Audio/Video Transport Working Group Internet Draft SPIRIT DSP Intended status: Informational February 25, 2009

RTP Payload Format for IP-MR Speech Codec draft-ietf-avt-rtp-ipmr-02.txt

Status of this Memo

This Internet-Draft is submitted to IETF in full conformance with the provisions of $\frac{\text{BCP}}{78}$ and $\frac{\text{BCP}}{79}$.

Copyright (c) 2009 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to $\underline{\sf BCP\ 78}$ and the IETF Trust's Legal Provisions Relating to IETF

Documents in effect on the date of publication of this document (http://trustee.ietf.org/license-

info). Please review these documents carefully, as they describe your rights and restrictions with respect to this document.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF), its areas,

and its working groups. Note that other groups may also distribute working documents as

Internet-Drafts.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated,

replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts

as reference material or to cite them other than as "work in progress."

The list of current Internet-Drafts can be accessed at http://www.ietf.org/1id-abstracts.html

The list of Internet-Draft Shadow Directories can be accessed at http://www.ietf.org/shadow.html

This Internet-Draft will expire on August 25, 2009.

Abstract

This document specifies the payload format for packetization of SPIRIT IP-MR encoded speech

signals into the Real-time Transport Protocol (RTP). The payload format supports transmission

of multiple frames per payload and introduced redundancy for robustness against packet loss.

Table of Contents

Introduction

<u>1</u> .	Introduction 2		
<u>2</u> .	IP-MR Codec Description 2		
<u>3</u> .	Payload Format 4		
<u>3.1</u> .	Payload Format Structure	4	
<u>3.2</u> .	Payload Header 4		
<u>3.3</u> .	Speech Table of Contents	5	
<u>3.4</u> .	Speech Data 5		
<u>3.5</u> .	Redundancy Header 5		
<u>3.6</u> .	Redundancy Table of Contents	6	
<u>3.7</u> .	Redundancy Data 6		
<u>4</u> .	Payload Examples 6		
<u>4.1</u> .	Payload Carrying a Single Fram	e 6	
<u>4.2</u> .	Payload Carrying Multiple Fram	es with Redundancy	7
<u>5</u> .	Media Type Registration 8		
<u>5.1</u> .	Registration of media subtype	audio/ip-mr_v2.5 8	
<u>5.2</u> .	Mapping Media Type Parameters	into SDP 9	
<u>6</u> .	Security Considerations 9		
<u>7</u> .	IANA Considerations 10		
<u>8</u> .	Normative References 10		
<u>9</u> .	Author's Information 10		
<u>10</u> .	Expiration date 10		
<u>11</u> .	Legal Terms 10		

This document specifies the payload format for packetization of SPIRIT IP-MR encoded speech $\,$

signals into the Real-time Transport Protocol (RTP). The payload format supports transmission

of multiple frames per payload and introduced redundancy for robustness against packet loss.

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.

IP-MR Codec Description

The IP-MR codec is scalable adaptive multi-rate wideband speech codec designed by SPIRIT for

use in IP based networks. These codec is suitable for real time communications such as

telephony and videoconferencing.

The codec operates on 20 ms frames at 16 kHz sampling rate and has an algorithmic delay of 25 ms.

The IP-MR supports six wide band speech coding modes with respective bit rates ranging from

about 7.7 to about 34.2 kbps. The coding mode can be changed at any 20 ms frame boundary

making possible to dynamically adjust the speech encoding rate during a session to adapt to the $\,$

varying transmission conditions.

The coded frame consists of multiple coding layers-base (or core) layer and several enhancement

layers which are coded independently. These enhancement layers can be omitted and remaining

base layer can be meaningfully decoded without artifacts. This making bit stream scalable and

allows reduce bit rate during transmission without re-encoding.

This memo specifies an optional form of redundancy coding within RTP for protection against

packet loss. It is based on commonly known scheme when previously transmitted frames are

aggregated together with new ones. Each frame is retransmitted once in the following RTP

payload packet. f(n-2)...f(n+4) denote a sequence of speech frames, and p(n-1)...p(n+4) a

sequence of payload packets:

But because of scalable nature of IP-MR codec there is no need to duplicate a whole previous

frame - only core layer may be retransmitted. This reduces redundancy overhead while keeping

efficiency. Moreover, the speech bits encoded in core layer are divided on six classes (from A to

F) of perceptual sensitivity to errors. Using these classes as introduced redundancy make

possible to adjust trade-off between overhead and robustness against packet loss.

The mechanism described does not really require signaling at the session setup. The sender is

responsible for selecting an appropriate amount of redundancy based on feedback about the $\,$

channel conditions.

