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Real-time Transport Protocol (RTP) Payload Format for Enhanced AC-3 (E-AC-3) Audio

Status of This Memo

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Abstract

This document describes a Real-time Transport Protocol (RTP) payload format for transporting Enhanced AC-3 (E-AC-3) encoded audio data. E-AC-3 is a high-quality, multichannel audio coding format and is an extension of the AC-3 audio coding format, which is used in US High-Definition Television (HDTV), DVD, cable and satellite television, and other media. E-AC-3 is an optional audio format in US and world wide digital television and high-definition DVD formats. The RTP payload format as presented in this document includes support for data fragmentation.

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1. Introduction

The Enhanced AC-3 (E-AC-3) [ETSI] audio coding system is built on a foundation of AC-3. It is an enhancement and extension to AC-3, which is an existing audio coding standard commonly used for DVD, broadcast, cable, and satellite television content. E-AC-3 is designed to enable operation at both higher and lower data rates than AC-3, provide expanded channel configurations, and provide greater flexibility for carriage of multiple audio program elements. The relationship between E-AC-3 and AC-3 provides for low-loss, low-cost conversion between the two and makes E-AC-3 especially suitable in applications that require compatibility with the existing broadcast-reception and audio/video decoding infrastructure. Dolby Digital Plus is a branded version of Enhanced AC-3.

E-AC-3 has been standardized within both the European Telecommunications Standards Institute (ETSI) and the Advanced Television Systems Committee (ATSC). It is an optional audio format for use in US (ATSC) and Digital Video Broadcasting (DVB) television transmission. It is also a required audio format for use in the High Definition (HD)-DVD optical-storage media format and included in the Blu-ray Disc format.

There is a need to stream E-AC-3 content over IP networks. E-AC-3 is primarily used in audio-for-video applications, so RTP serves well as a transport solution with its mechanism for synchronizing streams. Applications for streaming E-AC-3 include Internet Protocol television (IPTV), video on demand, interactive features of next generation DVD formats, and transfer of movies across a home network.

Section 2 gives a brief overview of the E-AC-3 algorithm. Section 3 specifies values for fields in the RTP header, and Section 4 specifies the E-AC-3 payload format, itself. Section 5 discusses media types and Session Description Protocol (SDP) usage. Security considerations are covered in Section 6, congestion control in Section 7, and IANA considerations in Section 8.

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].

2. Overview of Enhanced-AC-3

Enhanced AC-3 (E-AC-3) is a frequency-domain perceptual audio coding system. Time blocks of an audio signal are converted from the time domain to the frequency domain by a transform (the Modified Discrete Cosine Transform (MDCT)) so that a model of the human auditory perceptual system can be applied. In this domain, quantization noise can be constrained to specific frequency regions. The perceptual model predicts in which frequency regions the auditory system will be least able to detect the quantization noise from data rate reduction. A more detailed technical description of E-AC-3 can be found in [2004AES].

E-AC-3 is built upon a foundation of AC-3. More background on AC-3 can be found in the AC-3 specification [ETSI], a technical paper [1994AES], and the AC-3 RTP payload format [RFC4184]. The frame structure and meta-data of AC-3 are maintained. E-AC-3 content is not directly compatible with AC-3 decoders, but it can be converted to the AC-3 format to provide compatibility with existing decoders. Because AC-3 is the foundation of E-AC-3, conversion between the two formats can be done in a way that minimizes the degradations associated with tandem coding. In addition, the computational cost of the conversion is reduced compared to a full decode and re-encode.

E-AC-3 exploits psychoacoustic phenomena that cause a significant fraction of the information contained in a typical audio signal to be inaudible. Substantial data reduction occurs via the removal of inaudible information contained in an audio stream. Source coding techniques are further used to reduce the data rate.

Like most perceptual coders, E-AC-3 operates in the frequency domain. A 512-point MDCT transform is taken with 50% overlap, providing 256 new frequency samples. Frequency samples are then converted to exponents and mantissas. Exponents are differentially encoded. Mantissas are allocated a varying number of bits depending on the audibility of the spectral components associated with them. Audibility is determined via a masking curve. Bits for mantissas are allocated from a global bit pool.

