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X. Duan
S. Wang
China Mobile Communications Corporation
M. Westerlund
K. Hellwig
I. Johansson
Ericsson AB
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RTP Payload Format for Global System for Mobile Communications Half Rate (GSM-HR)

Abstract

This document specifies the payload format for packetization of Global System for Mobile Communications Half Rate (GSM-HR) speech codec data into the Real-time Transport Protocol (RTP). The payload format supports transmission of multiple frames per payload and packet loss robustness methods using redundancy.

Status of This Memo

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1. Introduction

This document specifies the payload format for packetization of GSM Half Rate (GSM-HR) codec [TS46.002] encoded speech signals into the Real-time Transport Protocol (RTP) [RFC3550]. The payload format supports transmission of multiple frames per payload and packet loss robustness methods using redundancy.

This document starts with conventions, a brief description of the codec, and payload format capabilities. The payload format is specified in Section 5. Examples can be found in Section 6. The media type specification and its mappings to SDP, and considerations when using the Session Description Protocol (SDP) offer/answer procedures are then specified. The document ends with considerations related to congestion control and security.

This document registers a media type (audio/GSM-HR-08) for the Real-time Transport Protocol (RTP) payload format for the GSM-HR codec. Note: This format is not compatible with the one provided back in 1999 to 2000 in early draft versions of what was later published as RFC 3551. RFC 3551 was based on a later version of the Audio-Visual Profile (AVP) draft, which did not provide any specification of the GSM-HR payload format. To avoid a possible conflict with this older format, the media type of the payload format specified in this document has a media type name that is different from (audio/GSM-HR).

2. Conventions Used in This Document

This document uses the normal IETF bit-order representation. Bit fields in figures are read left to right and then down. The leftmost bit in each field is the most significant. The numbering starts from 0 and ascends, where bit 0 will be the most significant.

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].

3. GSM Half Rate

The Global System for Mobile Communications (GSM) network provides with mobile communication services for nearly 3 billion users (statistics as of 2008). The GSM Half Rate (GSM-HR) codec is one of the speech codecs used in GSM networks. GSM-HR denotes the Half Rate speech codec as specified in [TS46.002].

Note: For historical reasons, these 46-series specifications are internally referenced as 06-series. A simple mapping applies; for example, 46.020 is referenced as 06.20, and so on.

The GSM-HR codec has a frame length of 20 ms, with narrowband speech sampled at 8000 Hz, i.e., 160 samples per frame. Each speech frame is compressed into 112 bits of speech parameters, which is equivalent to a bit rate of 5.6 kbit/s. Speech pauses are detected by a standardized Voice Activity Detection (VAD). During speech pauses, the transmission of speech frames is inhibited. Silence Descriptor (SID) frames are transmitted at the end of a talkspurt and about every 480 ms during speech pauses to allow for a decent comfort noise (CN) quality on the receiver side.

The SID frame generation in the GSM radio network is determined by the GSM mobile station and the GSM radio subsystem. SID frames come during speech pauses in the uplink from the mobile station about every 480 ms. In the downlink to the mobile station, when they are generated by the encoder of the GSM radio subsystem, SID frames are sent every 20 ms to the GSM base station, which then picks only one every 480 ms for downlink radio transmission. For other applications, like transport over IP, it is more appropriate to send the SID frames less often than every 20 ms, but 480 ms may be too sparse. We recommend as a compromise that a GSM-HR encoder outside of the GSM radio network (i.e., not in the GSM mobile station and not in the GSM radio subsystem, but, for example, in the media gateway of the core network) should generate and send SID frames every 160 ms.

4. Payload Format Capabilities

This RTP payload format carries one or more GSM-HR encoded frames -- either full voice or silence descriptor (SID) -- representing a mono speech signal. To maintain synchronization or to indicate unsent or lost frames, it has the capability to indicate No_Data frames.

4.1. Use of Forward Error Correction (FEC)

Generic forward error correction within RTP is defined, for example, in RFC 5109 [RFC5109]. Audio redundancy coding is defined in RFC 2198 [RFC2198]. Either scheme can be used to add redundant information to the RTP packet stream and make it more resilient to packet losses, at the expense of a higher bit rate. Please see either RFC for a discussion of the implications of the higher bit rate to network congestion.

