RTCP Extensions for VoIP

- Draft-clark-avt-rtcpvoip-00.txt
- Provides capability to send voice quality related metrics from VoIP endpoint to
  - Remote endpoint – for active control
  - Midpoints – for monitoring/test/fault isolation

Why RTCP Ext for VoIP?

- VoIP very sensitive to distribution/ time varying nature of impairments
- Endpoint contains active/adaptive functions such as jitter buffer, packet loss mitigation – easiest to incorporate these effects accurately in the endpoint itself
- RTCP does not provide enough information to accurately estimate voice quality

Issues

- Based on Friedman RTCP report extensions draft but uses alternative description of packet loss distribution
  - Friedman – RLC encoded (complete but needs processing to determine metrics)
  - Clark – Burst/Gap length and density based on 4-state Markov model (compact, descriptive)
- Proposed frequency of sending RTCP reports – 20-30 seconds?

Issues

- Metrics
  - Loss rate, Discard rate (from jitter buffer)
  - Burst length, density, gap length, density
  - Round trip delay, end-system delay
  - Signal level, echo level, noise level, distortion
  - Voice quality metrics – VoIP segment R Factor and MOS, External network R (if relevant)
    - Note combine R's for different network segments use $R = Ro - (Ro - R_1) - (Ro - R_2)$ assuming Ro's are the same, to estimate overall R

Request

- Comments?
  - Suitability/ completeness of metrics
  - Implementation issues