

## Live Streaming using SRT with QUIC Datagrams

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# What is SRT?

- A sub-second latency live contribution protocol on top of UDP (unicast)
- Stream multiplexing
- Bidirectional transmission
- Packet loss recovery (ARQ and/or FEC) within a fixed end-to-end latency constraint
- Connection bonding (or path redundancy)
- Content agnostic
- An open-source library is available on GitHub

More info: SRT Protocol Overview (SVA 2020)  
<https://www.youtube.com/watch?v=MFJeyInLKZY>



*Enabling low-latency video contribution & distribution and fast file transfer over unpredictable networks*

**SRT ALLIANCE**  
SECURE RELIABLE TRANSPORT



SRT Alliance is More than 600 Members Now!

Webinar: Tuesday, May 9th at 10am ET  
Plugfest: The whole week



The graphic features a dark blue background with a hexagonal grid pattern and glowing blue lines. At the top center is the SRT logo in a white rounded rectangle. Below it, the text 'InterOp Plugfest Webinar' is written in a large, bold, white sans-serif font. Underneath, the text 'Join the InterOp Plugfest and get the latest on SRT' is centered in a smaller white font. A horizontal line with the word 'FEATURING' in the center is positioned below. To the left of the line is the HAIVISION logo in white, and to the right is the YouTube logo (a red play button icon followed by the word 'YouTube' in white). At the bottom center, the SRT ALLIANCE logo is displayed in white, with the tagline 'SECURE RELIABLE TRANSPORT' underneath it.

[Mark Your Calendars for the Next SRT InterOp Plugfest with YouTube](#)

- DATAGRAM frames (like all QUIC frames) must fit completely inside a QUIC packet. In turn, QUIC packets must fit completely inside a UDP datagram since fragmentation is disabled in QUIC.
- To tunnel SRT over QUIC datagrams, a single SRT packet should be encapsulated into a single DATAGRAM frame (within the Datagram Data field of a QUIC datagram).
- See [Tunnelling SRT over QUIC](#) Internet-Draft (draft-sharabayko-srt-over-quic-00) for details.

## Listing 1: DATAGRAM frame format

```
DATAGRAM Frame {  
    Type ( i ) = 0x30 .. 0x31 ,  
    [ Length ( i ) ] ,  
    Datagram Data ( .. )  
}
```

- **The quicly library by Fastly** was selected for the project as it supports both QUIC STREAM and DATAGRAM frames.
- **srt-xtransmit** is a testing utility that
  - supports the UDP, TCP, SRT, and QUIC transport protocols,
  - implements generate, receive, and route commands which allow the simulation of live media transmission at a constant or variable bitrate without the need for a media encoder and decoder.
- The transmission was made from a MacBook Pro laptop located in Rendsburg, Germany (client/sender side), to a Raspberry Pi 3 Model A+ computer based in Madrid, Spain (server/receiver side). Both devices were connected to the Internet via Wi-Fi.

