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RTP Payload Format for European Telecommunications Standards Institute (ETSI) European Standard ES 201 108 Distributed Speech Recognition Encoding

Status of this Memo

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Abstract

This document specifies an RTP payload format for encapsulating European Telecommunications Standards Institute (ETSI) European Standard (ES) 201 108 front-end signal processing feature streams for distributed speech recognition (DSR) systems.

Table of Contents

1. Conventions 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 [RFC2119].

The following acronyms are used in this document:

DSR - Distributed Speech Recognition

ETSI - the European Telecommunications Standards Institute

FP - Frame Pair

DTX - Discontinuous Transmission

2. Introduction

Motivated by technology advances in the field of speech recognition, voice interfaces to services (such as airline information systems, unified messaging) are becoming more prevalent. In parallel, the popularity of mobile devices has also increased dramatically.

However, the voice codecs typically employed in mobile devices were designed to optimize audible voice quality and not speech recognition accuracy, and using these codecs with speech recognizers can result in poor recognition performance. For systems that can be accessed from heterogeneous networks using multiple speech codecs, recognition system designers are further challenged to accommodate the characteristics of these differences in a robust manner. Channel errors and lost data packets in these networks result in further degradation of the speech signal.

In traditional systems as described above, the entire speech recognizer lies on the server. It is forced to use incoming speech in whatever condition it arrives after the network decodes the vocoded speech. To address this problem, we use a distributed speech recognition (DSR) architecture. In such a system, the remote device acts as a thin client, also known as the front-end, in communication with a speech recognition server, also called a speech engine. The remote device processes the speech, compresses the data, and adds error protection to the bitstream in a manner optimal for speech recognition. The speech engine then uses this representation directly, minimizing the signal processing necessary and benefiting from enhanced error concealment.

To achieve interoperability with different client devices and speech engines, a common format is needed. Within the "Aurora" DSR working group of the European Telecommunications Standards Institute (ETSI), a payload has been defined and was published as a standard [ES201108] in February 2000.

For voice dialogues between a caller and a voice service, low latency is a high priority along with accurate speech recognition. While jitter in the speech recognizer input is not particularly important, many issues related to speech interaction over an IP-based connection are still relevant. Therefore, it is desirable to use the DSR payload in an RTP-based session.

2.1 ETSI ES 201 108 DSR Front-end Codec

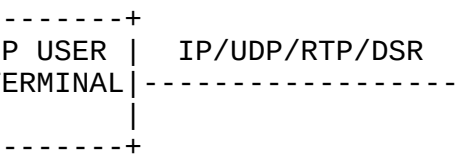
The ETSI Standard ES 201 108 for DSR [ES201108] defines a signal processing front-end and compression scheme for speech input to a speech recognition system. Some relevant characteristics of this ETSI DSR front-end codec are summarized below.

The coding algorithm, a standard mel-cepstral technique common to many speech recognition systems, supports three raw sampling rates: 8 kHz, 11 kHz, and 16 kHz. The mel-cepstral calculation is a frame-based scheme that produces an output vector every 10 ms.

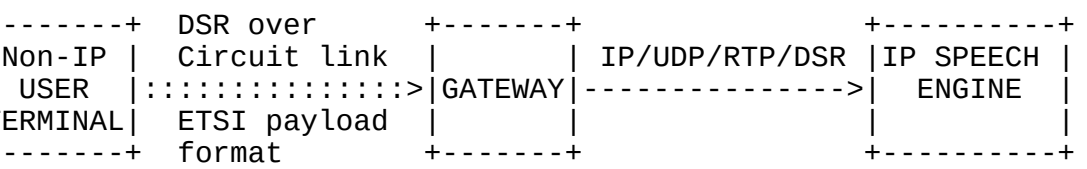
After calculation of the mel-cepstral representation, the representation is first quantized via split-vector quantization to reduce the data rate of the encoded stream. Then, the quantized vectors from two consecutive frames are put into an FP, as described in more detail in Section 4.1.

2.2 Typical Scenarios for Using DSR Payload Format

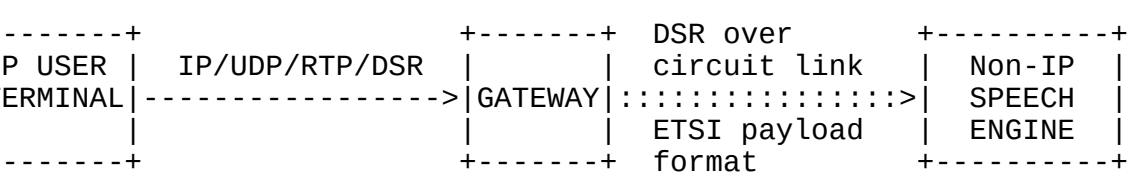
The diagrams in Figure 1 show some typical use scenarios of the ES 201 108 DSR RTP payload format.



a) IP user terminal to IP speech engine



b) non-IP user terminal to IP speech engine via a gateway



c) IP user terminal to non-IP speech engine via a gateway

Figure 1: Typical Scenarios for Using DSR Payload Format.

