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T.140 Text Conversation over Data Channels
draft-schwarz-mmusic-t140-usage-data-channel-03.txt

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Abstract

   This document specifies how the ITU-T Protocol for multimedia
   application text conversation (Recommendation ITU-T T.140) can be
   instantiated as a data channel sub-protocol, using the SDP
   offer/answer exchange-based external negotiation defined in [I-
   D.ietf-mmusic-data-channel-sdpneg]. Two network configurations are
   documented: a WebRTC end-to-end configuration (connecting two T.140
   over data channel endpoints), and a gateway configuration
   (connecting an T.140 over data channel endpoint with an T.140 over
   RTP/UDP endpoint).

Table of Contents

1. Introduction

1.1. Motivation

Recommendation ITU-T T.140 [ITU-T T.140] defines a protocol for text
conversation, also known as realtime text or text telephony. The
native transport for IP networks is based on the Real-time Transport
Protocol (RTP; see [RFC4103]) due to its inherent realtime
characteristics, similar as conversational audio and video services.

WebRTC text conversation is based on T.140, but considered as "data traffic" component (despite the fact of the native RTP-based transport in non-WebRTC environments), see [I-D.ietf-rtcweb-data-channel].

  NOTE - The decision to transport realtime text over a data channel in WebRTC (instead of RTP based transport) is constituted by use case "U-C 5: Realtime text chat during an audio and/or video call with an individual or with multiple people in a conference", see clause 3.2/[I-D.ietf-rtcweb-data-channel].

This document defines the SDP-based negotiation and transport of the T.140 protocol over data channels, where a data channel is a bi-directional communication channel running on top of SCTP/DTLS (as per [I-D.ietf-rtcweb-data-channel]) and where T.140 is instantiated as a sub-protocol of this data channel.

Considering an T.140 endpoint being an T.140 application that uses data channel from WebRTC specifications [I-D.ietf-rtcweb-data-channel], this document describes two configurations where the other endpoint is respectively either another T.140 over data channel endpoint (e.g., a WebRTC application) or an T.140 endpoint using native RTP transport.

1.2. Framework of WebRTC data applications

There are multiple IP application protocols which using WebRTC data channels as transport, such as MSRP or BFCP besides T.140. The SDP-based indication and negotiation of such WebRTC data applications at call control signalling level follows common principles. The first WebRTC application from this suite is/was the MSRP-based instant messaging service for WebRTC, see [I-D.ietf-mmusic-msrp-usage-data-channel]. This specification for T.140 was derived from that document and uses an aligned clause structuring.

It may be noted that the T.140 protocol as such is much simpler in comparison to the MSRP, which requires an extensive set of SDP elements (in comparison to T.140) for the description of specific MSRP services and their protocol parameter settings.

2. Conventions

2.1. Prescriptive language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC-2119 [RFC2119].

2.2. Notation

The brief notation "T.140" is used as a synonym for the text
conversation protocol according to [ITU-T T.140].

3. Terminology and abbreviations

3.1. Terminology used

This document uses the following terms:

Data channel: A WebRTC data channel as specified in [I-D.ietf-
rtcweb-data-channel].

T.140 data channel: A data channel specifically used to transport
the text and presentation control information of one T.140 session.

External negotiation: Data channel negotiation based on out-of-band
or in-band mechanisms other than the Data Channel Establishment
Protocol specified in [I-D.ietf-rtcweb-data-protocol].

In-band: Transmission through the peer-to-peer SCTP association.

Out-of-band: Transmission through the call control signaling path,
e.g., using JSEP [I-D.ietf-rtcweb-jsep] and the SDP Offer/Answer
model [RFC3264].