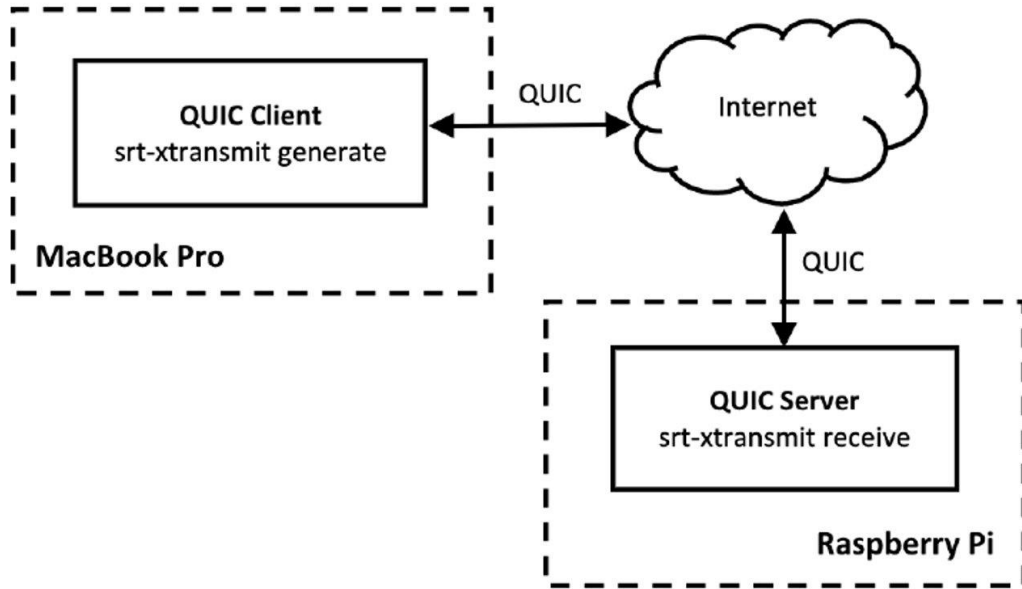


Figure 1: Test setup when sending data via QUIC datagrams.

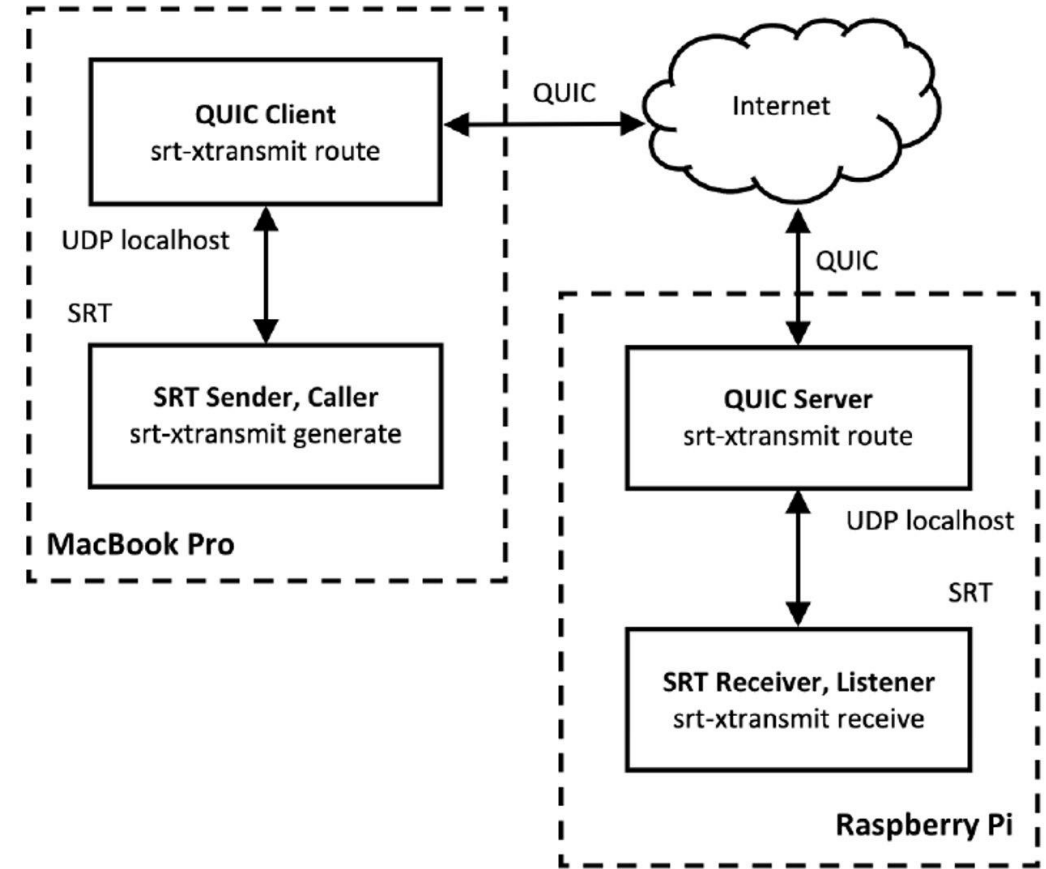


Figure 2: Test setup when tunnelling SRT over QUIC datagrams.



