TFRC for Voice:
the VoIP Variant

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http://www.icir.org/floyd/papers/
draft-ietf-dccp-tfrc-voip-01.txt

http://www.icir.org/floyd/talks.html
First, four repeat viewgraphs from last IETF...
VoIP: fairness in Bps.

• Standard TFRC has the goal of fairness in pps with TCP flows using the same packet size.

• The VoIP variant of TFRC has the goal of fairness in Bps with TCP flows using 1500-byte packets, (following RFC 3714).

• The VoIP variant assumes optimistically that the network limitation is in Bps, not in pps.
VoIP: fairness in Bps.

- In the TCP throughput equation, use the measured loss event rate and a packet size of 1460 bytes.

- Reduce the allowed transmit rate to account for the fraction of the VoIP bandwidth that would be used by 40-byte headers:
  - $X \leftarrow X \times \frac{\text{TruePktSize}}{\text{TruePktSize} + \text{Header}}$
  - TruePktSize = average segment size in bytes
  - Header = 40 bytes

- Enforce a Min Interval between packets of 10 ms.
Measuring Congestion:

- The VoIP variant of TFRC uses the loss event rate.
  - RFC 3714 uses the packet drop rate.

- These are both affected by packet size and by the smoothness of the sending rate.

- The effect of packet size on the packet drop rate could use more investigation.
The VoIP variant of TFRC:

- As it stands now, it sometimes favors the VoIP TFRC flow over the large-packet TCP flow.

- This needs to be quantified and evaluated.
Changes from draft-ietf-dccp-tfrc-voip-00.txt:

- Added more simulations.
- Added a Related Work section.
Small Packets, Standard TFRC:

• The problem:
  – TCP flows with 1460-byte packets, competing against:
  – Standard TFRC flows with 200-byte packets, 10 ms between packets (e.g., 160 Kbps)
• Another set of simulations use TFRC flows with 14-byte packets and 20 ms between packets (e.g., 5.6 Kbps), and a 1.5 Mbps instead of 10 Mbps shared link.
Small Packets, Standard TFRC:

160 Kbps TFRC flows, 10 Mbps link
Small Packets, Standard TFRC:

5.6 Kbps TFRC flows, 1.5 Mbps link
Simulations with VoIP TFRC:

• The algorithm so far in the draft:
  – TCP flows with 1460 bytes, competing against
  – VoIP TFRC flows with 200-byte packets.
Simulations with VoIP TFRC:

Table 1 from draft-ietf-dccp-tfrc-voip-01.txt, 160 Kbps TFRC flows.
Simulations with VoIP TFRC:

5.6Kbps TFRC flow, 1.5 Mbps shared link.
This won’t do.

- **The problem** [Widmer, Boutremans, and Le Boudec, 2004; Vasallo 2000]:
  - TCP and TFRC can’t send multiple packets per RTT in the face of high congestion; VoIP TFRC can.
  - The loss event rate used by TFRC follows TCP by responding to at most one loss per RTT.
  - When TCP has one “loss event” for every two 1460-byte packets, a rate-equivalent TFRC VoIP might have one “loss event” for every 30 100-byte packets.
  - So the “loss event rates” computed by VoIP TFRC can be too low in times of high congestion.

- **One solution:**
  - Loss rates have to be calculated differently, at least in times of high loss.
Solutions:

• Previously-explored solutions [WBB04]:
  – Count virtual packets [V00].
  – Do random sampling of arriving packets.
  – Shorten part of the Loss Interval.

• One possible solution for VoIP TFRC:
  – For short loss intervals (at most two RTTs), count the actual packet loss rate (but don’t increase the number of loss intervals).
  – Somewhat like “random sampling” above, but only for short loss intervals.
  – Examples: “./test-all-friendly HighLossShort” and “./test-all-friendly printLossesShort” in tcl/test in NS.
The Modified TFRC VoIP:

160 Kbps TFRC flow, 10 Mbps shared link.
The Modified TFRC VoIP:

5.6 Kbps TFRC flow, 1.5 Mbps shared link.
Issues remaining:

• More exploration needs to be done.

• There were two problems:
  – VoIP TFRC can send many packets in one RTT, even in the face of heavy congestion. Taken care of.
  – VoIP TFRC, with small packets, sees different packet drops that it would have with larger packets. When is this a problem?

• For simulations with RED in byte mode (where small packets are less likely to be dropped than large packets):
  – Even the modified VoIP TFRC gets much more than its share of the bandwidth in times of high congestion.
  – Under investigation.
Assumptions:

- **The sender doesn’t know** the packet-marking mechanisms used by the routers.
  - E.g., Drop-Tail? Queue in bytes or in packets? AQM in byte mode or in packet mode?
- **The sender can’t assume** that packets or bytes are being dropped with some relatively stable dropping probability p.
  - This is not necessarily the case.
Assumptions:

- The number of packets dropped in one round-trip time is not necessarily an indication of the level of congestion.
Extra Viewgraphs:
Measuring Congestion:

• Packet size in a Drop-Tail world:
  – Queue measured in bytes, packets, or in-between?
  – Smooth or bursty sending rates?
  – High or low levels of statistical multiplexing?

• RED in packet mode:
  – Same packet drop rate for big and small packets.
  – TFRC measures the loss interval in packets.

• RED in byte mode:
  – Same byte drop rate for big and small packets.
The state of TFRC in NS:

• Includes the VoIP variant.

• Includes RFC 3390 initial sending rates.

• More updating is needed.
  – Add RFC 3390 sending rates after idle periods.
  – Add Faster Restart.
  – Add overhead for packet headers.