Peer: From the perspective of one of the agents in a session, its
peer is the other agent. Specifically, from the perspective of the
SDP offerer, the peer is the SDP answerer. From the perspective of
the SDP answerer, the peer is the SDP offerer

3.2. Abbreviations used

DS0   Digital Signal 0 (1 x 64 kilobits per second)

DTLS   Datagram Transport Layer Security

GCP   Gateway Control Protocol

ITU-T   International Telecommunication Union Telecommunication
          Standardization Sector

IWF   Interworking Function

JSEP   Javascript Session Establishment Protocol

MG   (H.248) Media Gateway

MGC   (H.248) Media Gateway Controller

PDU   Protocol Data Unit

RTP   Real-time Transport Protocol

SCTP   Stream Control Transmission Protocol

SDP   Session Description Protocol

SIP   Session Initiation Protocol

TCP   Transmission Control Protocol

TDM   (Synchronous) Time Division Multiplexing

ToIP   Text over IP

TLS   Transport Layer Security

UA   User Agent

UDP   User Datagram Protocol

VBD   Voiceband Data

4. Prinicples

4.1. T.140 Data Channel

In this document, an T.140 data channel is a data channel for which
the instantiated sub-protocol is "t140", and where the T.140-related
negotiation is done as part of the SDP-based external negotiation
method defined in [I-D.ietf-mmusic-data-channel-sdpneg].

   NOTE - This WebRTC term of a "T.140 data channel" is actually
   synonym to the originally introduced concept of a "T.140 data
   channel"] for the T.140 protocol back in 1998, see [ITU-T T.140],
   clause 4.3.

4.2. Session Mapping

In this design, the T.140 session maps to the SCTP association and
the "SCTP stream pair" assigned to the data channel, and each T.140
session maps to one data channel exactly.

5. End-to-End Configuration

   This section describes the network configuration where each T.140
   endpoint is running T.140 over a data channel.

5.1. Basic T.140 Support

5.1.1. Session Negotiation

5.1.1.1. Use of dcmap Attribute

   The SDP offer SHALL include a dcmap attribute line (defined in [I-
   D.ietf-mmusic-data-channel-sdpneg], within the media description for
   the SCTP association for each T.140 data channel session to be
   negotiated.

   The attribute includes the following data channel parameters:

   o "label=" labelstring

   o "subprotocol=" "t140"

   The labelstring is set by the T.140 application according to [I-
   D.ietf-mmusic-data-channel-sdpneg]. The max-retr, max-time and
   ordered parameters SHALL not be used.

   Rest of the SDP offer/answer procedures are per [I-D.ietf-mmusic-
   data-channel-sdpneg].

   The following is an example of the dcmap attribute for an T.140
   session to be negotiated (on default SCTP port 5000) with stream=3
   and without any label:

   a=dcmap:3 subprotocol="t140"

5.1.1.2. Use of dcsa Attribute

   The SDP offer MAY also include a dcsa attribute line (defined in [I-
   D.ietf-mmusic-data-channel-sdpneg]) within the media description for
   the SCTP association for each T.140-specific SDP attribute to be
   negotiated for each T.140 data channel being negotiated.

   The T.140-specific items that can be negotiated include at least
   following attribute:

   o defined in [RFC4103], clause 6: the media format parameter
      "character transmission rate", indicated with "cps".

   NOTE - The "cps" parameter is optional and only required for
   specific network interworking cases, see clause 6/[RFC4103]. It is
   expected that this SDP parameter is for instance not required for
   end-to-end text/T.140 data channels between two or multiple WebRTC
   clients, but might be for instance required in case of
   interworking with PSTN text telephony (see clause 6.2.4).

   The SDP answer SHALL include zero or more corresponding dcsa
   attribute lines for each negotiated T.140 session, according to the
   T.140-specific attribute negotiation rules in the corresponding
   specifications.

   A new SDP offer/answer MAY update the T.140 subprotocol attribute(s)
   while keeping the same subprotocol a=dcmap description.

