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Media Content Delivery Protocols Performance and Reliability Evaluation in Cellular Mobile Networks

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PhD, Software Engineering

A member of the Institute of electrical and electronics engineers (IEEE) and the Association for computing machinery (ACM).

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Introduction

The most people access to social networks from the mobile devices via cellular networks. The most common access pattern is the request media (image and video thumbnails) for the content feed.

The relevance of the problem lies in the complexity of choosing the appropriate protocol for transferring content, since each protocol has its own set of features, in particular, different resistance to network interference. In accordance with this, it is necessary to study the stability of TCP and UDP protocols to the effects of network interference, as well as to compare them with the new protocol from Google - Quick UDP Internet Connections (QUIC)

It motivated us to study the performance and reliability of the content delivery protocols via cellular networks (2G, 3G, 4G, LTE, etc).

Evaluation approach

The NetPacket Simulator tool was developed to solve the problem of investigating the stability of protocols to network interference. This tool could be used to describe various network connection configurations and emulate possible network interference.

The following parameters are supported:

- RTT
- Bandwidth
- Congestion control window
- Packet Loss
- Upload/Download rate
- Multiple clients
- Setting amount of requests

The developed tool works on the basis of virtual network drivers TUN/TAP. This allows you to organize a virtual network within a single physical device, which eliminates the occurrence of uncontrolled network interference, if the network is organized between several physical devices.



Evaluation

To study the stability of protocols to network interference, the following experimental scenarios were identified. Among network interference cases were selected: RTT, Bandwidth, Packet Loss, Upload/Download rate, Congestion Control window size. These scenarios reflect the most popular network connections.

	Wi-Fi	LTE	3G	2G	RTT	Packet Loss	Bandwidth
RTT	110 ms	250 ms	550 ms	900 ms	10, 50, 100, 250, 500, 750, 1000 ms	100 ms	100 ms
Packet Loss	0.5 %	0.7 %	0.5 %	2.5 %	0.5 %	0.5, 1.0, 1.5, 2.0, 2.5%	0.5 %
Bandwidth	2.2 Mbit/s	2.0 Mbit/s	1.0 Mbit/s	0.2 Mbit/s	2.2 Mbit/s	2.2 Mbit/s	0.2, 0.6, 1.0, 1.4, 1.8, 2.2 Mbit/s
UP/DOWN rate	0.7	0.7	0.7	0.7	0.7	0.7	0.7
CC window	1 mbyte	1 mbyte	1 mbyte	1 mbyte	1 mbyte	1 mbyte	1 mbyte
File Size	50, 100, 250, 500, 1000 Kbytes				250 Kbytes		4096 Kbytes

Evaluation

Based on these scenarios, network connection configurations were described and the average time required for query execution to get the required amount of data was measured. The average time per request was calculated based on data on the execution time of five repeated simulations.

After the experiments had been performed, the results were summarized and converted to diagrams.

