Can you GET Me Now? Estimating the Time-to-First-Byte of HTTP Transactions with Passive Measurements

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ABSTRACT

Cellular network operators have a compelling interest to monitor HTTP transaction latency because it is an important component of the user experience. Existing techniques to monitor latency require active probing or use passive analysis to estimate round-trip time (RTT). Unfortunately, it is impractical to use active probing to monitor entire cellular networks, and RTT is only one component of HTTP latency in cellular networks. This paper presents a new passive technique to estimate HTTP transaction latency that overcomes the scaling and completeness limitations of prior approaches. We validate our technique in an operational cellular network and present results for traffic in the wild.

Categories and Subject Descriptors

C.2.3 [**Computer-Communication Networks**]: Network Operations

Keywords

Time To First Byte, Round-Trip Time, Network measurement, Cellular, Wireless, Mobile

1. INTRODUCTION

Quick application response time has always been a crucial component of a satisfactory user experience [10], and recent studies suggest that differences of only 250 milliseconds will cause a user to visit a Web site less often [2]. As more and more applications migrate to the Web and interact with the cloud, the latency of network protocol transactions is an increasingly important component of overall user response time. In particular, this trend implies a growing importance of HTTP transaction latency, as HTTP is the dominant protocol for the mobile Web, RESTful APIs, and video streaming [3].

Cellular network operators have a compelling interest to monitor the latency of HTTP transactions because changes in their networks can directly influence these latencies. A standard measure of HTTP transaction latency is the total time-to-first-byte (TTFB). The total TTFB is informally defined as the time elapsed from a user's request for an object to the reception of the first byte of that object. This time represents the lower bound on the delay the user will experience before an application can start rendering the requested content. Of particular interest to network operators is the TTFB of a single HTTP transaction, i.e., time between the start of the TCP handshake and the arrival of the HTTP response at the user device. This elapsed time, which we refer to as TTFB hereafter for brevity, captures all the latency components that a cellular network can directly influence.[1] Unfortunately, conventional methods to measure TTFB rely on using active probing tools,[2] which in practice limits the scale and representativeness of measurements to a small number of vantage points. This limitation is particularly problematic for wireless network operators because it is impractical to run probes at all physical locations that their users will visit and physical context plays a large role in the wireless network latency users experience.

As a consequence, passive estimation techniques are generally preferred to measure latency at scale. For example, well-known passive techniques estimate round-trip time (RTT) by measuring the arrival times of packets at the probe location [7, 8, 9, 16]. However, these prior approaches have drawbacks that limit their usefulness as a proxy for TTFB. First, prior approaches do not capture additional delays that may occur prior to the first packet being sent over the network. These delays are common and significant in cellular networks due to layer 2 connection setup signaling [13]. Second, approaches that only measure TCP handshake time don't account for the transmission delay of the HTTP request and response, which can be significantly larger than TCP handshake packets. Finally, the same approaches don't account for server delay in processing the HTTP request. As a result, prior passive latency estimation techniques provide only limited visibility into changes that can impact HTTP response time in cellular networks.

In this paper, we design and evaluate a new passive measurement technique to estimate TTFB, taking into account the signaling and large packet transmission delays specific to cellular networks. Our approach works at a large scale and only requires information available within the network

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[1]The TTFB of a single HTTP transaction excludes potential delays due to DNS and HTTP redirects included in the total TTFB, but we find that a large fraction of HTTP transactions are not preceded by these delays (Section 3.3).

[2]E.g.: *keynote.com, webpagetest.org, loadimpact.com*

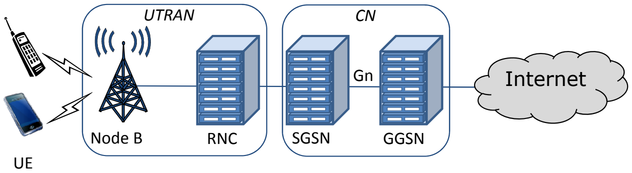


Figure 1: UMTS network architecture.

core, which meets two key challenges faced by a network operator. The main insight in our TTFB estimation approach is to utilize TCP timestamps (user domain) as well as the arrival times of packets at the probe location (network domain). Our approach addresses the limitations of previous techniques and challenges caused by peculiarities of traffic in the wild. This allows the operator to fully quantify the influence on TTFB of RTT, cellular network latency, connection setup delay, transmission time of HTTP messages, and server processing delay.