For the different scenarios in Figure 1, the speech recognizer always resides in the speech engine. A DSR front-end encoder inside the User Terminal performs front-end speech processing and sends the resultant data to the speech engine in the form of "frame pairs" (FPs). Each FP contains two sets of encoded speech vectors representing 20ms of original speech.

3. ES 201 108 DSR RTP Payload Format

An ES 201 108 DSR RTP payload datagram consists of a standard RTP header [RFC3550] followed by a DSR payload. The DSR payload itself is formed by concatenating a series of ES 201 108 DSR FPs (defined in Section 4).

FPs are always packed bit-contiguously into the payload octets beginning with the most significant bit. For ES 201 108 front-end, the size of each FP is 96 bits or 12 octets (see Sections 4.1 and 4.2). This ensures that a DSR payload will always end on an octet boundary.

The following example shows a DSR RTP datagram carrying a DSR payload containing three 96-bit-long FPs (bit 0 is the MSB):

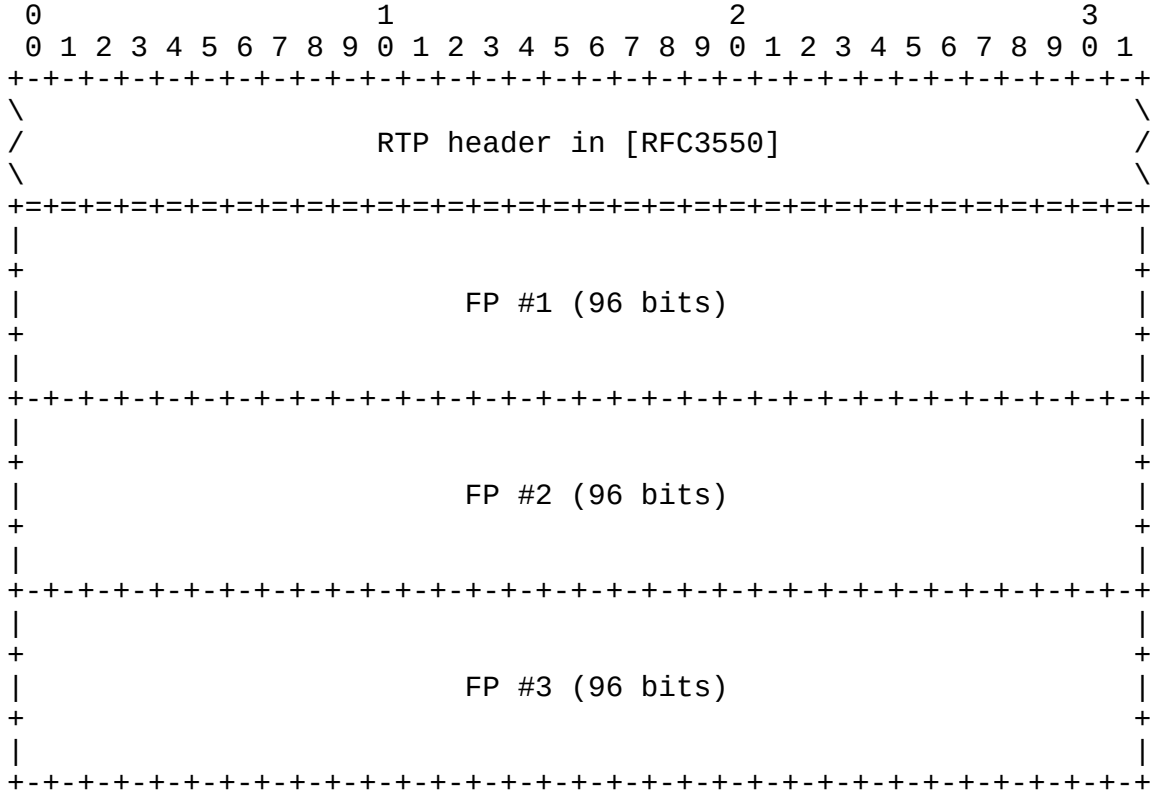


Figure 2. An example of an ES 201 108 DSR RTP payload.

