A Delay Based Congestion Control Candidate

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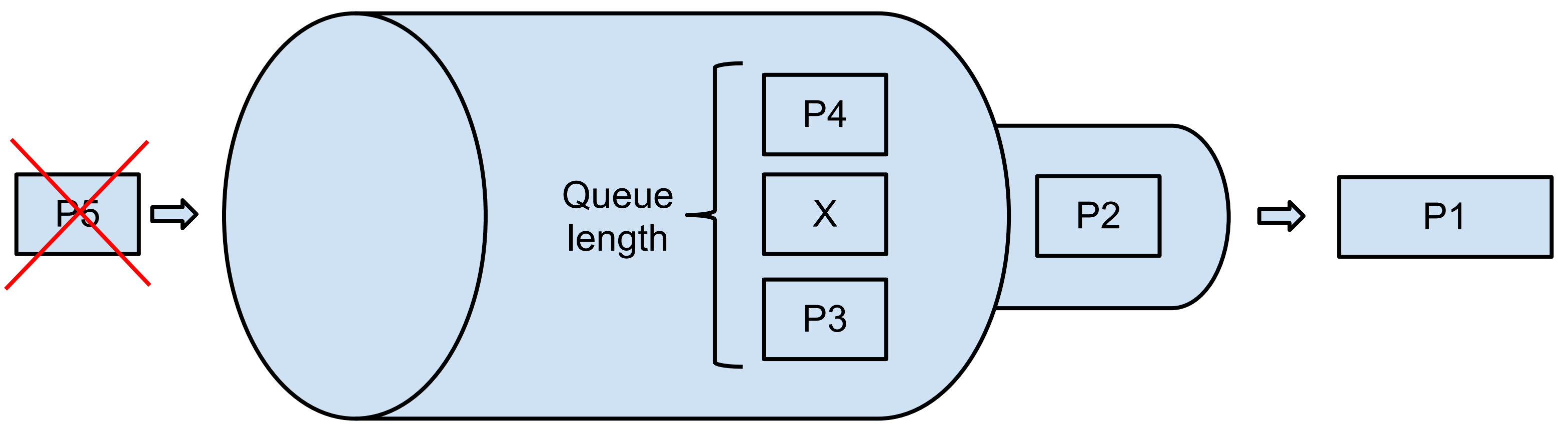
Details in: draft-alvestrand-rtcweb-congestion

Problem

At what bitrate can a sender transmit data while still keeping the end-to-end delay low and avoiding packet loss?

Model - Queuing

- Packets transmitted at a too high rate will be queued.
- The one-way delay increases with queue length.
- If a queue is full, packets are dropped.
  - Other schemes exist



Modeling

Over-using:

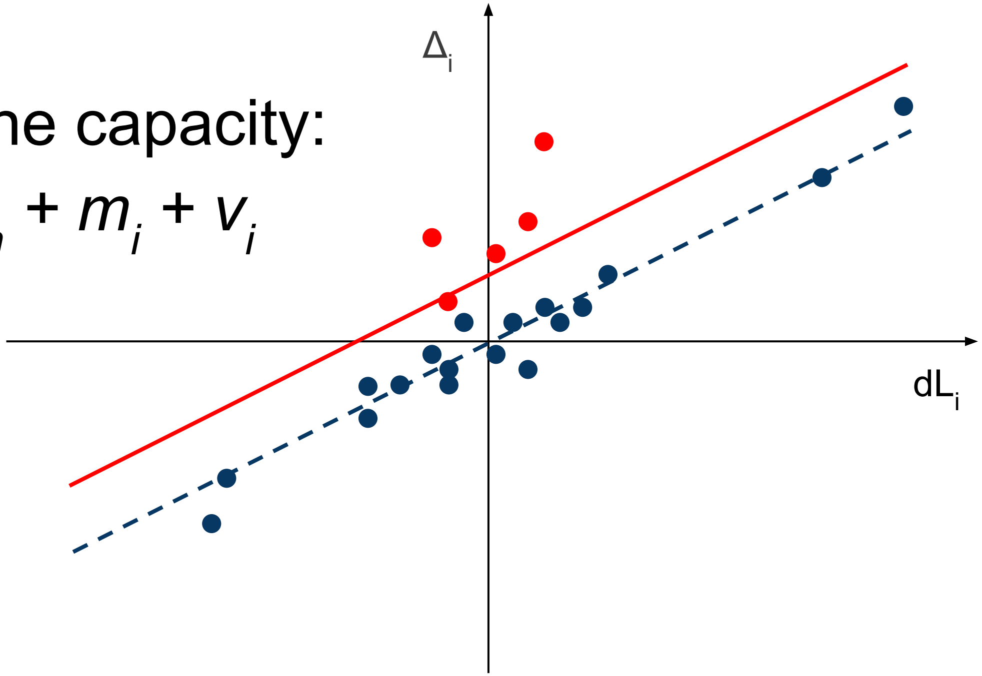
The one-way delay d_i increases as queues are being filled.

$\Delta_i = d_i - d_{i-1} > 0$

Inversely proportional to the capacity:

$\Delta_i = dL_i/C_{min} + w_i = dL_i/C_{min} + m_i + v_i$

$E\{v\} = 0$



Estimation/Detection

- Estimate the slope and offset.
- Most filters will do the job. We chose the Kalman filter.
  - Adaptive, handles random jitter as noise.
- Measure incoming rate when offset > threshold.
- Adjust target rate to some factor of the incoming rate.

Signaling

- Both estimation and control at the receiver.
- Transmit bandwidth estimates to the sender.
- The sender chooses to transmit at any rate < BW estimate. Employs own simpler algorithm to avoid problems of lost feedback messages.

Results

- Been run through a set of emulated network scenarios with success.

- Used in the wild with Chrome's WebRTC implementation.

- Working on getting some performance numbers.

TODO

- AQM/ECN.
- Packet-loss based detection at the receiver.
- Multiple streams.