211–222, 2008.

[16] M. Mathis, J. Heffner, and R. Reddy. Web100: Extended TCP Instrumentation for Research, Education and Diagnosis. *AMC SIGCOMM Computer Communications Review*, 33:69–79, July 2003.

[17] S. Ostermann. tcptrace. http://www.tcptrace.org/.

[18] F. Qian, A. Gerber, Z. M. Mao, S. Sen, O. Spatscheck, and W. Willinger. TCP Revisited: a Fresh Look at TCP in the Wild. In *Proceedings of the 9th ACM SIGCOMM IMC*, pages 76–89, 2009.

[19] Z. Wang, Z. Qian, Q. Xu, Z. Mao, and M. Zhang. An untold story of middleboxes in cellular networks. In *Proceedings of the ACM SIGCOMM*, 2011.

[20] D. X. Wei, C. Jin, S. H. Low, and S. Hegde. FAST TCP: Motivation, Architecture, Algorithms, Performance. *IEEE/ACM Transactions on Networking*, 14:1246–1259, December 2006.

[21] L. Xu, K. Harfoush, and I. Rhee. Binary Increase Congestion Control (BIC) for Fast Long-distance Networks. In *Proceedings of IEEE INFOCOM*, 2004.

[22] P. Yang, W. Luo, L. Xu, J. Deogun, and Y. Lu. TCP Congestion Avoidance Algorithm Identification. In *Proceedings of ICDCS*, 2011.