The main codec characteristics can be summarized as follows:

- * Wideband, 16 kHz, speech codec
- * Adaptive multi rate with six modes from about 7.7 to about 34.2 kbps
- * Bit rate scalable
- * Variable bit rate changing in accordance with actual speech content
- * Discontinuous Transmission (DTX), silence suppression and comfort noise generation
- * In-band redundancy scheme for protection against packet loss

3. Payload Format

3.1. Payload Format Structure

The IP-MR payload format consists of a payload header with general information about packet, a

speech table of contents (TOC), and speech data. An optional redundancy section follows after

speech data. The redundancy section consists of redundancy header, redundancy TOC and

redundancy data payload.

The following diagram shows the standard payload format layout:

```
+-----+ | payload | speech | speech | redundancy | redundancy | redundancy | | header | TOC | data | | +-----+
```

3.2. Payload Header

The payload header has the following format:

- * T (1 bit): Reserved compatibility with future extensions. SHOULD be set to 0.
- * CR (3 bits): coding rate of frame(s) in this packet, as per the following table:

+-		++
	CR	avg. bitrate
+-		++
	0	7.7 kbps
	1	9.8 kbps
	2	14.3 kbps
	3	20.8 kbps
	4	27.9 kbps
	5	34.2 kbps
	6	(reserved)
	7	NO_DATA
+-		++

The CR value 7 (NO_DATA) indicates that there is no speech data (and speech TOC accordingly) in the payload. This MAY be used to transmit redundancy data only. The value 6 is

reserved. If receiving this value the packet SHOULD be discarded.

 * BR (3 bits): base rate for core layer of frame(s) in this packet. Values in the range 0-5

indicate bitrates for core layer, same as for CR. Values 6 and 7 are reserved. If one of

these values is received the packet SHOULD be discarded. The base rate is the lowest

rate for scalability, so speech payload can be scaled down not lower than $\ensuremath{\mathsf{BR}}$ value. If a

received packet has ${\sf BR} > {\sf CR}$ then during decoding it will be assumed that ${\sf BR} = {\sf CR}$.

- * D (1 bit): indicates if the DTX mode is allowed or not.
- * A (1 bit): byte-aligned payload. If A=1 then all speech frames MUST be

byte-aligned.

This mode speeds up speech data access. The A=0 value specifies bandwidth-efficient

mode with no byte alignment (including end of header).

- * GR (2 bits): number of frames in packet (grouping size). Actual grouping size is GR + 1, thus maximum grouping supported is 4.
- * R (1 bit): redundancy presence bit. If R=1 then the packet contains redundancy information for lost packets recovery. In this case after speech data the redundancy section is present.

3.3. Speech Table of Contents

The speech TOC contains entries for each frame in packet (grouping size in total). Each entry contains a single field:

0

+-+

|E|

+-+

 * E (1 bit): frame existence indicator. If set to 0, this indicates the corresponding frame is

absent and the receiver should set special LOST_FRAME flag for decoder. This can be

followed by the lost frame itself or by empty frames generated by the encoder during

silence intervals in DTX mode.

Note that if CR flag from payload header is 7 (NO_DATA) then speech TOC is empty.

3.4. Speech Data

Speech data of a payload contains one or more speech frames or comfort noise frames, as

specified in the speech TOC of the payload.

Each speech frame represents 20 ms of speech encoded with the rate indicated in the CR and

base rate indicated in BR field of the payload header. The length of the speech frame is variable

due to the nature of the codec and can be calculated after decoding.

3.5. Redundancy Header

If a packet contains redundancy (R field of payload header is 1) the speech

data is followed by redundancy header:

```
0 1 2 3 4 5
+-+-+-+-+
| CL1 | CL2 |
+-+-+-+
```

Redundancy header consists of two fields. Each field contains class specifier for amount of

redundancy partly taken from the preceding packet (CL1) and pre-preceding packet (CL2), e.g.

distant from the current packet by 1 and 2 packets accordingly. The values are listed in the table below:

+ -		+	+
	CL	amount redundancy	
+-		+	t
	0	NONE	
	1	CLASS A	
	2	CLASS B	
	3	CLASS C	
	4	CLASS D	
	5	CLASS E	
	6	CLASS F	
	7	(reserved)	
+-		+	+

Each specifier takes 3 bits, thus the total redundancy header size is 6 bits.

3.6. Redundancy Table of Contents

The redundancy TOC contains entries for redundancy frames from preceding and pre-preceding

packets. Each entry takes 1 bit like speech TOC entry (3.3):

0 +-+ |E| +-+

- * E (1 bit): frame existence indicator. If set to 0, this indicates the corresponding frame is absent.
- * For each preceding and pre-preceding packet the number of entries is equal to the

grouping size of the current packet. E.g. maximum number of entries is 4*2 = 8.