5.1.1.3. Example SDP Negotiation

   The following is an example of an "m"-line for data channels in an
   SDP offer that includes the attributes needed to establish a T.140
   session:

        m=application 911 <L4>/DTLS/SCTP webrtc-datachannel
        c=IN IP4 11.9.19.65
        a=max-message-size:1000 ; "much smaller than e.g. MSRP"
        a=sctp-port 5000
        a=dcmap:1 label="text conversation";subprotocol="t140";
                ordered=true;max-time=???;max-retr=??? ; NOTE 1
        a=dcsa:1 fmtp:- cps=30 ; NOTE 2

     NOTE 1 - Maximum number of retransmissions = ??? (because FIXTHIS
     <Gunnar>);
             Maximum retransmission timing   = ??? (because FIXTHIS
     <Gunnar>).

     NOTE 2 - "the RFC4103 default value as example".

5.1.2. Session Opening

   [ITU-T T.140], clause 6.1 describes the generic T.140 session
   control functions at high-level and a signalling protocol
   independent manner. WebRTC-embedded T.140 sessions uses the data
   channel established for this T.140 session by the generic data
   channel opening procedure defined in [I-D.ietf-mmusic-data-channel-
   sdpneg].

As soon as this data channel is opened, the T.140 session is
actually in the (virtual) state "data transfer ready".

NOTE - The user plane protocol T.140 itself is stateless. The
(T.140)-session-endpoint could be abstracted by a two-state model
from signalling plane perspective (with states "Idle" and "Data
Transfer Ready").

5.1.3. Data Framing

Each T140block is sent on the corresponding SCTP stream using
standard T.140 transmission procedures, as defined in [ITU-T T.140],
with each T.140 chunk delivered in a single SCTP user message.

NOTE - A T140block as service data unit represents one or multiple
characters according to the syntax of clause 8 in [ITU-T T.140].

5.1.4. Data Sending and Reporting

Data sending and reporting procedures SHALL conform to [ITU-T
T.140].

5.1.5. Session Closing

Closing of an T.140 session SHALL be done using the generic data
channel closing procedure defined in [I-D.ietf-mmusic-data-channel-
sdpneg].

The port value for the "m="-line SHOULD NOT be changed (e.g., to
zero) when closing an T.140 session (unless all data channels are
being closed and the SCTP association is no longer needed), since
this would close the SCTP association and impact all of the data
channels.

The SDP answerer SHALL ensure that no dcmap or dcsa attributes are
present in the SDP answer if no corresponding attributes are present
in the received SDP offer.

5.2. Support of T.140-specific Functions

In general, the recommended procedures of clause 5 from [RFC4103]
SHOULD be applicable as well for SCTP-based transport of T.140.

NOTE - [RFC4103] defines T.-140-over-RTP: the procedures of
clauses 5.1 and 5.2 are transport protocol independent, hence

valid for SCTP too. Clauses 5.3 and 5.4 are RTP specific due
unreliable transport, hence not expected to be relevant for SCTP
(to be confirmed).

6. Gateway Configurations

6.1. Introduction

This section describes the network configuration where one T.140
endpoint uses data channels as T.140 transport, the other T.140
endpoint uses a different protocol stack. There is consequently an
interworking function (IWF) in the media plane required, associated
with correspondent interworking in the signalling plane. Such type
of IWFs are provided by gateway devices, given here by WebRTC
gateways as defined by [I-D.ietf-rtcweb-gateways], [ITU-T H.248.94],
[3GPP 29.334], etc.

6.2. Gateway-embedded Interworking Functions for T.140

There are a plethora of interworking functions knowns for T.140
[ITU-T T.140] due to the long history and usage of this service in
legacy packet-switched and circuit-switched networks. Some examples
are indicated below.

Editor's note: clause 6.2 should be moved in an informative
Appendix. Normative scope of this draft is 6.2.1 only.

6.2.1. WebRTC T.140 text to IP text telephony in text conversation mode

Service "IP text telephony in text conversation mode":

   Transport service according to [RFC4103].

