RTP packetization for text conversation. "RFC 2793bis" and related specifications
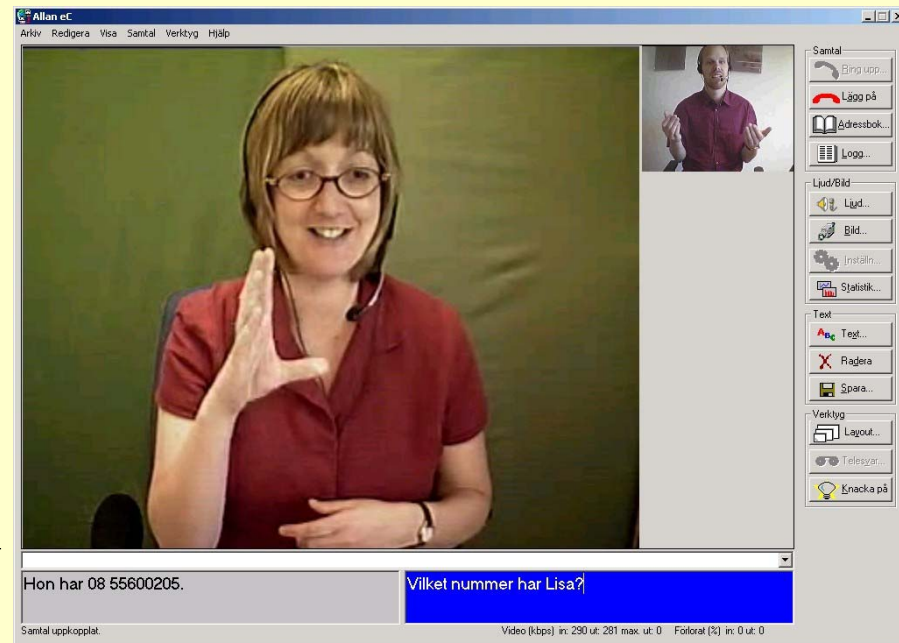
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IETF 59, march 2004

RFC 2793 text/t140 is revised

- Originally 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

RFC2793bis modifications –01 -- 02

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

1. Abstract: You need to remove the [1] reference.
<GH<Done

2. Section 1: There is a MUST in the introduction. It should be changed
to lower case must of two reasons.
<GH<Done

3. Section 3.2: Last sentence of second paragraph. "This T140block
counter may be   utilized to detect lost blocks." change this to:
"This T140block counter is intended to be utilized to detect lost blocks and
   avoid duplication of blocks."
<GH<Done

4. section 3.3, last sentence: Is this SHOULD correct? Why does the
paragraph implies that CCS SHOULD be place in one block, why not MAY or
SHALL?
<GH<No, SHOULD is suitable here. Nothing drastic happens if part of a CCS is
lost, and it depends a lot on the structure of the transmitter if it is
feasible to expect transmission of CCS in one block.

RFC2793bis modifications

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5. The robustness features of the draft. The robustness features could be collected into a single chapter.
<GH<I think the other way, it is good to have all code examples in one
chapter, and to get the information about robustness while you are reading
about the block basics. - No action.

6. Section 3.6: It should be more explicit about the fact that
the rendering of (reasonably) late blocks is recommend and very beneficial.
<GH<Done

7. Section 3.6: End of last paragraph: recommendation that voice is discarded rather than the
    T.140 text when going from IP to PSTN at a gateway. However I think it
   should reflect somewhat on the effects on the audio also.
<GH<Done

8. Section 3.7: Timestamp. "For audio/T140, the clock frequency MAY be set to any value. If not
   specified by out of band mechanism, the frequency value is generally set to be 8000 Hz, as
   that is most common for audio." There needs to be strict rules on what to do, otherwise you
   are in trouble. I think that if not out-of-band signalling, where the interleaved audio
    can be matched in rate, a fixed rate shall be set. There is also missing a sentence saying:
   "When using this payload format interleaved with an audio codec, the rate SHOULD be chosen to
   be the same as the audio codec(s) to avoid RTP timestamp rate switching."
<GH<Done, but in version -02 an incomplete sentence was creatyed by mistake. Will be correcteed
    in -03.