Finally, we present measurement results of TTFB in a deployment on a large U.S. cellular network. We find that TTFB can better reflect user response time than RTT, as connection setup signaling, which RTT ignores, comprises up to 68% of total TTFB. Comparison of TTFB values reveals up to 72.5% difference across domains, and up to 36% across applications, whereas RTT shows no appreciable difference. We believe our work demonstrates a practical method for operators to monitor a critical component of the mobile user experience.

2. BACKGROUND AND RELATED WORK

The most significant component of TTFB is the network latency. In cellular data networks, the network latency of delivering a packet comprises of three components: radio connection setup time, radio access network (RAN) scheduling and transmission, and core network transmission. To illustrate, consider a UMTS network (Figure 1). When a User Equipment (UE) such as a smartphone wants to send a packet, the UMTS Terrestrial Radio Access Network (UTRAN) must first allocate a radio resource control (RRC) connection for the UE if it has not already done so. Second, the UTRAN must schedule and then deliver the packet from the UE to the Core Network (CN). Finally, the packet is sent through the core network and the Internet to the destination. RAN scheduling and core network transmission typically incur RTTs of 100s of milliseconds on 3G networks [15] and 50-60 milliseconds in 4G Long Term Evolution (LTE) networks [5]. A new HTTP connection will typically incur two RTTs in a TTFB, one for the TCP 3-Way Hand Shake (3WHS) and one for the HTTP request and response. While these latencies are significant, radio connection setup time often adds an order of magnitude higher delay.

Radio connection setup delay is the time it takes for the UTRAN to allocate a radio connection to the UE. To understand why the connection setup delay is significant, we must understand the RRC state machine [11]. To efficiently utilize the radio resources, both UTRAN and UE maintain the state machine, which is synchronized via signaling on the control channel. There are typically three RRC states in a UMTS network: IDLE, CELL-FACH, CELL-DCH.[3] The state machine transitions to a higher resource state (state promotion), i.e., IDLE to CELL-DCH, when the UE has more data to send/receive and to a lower resource state (state demotion), i.e., CELL-DCH to CELL-FACH, when the UE has less data to send/receive. The reason for a significant connection setup delay is a state promotion, which involves signaling latency between the UE and the UTRAN. This latency is variable because signaling messages can be lost due to contention. Since RRC state is demoted after only a few seconds [12], a connection setup delay will typically occur for each user-initiated request.

The state promotion time is 1.5 to 2 seconds in UMTS [12], 260 milliseconds in LTE [5], and 80 ms in Wi-Fi [5]. These delays are an order of magnitude larger than the corresponding RAN scheduling and core network transmission delays, and they grow even larger when there is contention for radio resources. In addition, operators will evolve existing cellular networks by introducing new RRC states and by tuning RRC timers, which will change the magnitude and frequency of the connection setup delay [11]. Hence, to effectively monitor TTFB in cellular data networks, it is imperative that any estimation technique has the capability to measure all three components of network latency.

2.1 Limitations of Existing Techniques

Previous passive techniques attempt to estimate the RTT of TCP connections that pass through a passive packet monitor along the connections' end-to-end path. In a UMTS cellular data network, the monitor is typically located on the Gn link (Figure 1), where all cellular data traffic passes, because they are usually located in a small number of physical locations [4, 14].[4]

There are several approaches to estimate RTT. SYN-ACK estimation uses the interval between the SYN and acknowledgment (ACK) packets of the TCP 3WHS, and Slow-Start estimation measures intervals between bursts of data packets within a minimum of 5 consecutive data packets [8]. Jaiswal, et al. [7] estimate RTT by replicating the sender's state machine to infer congestion window size. Veal, et al. [16] estimate RTT as inter-arrival time between data and ACK packets matched by TCP timestamps.

