Managing Real-Time Video and Data Flows with Coupled Congestion Control Mechanism

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WebRTC enables interactive real-time communication between web browsers, facilitating a range of applications such as seamless video conferencing, telephony and interactive gaming. It allows a user to simultaneously transfer media (over the Real-Time Transport Protocol (RTP)) and data (over the Stream Control Transmission Protocol (SCTP)), multiplexed onto a single UDP port pair. When multiple flows are multiplexed over a single UDP port pair, they are normally regarded as a single flow and hence they are treated in the same way; because, routers or other middle-boxes usually identify flows using a five-tuple of source and destination IP addresses, transport protocol, and the transport protocol's source and destination port numbers and send their packets to the same path.

The poor design choice of using two different congestion control mechanisms by two different transport protocols in the WebRTC can lead to a competition between the flows, resulting in undesirable spikes in queuing delay and packet losses. Such competition can be eliminated by using a coupled congestion control mechanism which combines the congestion control mechanisms of all the flows sharing a common path. In [1, 2], we have shown that coupling can significantly improve the overall performance of multiple congestion-controlled Real-time Transport Protocol (RTP) sessions by reducing overall delay and loss and exerting precise allocation of the available bandwidth. However, this mechanism only combines a set of homogeneous congestion control mechanisms and therefore cannot be readily applied to combine the data and video flows in WebRTC, since they use two different congestion control mechanisms: a delay-based congestion control mechanism for media and a loss-based control mechanism for arbitrary data.

Combining a heterogeneous set of congestion control mechanisms can yield several performance benefits, especially when one of the mechanisms reacts to a congestion event earlier than the others. This would be a perfect choice for the WebRTC because the delay-based congestion control mechanism will react to the increasing delay as soon as the queue grows, allowing the coupling mechanism to react to this signal. This will ensure that the queue does not grow even for the loss-based congestion control mechanism.

The overarching objective of this thesis is to extend the coupled congestion control from [3], and propose and implement a solution in the chromium browser that combines a heterogeneous set of congestion control mechanisms. Finally, this thesis will show that this can reduce the negative impact that the data channel can have it on the video channel.

References

[1] Safiqul Islam, Michael Welzl, Stein Gjessing, and Naeem Khademi. Coupled congestion control for rtp media. *SIGCOMM Comput. Commun. Rev.*, 44(4), August 2014.

[2] S. Islam, M. Welzl, D. Hayes, and S. Gjessing. Managing real-time media flows through a flow state exchange. In *NOMS 2016 - 2016 IEEE/IFIP Network Operations and Management Symposium*, pages 112–120, 2016.

[3] Safiqul Islam, Michael Welzl, and Stein Gjessing. Coupled Congestion Control for RTP Media. RFC 8699, January 2020.