

Audio/Video Transport Working Group
Internet Draft
Intended status: Informational

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 [BCP 79](#).

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

This document is subject to [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/charter/charter.html>). Please review these documents carefully, as they describe your rights and obligations with respect to this document.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF) 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 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/lid-abbrev/>

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 signals into the Real-time Transport Protocol (RTP). The payload format supports multiple frames per payload and introduced redundancy for robustness against

Table of Contents

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

This document specifies the payload format for packetization of SPIRIT IP-MR signals into the Real-time Transport Protocol (RTP). The payload format supports multiple frames per payload and introduced redundancy for robustness against

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](#).

[2.](#) IP-MR Codec Description

The IP-MR codec is scalable adaptive multi-rate wideband speech codec designed for use in IP based networks. This codec is suitable for real time communications, telephony and videoconferencing.

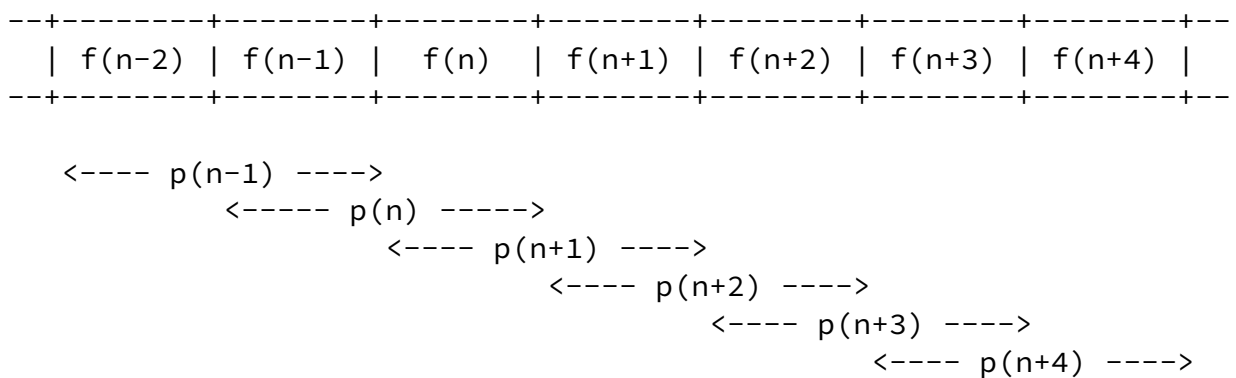
The codec operates on 20 ms frames at 16 kHz sampling rate and has an algorithm

ms.

The IP-MR supports six wide band speech coding modes with respective bit rates about 7.7 to about 34.2 kbps. The coding mode can be changed at any 20 ms frame making possible to dynamically adjust the speech encoding rate during a session varying transmission conditions.

The coded frame consists of multiple coding layers-base (or core) layer and several layers which are coded independently. These enhancement layers can be omitted and the base layer can be meaningfully decoded without artifacts. This making bit stream 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 payload packet. $f(n-2) \dots f(n+4)$ denote a sequence of speech frames, and $p(n-1) \dots p(n+4)$ denote a sequence of payload packets:



But because of scalable nature of IP-MR codec there is no need to duplicate a whole frame - only core layer may be retransmitted. This reduces redundancy overhead and improves efficiency. Moreover, the speech bits encoded in core layer are divided into six classes of perceptual sensitivity to errors. Using these classes as introduced redundancy makes 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 from the receiver and 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

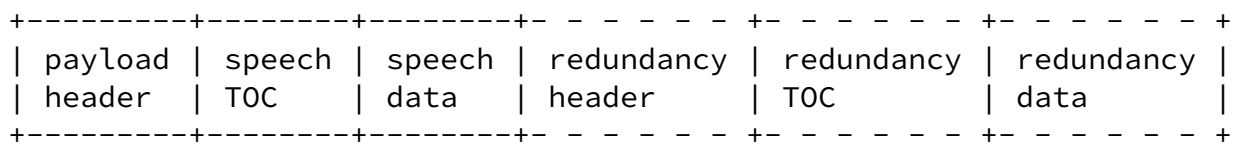
* In-band redundancy scheme for protection against packet loss

3. Payload Format

3.1. Payload Format Structure

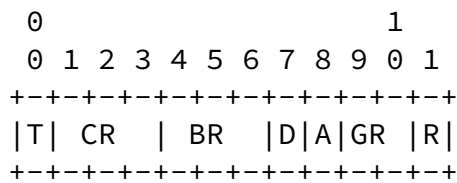
The IP-MR payload format consists of a payload header with general information speech table of contents (TOC), and speech data. An optional redundancy section speech data. The redundancy section consists of redundancy header, redundancy T redundancy data payload.

