RTP Congestion Control: Circuit Breakers for Unicast Sessions
draft-perkins-avtcore-rtp-circuit-breakers-00

Colin Perkins – University of Glasgow
Varun Singh – Aalto University
Background and Goals

• RTP requires applications to be congestion aware, but lacks a standard congestion control algorithm

• Interest in developing and standardising congestion control algorithms for WebRTC
  - Discussion in IRTF ICCRG and IETF RTCWeb WG sessions this week
  - These algorithms are new, and will take time to develop and be validated
  - WebRTC, and other, applications need an immediate safety-net, to allow initial deployment before sophisticated congestion control is developed

• Goal of this draft is to define the envelope within which these algorithms must work
  - Define “circuit breaker” conditions for an RTP session – limits that should not be met under normal operation, and can be used to stop errant flows
Congestion Signals for RTP/AVP Flows

• Potential congestion signals available from RTCP:
  • RTT estimate once per reporting interval
  • Jitter estimate once per reporting interval (limited use for video flows)
  • Fraction of packets lost during the reporting interval, plus cumulative number of packets lost over the entire RTP session

• Not (yet) considering RTP extensions – looking for a baseline mechanism

• Applicability as circuit breakers:
  • RTT/jitter estimates too infrequent to be useful with RTP/AVP timing rules
  • Packet loss statistics too infrequent for adaptation, but useful for detecting overload situations – use as the basis for a limiting condition
Circuit Breakers for Unicast RTP/AVP

• Circuit breaker #1: Timeout
  • RTP data packets being sent, but corresponding RTCP RR packets report non-increasing extended highest sequence number received
  • Indication of significant connectivity problem if persistent for ≥ 2 reporting intervals → cease transmission

• Circuit breaker #2: Congestion
  • RTP data sent, corresponding RR packets have increasing extended highest sequence number received, but non-zero packet loss fraction
  • Indication of network congestion – estimate equivalent TCP throughput:

\[ T = \frac{s}{R \sqrt{\frac{2p}{3}} + \left( t_{RTO} \left( 3 \sqrt{\frac{3p}{8}} \right) p(1 + 32p^2) \right)} \]

where \( t_{RTO} \approx 4R \), and cease transmission if RTP sending rate \( \geq 10T \) for 2 reporting intervals (similar to TFRC definition – see draft for rationale and assumptions)
Open Issues

• Are these appropriate rate limiting conditions for RTP/AVP sessions?

• What is the impact of RTP extensions?
  • E.g., RTCP XR, RTP/AVPF profile, and ECN
Next Steps

• Presentation and further discussion in IRTF ICCRG

• Consider for adoption as a working group draft