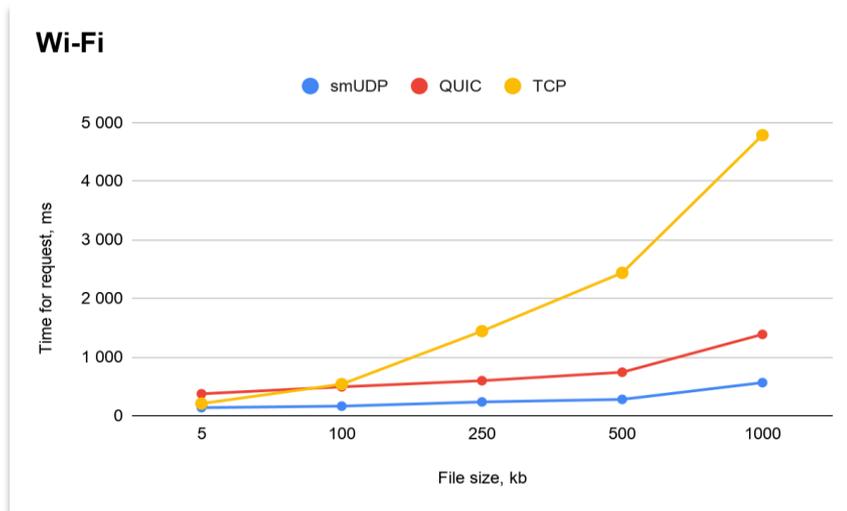


Figure 1. Wi-Fi

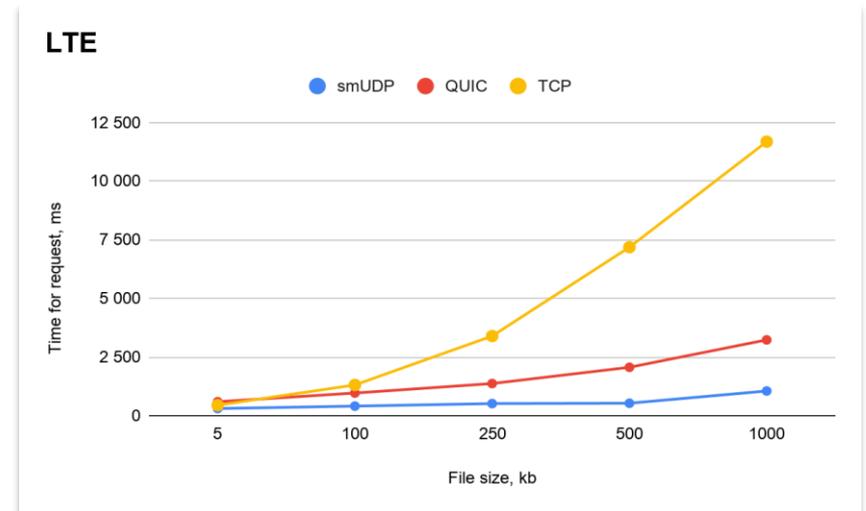


Figure 2. LTE

Evaluation

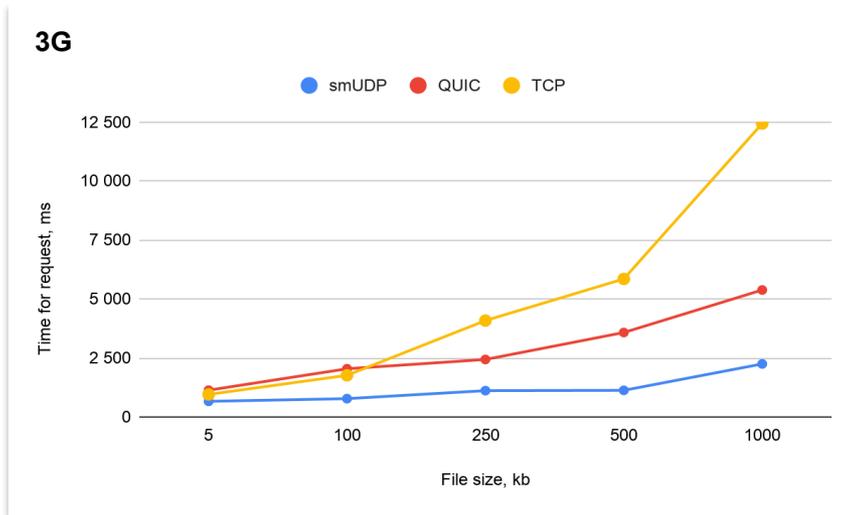


Figure 3. 3G

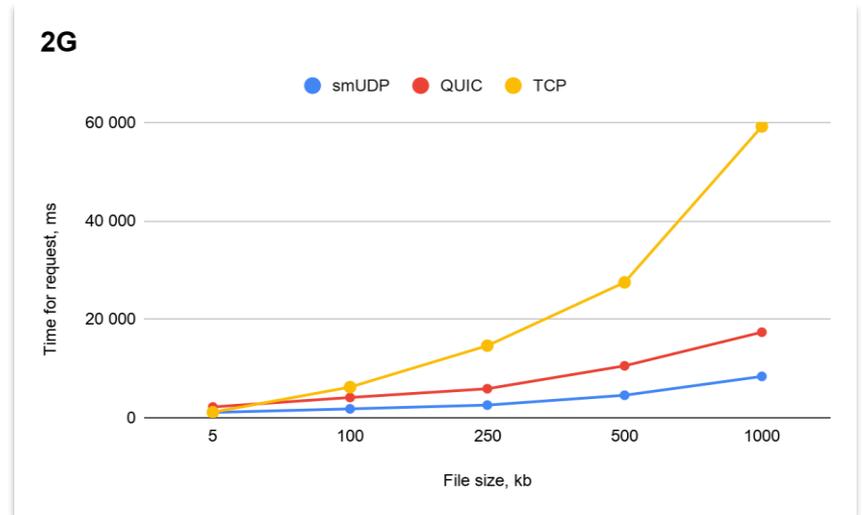


Figure 4. 2G

In Figures 1-4, it can be found that the TCP protocol has worse behavior than other protocols. While for QUIC and self-made UDP, the increase in request time increases linearly with the increase in the size of the requested file, for TCP, there is an exponential increase.