In addition to these media-unaware mechanisms, this memo specifies an optional-to-use GSM-HR-specific form of audio redundancy coding, which may be beneficial in terms of packetization overhead. Conceptually, previously transmitted transport frames are aggregated together with new ones. A sliding window can be used to group the frames to be sent in each payload. Figure 1 below shows an example.

```
   --+--------+--------+--------+--------+--------+--------+--------+--
     | f(n-2) | f(n-1) |  f(n)  | f(n+1) | f(n+2) | f(n+3) | f(n+4) |
   --+--------+--------+--------+--------+--------+--------+--------+--

      <---- p(n-1) ---->
               <----- p(n) ----->
                        <---- p(n+1) ---->
                                 <---- p(n+2) ---->
                                          <---- p(n+3) ---->
                                                   <---- p(n+4) ---->
```

Figure 1: An Example of Redundant Transmission

Here, each frame is retransmitted once in the following RTP payload packet. f(n-2)...f(n+4) denote a sequence of audio frames, and p(n-1)...p(n+4) a sequence of payload packets.

The mechanism described does not really require signaling at the session setup. However, signaling has been defined to allow the sender to voluntarily bound the buffering and delay requirements. If nothing is signaled, the use of this mechanism is allowed and unbounded. For a certain timestamp, the receiver may acquire multiple copies of a frame containing encoded audio data. The cost of this scheme is bandwidth, and the receiver delay is necessary to allow the redundant copy to arrive.

This redundancy scheme provides a functionality similar to the one described in RFC 2198, but it works only if both original frames and redundant representations are GSM-HR frames. When the use of other media coding schemes is desirable, one has to resort to RFC 2198.

The sender is responsible for selecting an appropriate amount of redundancy, based on feedback regarding the channel conditions, e.g., in the RTP Control Protocol (RTCP) [RFC3550] receiver reports. The sender is also responsible for avoiding congestion, which may be exacerbated by redundancy (see Section 9 for more details).

5. Payload Format

The format of the RTP header is specified in [RFC3550]. The payload format described in this document uses the header fields in a manner consistent with that specification.

The duration of one speech frame is 20 ms. The sampling frequency is 8000 Hz, corresponding to 160 speech samples per frame. An RTP packet may contain multiple frames of encoded speech or SID parameters. Each packet covers a period of one or more contiguous

20-ms frame intervals. During silence periods, no speech packets are sent; however, SID packets are transmitted every now and then.

To allow for error resiliency through redundant transmission, the periods covered by multiple packets MAY overlap in time. A receiver MUST be prepared to receive any speech frame multiple times. A given frame MUST NOT be encoded as a speech frame in one packet and as a SID frame or as a No_Data frame in another packet. Furthermore, a given frame MUST NOT be encoded with different voicing modes in different packets.

The rules regarding maximum payload size given in Section 3.2 of [RFC5405] SHOULD be followed.

5.1. RTP Header Usage

The RTP timestamp corresponds to the sampling instant of the first sample encoded for the first frame in the packet. The timestamp clock frequency SHALL be 8000 Hz. The timestamp is also used to recover the correct decoding order of the frames.

The RTP header marker bit (M) SHALL be set to 1 whenever the first frame carried in the packet is the first frame in a talkspurt (see definition of the talkspurt in Section 4.1 of [RFC3551]). For all other packets, the marker bit SHALL be set to zero (M=0).

The assignment of an RTP payload type for the format defined in this memo is outside the scope of this document. The RTP profiles in use currently mandate binding the payload type dynamically for this payload format.

The remaining RTP header fields are used as specified in RFC 3550 [RFC3550].

5.2. Payload Structure

The complete payload consists of a payload table of contents (ToC) section, followed by speech data representing one or more speech frames, SID frames, or No_Data frames. The following diagram shows the general payload format layout:

```
+-------------+-----------------------
| ToC section | speech data section ...
+-------------+-----------------------
```

Figure 2: General Payload Format Layout

Each ToC element is one octet and corresponds to one speech frame; the number of ToC elements is thus equal to the number of speech frames (including SID frames and No_Data frames). Each ToC entry represents a consecutive speech or SID or No_Data frame. The timestamp value for ToC element (and corresponding speech frame data) N within the payload is (RTP timestamp field + (N-1)*160) mod 2^32. The format of the ToC element is as follows.

```
 0 1 2 3 4 5 6 7
+-+-+-+-+-+-+-+-+
|F| FT  |R R R R|
+-+-+-+-+-+-+-+-+
```

Figure 3: The TOC Element

F: Follow flag; 1 denotes that more ToC elements follow; 0 denotes the last ToC element.

