IAB Concerns Regarding Congestion Control for Voice Traffic in the Internet

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draft-iab-congestion-00.txt

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Concerns about end-to-end congestion control for best-effort voice traffic in the Internet

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- This draft does not recommend any particular deployment path for VoIP in the Internet (e.g., best-effort, QoS, reservations, etc.).
- The draft observes that in fact, some VoIP traffic ends up competing as best-effort traffic with other best-effort traffic over some link in the Internet.
- The draft recommends that such flows with a minimum sending rate should terminate in the presence of sufficiently-high, persistent packet drop rates.
- The draft further observes that adaptive codecs can expand the available range for VoIP.

The Reality:

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- A VoIP flow between the Atlanta IETF and Nairobi, Kenya:
  – 64 kbps plus FEC plus framing.
  – A shared, congested 128 Kbps access link.
  – Good voice quality in the presence of 5-40% drop rates.

- The problems:
  – Congestion collapse;
  – User quality;
  – Fairness.

Developments in the IETF:

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- RTP:

  – RTP Profile for Audio and Video Conferences with Minimal Control

- TFRC:

  – TFRC-PS, still under development, would be for applications with a fixed sending rate but varying packet sizes.

  – DCCP.

- Adaptive Rate Audio Codecs:

  – RFC 3267: Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) audio codecs.

  – iLBC: Internet Low Bit Rate Codec.

  – Ivox: Interactive VOice eXchange.

Minimum Acceptable Sending Rates for Best-Effort Traffic

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- Assume (generously) a network limited in bandwidth, not CPU cycles.
- Consider fairness with TCP flows with the same RTT and 1500-byte packets.
- Take into account packet header size.
- Don't assume that N small packets dropped equals one large packet dropped.

Minimum Acceptable Sending Rates: the details

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- For a VoIP flow at 4.75 kbps, 20 pps, 100 ms RTT:
  – A TCP flow sendng at the same rate in bps would have a persistent packet drop rate between 35 and 40%.

- For a VoIP flow at 64 kbps, 50 pps, 100 ms RTT:
  – A TCP flow sendng at the same rate in bps would have a persistent packet drop rate between 20 and 25%.

Recommendations:

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- (1) In IETF standards for protocols regarding best-effort flows with a minimum sending rate, a packet drop rate must be specified, such that the best-effort flow terminates when the steady-state packet drop rate significantly exceeds the specified drop rate.
- (2) The specified drop rate for the minimum sending rate should be consistent with the use of Tables 1 and 2 as illustrated in this document.

Extra viewgraphs:

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More on minimum sending rates.

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- For the heavy packet drop regime, the standard TCP response function overestimates TCP's sending rate.

- The standard TCP response function:
  – For a packet drop rate of 50%, a sending rate of 0.1 ppr.

- From simulations:
  – For a packet drop rate of 50%, a sending rate of 0.018 ppr.

(For an RTO set to twice the RTT.)