All these techniques estimate network latency by recording the arrival time of IP packets at the monitor. Thus, a crucial limitation of these approaches is that they can not capture the connection setup delay that occurs before the first IP packet of a transaction leaves the UE for the UTRAN. Moreover, they don't measure the transmission time incurred for the HTTP request and response. In this paper, we design a new approach that overcomes this limitation by passively inferring the timing of packets at the UE.

3. DESIGN

In order to capture all potential components of cellular network latency, we focus on measuring the TTFB of HTTP transactions that begin with new TCP connections.[5] Figure 2 depicts the typical packet exchange at the beginning

[3]The same state machine mechanism applies when additional states are used, such as URA-PCH, and in LTE and Wi-Fi with PSM [5].

[4]In LTE, the equivalents to the Gn link are the S1u and S11 links.

[5]These transactions are most likely to be user initiated because typical web servers tear down persistent HTTP connections after only a few seconds [1].

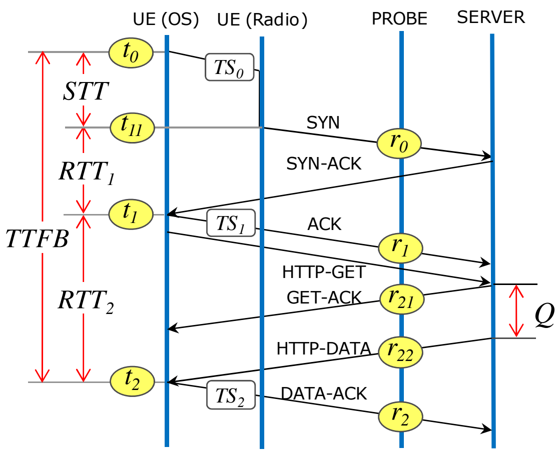


Figure 2: Typical HTTP data transfer.

Table 1: TTFB and its components.

Metric	Actual	Estimate
$TTFB$	$t_2 - t_0$	$G(TS_2 - TS_0)$
STT	$t_{11} - t_0$	$G(TS_1 - TS_0) - (r_1 - r_0)$
RTT_1	$t_1 - t_{11}$	$(r_1 - r_0)$
RTT_2	$t_2 - t_1$	$(r_2 - r_1)$
Q	unknown	$(r_{22} - r_{21})$
G	unknown	$\frac{r_2 - r_1}{TS_2 - TS_1}$

of such HTTP transaction. The vertical lines represent elapsed time at the UE operating system (OS), radio interface, probe, and server. Bubbles labeled t_i and r_i represent arrival times of packets, and TS_i are TCP timestamps in packets, which we describe in Section 3.1. The SYN, SYN-ACK, and ACK packets comprise the TCP 3WHS, while the HTTP-GET and HTTP-DATA packets correspond to the HTTP request and the first packet of the response. As RRC state transition is the largest part of the radio connection setup, we refer to this delay as State Transition Time (STT) hereafter. Q is the server processing delay.

We define the actual TTFB as the time elapsed between the SYN packet departure and the arrival of the first HTTP-DATA packet at the UE, i.e. $t_2 - t_0$. We stress that our definition of TTFB differs from the conventional one by not including DNS latency, as explained later. However, it includes the complete TCP 3WHS, STT when it occurs, as well as the HTTP request and the first packet of the response. Previous estimation techniques only used arrival times r_i to estimate latency (e.g., RTT $= r_1 - r_0$ [14, 8]), thereby missing the crucial STT that occurs between t_0 and r_0. Unfortunately, t_0 and t_2 are only directly measurable at the UE and available only for active probing techniques. In this section, we describe how we can estimate $t_2 - t_0$ and break down its constituent latency components using only information measured at the probe. Our approach is summarized in Table 1.

