RTP packetization for text conversation. ”RFC 2793bis”

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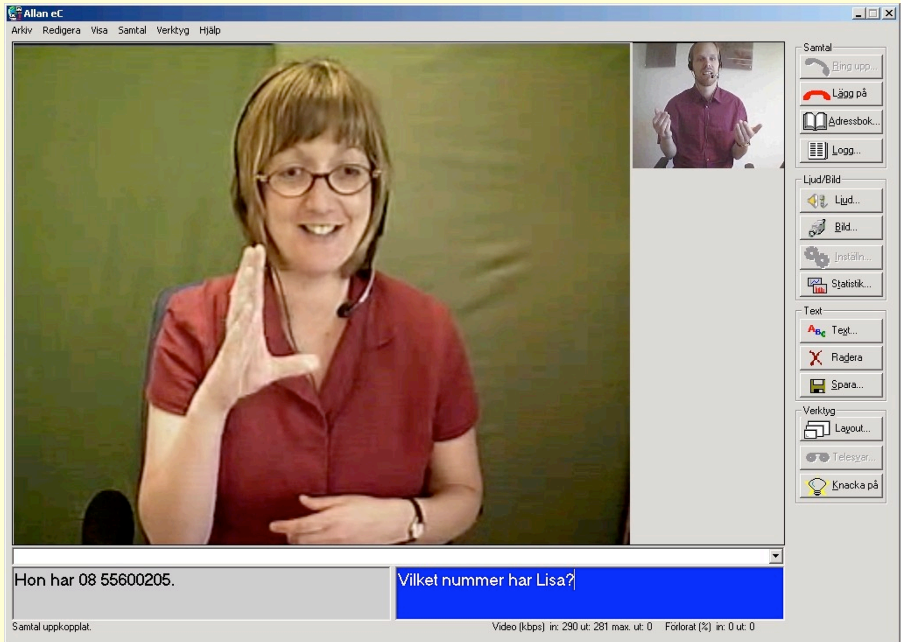
RFC 2793 text/t140 needs a revision

- Published 2000
- Used for real time character by character text conversation transmission during call.
- Used in SIP and H.323 and megaco/H.248.2 and 3GPP
- Mature and works fine
- Enables calls
  - In text only
  - Combined with
    - Voice
    - Video
- Accessible conversation for all



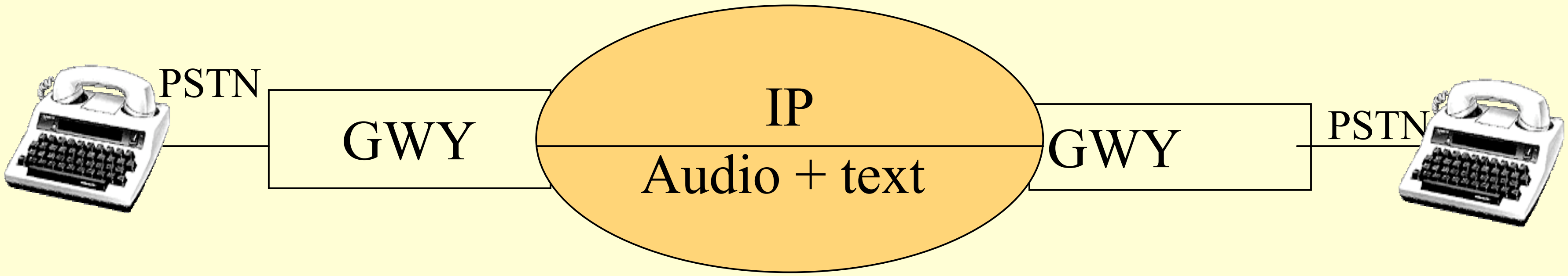
RFC2793 functions and Modifications

- http://www.ietf.org/internet-drafts/draft-hellstrom-avt-rfc2793bis-02.txt

- Straightforward rtp packetization of T.140 text
- Optional use of RFC2198 redundancy
- Ambiguities removed in timing procedures
- Default transmission frequency 300 ms
  - Limits overhead, maintains real time feeling
- Default redundancy usage: 3 redundant generations
  - Meet reliability goals in lossy networks. (char loss is marked)
- SDP examples included. Assure interoperability.
  - Acknowledged use of RTP/AVP even for real time text
- MIME registration of media type text/red

Addition for transit gateways

- Text in PSTN is transmitted by modem tones.
- IP transit gateways for PSTN traffic must transmit text correctly. Audio coding in IP is unreliable. 0.2% packet loss gives 1% character corruption.
- RFC2793 is already bit-economic and reliable
- But multiplexing with audio is wanted for large gateways.



Audio/t140 addition for transit gateways

- New MIME subtype audio/t140
  - Enable Payload Type multiplexing with audio
- Sequence numbering in data. Enable loss detection in multiplexed session
- New fmtp parameter cps= for limiting transmission rate from peer gateway
- Same timestamp clock rate as for audio may be used

RFC2793bis dependencies

- RFC2793bis is needed
  - For gateway work in ITU-T SG 16 in feb 2004.
  - For sip based text over ip framework proposed to sipping
  - Asap, to ensure interoperability in implementations by giving sdp examples