- We chose to limit the generated constant bitrate (CBR) stream to 3 Mbps for both streaming with QUIC datagrams and tunnelling SRT over QUIC datagrams (giving 6 Mbps in total) to ensure that link capacity would be enough for concurrent transmission of both streams.
- Streaming was done simultaneously for each experiment to equally capture the effect of possible network congestion or packet loss in both datasets.

**Table 1: Summary of experiments**

Experiment	SRT Latency	Duration
Experiment 1	400 ms	15 minutes
Experiment 2	600 ms	15 minutes
Experiment 3	800 ms	15 minutes
Experiment 4	800 ms	60 minutes

\* Note that SRT Latency setting was applied for tunnelling SRT over QUIC transmission only



# Streaming via QUIC Datagrams (Experiment 3)

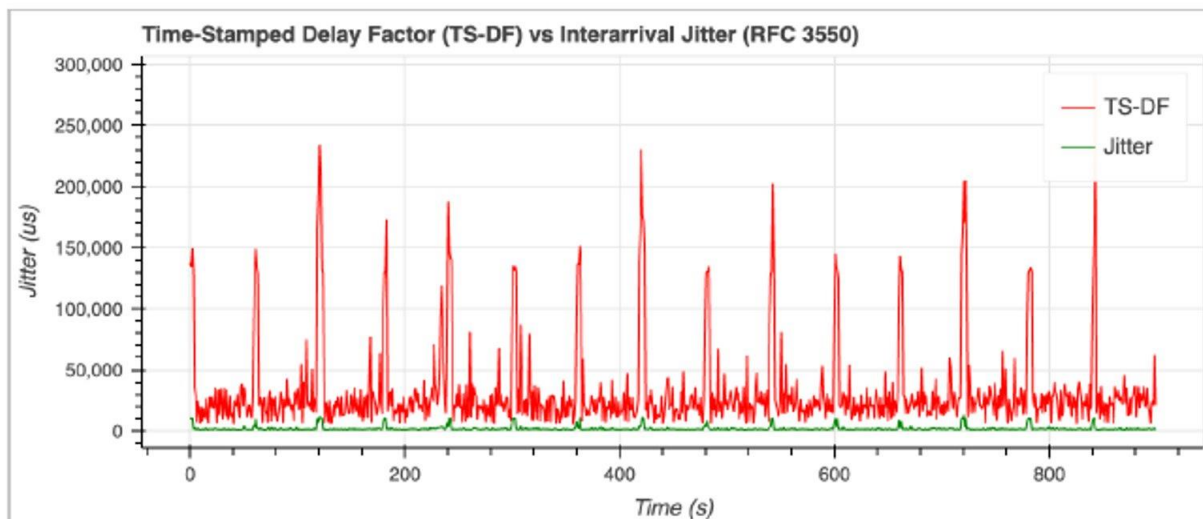


Figure 3: TS-DF vs RFC3550 jitter, in microseconds, observed at the server side when streaming via QUIC datagrams during the 3rd experiment.