TCP is designed in such a way that TCP generally uses a TCP 3-way handshake, while UDP and protocols based on it do not spend time on this and can send data in the first packet (see Figure 5).

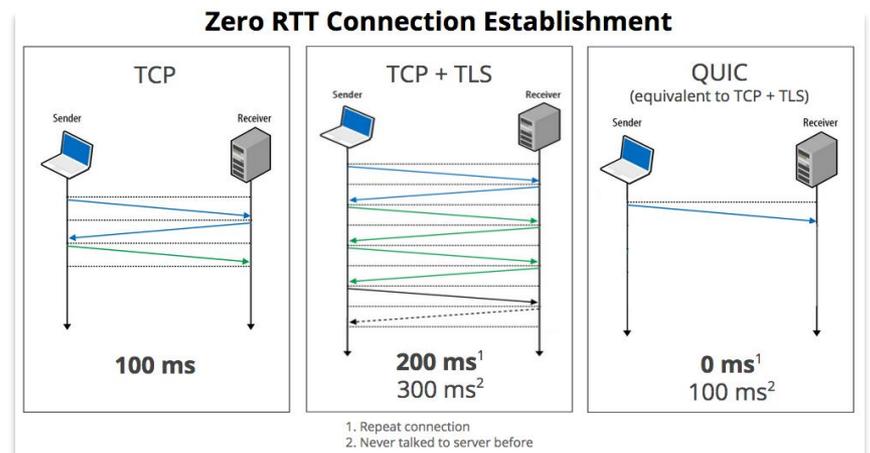


Figure 5. zeroRTT vs TCP

Evaluation

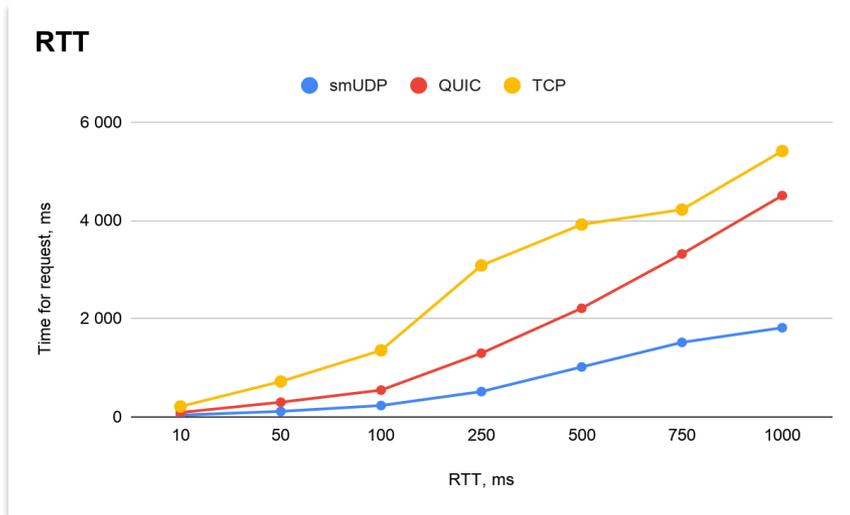


Figure 6. RTT impact comparison

The RTT change is not expected to significantly affect the delivery speed – for TCP the growth is similar to that of UDP-based protocols, as shown in Figure 6. However, due to the fact that some packets with ACK or the data itself may be lost due to network connection instability in TCP, the query execution time is greatly increased. This disadvantage can be clearly observed in Figure 7. In contrast to UDP-based protocols, with an increase in the percentage of lost packets, TCP also increases the time of data delivery.