E-AC-3 adds new coding tools, such as a longer filter bank, vector quantization, and spectral extension, to provide greater data efficiency and to operate at lower data rates than AC-3. In the other direction, an expanded bit stream syntax and new frame constraints permit operation at higher data rates than AC-3. The E-AC-3 syntax also allows a larger number of audio channels in one bit stream. E-AC-3 operates at data rates from 32 kbps to 6.144 Mbps and at three sampling rates: 32 kHz, 44.1 kHz, and 48 kHz.

E-AC-3 supports the carriage of multiple programs and the carriage of programs with more than a baseline of 5.1 audio channels. Both of these extensions beyond AC-3 are accomplished by time multiplexing additional data with baseline data. In the case of multiple programs, frames with data for the programs are interleaved. In the case of more than 5.1 channels, frames from substreams carrying the extra channels are interleaved with the independent substream that carries a 5.1-channel compatible mix. Both of these forms of multiplexing can occur in the same bit stream. In other words, mixing multiple programs, some or all with more than 5.1 channels, is permitted.

Additional channel capacity is enabled by adding substreams to a program. One primary substream, called the "independent substream", is required for each program. This substream carries a self-contained mix of the audio, using a maximum of 5.1 channels, which makes its channel configuration compatible with AC-3. Then, additional, optional substreams are used in the program to carry additional channels. The data for each additional channel carries an indication of whether that channel provides data for an additional speaker location or replacement data for one of the speaker locations already defined by a previous substream. For example, one common 7.1-channel format uses three front channels and four surround channels. It is packaged with a primary substream, which contains a 5.1-channel downmix of the 7.1-channel content, using left, center, right, left surround, right surround, and low-frequency effects channels. One dependent substream supplies four channels: replacements for left surround and right surround, along with two additional surround channels (left back and right back).

The specification for E-AC-3 [ETSI] requires that all E-AC-3 decoders be capable of decoding at least a baseline portion of any E-AC-3 bit stream, which consists of the first independent substream of the first program, and of ignoring the other elements of the bit stream. This baseline is limited to 5.1 channels, and a system is also able to convert to configurations with fewer channels for a presentation that matches its output capabilities, if needed. More capable decoders can optionally choose among and mix multiple programs, and also decode configurations with more channels than the baseline by decoding dependent substreams.

2.1. E-AC-3 Bit Stream

2.1.1. Sync Frames and Audio Blocks

The basic organizational building block in an E-AC-3 bit stream is the sync frame (also called a frame in this document). A sync frame contains the data necessary to decode time domain audio samples for one or more channels over a time of one or more audio blocks, so a frame is an Application Data Unit (ADU). Each E-AC-3 frame contains a Sync Information (SI) field, a Bit Stream Information (BSI) field, an Audio Frame (AF) field, and up to six audio blocks (ABs). Each AB represents 256 Pulse Code Modulation (PCM) samples for each channel. The frame ends with an optional auxiliary data field (AUX) and an error correction field (CRC). Figure 1 shows the structure of an E-AC-3 frame, where N is the number of blocks in the frame.

```
+---+---+---+---------+- ... -+---------+---+---+
|SI |BSI|AF |  AB(0)  |  ...  |  AB(N)  |AUX|CRC|
+---+---+---+---------+- ... -+---------+---+---+
```

Figure 1. E-AC-3 frame format with more than one block

The SI field contains information needed to acquire and maintain codec synchronization. The BSI field contains parameters that describe the coded audio service. It carries an indication of the size of the frame in 16-bit words ('frmsiz', Section E.1.3 of [ETSI]) and an indication of the sampling rate ('fscod'). It also carries an indication of the number of blocks in the frame ('numblkscod'); permitted values are one, two, three, or six blocks. The AF field contains information about coding tools that applies to the entire frame. Each block has a duration of 256 samples, so a frame's duration is the corresponding multiple of 256 samples. The time duration of the frame is also dependent on the sampling rate, as shown in Table 1.