3.1 Challenges

The naïve approach to estimating TTFB at the probe would be as the inter-arrival time between the SYN and DATA-ACK, i.e., $r_2 - r_0$. This estimate would be correct if the latency between the UE and the probe is always the same. However, as we already explained, the delay $r_0 - t_0$ is often much larger than the delays $r_1 - t_1$ and $r_2 - t_2$ due to STT. To overcome this limitation of the naïve approach, we need to estimate t_0 and t_2 directly.

To do this estimation, we leverage the observation that many TCP connections include the TCP timestamp option (75% of the top 500 servers in previous work [16] and 85% of TCP connections in our dataset). This option adds two fields to the TCP header: Timestamp Value (TSval) and Timestamp Echo Reply (TSecr) [6]. The sender sets TSval to the time at which the segment was sent, and the receiver echoes the TSval of the most recently received segment in TSecr. Therefore, each t_i is represented by the corresponding TS_i in each TCP packet (Figure 2).

Under the assumption that the clock the sender uses to compute TS_0 and TS_2 has a known frequency $\frac{1}{G'}$ ticks/sec, we would have the simple equivalence $t_2 - t_0 = G'(TS_2 - TS_0)$. Unfortunately, $\frac{1}{G'}$ is not known and it depends on the device generating the timestamp. Moreover, we will not be able to measure TTFBs with any finer time resolution than G' because the timestamps are quantized to integral values. Thus, two challenges are to reliably estimate G' using only information at the probe and to evaluate whether G' is sufficiently granular in real traffic to effectively measure TTFB. We also want to ensure that TTFB estimation is robust against changes in the rapidly evolving mobile technology.

Finally, typical probes at the Gn interface will encounter traffic rates of tens of Gbps, if not larger. Thus, in order to capture as many real-time samples as possible, the TTFB estimation approach should be light-weight in terms of processing cost and memory footprint. While sampling can be employed, a large number of samples is required to monitor the real-time health of the individual elements in the entire spatial extent of a cellular network, e.g., NodeBs (Figure 1) can number in the hundreds of thousands.

3.2 TTFB Estimation

Recall that the actual TTFB $= t_2 - t_0$. We would like to estimate TTFB directly as $TS_2 - TS_0$. However, as we explained in the previous section, the units of TS_i, i.e., the *timestamp granularity*, are unknown and UE-specific, so we have to convert them into comparable time units. For this purpose, we define the granularity factor, G, as an estimate of timestamp granularity:

$$G = \frac{r_2 - r_1}{TS_2 - TS_1} \ . \tag{1}$$

The granularity factor G is effective in estimating the timestamp granularity because we expect that the time elapsed between the HTTP-GET and DATA-ACK at the UE ($t_2 - t_1$) is the same as at the probe ($r_2 - r_1$), and we record r_2 and r_1 in known time units. Since we use sender's TCP timestamps only, there is no need for any kind of clock synchronization between the sender and receiver.

Our TTFB estimate, thus, is defined as follows:

$$TTFB \quad = \quad G(TS_2 - TS_0) \ . \tag{2}$$

We can also estimate the TTFB's constituent components using the formulas in Table 1.

With the above formulation, we need only collect the following values for each TCP connection: r_0, r_1, r_2, TS_0, TS_1, and TS_2. Since r_i times will almost never differ by more than several seconds, we can use a 16 bit clock to generate the r_i timestamps with millisecond granularity and compute the differences modulo 2^{16}. The TCP timestamps themselves are 4 bytes each, so the total state tracked per TCP connection can be only 18 bytes. We identify packets belonging to distinct TCP connections using the standard IP 4-tuple (12 bytes for IPv4). The tuple can be hashed to 8 bytes if memory is more constrained than processing resources.

Alternative Formulation. We also consider an alternative formulation which requires tracking less state. This formulation uses twice the conventional estimate of RTT and the state transition delay STT in connection setup, i.e.

$$TTFB' \; = \; 2(r_1 - r_0) + STT \; . \tag{3}$$

STT is the difference between $G(TS_1 - TS_0)$, which includes STT, and $(r_1 - r_0)$, the conventional RTT estimate without STT. This option assumes that $RTT_1 \approx RTT_2$. This option does not need to track the HTTP request/response packets if we can obtain the timestamp granularity G without the HTTP request/response.

