

Version: Draft 0.4: 12.5.2015

History:

Version		
0.1	16.10.2014	Initial Version
0.2	27.10.2014	Editorial Team's review
0.3	13.03.2015	Ralf Dux: Corrections & clarifications, comments
0.4	12.5.2015	Audio Signal Relevance Indication to ease LM-END QOS 026 implementation for the DR-GW

This document is written from the formal point of view like an IETF draft so that in the future a person familiar with the tooling could create an IETF draft, which fulfills the IETF formal criteria (<http://tools.ietf.org/inventory/author-tools.shtml>).

The document is written based on following instructions:



draft-ietf-avt-rtp-howto-02 - How to Write an RTP Payload Format.mht

This winword document contains items of the template as "Hidden Text.", which is currently not filled out yes so that the reader can see what is left to be done to polish the document so that it might get sent to the IETF.

1. Title

Real-Time Transport Protocol (RTP) Payload Format for the TETRA Audio Codec

2. Front page boilerplate

This Memo contains the RTP payload and SDP definition for TETRA coded audio in the context of „Digitalradio-Gateway-Interface“ initiative.

3. Abstract

This document specifies a Real-time Transport Protocol (RTP) payload format to be used for TETRA encoded speech signals. The payload format is designed to be able to interoperate with existing TETRA transport formats on non-IP networks. This version of the document does not specify a file format for transport of TETRA speech data in storage mode applications such as email as would be required by the IETF. A media type registration is included, specifying the use of the RTP payload format and the storage format.

4. Table of Content

Todo for an IETF tooling guru.

5. Introduction

This document specifies the payload format for packetization of TErrestrial Trunked Radio (TETRA) encoded speech signals into the Real-time Transport Protocol (RTP) [1]. The payload format supports transmission of multiple channels, multiple frames per payload, robustness against packet loss, and interoperation with existing TETRA transport formats on non-IP networks, as described in Section 3.

The payload format itself is specified in Section 8.

6. Conventions, Definitions and Acronyms

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.

The following acronyms are used in this document:

ETSI	- European Telecommunications Standards Institute
TETRA	- TErrestrial Trunked Radio

The byte order used in this document is network byte order, i.e., the most significant byte first. The bit order is also the most significant bit first. This is presented in all figures as having the most significant bit leftmost on a line and with the lowest number. Some bit fields may wrap over multiple lines in which cases the bits on the first line are more significant than the bits on the next line.

7. Media Format Background

The TETRA codec is used as vocoder for TETRA systems. The TETRA codec is designed for compressing 30ms of audio speech data into 137 bits. The TETRA codec is designed in such a way that on the air interface two of these 30ms samples are transported together (sub-block 1 and sub-block 2). The codec allows that data of the first 30ms voice frame can be stolen and used for other purposes, e.g. for the exchange of dynamically updated key-material in end-to-end encrypted voice sessions. For E1 lines there are two optional formats defined [3], the first format is called FSTE (First Speech Transport Encoding Format), the other format is called OSTE (Optimized Speech Transport Encoding Format). These two formats differ mainly insofar that the OSTE format transports an additional 5 bit frame number, which provides timing information from the air interface to the receiving side in order to save the need for buffering due to different transport speeds on air and in 64 kbit/s circuit switched networks. The RTP payload format is defined such that the value of this frame number can be transported.

8. Payload format

The RTP payload format is designed in such a way that it can carry the information needed to map the FSTE and OSTE format from [1]. The RTP format is defined such that both of the independent sub-blocks can be transferred separately or together within one RTP frame. Both of them contain the same information in terms of control bits - the information is propagated redundantly. This redundancy is driven by on one hand to simplify the encoding process in direction from E1 to RTP on the other to provide the option to go for either 30ms or 60ms packet size. The redundant information SHALL be propagated consistently equal - otherwise the behavior of the receiver is unspecified. The payload format is chosen such that the TETRA data bits are octet aligned.

I bit: Frame Indicator

1: The following frame contains a first block of two sub-blocks

0: The following frame contains a separated sub-block. A sub-block marked as such could either be a second sub-block, or an independent block, which does not have a relation with any first block. To distinguish between the one and the other the information of the Control bits has to be evaluated.

F bit: Frame Type

0: Frame contains FSTE encoded data

1: Frame contains OSTE encoded data

CTRL: Control bit(5 bits)

Ctrl 1..3 according table 2 of [2].

000 Sub block1 normal; sub block2 normal

001 Sub block1 C stolen; sub block2 normal

010 Sub block1 U stolen; sub block2 normal

011 Sub block1 C stolen; sub block2 C stolen

100 Sub block1 C stolen; sub block2 U stolen

101 Sub block1 U stolen; sub block2 C stolen

110 Sub block1 U stolen; sub block2 U stolen

111 O&M ISI block

Ctrl 4..5 according table 3 of [2].