Table 1. Time duration of E-AC-3 frame (number of blocks vs. sampling rate)

blocks per frame	32 kHz	44.1 kHz	48 kHz
1	8 ms	approx. 5.8 ms	approx. 5.3 ms
2	16 ms	approx. 11.6 ms	approx. 10.7 ms
3	24 ms	approx. 17.4 ms	16 ms
6	48 ms	approx. 34.8 ms	32 ms

Each audio block contains header fields that indicate the use of various coding tools: block switching, dither, coupling, spectral extension, and exponent strategy. They also contain metadata, optionally used to enhance playback, such as dynamic range control. Finally, the exponents and bit allocation data needed to decode the mantissas into audio data, and the mantissas themselves, are included. The format of audio blocks is described in detail in [ETSI].

2.1.2. Programs and Substreams

An E-AC-3 bit stream is logically arranged into programs. A bit stream contains one or more programs, up to a maximum of eight. When multiple programs are present in a bit stream, the frames that constitute them are interleaved in time.

```
+----------+-     -+----------+----------+-     -+----------+-
|Program(1)|  ...  |Program(N)|Program(1)|  ...  |Program(N)| ...
| Frame 0  |       | Frame 0  | Frame 1  |       | Frame 1  |
+----------+-     -+----------+----------+-     -+----------+-
```

Figure 2. Interleaving of multiple programs in an E-AC-3 bit stream

Each program contains one independent substream and optionally contains up to eight dependent substreams. The independent substream carries a soundtrack of up to 5.1 channels, the multichannel format that matches the capabilities of AC-3, and can be meaningfully decoded and presented without any of the associated dependent substreams. The dependent substreams are used to provide alternate channel data that enable different channel configurations, for example, to increase the number of channels beyond 5.1. A frame of a dependent substream can be decoded by itself, but its content can only be meaningfully presented in conjunction with the corresponding independent substream. The type and identity of the substream to which a frame belongs can be determined from parameters in the frame's BSI (strmtyp and substreamid, in Section E.1.3.1 of [ETSI]).

When a program contains more than one substream, the frames belonging to those substreams are interleaved in time, and taken together, the frames of a program that correspond to the same time period are called a 'program set'. Figure 3 shows the interleaving of substreams for a single program.

```
  / --------- program set for frame 0 ------- \
  :                                           :
+-------------+-------------+-   -+-------------+-------------+-
|  Program(1) |  Program(1) |     |  Program(1) |  Program(1) |
| Independent |  Dependent  | ... |  Dependent  | Independent | ...
|  Substream  | Substream(0)|     | Substream(n)|  Substream  |
|   Frame 0   |   Frame 0   |     |   Frame 0   |   Frame 1   |
+-------------+-------------+-   -+-------------+-------------+-
```

Figure 3. Interleaving of multiple substreams in an E-AC-3 program

2.1.3. Frame Sets

A further logical organization of the E-AC-3 bit stream is applied to facilitate conversion of E-AC-3 bit streams to AC-3 bit streams. In this organization, the frames carrying six consecutive audio blocks are treated as a group, called a 'frame set', regardless of the number of frames needed to carry six audio blocks. This grouping extends across all programs and substreams that cover the time period of the six blocks. Since E-AC-3 frames may carry one, two, three, or six blocks, a frame set will consist of six, three, two, or one frames. AC-3 frames always carry six blocks, so the frame set provides framing synchronization between an E-AC-3 bit stream and an AC-3 bit stream. Metadata that indicates the alignment is carried in the first frame (which will be part of an independent substream) of each frame set in an E-AC-3 stream. This first frame can be identified by a parameter in the BSI field of the bit stream: the Converter Synchronization flag (convsync, in Section E.1.3.1.34 of [ETSI]) is set to true (1).

3. RTP E-AC-3 Header Fields

The RTP header is defined in the RTP specification [RFC3550]. This section defines how a number of fields in the header are used.

- Payload Type (PT): The assignment of an RTP payload type for this packet format is outside the scope of this document; it is specified by the RTP profile under which this payload format is used, or signaled dynamically out-of-band (e.g., using SDP).