One way to obtain G without looking at the HTTP request/response is to maintain a database of timestamp granularities for each model of UE, since G is typically determined by the operating system and the device. We can then map each flow to the device model that generated it to obtain the correct G.[6] This requires a post-processing step after the measurement, but enables us to avoid tracking HTTP-GET and DATA-ACK packets.

The state required for the alternative option is only: r_0, r_1, TS_0, TS_1 (12 bytes), and there would be no need to track the HTTP-GET and DATA-ACK, allowing the probe to release memory resources sooner for each connection. Unfortunately, we found that $RTT_2 = 2.2 RTT_1$ on average, which violates the underlying assumption of this approach and significantly alters the TTFB estimate (by up to 60% without STT).

Distinguishing State Transitions. Not all TTFB measurements will include STT, but we need to know which ones do to effectively understand if the change in STT impacted the observed TTFB. To estimate if a packet incurred a state transition, we use a Finite State Machine (FSM) to emulate a simplified RRC state machine for each UE IP address. We use the FSM model from [13], but only estimate transitions from IDLE to CELL-DCH, the most common transition for new user-initiated transactions. We validate the accuracy of this approach in the next section.

3.3 Limitations

Our approach assumes that TTFB transactions will look like Figure 2. While there are exceptions, which we describe here, the next section shows that enough follow this pattern to accurately estimate TTFB. First, our estimate assumes that the DATA-ACK immediately follows HTTP-DATA packet. The UE could use a delayed ACK after the first HTTP-DATA packet, but the error introduced would likely be small (e.g., 3.7% for a 40 ms delayed-ACK timer, in our data described in the next section). Second, we assume that the SYN packet is not lost before reaching the probe, so that the timestamp will represent initial user activity. Loss rather than delay of the SYN at the IP layer is unlikely, as the cellular link layer employs hybrid ARQ to retransmit lost packets. The delay of the uplink ACK that completes the 3WHS on the radio link may reduce the observed interval $r_2 - r_1$ and, in turn, introduce error into G, because G assumes this interval is proportional to $TS_2 - TS_1$. In the next section, we show error in G introduced by typical delay variance does not result in appreciable error in TTFB estimates.

Since TTFB estimation is performed per TCP connection, distinguishing between HTTP responses that carry content vs. redirect instructions is not a part of the estimation technique. This does not diminish the value of the estimates, which may be filtered or aggregated in the post-processing stage to produce the estimates for different case studies, such as video stream start-ups, aggregated redirects and initial content, discarding advertising and analytics, etc. (e.g., see Section 5).

Finally, our TTFB definition does not include the DNS request and response latency that precede many HTTP transactions (30-39% in our data). While our estimate will be affected by all network latency components that can also affect the DNS exchange, it may miss state transitions that are initiated by the DNS request. We note that our definition already captures state transitions for 7-14% of HTTP transactions in our data. We are currently exploring ways to estimate state transition delays caused by the DNS exchange, e.g., by leveraging the fact that DNS clients retry requests after just 1 second, which is shorter than STT in today's UMTS networks. DNS packets do not include any data comparable to TCP timestamps, hence the technique to estimate TTFB is not applicable to DNS delay.

4. VALIDATION

To validate the estimation of TTFB on real traffic, we collect two packet traces at Gn links of a major U.S. cellular carrier (Figure 1), which we refer to as "network traces" hereafter. The traces include the first 128 bytes of each data packet, capturing extended TCP/IP headers. The traces do not include any personally identifiable information. Each TCP connection is identified by the standard IP 4-tuple. The arrival timestamp, TCP flags and timestamps are extracted from packet headers for TTFB estimation. All other data is discarded. The first trace captures 2 minutes of traffic on September 2, 2011 (754,082 TCP connections and 61,362 unique UE IPs), while the second captures 9 hours and 5 minutes of traffic March 2, 2012 (5,626,320 TCP connections and 73,058 unique UE IPs). The two Gn links where traces were obtained serve large regions of the western U.S. with significantly different traffic load.