R: Reserved bits; MUST be set to zero, and MUST be ignored by receiver.

FT: Frame type
- 000 = Good Speech frame
- 001 = Reserved
- 010 = Good SID frame
- 011 = Reserved
- 100 = Reserved
- 101 = Reserved
- 110 = Reserved
- 111 = No_Data frame

The length of the payload data depends on the frame type:

Good Speech frame: The 112 speech data bits are put in 14 octets.

Good SID frame: The 33 SID data bits are put in 14 octets, as in the case of Speech frames, with the unused 79 bits all set to "1".

No_Data frame: Length of payload data is zero octets.

Frames marked in the GSM radio subsystem as "Bad Speech frame", "Bad SID frame", or "No_Data frame" are not sent in RTP packets, in order to save bandwidth. They are marked as "No_Data frame", if they occur within an RTP packet that carries more than one speech frame, SID frame, or No_Data frame.

5.2.1. Encoding of Speech Frames

The 112 bits of GSM-HR-coded speech (b1...b112) are defined in TS 46.020, Annex B [TS46.020], in their order of occurrence. The first bit (b1) of the first parameter is placed in the most significant bit (MSB) (bit 0) of the first octet (octet 1) of the payload field; the second bit is placed in bit 1 of the first octet; and so on. The last bit (b112) is placed in the least significant bit (LSB) (bit 7) of octet 14.

5.2.2. Encoding of Silence Description Frames

The GSM-HR codec applies a specific coding for silence periods in so-called SID frames. The coding of SID frames is based on the coding of speech frames by using only the first 33 bits for SID parameters and by setting all of the remaining 79 bits to "1".

5.3. Implementation Considerations

An application implementing this payload format MUST understand all the payload parameters that are defined in this specification. Any mapping of the parameters to a signaling protocol MUST support all parameters. So an implementation of this payload format in an application using SDP is required to understand all the payload parameters in their SDP-mapped form. This requirement ensures that an implementation always can decide whether it is capable of communicating when the communicating entities support this version of the specification.

5.3.1. Transmission of SID Frames

When using this RTP payload format, the sender SHOULD generate and send SID frames every 160 ms, i.e., every 8th frame, during silent periods. Other SID transmission intervals may occur due to gateways to other systems that use other transmission intervals.

5.3.2. Receiving Redundant Frames

The reception of redundant audio frames, i.e., more than one audio frame from the same source for the same time slot, MUST be supported by the implementation.

5.3.3. Decoding Validation

If the receiver finds a mismatch between the size of a received payload and the size indicated by the ToC of the payload, the receiver SHOULD discard the packet. This is recommended, because decoding a frame parsed from a payload based on erroneous ToC data could severely degrade the audio quality.

6. Examples

A few examples below highlight the payload format.

6.1. 3 Frames

Below is a basic example of the aggregation of 3 consecutive speech frames into a single packet.

```
   The first 24 bits are ToC elements.

   Bit 0 is '1', as another ToC element follows.
   Bits 1..3 are 000 = Good speech frame
   Bits 4..7 are 0000 = Reserved
   Bit 8 is '1', as another ToC element follows.
   Bits 9..11 are 000 = Good speech frame
   Bits 12..15 are 0000 = Reserved
   Bit 16 is '0'; no more ToC elements follow.
   Bits 17..19 are 000 = Good speech frame
   Bits 20..23 are 0000 = Reserved

    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 3 4 5 6 7 8 9 0 1
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |1|0 0 0|0 0 0 0|1|0 0 0|0 0 0 0|0|0 0 0|0 0 0 0|b1           b8|
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+               +
   |b9   Frame 1                                                b40|
   +                                                               +
   |b41                                                         b72|
   +                                                               +
   |b73                                                        b104|
   +               +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |b105       b112|b1                                          b24|
   +-+-+-+-+-+-+-+-+                                               +
   |b25  Frame 2                                                b56|
   +                                                               +
   |b57                                                         b88|
   +                                                +-+-+-+-+-+-+-+-+
   |b89                                         b112|b1           b8|
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+               +
   |b9   Frame 3                                                b40|
   +                                                               +
   |b41                                                         b72|
   +                                                               +
   |b73                                                        b104|
   +               +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |b105       b112|
   +-+-+-+-+-+-+-+-+
```

