Media Transport Protocols Testing and comparison

Dr. Alex.

Disclaimer

At this stage this presentation is just but an initial draft, with more questions than anything else.

The current goal is to establish the right scope for a fair, and informative test suite. Fair is a key for the data to be usable in a logical decision process. Informative meaning that some question might not be fair, but are still of interest to the streaming community at large, and given the right warnings, corresponding tests should/could be added.

A good outcome of this meeting or presentation is to have some answers to the questions to narrow the scope and reach consensus on that scope. Consensus can be reached on the mailing list.

Versioning and Maturity of specifications

Goal: Strong base for comparison (not a moving target, not subject to interpretations, testing the spec and not the implementation itself)

- There is no standard spec for SRT. We could fix one version with help of the SRT team. The versioning is not a big problem in itself.
- Only one implementation, which prevent testing maturity of the technology through interop. We would practically test the implementation.

Note: a good spec is a specification that is accurate and complete enough that two software implementations from scratch starting uniquely from the specs would be interoperable. Part of the maturation of a spec is to have implementor feedback, separate implementations and interoperability events (see e.g. QUIC).

OSI layer?

Goal: Fair comparison

- TCP/UDP/QUIC are network protocols, they are used by RTP/WS and other higher layer protocols in their stack.
- RTMP/RTP/RTSP are media transport protocols: they only carries media and have core features explicitly dedicated to handle bitstreams.
- HTTP is an application level protocol, but one could argue it is used as a transport protocol by HLS, or Websocket.

Q2: where does SRT stand?

OSI layer?

Goal: Fair comparison - Q2: where does SRT stand?

Gut feeling: not a media transport as the media itself is separated from transport (good or bad is not the question). RTP could actually be send over SRT.

Testing against RTMP or RTP (and RTP-based protocol) is not direct and require fixing the media part externally in an implementation. Not impossible, but the conclusion could not be a comparison between protocols, but a comparison between say RTP and SRT used in a certain configuration. Not fair, but very possibly informative.

OSI layer?

Goal: Fair comparison - Q2: where does SRT stand?

Media transport, because they explicitly connect with media / encoders / decoders, usually have a limitation in the highest resolution, codec, and other media feature they support. E.G. RTMP will not support VP9, H265, AV1 and latest codecs.

Some Media Transport, like RTP, have a mechanism (RTP payload) to add codecs as they become available and keep the other advantages. They can also have a mechanism to extend capacity of the language and stay compliant (RTP Header Extensions).

Pure transport do not deal with media and can carry anything, at the cost of no feedback loop between media and network.

Features to test?

Goal: Fair comparison

- Reliability (full, none, partial), per layer
- Resilience (jitter, delay, packet loss), per layer
- Security (hard to test, better do an exert review. Initial comments by Ekr and others). (hop-by-hop, end-to-end, double layer, ….), per layer
- Firewall / NAT traversal

The per-layer here is important, because often the feature comes from a lower layer. In the original HLS RFC, the security was delegated to the lower layer (HTTP), and so was the NAT traversal.

Features to test? NAT/FW: P2P or C2S use case?

Goal: Fair comparison

The question is not benign. In IETF webTransport it has been acknowledged that p2p and c2s cases have fundamental differences and expectation.

Advanced NAT traversal and firewall traversal (including support for IPv6), are most important for p2p cases. All the work of the IETF ICE group is dedicated to this. Older approach consisted in using TCP or HTTP to achieve this goal, with the corresponding cost associated (network level reliability, CC, BWE, and end-of-line problems).

However, in the c2s case, not all the ICE complexity is needed. It looks that the usage of SRT / RTMP today is mainly first leg / upload to server use case. Last week Microsoft lead QuicTransport Hackathon results should be taken into account as well.

Gut Feeling: SRT vs QUIC

Goal: Fair comparison

it looks like SRT would not be a media transport (with the definition provided before), even though it can transport media.

It looks like it has some advantages over RTMP, however those are not exclusive to SRT, and better alternative exist for some. We could start some tests, but I think it might make sense to wait for Magnus and the group answer with respect to SRT / QUIC respective qualities.

Side question

109? BKK or online?