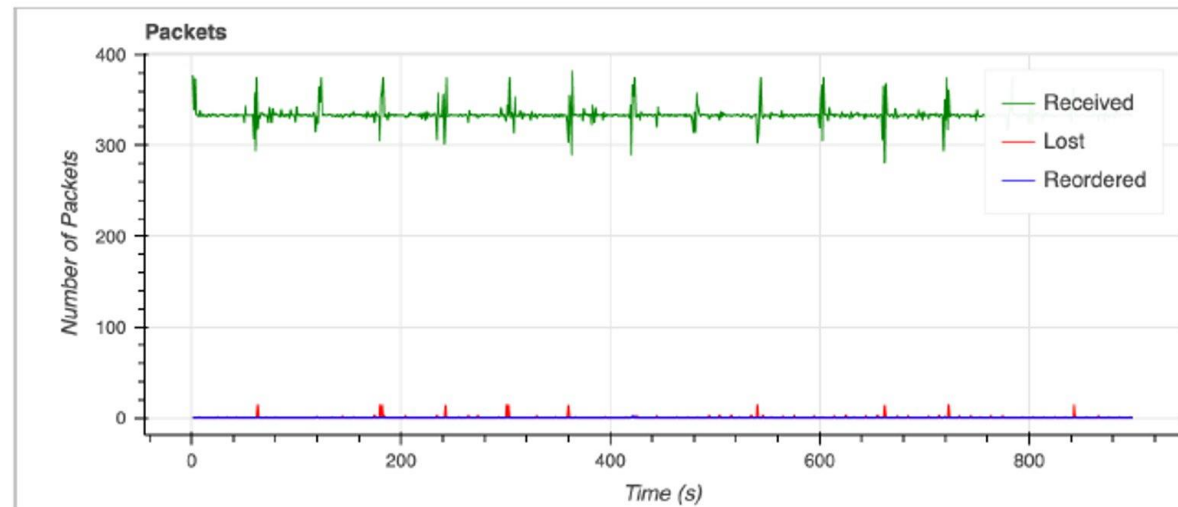


Figure 4: Number of received, lost, and reordered packets observed at the server side when streaming via QUIC datagrams during the 3rd experiment.

Table 2: Percentage of lost and reordered packets per experiment when streaming via QUIC datagrams

Statistic	Experiment 1	Experiment 2	Experiment 3	Experiment 4
Lost Packets (%)	0.001058	0.001219	0.000879	0.001083
Reordered Packets (%)	0.000019	0.000035	0.000016	0.000019

# Tunnelling via SRT over QUIC (Experiment 3)

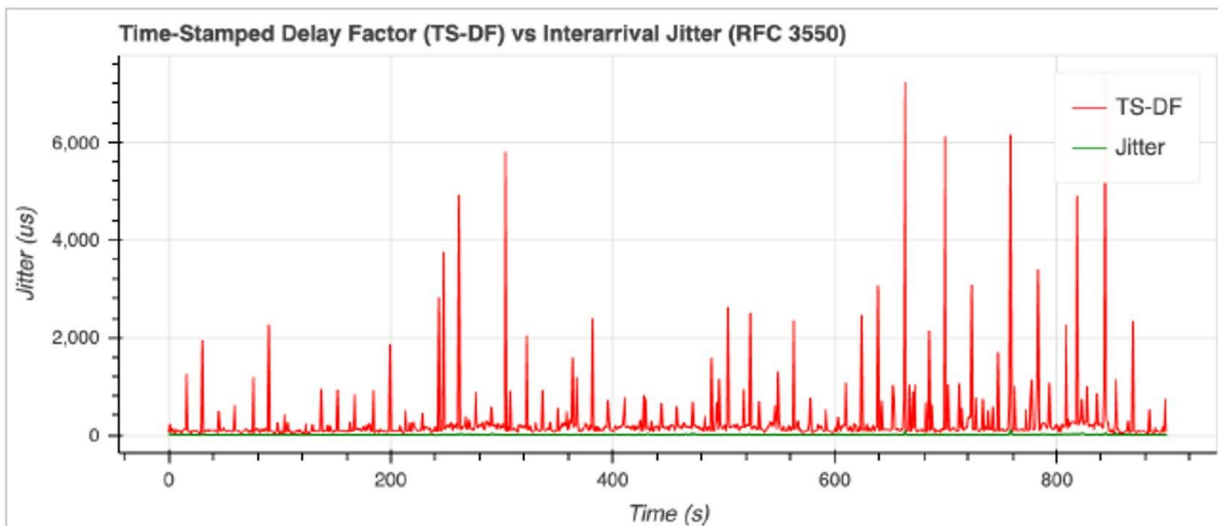


Figure 7: TS-DF vs RFC3550 jitter, in microseconds, observed at the server side when tunnelling via SRT over QUIC during the 3rd experiment.