- o Marker (M) bit: The M bit is set to one to indicate that the RTP packet payload contains at least one complete E-AC-3 frame or contains the final fragment of an E-AC-3 frame.

- o Extension (X) bit: Defined by the RTP profile used.

- o Timestamp: A 32-bit word that corresponds to the sampling instant for the first E-AC-3 frame in the RTP packet. Packets containing fragments of the same frame MUST have the same timestamp. The timestamp of the first RTP packet sent SHOULD be selected at random; thereafter, it increases linearly according to the number of samples included in each frame. Note that the number of samples in a frame depends on the number of blocks in the frame, with 256 samples in each block. Also note that more than one frame might correspond to the same time period when multiple channel configurations or programs are present. If these frames occupy multiple packets, it is possible that the resulting packets will have the same timestamp value.

4. RTP E-AC-3 Payload Format

This payload format is defined for E-AC-3, as defined in Annex E of [ETSI]. Note that E-AC-3 decoders are required to be capable of decoding AC-3 bit streams, so a receiver capable of receiving the E-AC-3 payload format defined in this document MUST also receive the payload format for AC-3 defined in [RFC4184].

According to [RFC2736], RTP payload formats should contain an integral number of application data units (ADUs). The E-AC-3 frame corresponds to an ADU in the context of this payload format. Each RTP payload MUST start with the two-byte payload specific header followed by an integral number of complete E-AC-3 frames, or a single fragment of an E-AC-3 frame.

If an E-AC-3 frame exceeds the MTU for a network, it SHOULD be fragmented for transmission within an RTP packet. Section 4.2 provides guidelines for creating frame fragments.