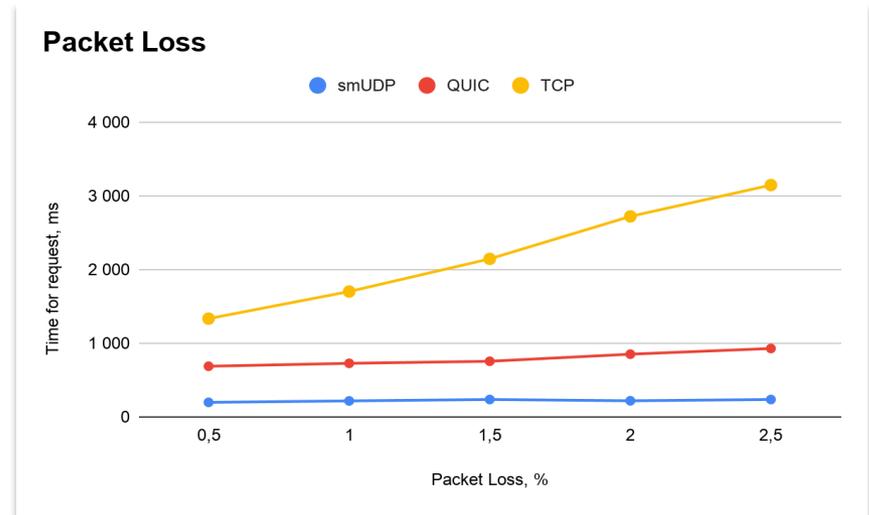


Figure 7. Packet Loss impact comparison

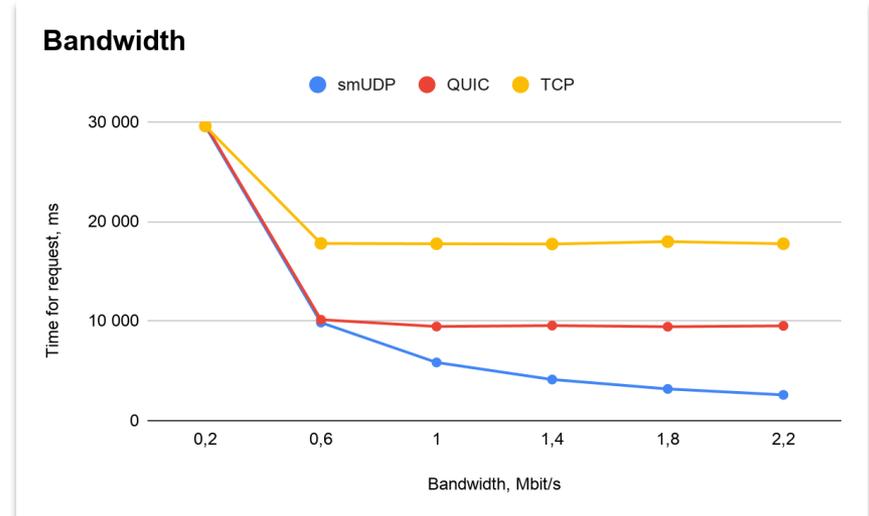


Figure 8. Bandwidth impact comparison

Evaluation

While QUIC and smUDP demonstrate stable delivery speed, despite the presence of problems in the connection. This development is due to the fact that QUIC calls two TLPs before the RTO works – even when the losses are very noticeable. This is different from TCP implementations. TLP mainly forwards the last packet (or a new one, if there is one) to start fast replenishment.

This is also due to the fact that TCP was originally developed as a protocol for wired network connections - which is more stable than wireless networks. Wireless networks are designed differently. To deal with fluctuations in bandwidth and loss, wireless networks usually use large buffers for traffic spikes. TCP very often treats the queue as a loss due to the increased response timeout, which is why TCP is forced to retransmit packets, which leads to a full buffer and a longer query execution time.

Also, while studying the stability of protocols to change bandwidth, it can be noted that in general, this parameter does not significantly affect the transfer speed. This is due to the fact that the TCP and QUIC protocols contain an implementation of Congestion Control that limits the channel load, as shown in Figure 8.

Conclusion

As a result of the research, we can conclude that the TCP data transfer protocol is poorly optimized for wireless networks, which are currently very widely used and are rapidly spreading around the world, even in the most remote corners of it. Poor performance is associated with unstable wireless network connections. As a result, the data reaches the end customer over a longer period of time. However, the protocol provides guaranteed data delivery to the client.

At the same time, UDP-based protocols demonstrate good performance in fast data delivery to the client due to the fact that new solutions are implemented in new protocols separately by each protocol developer, and to use the protocol, it is enough to update the versions on the server and client. However, as for data integrity, this task falls on the shoulders of the one who will use this protocol for their own purposes.

It is also worth noting that there are already ready-made implementations of UDP-based protocols that are rapidly gaining popularity, such as QUIC. It is worth noting that QUIC is already used on 4.3% of all websites.

Thus, if the content on a service or resource is mostly consumed via wireless mobile networks, it is recommended to use a UDP-based protocol with its own implementation of the necessary mechanisms. However, if you don't have enough resources to implement your own protocol, you can use ready-made solutions. However, TCP can be used as a backup connection.