* If class specifier in the redundancy header is CL=0 (NO_DATA) then there is no entries for corresponding packet redundancy.

3.7. Redundancy Data

Redundancy data of a payload contains redundancy information for one or more speech frames

or comfort noise frames that may be lost during transition, as specified in the redundancy TOC

of the payload. Actually redundancy is the most important part of preceding frames representing

<u>20</u> ms of speech. This data MAY be used for partial reconstruction of lost frames. The amount of available redundancy is specified by CL flag in redundancy header section (3.5). This flag

SHOULD be passed to decoder. The length of redundancy frame is variable and can be

calculated after decoding.

4. Payload Examples

A few examples to highlight the payload format follow.

4.1. Payload Carrying a Single Frame

The following diagram shows a standard IP-MR payload carrying a single speech frame without redundancy:

0	1			2			3
0 1 2 3 4	5 6 7 8 9 0	1 2 3 4	5 6 7	8 9 0 1 2	2 3 4 5	6 7 8	9 0 1
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+	+-+-+-+
0 CR=1 BR	=0 0 0 0 0	0 1 sp(9)				
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+	+-+-+
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+-	+-+-+
1							
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+-	+-+-+-+
1							
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+-	+-+-+
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+-	+-+-+
1							1
+-+-+-+-+	-+-+-+-+-	+-+-	+-+-+-+	-+-+-+-	-+-+	+-+-+-	+-+-+
1		sp(193)	P				
+-+-+-+-+	-+-+-+-+-	+-+-+-	+-+				

In the payload the speech frame is not damaged at the IP origin (E=1), the coding rate is $9.7~{\rm kbps}$

(CR=1), the base rate is 7.8 kbps (BR=0), and the DTX mode is off. There is no byte alignment $\frac{1}{2}$

(A=0) and no redundancy (R=0). The encoded speech bits - s(0) to s(193) - are placed

immediately after TOC. Finally, one zero bit is added at the end as padding to make the payload byte aligned.

4.2. Payload Carrying Multiple Frames with Redundancy

The following diagram shows a payload that contains three frames, one of them with no speech

data. The coding rate is 7.7 kbps (CR=0), the base rate is 7.7 kbps (BR=0), and the DTX mode

is on. The speech frames are byte aligned (A=1), so 1 zero bit is added at the end of the header.

Besides the speech frames the payload contains six redundancy frames (three per each delayed packet).

The first speech frame consists of bits sp1(0) to sp1(92). After that 3 bits are added for byte

alignment. The second frame does not contain any speech information that is represented in the

payload by its TOC entry. The third frame consists of bits sp3(0) to sp3(171).

The redundancy header follows after speech data. The one-packet- delayed redundancy contains

class A+B bits (CL1=2), and two-packet- delayed redundancy contains class A bits (Cl2=1). The $\,$

one-packet- delayed redundancy contains three frames with 20, 39 and 35 bits respectively. The

first frame of two-packet-delayed redundancy is absent, it is represented in its TOC entry, and

two other frames have sizes 15 and 19 bits.

Note that all speech frames are padded with zero bits for byte alignment.

0	1	2	3
0 1 2 3 4 5	67890123	4 5 6 7 8 9 0 1 2	3 4 5 6 7 8 9 0 1
+-+-+-+-+-+	-+-+-+-+-+-+	-+-+-+-+-+-	+-+-+-+-+-+-+-+
0 CR=0 BR=0	1 1 1 0 1 1 0	1 P sp1(0)	
+-+-+-+-+	-+-+-+-+-+	-+-+-+-+-+-+-	+-+-+-+-+-+-+-+-+-
1			
+-+-+-+-+	-+-+-+-+-+-+	-+-+-+-+-	+-+-+-+-+-+-+-+
1			
+-+-+-+-+-+	-+-+-+-+-+-+-+	-+-+-+-+-+-	+-+-+-+-+-+-+-+
1	sp1(92) P	P P sp3(0)	
+-+-+-+-+-+	-+-+-+-+-+-+-+	-+-+-+-+-+-	+-+-+-+-+-+-+-+
1			
+-+-+-+-+-+	-+-+-+-+-+-+	-+-+-+-+-+-	+-+-+-+-+-+-+-+
1			
+-+-+-+-+-+	-+-+-+-+-+-+	-+-+-+-+-+-	+-+-+-+-+-+-+-+-+

Media Type Registration

Published specification:

Applications that use this media type:

RFC XXXX

multi-media streaming.

This section describes the media types and names associated with this payload format.