00 Sub block1 BFI no errors; sub BLOCK2 BFI no errors

01 Sub block1 BFI no errors; sub Block2 BFI with error(s)

10 Sub block1 BFI with error(s); sub block2 BFI no error(s)

11 Sub block1 BFI with error(s); sub block2 BFI with error(s)

NOTE: The meaning of C4 and C5 is outside the scope of the present

C bit: Failed Crypto operation indication: This bit may be set to "1" if an encryption or a decryption operation could not be performed successfully for the specific half-block. Consequently, the encryption status of the half-block audio data is unknown. If a receiver decides to forward the TETRA audio data to OSTE or FSTE or to directly hand over the TETRA audio data to a TETRA audio decoder, the contained audio might be scrambled - depending if the audio originally was generated as a plain-override half-block or as an encrypted half-block.

FRAME_NR: FN (5 bits): contains an uplink frame number as defined in table 8 of [2]. If no frame number is available the FRAME_NR value SHALL be set to 00000.

R: Audio Signal Relevance (3 bits): contains information about the Relevance of the voice packet contained here.

```
R 1
0  no audio signal relevance propagated (R2 and R3 do not contain any valid information)
1  audio signal relevance propagated in R2 and R3
```

```
R 2..3
According to table 1 of [7]
00 no audio signal relevance ( $\text{level} \leq -72 \text{ dBm0}$ )
01 low audio signal relevance ( $-52\text{dBm0} \geq \text{level} > -72\text{dBm0}$ )
10 medium audio signal relevance ( $-32\text{dBm0} \geq \text{level} > -52\text{dBm0}$ )
11 high audio signal relevance ( $0\text{dBm0} \geq \text{level} > -32\text{dBm0}$ )
```

S: Spare bit 0

D() bits: Data bits (i.e. TETRA ACELP coded speech bits) according to table 4 of [3].

Payload definition:

[illegible]

The following example shows how a first and a consecutive 30 ms frame is combined into a single 60ms RTP packet. Note: This example shows of usage of OSTE mapping.

[illegible]

Both halves of information contain exact the same CTRL bits

11.1. Media Type Definition

The media type for the TETRA codec is expected to be allocated from the IETF tree once this draft turns into an RFC. This media type registration covers both real-time transfer via RTP and non-real-time transfers via stored files.

```
[Here the media type registration template from RFC 4288 is placed
 and filled out.  This template is provided with some common RTP
 boilerplate.]
```

Media Type name: audio

Media Subtype name: TETRA

Required parameters: none

Optional parameters:

These parameters apply to RTP transfer only.

maxptime: The maximum amount of media which can be encapsulated in a payload packet, expressed as time in milliseconds. The time is calculated as the sum of the time that the media present in the packet represents. The time SHOULD

be an integer multiple of the frame size. If this parameter is not present, the sender MAY encapsulate any number of speech frames into one RTP packet.

ptime: see RFC 4566 [4].

channels: The number of audio channels. The possible values (1-6) and their respective channel order is specified in Section 4.1 in [5]. If omitted, it has the default value of 1.

drgw-fe: As long as there is no official RTP payload definition from IETF this proprietary parameter ("digital radio gateway forum of experts") is marked with the only possible value 1. It marks the session to be established according to this specification.

Encoding considerations:

The Audio data is binary data, and must be encoded for non-binary transport; the Base64 encoding is suitable for email. When used in RTP context the data is framed as defined in [6].

Security considerations:

See Section 7 of RFC 4867.

Interoperability considerations:

Published specification:

Applications that use this media type:

This media type is used in applications needing transport or storage of encoded voice. Some examples include; Voice over IP, streaming media, voice messaging, and voice recording on recording systems.

Person & email address to contact for further information:

See Authors

Intended usage: (One of COMMON, LIMITED USE or OBSOLETE.)

Restrictions on usage:

When this media type is used in the context of transfer over RTP, the RTP payload format specified in Section 8 SHALL be used. In all other contexts, the file format defined in Section 5 SHALL be used.

This media type depends on RTP framing, and hence is only defined for transfer via RTP [RFC3550]. Transport within other framing protocols is not defined at this time.

11.2. Mapping to SDP

The information carried in the media type specification has a specific mapping to fields in the Session Description Protocol (SDP) [4], which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing the TETRA codec, the mapping is as follows:

Media Type name: audio
Media subtype name: TETRA
Required parameters: none
Optional parameters: none

Mapping MIME Parameters into SDP: The information carried in the MIME media type specification has a specific mapping to fields in the Session Description Protocol [RFC2327], which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing the TETRA codec, the mapping is as follows:

- The MIME type ("audio") goes in SDP "m=" as the media name.
- The MIME subtype (payload format name) goes in SDP "a=rtpmap" as the encoding name. The RTP clock rate in "a=rtpmap" MUST be 8000.
- The parameters "ptime" and "maxptime" go in the SDP "a=ptime" and "a=maxptime" attributes, respectively.
- Any remaining parameters go in the SDP "a=fmtp" attribute by copying them directly from the media type parameter string as a semicolon-separated list of parameter=value pairs.