3.1 Consideration on Number of FPs in Each RTP Packet

The number of FPs per payload packet should be determined by the latency and bandwidth requirements of the DSR application using this payload format. In particular, using a smaller number of FPs per payload packet in a session will result in lowered bandwidth efficiency due to the RTP/UDP/IP header overhead, while using a larger number of FPs per packet will cause longer end-to-end delay and hence increased recognition latency. Furthermore, carrying a larger number of FPs per packet will increase the possibility of catastrophic packet loss; the loss of a large number of consecutive FPs is a situation most speech recognizers have difficulty dealing with.

It is therefore RECOMMENDED that the number of FPs per DSR payload packet be minimized, subject to meeting the application's requirements on network bandwidth efficiency. RTP header compression techniques, such as those defined in [RFC2508] and [RFC3095], should be considered to improve network bandwidth efficiency.

3.2 Support for Discontinuous Transmission

The DSR RTP payloads may be used to support discontinuous transmission (DTX) of speech, which allows that DSR FPs are sent only when speech has been detected at the terminal equipment.

In DTX a set of DSR frames coding an unbroken speech segment transmitted from the terminal to the server is called a transmission segment. A DSR frame inside such a transmission segment can be either a speech frame or a non-speech frame, depending on the nature of the section of the speech signal it represents.

The end of a transmission segment is determined at the sending end equipment when the number of consecutive non-speech frames exceeds a pre-set threshold, called the hangover time. A typical value used for the hangover time is 1.5 seconds.

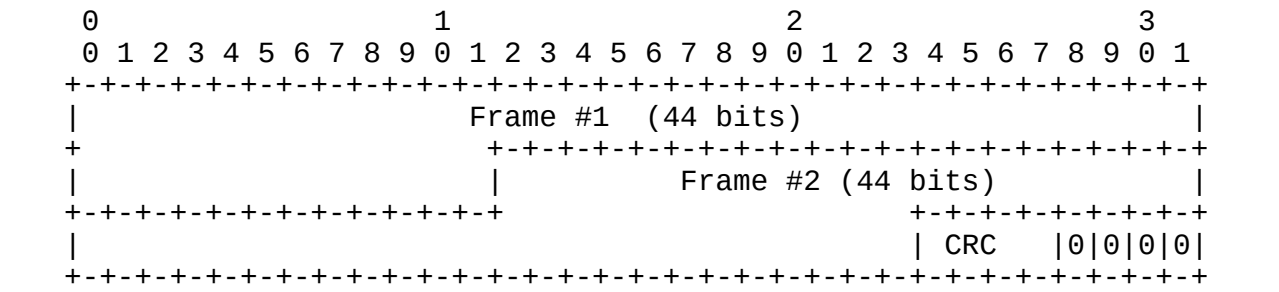
After all FPs in a transmission segment are sent, the front-end SHOULD indicate the end of the current transmission segment by sending one or more Null FPs (defined in Section 4.2).

4. Frame Pair Formats

4.1 Format of Speech and Non-speech FPs

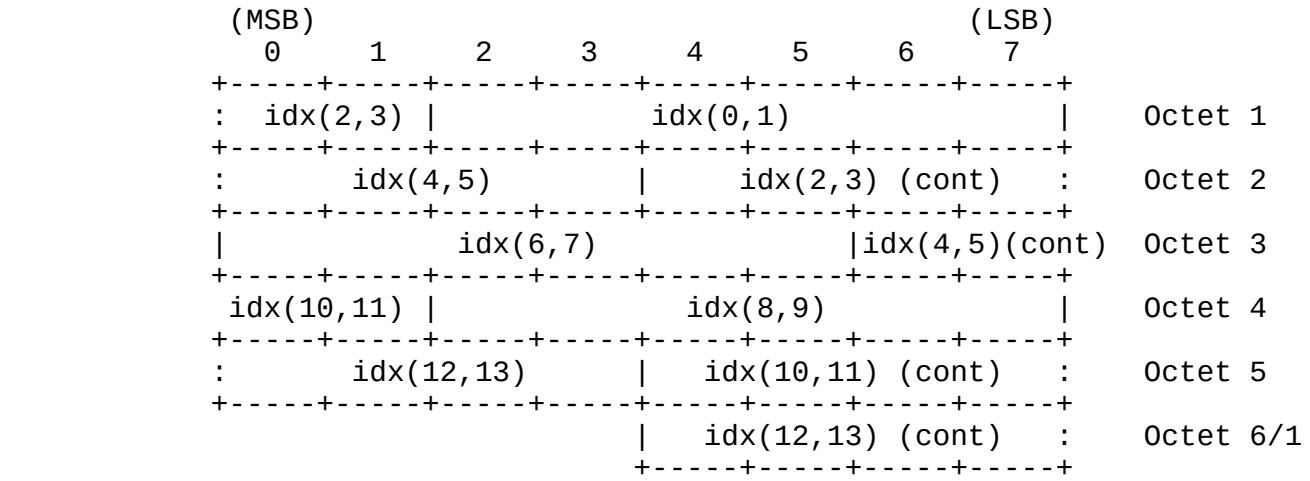
The following mel-cepstral frame MUST be used, as defined in [ES201108]:

