RFC 2833bis

Henning Schulzrinne
Columbia University
hgs@cs.columbia.edu
AVT working group
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Overview (-02)

- Goal: carry telephony signals as "named events"
  - DTMF
  - trunk events (e.g., MF R1, R2)
  - subscriber line tones (dial tone, busy, …)
  - fax and low-rate data (e.g., V.21)
- can be considered a highly-compressed special purpose audio codec
- uses RFC 2198 for redundancy, as well as triple transmission
- also tone generator (frequencies)

Changes since -00

- formal notion of state (ABCD, hook)
  - zero duration value
- long events (> 8 s), composed of multiple events
  - absence of gaps and E bit signals long event
- clarify which tones can have audio volume
- additional data tones
- ANS signals clarified

Open issues

- Need clear description of MF R1 signals
  - e.g., same signal (1300/1700 Hz) = KP2, S2, ST2, ST2P, KP2P → causes confusion
- V.21 data (for fax negotiation) currently very inefficient:
  - whole packet for each bit → 384 bits/V.21 symbol
  - with RFC 2198 → 64 bits/symbol
  - with multiple named events/RTP packet packing → 32 bits/symbol
  - define new PT that's just raw bits?
    - don't need volume, duration (implied), E bit
    - can we define one payload type for all "voice data" with different (dynamic) sampling rates, rather than one per V.x?

Open issues: general

- Mission creep – 2833bis is not an MGCP or SIP replacement
  - also not a PWE replacement
- Interoperability matrix started at SIPit
  - Robert Sparks has started draft
  - Nicholas Cutaia and Art Allison volunteered earlier
  - need additional interop reports
  - anybody implementing tones?