Here is an example SDP session of usage of TETRA:

```
m=audio 49120 RTP/AVP 99
a=rtpmap:99 TETRA/8000
a=maxptime:60
a=ptime:60
a=fmtp:99
```

11.2.1. Offer/Answer Considerations

The following considerations apply when using SDP Offer-Answer procedures to negotiate the use of TETRA payload in RTP:

- In most cases, the parameters "maxptime" and "ptime" will not affect interoperability; however, the setting of the parameters can affect the performance of the application. The SDP offer-answer handling of the "ptime" parameter is described in RFC 3264 [13]. The "maxptime" parameter MUST be handled in the same way.
- Any unknown parameter in an offer SHALL be removed in the answer.

13. Security Considerations

[See Section Section 7.1]

RTP packets using the payload format defined in this specification

are subject to the security considerations discussed in the RTP specification [RFC3550] , and in any applicable RTP profile. The main security considerations for the RTP packet carrying the RTP payload format defined within this memo are confidentiality, integrity and source authenticity. Confidentiality is achieved by encryption of the RTP payload. Integrity of the RTP packets through suitable cryptographic integrity protection mechanism. Cryptographic systems may also allow the authentication of the source of the payload. A suitable security mechanism for this RTP payload format should provide confidentiality, integrity protection and at least source authentication capable of determining if an RTP packet is from a member of the RTP session or not.

Note that the appropriate mechanism to provide security to RTP and payloads following this memo may vary. It is dependent on the application, the transport, and the signaling protocol employed. Therefore a single mechanism is not sufficient, although if suitable the usage of SRTP [RFC3711] is recommended. Other mechanism that may be used are IPsec [RFC4301] and TLS [RFC4346] (RTP over TCP), but also other alternatives may exist.

14. References

- [1] Schulzrinne, H. and S. Casner, " RTP: A Transport Protocol for Real-Time Applications ", RFC 3550, July 2003.
- [4] Handley, M. and V. Jacobson, "SDP: Session Description Protocol", RFC 4566, July 2006.
- [5] Josefsson, S., "The Base16, Base32, and Base64 Data Encodings", RFC 3548, July 2003.
- [6] Handley, M., Perkins, C., and E. Whelan, "Session Announcement Protocol", RFC 2974, October 2000.
- [7] Bundesanstalt für den Digitalfunk der Behörden und Organisationen mit Sicherheitsaufgaben, „BIP 20 QOS Dienstgüte-Parameter BOS-Interoperabilitätsprofil für Endgeräte zur Nutzung im Digitalfunk BOS“, Version 2014-04 - Revision 2

14.1. Normative References

- [2] ETSI TS 100 392-3-6 V1.1.1 (2003-12) Part 2: TETRA Interworking at the Inter -System interface (ISI) Sub-part 6: Speech format implementation for circuit mode transmission
- [3] ETSI TS 300 395-2 February 1998 Terrestrial Trunked Radio (TETRA); Speech codec for full-rate traffic channel; Part 2: TETRA codec

14.2. Informative References

15. Author Addresses

Udo Brandhuber
eurofunk Kappacher GmbH
Germany

Email: ubrandhuber@eurofunk.com

Thomas Haas
Thales Group
Germany

EMail: THOMAS.HAAS@thalesgroup.com>

Joachim Hagedorn
Hagedorn Informationssysteme GmbH
Germany

EMail: joachim@hagedorn-infosysteme.de

Klaus-Peter Höhnsch (
T-Systems International GmbH
Germany

EMail: klaus-peter.hoehnsch@t-systems.com

Andreas Reisenbauer
Frequentis AG
Austria

EMail: andreas.reisenbauer@frequentis.com

Stefan Wenk
Frequentis AG
Austria

EMail: stefan.wenk@frequentis.com

16. IPR Notice

17. Copyright Notice

Copyright (C) The IETF Trust (2008).

This document is subject to the rights, licenses and restrictions contained in BCP 78, and except as set forth therein, the authors retain all their rights.

This document and the information contained herein are provided on an "AS IS" basis and THE CONTRIBUTOR, THE ORGANIZATION HE/SHE REPRESENTS OR IS SPONSORED BY (IF ANY), THE INTERNET SOCIETY, THE IETF TRUST AND THE INTERNET ENGINEERING TASK FORCE DISCLAIM ALL WARRANTIES, EXPRESS OR IMPLIED, INCLUDING BUT NOT LIMITED TO ANY WARRANTY THAT THE USE OF THE INFORMATION HEREIN WILL NOT INFRINGE ANY RIGHTS OR ANY IMPLIED WARRANTIES OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.