As defined in [ES201108], pairs of the quantized 10ms mel-cepstral frames MUST be grouped together and protected with a 4-bit CRC, forming a 92-bit long FP:

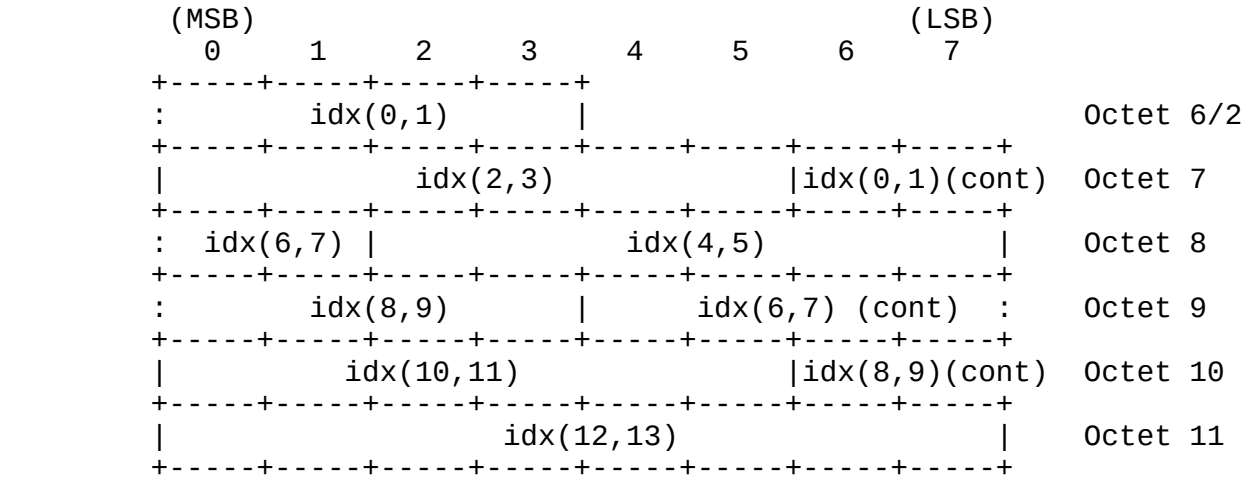


The length of each frame is 44 bits representing 10ms of voice. The following mel-cepstral frame formats MUST be used when forming an FP:

Frame #1 in FP:



Frame #2 in FP:
===============



Therefore, each FP represents 20ms of original speech. Note, as shown above, each FP MUST be padded with 4 zeros to the end in order to make it aligned to the 32-bit word boundary. This makes the size of an FP 96 bits, or 12 octets. Note, this padding is separate from padding indicated by the P bit in the RTP header.

The 4-bit CRC MUST be calculated using the formula defined in 6.2.4 in [ES201108]. The definition of the indices and the determination of their value are also described in [ES201108].

4.2 Format of Null FP

A Null FP for the ES 201 108 front-end codec is defined by setting the content of the first and second frame in the FP to null (i.e., filling the first 88 bits of the FP with 0's). The 4-bit CRC MUST be calculated the same way as described in 6.2.4 in [ES201108], and 4 zeros MUST be padded to the end of the Null FP to make it 32-bit word aligned.

4.3 RTP header usage

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

The RTP timestamp corresponds to the sampling instant of the first sample encoded for the first FP in the packet. The timestamp clock frequency is the same as the sampling frequency, so the timestamp unit is in samples.

As defined by ES 201 108 front-end codec, the duration of one FP is 20 ms, corresponding to 160, 220, or 320 encoded samples with sampling rate of 8, 11, or 16 kHz being used at the front-end, respectively. Thus, the timestamp is increased by 160, 220, or 320 for each consecutive FP, respectively.

The DSR payload for ES 201 108 front-end codes is always an integral number of octets. If additional padding is required for some other purpose, then the P bit in the RTP in the header may be set and padding appended as specified in [RFC3550].