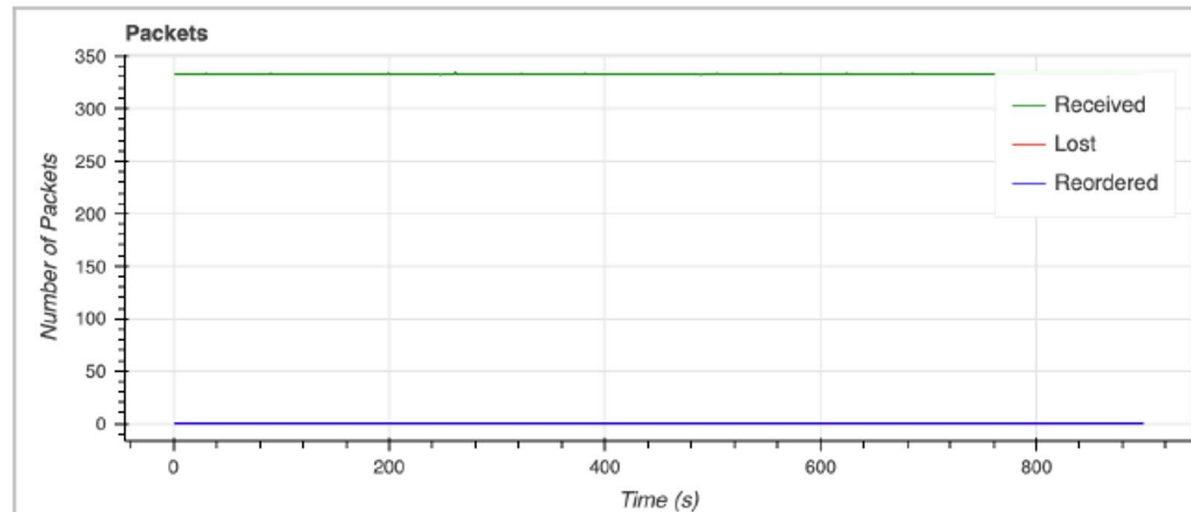
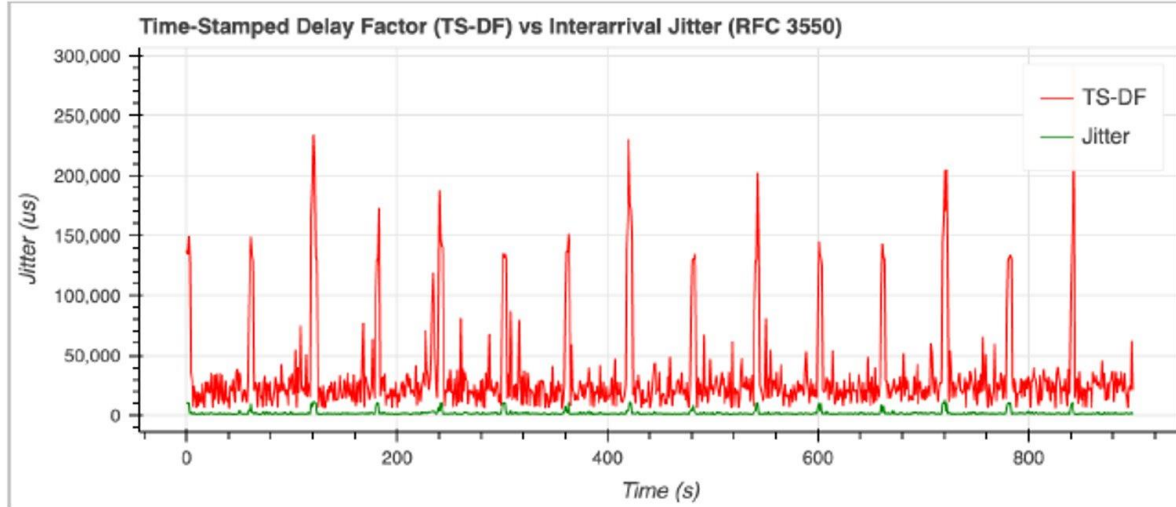


Figure 8: Number of received, unrecovered (labeled "Lost" on the graph), and reordered packets observed at the server side when tunnelling via SRT over QUIC during the 3rd experiment.