During the capture of network traces, we also capture `tcpdump` traces from active experiments on four popular UE devices. These traces, referred to as "UE traces" hereafter, record packets as they enter the UE's operating system. During the time of the second network trace, we simultaneously capture radio event logs from the RNC that our UE was connected to, which will be used to validate state transitions.

4.1 Validation of TTFB estimates

Timestamp Granularity Validation. The key ingredient of our estimation technique is the TCP timestamp granularity, estimated according to Equation 1. Since the timestamp granularity determines the time resolution at which we

[6]In a UMTS or LTE network, the device type can be obtained from GTP-C messages [4].

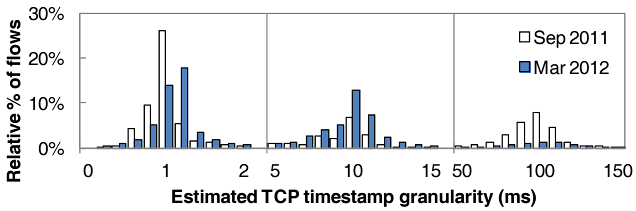


Figure 3: Distribution of G estimates.

can resolve TTFB estimates, we want to determine whether most TCP connections have a fine enough granularity. Resolutions from 1 to 10 ms are sufficient to differentiate changes in TTFB latencies since they are several multiples of this resolution. By examining packet traces of several popular UE devices, we find that common timestamp granularities are 1, 5, 10, and 100 ms. Next, we estimate G for TCP connections from the network traces and plot the distribution in Figure 3. Our estimates of G using Equation 1 correspond to the expected granularities, though the 5 ms granularity does not appear in a significant number of connections. For both network traces, we are able to see clusters around expected values of G from 96% of connections. In our validation below, we evaluate whether quantizing these estimated G values to 1, 5, 10, and 100 ms would result in higher accuracy, since other values are most likely due to estimation errors. Threfore, G is indeed accurate in the face of actual variance in scheduling and transmission delays.

We see a shift toward finer timestamp granularity over 6 months between trace collections, which corresponds to newer devices. If this trend continues, estimation accuracy will improve.

TTFB Validation. In order to determine the accuracy of our approach, we compare the TTFB estimates derived from the network traces against the actual TTFB as seen in the UE traces. Latencies from UE traces closely reflect the approach used in most popular active probing tools that execute within the Web browser or a standalone application, with the exception of the DNS exchange that we exclude. We use a UE with a 1 ms timestamp granularity in three scenarios: a) manual browsing, b) scripted browsing (without state transitions), and c) scripted browsing with state transitions. We find similar scripted browsing results for a UE with a 100 ms granularity (omitted for brevity). Scenario (a) consists of 731 connections to arbitrary Web sites, and (b) and (c) have 5,149 and 278 connections (respectively) to 26 popular Web sites (20% are mobile versions). Manual browsing connections may or may not include STT as the delay between page loads is arbitrary. Scripted browsing does not include STT as the delay is 1 second between loads. Scripted browsing with state transitions does include STT induced by a 20-second delay between loads.

The CDFs of the actual and estimated TTFB are shown in Figure 4 for all three scenarios. For competitive reasons, latency values in all plots are consistently normalized by the same constant. All scenarios show that our estimate (Equation 2) produces a distribution of estimates fairly close to the actual TTFB. A question posed earlier is how much the estimates would improve if we quantized estimated G values, since we know the actual timestamp granularity is usually 1 or 10 ms. To answer this question, we apply the following heuristic: when the G estimate falls in the range $(0.5, 1.5)$ ms, quantize G to 1 ms. This range is determined empirically from Figure 3. Figure 4 shows that using this approach (labeled Quantized) provides a slightly better estimate of TTFB. However, since the improvement is small and quantizing would result in incorrect results for fractional timestamp granularities, which are allowed, we do not use quantizing in our implementation.

Table 2: MSE for TTFB estimations.