5.1. Registration of media subtype audio/ip-mr_v2.5

```
Type name: audio

Subtype name: ip-mr_v2.5

Required parameters: none

Optional parameters:

* ptime: Gives the length of time in milliseconds represented by the media in a packet.

Allowed values are: 20, 40, 60 and 80.

Encoding considerations:
   This media type is framed binary data (see RFC 4288, Section 4.8).

Security considerations:
   See RFC 3550

Interoperability considerations: none
```

Real-time audio applications like voice over IP and teleconference, and

Additional information: none

Person & email address to contact for further information:

Elena Berlizova

berlizova@spiritdsp.com

Intended usage: COMMON

Restrictions on usage:

This media type depends on RTP framing, and hence is only defined for transfer via RTP (RFC 3550).

Author:

Sergey Ikonin <ikonin@spiritdsp.com>

Change controller:

IETF Audio/Video Transport working group delegated from the IESG.

5.2. Mapping Media Type Parameters into SDP

The information carried in the media type specification has a specific mapping to fields in the

Session Description Protocol (SDP) [$\underline{\mathsf{RFC4566}}$], which is commonly used to describe RTP

sessions. When SDP is used to specify sessions employing the IP-MR codec, the mapping is as $\frac{1}{2} \frac{1}{2} \frac{1}{2}$

follows:

- * The media type ("audio") goes in SDP "m=" as the media name.
- * The media subtype (payload format name) goes in SDP "a=rtpmap" as the encoding name. The

RTP clock rate in "a=rtpmap" MUST 16000.

* The parameter "ptime" goes in the SDP "a=ptime" attributes.

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.

Note that the payload format (encoding) names are commonly shown in upper case. Media

subtypes are commonly shown in lower case. These names are case-insensitive in both places.

Security Considerations

RTP packets using the payload format defined in this specification are subject to the security

considerations discussed in the RTP specification [RFC3550], and any appropriate RTP profile.

This implies that confidentiality of the media streams is achieved by encryption. Encryption

may be performed after compression so there is no conflict between the two operations.

This payload format does not exhibit any significant non-uniformity in the receiver side

computational complexity for packet processing, and thus is unlikely to pose a denial-of-service

threat due to the receipt of pathological data.

7. IANA Considerations

One media type has been defined and needs registration in the media types registry.

8. Normative References

- [1] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", $\underline{\rm BCP}\ 14,\ \rm RFC$
- 2119, March 1997.
- [2] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, <u>RFC 3550</u>, July 2003.
- [3] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", RFC 4566, July 2006.

9. Author's Information

Sergey Ikonin

Russia

109004 B.Kommunisticheskaya st. 27

Tel: +7 495 661-2178 Fax: +7 495 912-6786

Email: ikonin@spiritdsp.com

Expiration date

This Internet-Draft will expire on August 25, 2009.

11. Legal Terms

All IETF Documents and the information contained therein 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 THEREIN WILL NOT INFRINGE ANY RIGHTS OR ANY IMPLIED

WARRANTIES OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.

The IETF Trust takes no position regarding the validity or scope of any Intellectual Property

Rights or other rights that might be claimed to pertain to the implementation or use of the

technology described in any IETF Document or the extent to which any license under such rights

might or might not be available; nor does it represent that it has made any independent effort to

identify any such rights.

Copies of Intellectual Property disclosures made to the IETF Secretariat and any assurances of

licenses to be made available, or the result of an attempt made to obtain a general license or

permission for the use of such proprietary rights by implementers or users of this specification

can be obtained from the IETF on-line IPR repository at http://www.ietf.org/ipr.

The IETF invites any interested party to bring to its attention any copyrights, patents or patent

applications, or other proprietary rights that may cover technology that may be required to

implement any standard or specification contained in an IETF Document. Please address the $\,$

information to the IETF at ietf-ipr@ietf.org.

The definitive version of an IETF Document is that published by, or under the auspices of, the

IETF. Versions of IETF Documents that are published by third parties, including those that are

translated into other languages, should not be considered to be definitive versions of IETF

Documents. The definitive version of these Legal Provisions is that published by, or under the

auspices of, the IETF. Versions of these Legal Provisions that are published by third parties,

including those that are translated into other languages, should not be considered to be definitive

versions of these Legal Provisions.

For the avoidance of doubt, each Contributor to the IETF Standards Process licenses each

Contribution that he or she makes as part of the IETF Standards Process to the IETF Trust

pursuant to the provisions of $\overline{\text{RFC }5378}$. No language to the contrary, or terms, conditions or

rights that differ from or are inconsistent with the rights and licenses granted under $\frac{\text{RFC }5378}{\text{FC }}$,

shall have any effect and shall be null and void, whether published or posted by such

Contributor, or included with or in such Contribution.