Table 3: Percentage of unrecovered and reordered packets per experiment when tunnelling via SRT over QUIC

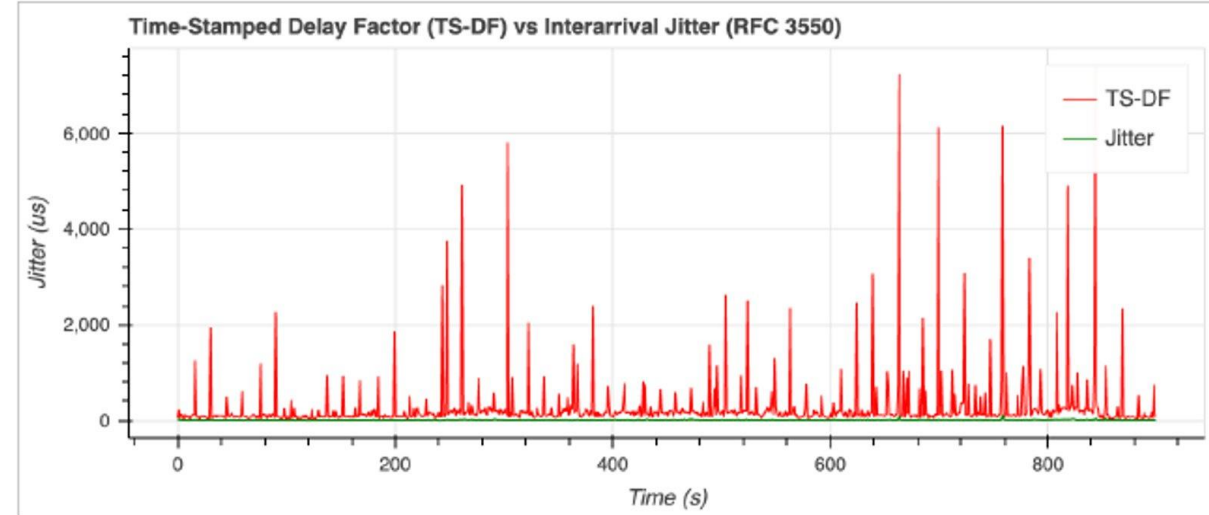
Statistic	Experiment 1	Experiment 2	Experiment 3	Experiment 4
Unrecovered Packets (%)	0.000161	0.000033	0	0
Reordered Packets (%)	0	0	0	0

# Side by Side Comparison (Time-Stamped Delay-Factor)



**Figure 3: TS-DF vs RFC3550 jitter, in microseconds, observed at the server side when streaming via QUIC datagrams during the 3rd experiment.**