Estimate	Quantized	Proposed	Alternative
Manual	3.95	4.86	7.05
Scripted	0.03	0.03	0.17
Scripted with STT	1.65	2.09	2.26

Table 3: Composition of TTFB (normalized).

State	STT	RTT_1	$RTT_2 - Q$	Q
No ST	0 (0%)	0.15 (32%)	0.31 (50%)	0.08 (18%)
With ST	0.79 (68%)	0.08 (7%)	0.29 (12%)	0.15 (13%)

The mean-squared-error (MSE) of estimates is shown in Table 2. The quantized estimate has the same error as the proposed one (Equation 2) in the scripted scenario, and a moderate advantage for other scenarios is indicated by lower MSE, as expected. For completeness, we also evaluate the alternative formulation (Equation 3) and it has higher error, especially for the scripted scenario. The error in scenarios that include STT is moderated by the relative magnitude of STT that dominates total TTFB. Due to its high error, we did not implement the alternative formulation.

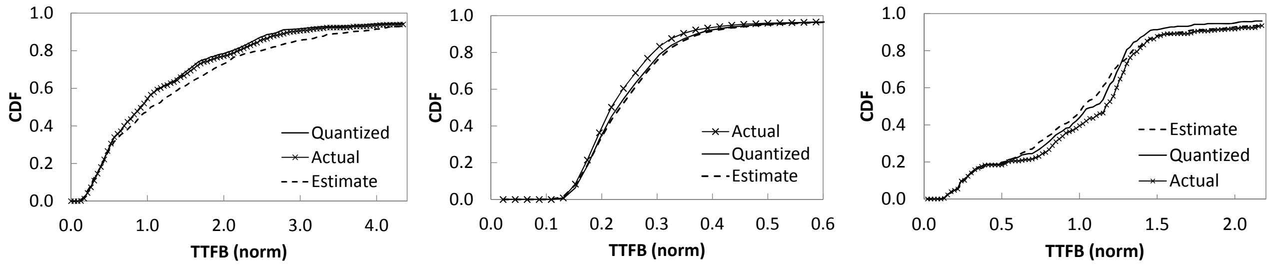
State Transition Validation. Finally, to validate that our FSM can approximately distinguish TTFBs with and without state transitions, we compare our FSM's estimated transitions, which are based on the network trace, to the RNC radio event logs, which record when the UE really has a state transition.[7] We find that over 76% of our estimated transitions are correct. 10% of incorrect estimates are edge cases, such as starting and ending a measurement session, while the remaining 14% are cases when signaling traffic that is not observable at the Gn keeps the UE in active state. Since the FSM classifies a large majority correctly, it is sufficient to determine whether changes in TTFB values are due to changes in the STT.

5. DEPLOYMENT AND RESULTS

We have implemented our TTFB estimation technique in a probe platform on the production network of a major U.S. cellular carrier. In addition to performance estimation, the platform categorizes TCP flows by application, domain, and device type using HTTP signatures and GTP-C information [18]. In this section, we describe some early results from this deployment.

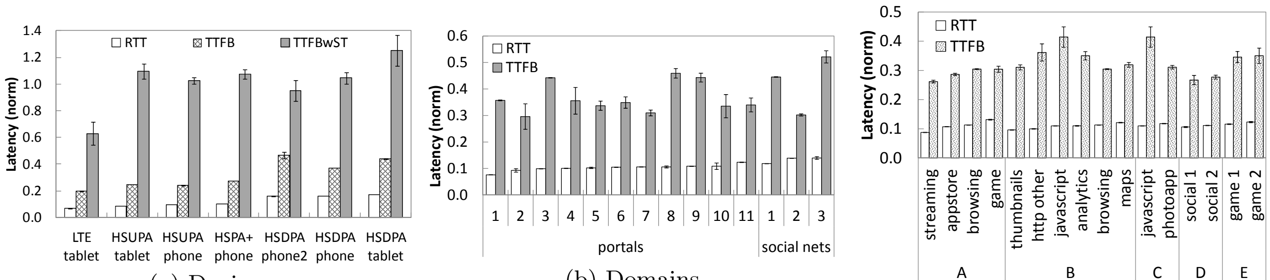
TTFB Latency Breakdown. The difference in TTFB for connections with and without a state transition is shown using CDFs in Figure 5. The averages of key constituent latencies in the TTFB, estimated using formulae in Table 1, are listed in Table 3. Both results demonstrate the large contribution of STT to total response time. STT comprises 68% of TTFB, on average, when a state transition occurs