6.2. 3 Frames with Lost Frame in the Middle

Below is an example of a payload carrying 3 frames, where the middle one is No_Data (for example, due to loss prior to transmission by the RTP source).

The first 24 bits are ToC elements.

```
Bit 0 is '1', as another ToC element follows.
Bits 1..3 are 000 = Good speech frame
Bits 4..7 are 0000 = Reserved
Bit 8 is '1', as another ToC element follows.
Bits 9..11 are 111 = No_Data frame
Bits 12..15 are 0000 = Reserved
Bit 16 is '0'; no more ToC elements follow.
Bits 17..19 are 000 = Good speech frame
Bits 20..23 are 0000 = Reserved
```

```
 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 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|1|0 0 0|0 0 0 0|1|1 1 1|0 0 0 0|0|0 0 0|0 0 0 0|b1           b8|
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+               +
|b9   Frame 1                                                b40|
+                                                               +
|b41                                                         b72|
+                                                               +
|b73                                                        b104|
+               +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|b105      b112|b1                                           b24|
+-+-+-+-+-+-+-+-+                                               +
|b25  Frame 3                                                b56|
+                                                               +
|b57                                                         b88|
+                                                +-+-+-+-+-+-+-+-+
|b89                                        b112|
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

7. Payload Format Parameters

This RTP payload format is identified using the media type "audio/GSM-HR-08", which is registered in accordance with [RFC4855] and uses [RFC4288] as a template. Note: Media subtype names are case-insensitive.

7.1. Media Type Definition

The media type for the GSM-HR codec is allocated from the IETF tree, since GSM-HR is a well-known speech codec. This media type registration covers real-time transfer via RTP.

Note: Reception of any unspecified parameter MUST be ignored by the receiver to ensure that additional parameters can be added in the future.

Type name: audio

Subtype name: GSM-HR-08

Required parameters: none

Optional parameters:

max-red: The maximum duration in milliseconds that elapses between the primary (first) transmission of a frame and any redundant transmission that the sender will use. This parameter allows a receiver to have a bounded delay when redundancy is used. Allowed values are integers between 0 (no redundancy will be used) and 65535. If the parameter is omitted, no limitation on the use of redundancy is present.

ptime: See [RFC4566].

maxptime: See [RFC4566].

Encoding considerations:

This media type is framed and binary; see Section 4.8 of RFC 4288 [RFC4288].

Security considerations:

See Section 10 of RFC 5993.

Interoperability considerations:

The media subtype name contains "-08" to avoid potential conflict with any earlier drafts of GSM-HR RTP payload types that aren't bit-compatible.

Published specifications:

RFC 5993, 3GPP TS 46.002

Applications that use this media type:

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

Additional information: none

Person & email address to contact for further information:

Ingemar Johansson <ingemar.s.johansson@ericsson.com>

Intended usage: COMMON

Restrictions on usage:

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.

Authors:

Xiaodong Duan <duanxiaodong@chinamobile.com>

Shuaiyu Wang <wangshuaiyu@chinamobile.com>

Magnus Westerlund <magnus.westerlund@ericsson.com>

Ingemar Johansson <ingemar.s.johansson@ericsson.com>

Karl Hellwig <karl.hellwig@ericsson.com>

Change controller:

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

7.2. Mapping to SDP

The information carried in the media type specification has a specific mapping to fields in the Session Description Protocol (SDP) [RFC4566], which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing the GSM-HR codec, the mapping is as follows:

- o 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 be 8000, and the encoding parameters (number of channels) MUST either be explicitly set to 1 or omitted, implying a default value of 1.

- 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.

7.2.1. Offer/Answer Considerations

The following considerations apply when using SDP offer/answer procedures to negotiate the use of GSM-HR payload in RTP:

- The SDP offerer and answerer MUST generate GSM-HR packets as described by the offered parameters.

- 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 [RFC3264]. The "maxptime" parameter MUST be handled in the same way.

- The parameter "max-red" is a stream property parameter. For sendonly or sendrecv unicast media streams, the parameter declares the limitation on redundancy that the stream sender will use. For recvonly streams, it indicates the desired value for the stream sent to the receiver. The answerer MAY change the value, but is RECOMMENDED to use the same limitation as the offer declares. In the case of multicast, the offerer MAY declare a limitation; this SHALL be answered using the same value. A media sender using this payload format is RECOMMENDED to always include the "max-red" parameter. This information is likely to simplify the media stream handling in the receiver. This is especially true if no redundancy will be used, in which case "max-red" is set to 0.

- Any unknown media type parameter in an offer SHALL be removed in the answer.

7.2.2. Declarative SDP Considerations

In declarative usage, like SDP in the Real Time Streaming Protocol (RTSP) [RFC2326] or the Session Announcement Protocol (SAP) [RFC2974], the parameters SHALL be interpreted as follows:

- o The stream property parameter ("max-red") is declarative, and a participant MUST follow what is declared for the session. In this case, it means that the receiver MUST be prepared to allocate buffer memory for the given redundancy. Any transmissions MUST NOT use more redundancy than what has been declared. More than one configuration may be provided if necessary by declaring multiple RTP payload types; however, the number of types should be kept small.

- o Any "maxptime" and "ptime" values should be selected with care to ensure that the session's participants can achieve reasonable performance.

8. IANA Considerations

One media type (audio/GSM-HR-08) has been defined, and it has been registered in the media types registry; see Section 7.1.

9. Congestion Control

The general congestion control considerations for transporting RTP data apply; see RTP [RFC3550] and any applicable RTP profiles, e.g., "RTP/AVP" [RFC3551].

The number of frames encapsulated in each RTP payload highly influences the overall bandwidth of the RTP stream due to header overhead constraints. Packetizing more frames in each RTP payload can reduce the number of packets sent and hence the header overhead, at the expense of increased delay and reduced error robustness. If forward error correction (FEC) is used, the amount of FEC-induced redundancy needs to be regulated such that the use of FEC itself does not cause a congestion problem.

10. 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 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, and integrity of the RTP packets through a suitable cryptographic integrity protection mechanism. A cryptographic system 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 whether or not an RTP packet is from a member of the RTP session.

Note that the appropriate mechanism to provide security to RTP and payloads following this 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 the Secure Real-time Transport Protocol (SRTP) [RFC3711] is recommended. Other mechanisms that may be used are IPsec [RFC4301] and Transport Layer Security (TLS) [RFC5246] (e.g., for RTP over TCP), but other alternatives may also exist.

This RTP payload format and its media decoder do not exhibit any significant non-uniformity in the receiver-side computational complexity for packet processing, and thus are unlikely to pose a denial-of-service threat due to the receipt of pathological data; nor does the RTP payload format contain any active content.

11. Acknowledgements

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Authors' Addresses

Xiaodong Duan
China Mobile Communications Corporation
53A, Xibianmennei Ave., Xuanwu District
Beijing, 100053
P.R. China
EMail: duanxiaodong@chinamobile.com

Shuaiyu Wang
China Mobile Communications Corporation
53A, Xibianmennei Ave., Xuanwu District
Beijing, 100053
P.R. China
EMail: wangshuaiyu@chinamobile.com

Magnus Westerlund
Ericsson AB
Farogatan 6
Stockholm, SE-164 80
Sweden
Phone: +46 8 719 0000
EMail: magnus.westerlund@ericsson.com

Karl Hellwig
Ericsson AB
Ericsson Allee 1
52134 Herzogenrath
Germany
Phone: +49 2407 575-2054
EMail: karl.hellwig@ericsson.com

Ingemar Johansson
Ericsson AB
Laboratoriegrand 11
SE-971 28 Lulea
Sweden
Phone: +46 73 0783289
EMail: ingemar.s.johansson@ericsson.com