QUIC Datagrams  
Average: 33.09 ms  
Spikes up to: 292.19 ms

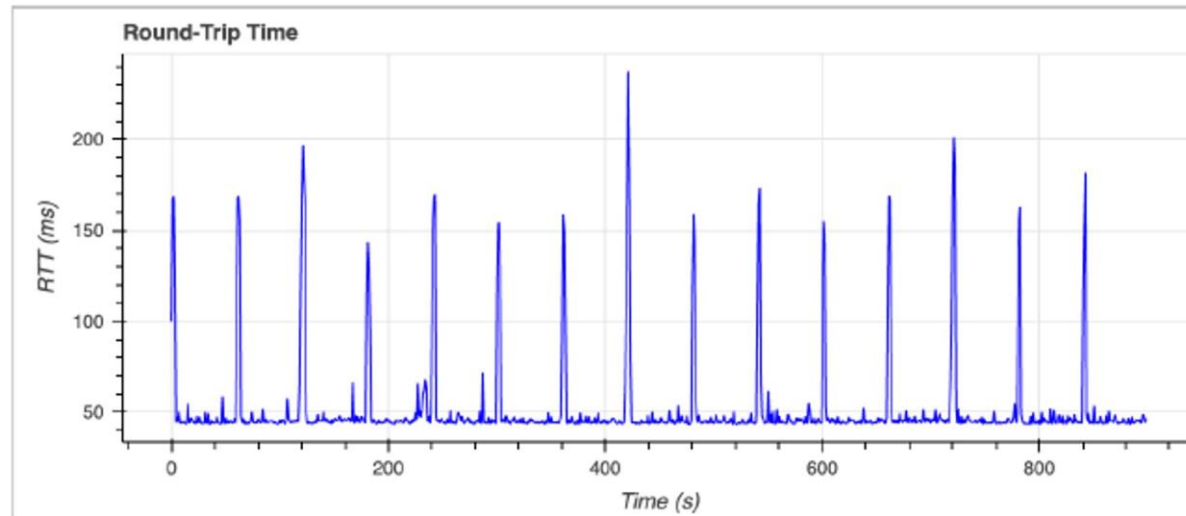


**Figure 7: TS-DF vs RFC3550 jitter, in microseconds, observed at the server side when tunnelling via SRT over QUIC during the 3rd experiment.**

SRT over QUIC Datagrams  
Average: ~0.28 ms  
Spikes up to: ~7.41 ms

## Smoothed Round-Trip Time (Experiment 3)

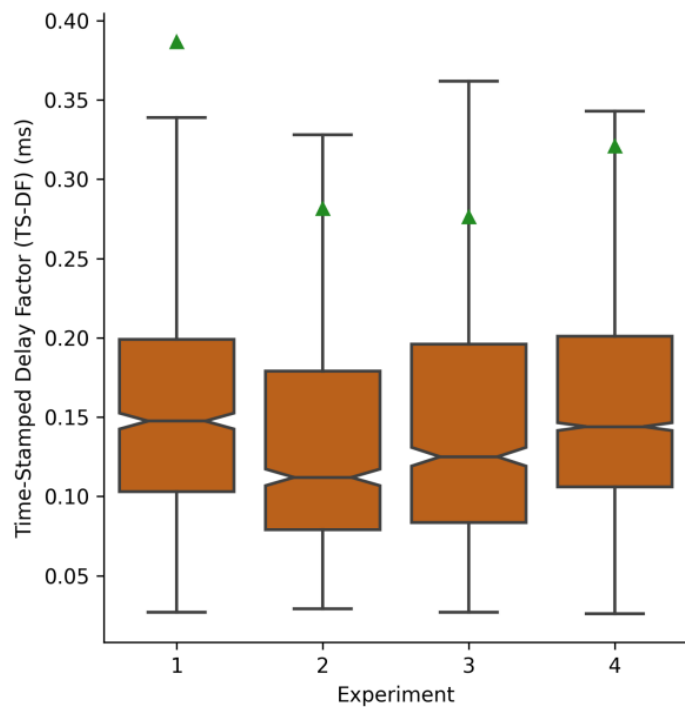
The RTT graph is built from the SRT protocol msRTT statistics observed at the receiver side and includes delay associated with transmission over QUIC datagrams.



**Figure 6: Smoothed round-trip time, in milliseconds, observed at the server side when streaming via SRT over QUIC during the 3rd experiment.**

See also [Examining SRT streaming over 4G connection](#)