Media plane IWF:

   "T.140-over-SCTP stream-over-SCTP association-over-DTLS-over-L4"
   to "T.140-over-RTP/UDP"

6.2.2. WebRTC T.140 text to IP text telephony in text relay mode

Service "IP text telephony in text relay mode":

Transport of T.140 over IP networks, according to [ITU-T V.151].
Also known as Text-over-IP (ToIP).

Media plane IWF:

"T.140-over-SCTP stream-over-SCTP association-over-DTLS-over-L4"
to "T.140-over-IP TLP-UDP".

NOTE: [ITU-T V.151] uses the concept of an "IP Transport Layer
Protocol" (IP-TLP) above the native IP transport layer, which
again allows different framing protocol options (such as RTP-based
or SPRT-based framing (Simple Packet Relay Transport, see Annex B
of [ITU-T V.150.1])).

6.2.3. WebRTC T.140 text to IP text telephony in text pass-through mode

Service "IP text telephony in text pass-through mode":

Transport of analogue PSTN text telephones signals over PCM
encoded voice over IP networks, according to [ITU-T V.152]. Also
known as Voiceband-over-IP (VBDoIP).

Media plane IWF:

"T.140-over-SCTP stream-over-SCTP association-over-DTLS-over-L4"
to "text/modem-over-G.711-over-RTP/UDP".

NOTE: This is the VBD mode according to [ITU-T V.152], using ITU-T
G.711 as VBD codec (i.e., a different codec configuration as in
comparison to G.711 as audio codec).

6.2.4. WebRTC T.140 text to PSTN text telephony

Service "PSTN text telephony":

Transport of analogue PSTN text telephones signals over a)
analogue lines or b) circuit-switched 64-kbit/s channels.

Media plane IWF:

"T.140-over-SCTP stream-over-SCTP association-over-DTLS-over-L4"
to "text/modem-over-DS0/TDM" (in case of circuit-switched
networks).

6.3. **Gateway configuration and procedures in more detail**

   The signalling plane and media plane interworking functions are
   elaborated in more detail at the example of a WebRTC gateway with
   support of interworking to IP text telephony in text conversation
   mode.

   This example describes the network configuration where one T.140
   endpoint uses data channels as T.140 transport, the other T.140
   endpoint uses RTP/UDP connections as T.140 transport, and the two
   T.140 endpoints interwork via an appropriate interworking function
   (see also [ITU-T H.248.94], Appendix <#>).

6.3.1. **Interworking support by WebRTC gateway**

   The WebRTC gateway interconnects two IP network domains, - there are
   specific considerations:

   1. WebRTC domain:

     Application specific configuration of the SCTP stream is necessary
     in order to satisfy T.140 requirements concerning reliable
     transport.

   2. Non-WebRTC domain:

     The WebRTC gateway needs to emulate a "T.140/RTP"-endpoint in the
     non-WebRTC domain, which implies an RTP source/RTP sink behaviour
     according to [RFC4103]. Hence, the WEBRTC gateway is required to
     be aware about the IP application protocol T.140 despite the fact
     of transparent forwarding mode. The "T.140 awareness" by the
     WEBRTC gateway is limited to the detection of active/silence
     periods related to the transfer of T.140 PDUs, as well as a
     "packet loss concealment" method related to incoming T.140/RTP
     packets.

   Next two sub-clauses refer to possible solutions.

6.3.2. **WebRTC domain: SCTP stream configuration**

   FIXTHIS

6.3.3. **Non-WebRTC domain: "RFC 4103/RTP"-endpoint emulation**

   There are three aspects of consideration:

R1: Inactivity of T.140 traffic in RTP send direction?

R2: Incoming RTP packets out of order?

R3: Incoming RTP packets lost?