The following diagram shows the standard payload format layout:



3.2. Payload Header

The payload header has the following format:



* T (1 bit): Reserved compatibility with future extensions. SHOULD be set

* CR (3 bits): coding rate of frame(s) in this packet, as per the followi

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

* BR (3 bits): base rate for core layer of frame(s) in this packet. Value indicate bitrates for core layer, same as for CR. Values 6 and 7 are reserved. these values is received the packet SHOULD be discarded. The base rate is t rate for scalability, so speech payload can be scaled down not lower than BR va received packet has BR > CR then during decoding it will be assumed that BR = C

- * 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 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 to) and contains a single field:

```

0
+--+
|E|
+--+

```

- * E (1 bit): frame existence indicator. If set to 0, this indicates the 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 payload header. The length of the speech data is determined 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 a redundancy header:

```

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

```

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

redundancy partly taken from the preceding packet (CL1) and pre-preceding packets distant from the current packet by 1 and 2 packets accordingly. The values are below:

CL	amount redundancy
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

Pkt1 Entries	Pkt2 Entries

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 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 $4 \times 2 = 8$.

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

3.7. Redundancy Data

Redundancy data of a payload contains redundancy information for one or more speech or comfort noise frames that may be lost during transition, as specified in the header of the payload. Actually redundancy is the most important part of preceding frames for 20 ms of speech. This data MAY be used for partial reconstruction of lost frames if available redundancy is specified by CL flag in redundancy header section (3.5). This data SHOULD be passed to decoder. The length of redundancy frame is variable and can be calculated after decoding.

Encoding considerations:

This media type is framed binary data (see [RFC 4288, Section 4.8](#)).

Security considerations:

See [RFC 3550](#)

Interoperability considerations: none

Published specification:

RFC XXXX

Applications that use this media type:

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

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 trans
([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
Session Description Protocol (SDP) [[RFC4566](#)], which is commonly used to describ
sessions. When SDP is used to specify sessions employing the IP-MR codec, the
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 encodin
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 direc
media type parameter string as a semicolon- separated list of parameter=value p

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

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

6. Security Considerations

RTP packets using the payload format defined in this specification are subject to security considerations discussed in the RTP specification [[RFC3550](#)], and any appropriate security considerations. 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's computational complexity for packet processing, and thus is unlikely to pose a security 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", RFC 2119, March 1997.
- [2] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, [RFC 3550](#).
- [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

10. 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. 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 this document.

technology described in any IETF Document or the extent to which any license un might or might not be available; nor does it represent that it has made any ind identify any such rights.

Copies of Intellectual Property disclosures made to the IETF Secretariat and an licenses to be made available, or the result of an attempt made to obtain a gen permission for the use of such proprietary rights by implementers or users of t 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, applications, or other proprietary rights that may cover technology that may be implement any standard or specification contained in an IETF Document. Please a information to the IETF at ietf-ipr@ietf.org.

The definitive version of an IETF Document is that published by, or under the a IETF. Versions of IETF Documents that are published by third parties, including translated into other languages, should not be considered to be definitive vers Documents. The definitive version of these Legal Provisions is that published b auspices of, the IETF. Versions of these Legal Provisions that are published by including those that are translated into other languages, should not be conside versions of these Legal Provisions.

For the avoidance of doubt, each Contributor to the IETF Standards Process lice Contribution that he or she makes as part of the IETF Standards Process to the pursuant to the provisions of [RFC 5378](#). No language to the contrary, or terms, rights that differ from or are inconsistent with the rights and licenses grante shall have any effect and shall be null and void, whether published or posted b Contributor, or included with or in such Contribution.