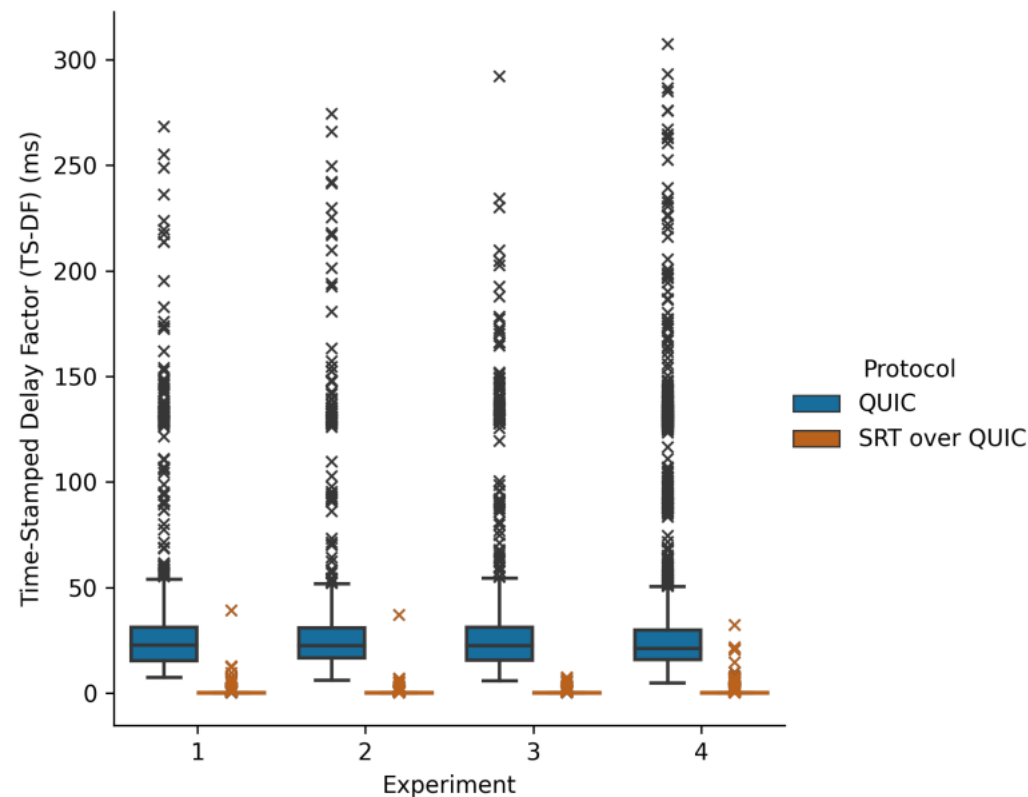
# Time-Stamped Delay Factor for all the Experiments



**Figure 8: Time-Stamped Delay Factor, in milliseconds, observed at the server side in each experiment. Extreme values are marked with crosses, average values are marked with green triangles.**

**Table 1: Summary of experiments**

Experiment	SRT Latency	Duration
Experiment 1	400 ms	15 minutes
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Experiment 4	800 ms	60 minutes



# Conclusions

- The research has shown that live streaming protocols such as SRT can be implemented on top of QUIC datagrams to achieve low latency streaming, while mechanisms such as SRT's latency-aware ARQ-based packet recovery can reduce packet loss.
- The resulting latencies and jitter can be constrained to sub-second values, depending on the network round-trip time.
- Lost packets can be recovered within the configured latency buffer, or dropped when latency boundaries are exceeded.

Get more info & share ideas:

- SRT Protocol Internet-Draft  
<https://datatracker.ietf.org/doc/html/draft-sharabayko-srt-01>
- Tunnelling SRT over QUIC Internet-Draft  
<https://datatracker.ietf.org/doc/draft-sharabayko-srt-over-quic/>
- Blog on Medium  
<https://medium.com/innovation-labs-blog/tagged/secure-reliable-transport>
- SRT Open-source Library  
<https://github.com/Haivision/srt>

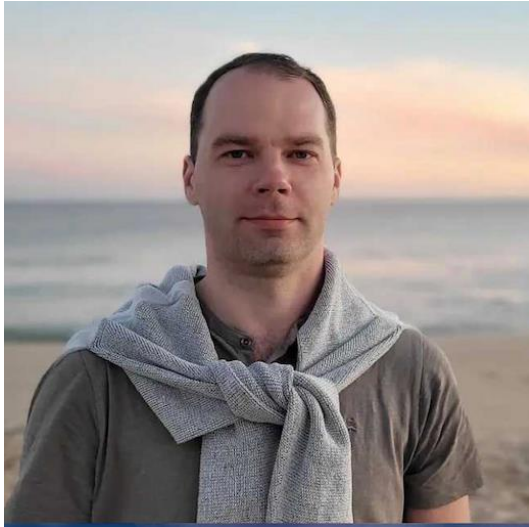
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<https://www.srtalliance.org/>
- SRT Slack Channels  
<https://srtalliance.slack.com/>

To join

<https://slackin-srtalliance.azurewebsites.net/>

Channels: #general, #develop, #quic-srt, #rfc

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