See [ITU-T H.248.94], Appendix <#> concerning possible WebRTC
gateway behaviour in order to address above events. The required
gateway T.140 protocol interventions SHALL be compliant to
[RFC4103], [ITU-T T.140] and [ITU-T T.140Add1].

Editor's note: it was mentioned that an "application level" keep
alive mechanism might need to be supported by the WebRTC gateway in
case of NAT traversal support requirements, i.e.,.a mechanism such
as outlined by RFC 6263 "Application Mechanism for Keeping Alive the
NAT Mappings Associated with RTP / RTP Control Protocol (RTCP)
Flows".

7. Security Considerations

FIXTHIS

8. IANA Considerations

FIXTHIS

9. References

9.1. Normative References

[RFC2119] RFC 2119 (03/1997), "Key words for use in RFCs to Indicate
          Requirement Levels", BCP 14.

[RFC3264] RFC 3264 (06/2002), "An Offer/Answer Model with the
          Session Description Protocol (SDP)".

[RFC4103] RFC 4103 (06/2005), "RTP Payload for Text Conversation".

[RFC4566] RFC 4566 (07/2006), "SDP: Session Description Protocol".

[I-D.ietf-rtcweb-data-protocol] draft-ietf-rtcweb-data-channel
          (##/2015), "WebRTC Data Channel Establishment Protocol".

[I-D.ietf-rtcweb-data-channel]
           http://tools.ietf.org/wg/rtcweb/draft-ietf-rtcweb-data-
          channel/draft-ietf-rtcweb-data-channel (##/2015), "WebRTC
          Data Channels".

   [I-D.ietf-mmusic-data-channel-sdpneg] draft-ietf-mmusic-data-
            channel-sdpneg (##/2015), "SDP-based Data Channel
            Negotiation".

   [I-D.ietf-mmusic-msrp-usage-data-channel] draft-ietf-mmusic-msrp-
            usage-data-channel (##/2015), "MSRP over Data Channels".

   [ITU-T T.140] Recommendation ITU-T T.140 (02/1998), "Protocol for
            multimedia application text conversation".
            Free copy via:
            https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-
            T.140-199802-I!!PDF-E&type=items

   [ITU-T T.140Add1] Recommendation ITU-T T.140 Addendum 1 (02/2000),
            "Protocol for multimedia application text conversation -
            Addendum 1".
            Free copy via:
            https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-
            T.140-200002-I!Add1!PDF-E&type=items

9.2. Informative References

   [I-D.ietf-rtcweb-gateways] draft-ietf-rtcweb-gateways (##/2015),
            "WebRTC Gateways".

   [ITU-T H.248.94] Recommendation ITU-T H.248.94 (10/2015), "Web-
            based real-time communication services - H.248 protocol
            support and profile guidelines".
            Status: still work in progress in ITU-T SG16 Question 3
            Free copy via: FIXTHIS

    [ITU-T V.151] Recommendation ITU-T V.151 (05/2006), "Procedures for
            the end-to-end connection of analogue PSTN text telephones
            over an IP network utilizing text relay".
            Free copy via:
            https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-
            V.151-200605-I!!PDF-E&type=items

   [ITU-T V.152] Recommendation ITU-T V.152 (09/2010), "Procedures for
            supporting voice-band data over IP networks".
            Free copy via:
            https://www.itu.int/rec/dologin_pub.asp?lang=e&id=T-REC-
            V.152-201009-I!!PDF-E&type=items

   [3GPP 29.334] 3GPP TS 29.334 Release 13 (2015), "Digital cellular
                  telecommunications system (Phase 2+); Universal Mobile
                  Telecommunications System (UMTS); LTE; IMS Application
                  Level Gateway (IMS-ALG) - IMS Access Gateway (IMS-AGW); Iq
                  Interface; Stage 3".
                  Free copy via:
                  ftp://ftp.3gpp.org/specs/archive/29_series/29.334/

10. Acknowledgments

   FIXTHIS

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11. CHANGE LOG

11.1. Initial draft-schwarz-mmusic-t140-usage-data-channel-00

11.1.1. Status

o The initial document represents a skeleton where still a number
   of clauses are still empty.

o The intention is to propose a document structure aligned with the
   MSRP draft, and to emphasize WebRTC gateway flavours.