RFC2793bis modifications

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9. Section 3.7: Missing marker bit definition. Even if not used it MUST be defined as being not used and set to 0.

<GH<Done

10. Section 3.8: I think there are no reason to have a section called "additional headers".It is mostly about redundancy anyway, please rename.

<GH<Sorry, missed that. Will be called "Structure of redundant data" in -03

11. section 3.9, second paragraph. This paragraph is very hard to interpret.

<GH<Simplified

12. In regards to the robustness options. As we now have available mechanism to report packet losses, RFC 3611, there is now possibilities to do reactive repairs. I think the possibility should at least be motivated.

<GH<Done

13. Section 5, Second paragraph: "To control the character transmission rate, the "fmtp" attribute [7] is used with the following syntax:" This sentence should be changed to explain that; first CPS is a MIME parameter, secondly that this is how it is mapped according to section x to SDP.

<GH<Done.

RFC2793bis modifications

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14. Section 6.3: The RED SDP examples. Why does one have a=fmtp lines like this?
<GH<This is how RFC2198 is specified in SDP

15. Section 7: I think the security section can be improved:
- Separate the three main issues, confidentiality, integrity, and source authentication. They are different, and have different severity and attacks, and different counters.
- The section should give recommendations to mechanism that allows one
to counter these problems.
<GH<Done

16. Section 8: Should be titled "IANA Consideration". And the
introduction to the chapter should request that the two mime types are registered.
<GH<Done

17. Section 8.1, and 8.2: The registration should mention "This type is
only defined for transfer via RTP."
<GH<Done

18. Section 8.1, and 8.2: Security consideration: Please point at
section 7 of RFC XXXX.
<GH<Done

RFC2793bis modifications

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

19. Section 8.2: The rate parameter for audio should contain the rule that it SHOULD be matched
    with interleaved audio codecs.
<GH<Done

20. Section 11: Are really all references normative. I would expect that 5, 6, and 8 are
    informative. I have however not looked closely at it.
<GH<They look normative.

21. As it seems there will be some more RFC editors notes, they may be moved into a separate RFC-
    editor section instead of being spread through the document.
<GH<done.

22. <GH<The explanation on the applicability of the two formats text/t140 vs
audio/t140 were requested to be refined to more clear show the difference
and application area. Sections 3.1 and 3.2 have been amended. Comments
already received on the new wording in section 3.2 in version -02 indicate
that there is a need for further adjustment to give a balanced view of the
application area of the audio/t140 format for IP transit of text between
PSTN equipment.

23. Layout improved, no block layout is divided by pages.

Proposal: Recommend last call

Additional documents related to interactive text conversation

draft-ietf-avt-text-red-01.txt

MIME Registration of the text/red media subtype for text with redundancy as RFC2198

Author: Paul Jones, CISCO

-00 version issued right after ietf 58 meeting

Now, minor adjustments.

Add the following into the bullet

about how to handle the "pt" parameter in section 4.

"Any dynamic payload type in the list, SHALL NOT include its

content-type, only the payload type number. The mapping of payload types to the content-type
is done using the normal SDP procedures with "a=rtpmap". "

<GH> Done

Proposal: Recommend Last Call

Additional documents related to interactive text conversation

- **draft-manyfolks-sipping-toip-01.txt**

A requirements spec for SIP and PSTN text communication with RFC2793 for text.

- **draft-sinnreich-sipdev-req-03.txt**

A draft for SIP telephone device requirements.

Includes requirement for RFC2793 for text

- **draft-ietf-mmusic-sdp-new-16.txt (not available ?)**

SDP used in SIP calls. The text medium is introduced.

- **draft-ietf-avt-rfc2833bis-04.txt**

The "DTMF" transport spec. Refers to RFC2793 for text data transport.

Conclusion

- The avt documents that specify interactive text conversation payloads are an important prerequisite for real time text to gain status as a normal part of a telephone call.

- IP based telephony can be accessible for all and fulfill urgent user needs

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