TFRC for Voice: the VoIP Variant

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March 2005

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)
Small Packets, Standard TFRC:

DropTail, Standard TFRC

Drop Rates

TCP drops
VoIP drops

Number of Flows of each Type

DropTail, Standard TFRC

Throughput (Per Flow)

TCP throughput
VoIP throughput

Number of Flows of each Type
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.
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:

DropTail, Modified Algorithm

Drop Rates

Number of Flows of each Type

Throughput (Per Flow)

Number of Flows of each Type
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.