11.1.2. Changes against "draft-ietf-mmusic-msrp-usage-data-channel-01"

o Replace protocol "MSRP" by "T.140" plus protocol specific
   adaptations

o Initial scope of gateway section

11.2. Changes against draft-schwarz-mmusic-t140-usage-data-channel-01

o Reference to rtcweb draft inserted which defines realtime text
   transport in WebRTC.

o Initial input text added to the majority of "placeholder
   sections" from the -00 version.

11.3. Changes against draft-schwarz-mmusic-t140-usage-data-channel-02

o uppercase of modal vers

o clause 5.1.1.2: clarification fixed

o clause 5.1.2: clarification fixed

o clause 5.1.3: clarification fixed

o clause 5.2: clarification fixed

o clause 6.3: initial description inserted

11.4. Changes against draft-schwarz-mmusic-t140-usage-data-channel-03

o Editorial: update of author addresses

o Indication of some issues for clarification (by editor notes)

12. **Backup material: Discussion of realtime text service handling within WebRTC**

   {Editor's comment: this clause is only temporary and will be deleted again as soon as draft becomes technically stable.}

   RTCWEB email threads (non-exhaustive list):

   o 2012-11 Subject: "[rtcweb] Text communication in RTCWEB sessions"
      https://mailarchive.ietf.org/arch/msg/rtcweb/Vx6ZbLPPh4vsG25UPZlR
      Xq_ffM0

   o 2012-11 Subject: "[rtcweb] Text communication requirements for
      RTCWEB"
      https://mailarchive.ietf.org/arch/msg/rtcweb/JXesT_A6vOMs_N05fWaF
      DGPKryo

   o 2012-11 Subject: "[rtcweb] Text communication SDP in RTCWEB"
      https://mailarchive.ietf.org/arch/msg/rtcweb/EQI3Trtzmhnlh15Jocil
      TxbWHt0

   o 2012-11 Subject: "[rtcweb] Consensus call on Text Communication"
      https://mailarchive.ietf.org/arch/msg/rtcweb/viAKLXAAs0mKXL8xwXRB
      1nthPhg

   o 2012-12 Subject: "[rtcweb] Real-time text implementations"
      https://mailarchive.ietf.org/arch/msg/rtcweb/YNlKMFuzbVo2SnZHzICy
      ruf4x9I

   o 2013-05 Subject: "[rtcweb] RTT (was Re: No Plan)"
      https://mailarchive.ietf.org/arch/msg/rtcweb/iDXc5DFMJIgiEPCWnyvV
      dNFlOPo

   o 2013-05 Subject: "[rtcweb] RTT Education: Neat Demonstration of
      NON-peer-to-peer RTT (for future webrtc standardization
      purposes)"
      https://mailarchive.ietf.org/arch/msg/rtcweb/bfUUQXP_cG-
      63wBpS0JH62vCf4c

   o 2013-12 Subject: "[rtcweb] Real-time text"
      https://mailarchive.ietf.org/arch/msg/rtcweb/Hsvm7gzybT1MNYXBzbJR
      EOvy3zY

   o 2014-08 Subject: "[rtcweb] draft-ietf-rtcweb-data-channel failure
      handling description"
      https://mailarchive.ietf.org/arch/msg/rtcweb/BqBrwBlrT4h9aAmEzQkY
      b-3on7Q

   o 2015-02 Subject: "[rtcweb] Use of redundancy and RFC 2119
      language in draft-ietf-rtcweb-fec-00"
      https://mailarchive.ietf.org/arch/msg/rtcweb/HYMplbKGmBOUUR_VQiAB
      BjMfDBY