[7]Due to load, resource, and vendor limitations, it is not practical to collect and process RNC logs continuously [17], which is why an estimation approach is needed.



(a) Manual browsing (b) Scripted browsing (c) Scripted browsing with STT

Figure 4: Validation of TTFB estimates.



(a) Devices (b) Domains (c) Applications

Figure 6: Comparison of mean RTT vs. mean TTFB. Error bars indicate 95% confidence intervals.

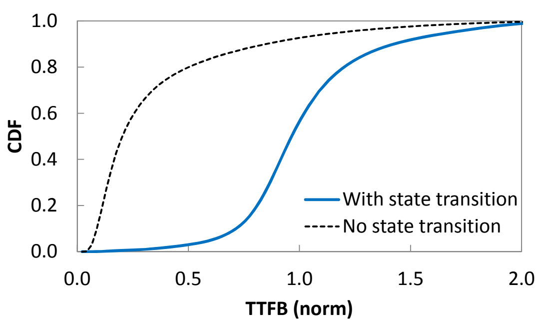


Figure 5: TTFB with and without STT.

(Table 3). In addition, we observe that the average server delay Q is non-negligible.

TTFB by Device, Domain, and Application. Next, we present three examples where TTFB offers a more complete picture of user experienced delay than RTT, using $5\frac{1}{2}$ hours of network data collected from 4:50PM ET on May 2, 2011. Figure 6 shows estimated RTT (as per [8]) and TTFB by device, domain, and application. Mobile devices with newer radio technology are expected to offer a better user experience, but we observe that the improvements can be subtle. Figure 6a shows that devices with HSUPA/HSPA+ have lower RTT and TTFB over those without (HSDPA), while there is no significant difference for TTFB with state transitions. However, the LTE device shows better performance across all metrics. This is due to the fact that LTE technology has both higher data rates and shorter state transition delays, while both HSDPA and HSUPA/HSPA+ devices have the same state transition delay.

Figure 6b compares the RTT and TTFB for different Web portals and social network domains. We observe up to 72.5% difference in TTFB between domains that have nearly identical measured RTT (social networks 2 and 3). This can be mostly attributed to varying processing delays and HTTP packet transmission delays, which RTT estimates do not capture. As cellular technologies with lower transmission delay are deployed, such as LTE, server processing delay will make up a larger faction of the TTFB (Table 3).

Figure 6c shows the RTT and TTFB of different applications, grouped by domain (all apps under the same letter are transactions to the same domain). We again see examples where TTFB differentiates user-perceived performance but RTT does not. For example, we have cases where there is no correlation between RTT and TTFB (domains B and C). The three application classes that stand out are analytics, http-other and javascript, which are served up to 36% slower than browsing and photoapp in domains B and C, respectively. The higher TTFB is likely the consequence of such transactions occurring in parallel with others.

6. CONCLUSION AND FUTURE WORK

This paper presented an accurate and light-weight technique to estimate TTFB of HTTP connections. Our early results from a deployment in a large cellular carrier demonstrate that TTFB, a large component of user response time, can vary substantially along many dimensions, even when measured RTT does not. Hence, measuring TTFB provides a more complete picture of the user experienced delay. In addition, our technique allows each component of TTFB to be studied separately, such as state transitions and server delay. While presented technique is applicable not only to cellular networks, there may be simpler methods available, depending on the transmission technologies in each particular case. In the future, we plan to improve the accuracy of our state estimation FSM by including more states. We are also evaluating ways to include the DNS exchange into our definition of TTFB.

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