The RTP header marker bit (M) should be set following the general rules defined in [RFC3551].

The assignment of an RTP payload type for this new packet format is outside the scope of this document, and will not be specified here. It is expected that the RTP profile under which this payload format is being used will assign a payload type for this encoding or specify that the payload type is to be bound dynamically.

5. IANA Considerations

One new MIME subtype registration is required for this payload type, as defined below.

This section also defines the optional parameters that may be used to describe a DSR session. The parameters are defined here as part of the MIME subtype registration. A mapping of the parameters into the Session Description Protocol (SDP) [RFC2327] is also provided in 5.1 for those applications that use SDP.

Media Type name: audio

Media subtype name: dsr-es201108

Required parameters: none

Optional parameters:

rate: Indicates the sample rate of the speech. Valid values include: 8000, 11000, and 16000. If this parameter is not present, 8000 sample rate is assumed.

maxptime: The maximum amount of media which can be encapsulated in each packet, expressed as time in milliseconds. The time shall be calculated as the sum of the time the media present in the packet represents. The time SHOULD be a multiple of the frame pair size (i.e., one FP <-> 20ms).

If this parameter is not present, maxptime is assumed to be 80ms.

Note, since the performance of most speech recognizers are extremely sensitive to consecutive FP losses, if the user of the payload format expects a high packet loss ratio for the session, it MAY consider to explicitly choose a maxptime value for the session that is shorter than the default value.

ptime: see RFC2327 [RFC2327].

Encoding considerations : This type is defined for transfer via RTP [RFC3550] as described in Sections 3 and 4 of RFC 3557.

Security considerations : See Section 6 of RFC 3557.

Person & email address to contact for further information:
Qiaobing.Xie@motorola.com

Intended usage: COMMON. It is expected that many VoIP applications (as well as mobile applications) will use this type.

Author/Change controller:
Qiaobing.Xie@motorola.com
IETF Audio/Video transport working group

5.1 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 (SDP) [RFC2327], which is commonly used to describe RTP sessions. When SDP is used to specify sessions employing ES 201 018 DSR codec, the mapping is as follows:

- The MIME type ("audio") goes in SDP "m=" as the media name.
- The MIME subtype ("dsr-es201108") goes in SDP "a=rtpmap" as the encoding name.
- The optional parameter "rate" also goes in "a=rtpmap" as clock rate.
- The optional parameters "ptime" and "maxptime" go in the SDP "a=ptime" and "a=maxptime" attributes, respectively.

Example of usage of ES 201 108 DSR:

```
m=audio 49120 RTP/AVP 101
a=rtpmap:101 dsr-es201108/8000
a=maxptime:40
```

6. Security Considerations

Implementations using the payload defined in this specification are subject to the security considerations discussed in the RTP specification [RFC3550] and the RTP profile [RFC3551]. This payload does not specify any different security services.

7. Contributors

The following individuals contributed to the design of this payload format and the writing of this document: Q. Xie (Motorola), D. Pearce (Motorola), S. Balasuriya (Motorola), Y. Kim (VerbalTek), S. H. Maes (IBM), and, Hari Garudadri (Qualcomm).

8. Acknowledgments

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9. References

9.1 Normative References

[ES201108] European Telecommunications Standards Institute (ETSI) Standard ES 201 108, "Speech Processing, Transmission and Quality Aspects (STQ); Distributed Speech Recognition; Front-end Feature Extraction Algorithm; Compression Algorithms," Ver. 1.1.2, April 11, 2000.

[RFC3550] Schulzrinne, H., Casner, S., Jacobson, V. and R. Frederick, "RTP: A Transport Protocol for Real-Time Applications", RFC 3550, July 2003.

[RFC2026] Bradner, S., "The Internet Standards Process -- Revision 3", BCP 9, RFC 2026, October 1996.

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC2327] Handley, M. and V. Jacobson, "SDP: Session Description Protocol", RFC 2327, April 1998.

9.2 Informative References

[RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", RFC 3551, July 2003.

[RFC2508] Casner, S. and V. Jacobson, "Compressing IP/UDP/RTP Headers for Low-Speed Serial Links", RFC 2508, February 1999.

[RFC3095] Bormann, C., Burmeister, C., Degermark, M., Fukushima, H., Hannu, H., Jonsson, L-E, Hakenberg, R., Koren, T., Le, K., Liu, Z., Martensson, A., Miyazaki, A., Svanbro, K., Wiebke, T., Yoshimura, T. and H. Zheng, "RObust Header Compression (ROHC): Framework and four profiles", RFC 3095, July 2001.

10. IPR Notices

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