4.1. Payload Specific Header

There is a two-octet Payload header at the beginning of each payload. Each E-AC-3 RTP payload MUST begin with the following Payload header.

```
 0                   1
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|    MBZ    |F|       NF        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 4. E-AC-3 RTP Payload header

- o Must Be Zero (MBZ): Bits marked MBZ SHALL be set to the value zero and SHALL be ignored by receivers. The bits are reserved for future extensions.
- o Frame Type (F): This one-bit field indicates the type of frame(s) present in the payload. It takes the following values: 0 - One or more complete frames. 1 - Fragment of frame. (Note that the M bit in the RTP header is set for the final fragment.)
- o Number of frames/fragments (NF): An 8-bit field whose meaning depends on the Frame Type (F) in this payload. For complete frames (F of 0), it is used to indicate the number of E-AC-3 frames in the RTP payload. For frame fragments (F of 1), it is used to indicate the number of fragments (and therefore packets) that make up the current frame. NF MUST be identical for packets containing fragments of the same frame.

When receiving E-AC-3 payloads with F = 0 and more than a single frame (NF > 1), a receiver needs to use the "frmsiz" field in the BSI header in each E-AC-3 frame to determine the frame’s length if the receiver needs to determine the boundary of the next frame. Note that the frame length varies from frame to frame in some circumstances.

4.2. Fragmentation of E-AC-3 Frames

The size of an E-AC-3 frame is signaled in the Frame Size (frmsiz) field in a frame’s BSI header. The value of this field is one less than the number of 16-bit words in the frame. If the size of an E-AC-3 frame exceeds the MTU size, the frame SHOULD be fragmented at the RTP level. The fragmentation MAY be performed at any byte boundary in the frame. RTP packets containing fragments of the same E-AC-3 frame SHALL be sent in consecutive order, from first to last fragment. This enables a receiver to assemble the fragments in the correct order.

4.3. Concatenation of E-AC-3 Frames

There are cases where E-AC-3 frame sizes are smaller than the MTU size and it is advantageous to include multiple frames in a packet.

It is useful to take into account the logical arrangement of the bit stream into program sets and frame sets to constrain the effects of the loss of a packet. It is desirable for a complete program set or a complete frame set to be included in one packet. Also, it is undesirable for frames from more than one program set or frame set to be in the same packet, unless the sets are complete. In this way, the loss of a packet is kept from causing the contents of another packet to be unusable.

Frames from more than one program set SHOULD NOT be included in the same packet unless all program sets in the packet are complete. Frames from more than one frame set SHOULD NOT be included in the same packet unless all frame sets in the packet are complete.

4.4. Carriage of AC-3 Frames

The E-AC-3 specification [ETSI] requires that E-AC-3 decoders be capable of decoding AC-3 frames. That specification also supports carriage of AC-3 frames in an E-AC-3 bit stream. Due to differences between E-AC-3 and AC-3 frames, there are restrictions placed on the use of AC-3 frames: they are only used for the independent substream of the first (or only) program in an E-AC-3 bit stream. Note that carriage of only E-AC-3 frames, only AC-3 frames, and a mixture of E-AC-3 and AC-3 frames are all legal configurations. It is legal to change among the configurations in a bit stream. The AC-3 frame format is described in [RFC4184] and specified in [ETSI].

5. Types and Names

5.1. Media Type Registration

This registration uses the template defined in [RFC4288] and follows [RFC3555].

To: ietf-types@iana.org
Subject: Registration of media type audio/eac3

Type name: audio

Subtype name: eac3

Required parameter:

o rate: The RTP timestamp clock rate that is equal to the audio sampling rate. Permitted rates are 32000, 44100, and 48000.

Optional parameter:

o bitStreamConfig: The configuration of programs and substreams in the bit stream, expressed as a sequence of ASCII characters. This parameter can serve two purposes. First, during the creation of a session, the bitStreamConfig parameter might be used to negotiate a match between the requirements of a bit stream and the capabilities of a receiver to avoid using network bandwidth for data that cannot be used. Second, it makes the configuration of the bit stream explicit to the receiver so that whenever a packet is lost, the receiver can identify which kind of frame(s) has been lost to aid error mitigation.

The format for the value for this parameter is to represent each substream of the bit stream by a single character indicating its type, immediately followed by the number of audio channels resulting if a frame of that substream (plus any other required substreams) is decoded. Note that even though Low-Frequency Effects (LFE) channels are often described as "fractional" channels (e.g., the ".1" in 5.1), for this parameter, an LFE channel is counted as one (e.g., a 5.1-channel configuration is indicated as 6). The configuration of the bit stream MUST match the value of this parameter for the duration of the session.

Allowed values for the substream type are as follows:

i - Independent substream.
d - Dependent substream.

The E-AC-3 specification [ETSI] defines which configurations of bit streams are legal, which constrains the values the bitStreamConfig parameter will take. Each program starts with, and contains exactly one, independent substream ('i'). Each independent substream is followed by between 0 and 8 dependent substreams ('d'), which belong to the same program. See Section 2.1.2 for more discussion of programs and substreams.

For example, consider a bit stream containing two programs:

* the first program with

  + a six-channel independent substream
  + a dependent substream containing the additional channels needed for eight channels
  + a second dependent substream containing the further channels needed for 14 channels

* along with a second program with

  + another six-channel independent substream
  + a dependent substream containing the additional channels needed for eight channels

Then the configuration of the bit stream is indicated as follows:

bitStreamConfig = i6d8d14i6d8

When the bitStreamConfig parameter is being used in an offer/answer exchange, zero (0) for the number of channels for a substream in an answer is used to indicate a substream that the answerer desires not to receive.

Encoding considerations:

This media type is framed and contains binary data.

Security considerations:

See Section 6 of RFC 4598.

Interoperability considerations:

To maintain interoperability with AC-3-capable end-points, in cases where negotiation is possible, an E-AC-3 end-point SHOULD declare itself also as AC-3 capable (i.e., supporting also "audio/ac3" as specified in RFC 4184 [RFC4184]). Note that all E-AC-3 end-points are required to be AC-3 capable.

Published specification:

RFC 4598 and ETSI TS 102.366 [ETSI].

Applications that use this media type:

Multichannel audio compression of audio, and audio for video.

Additional information:

Magic number(s): The first two octets of an E-AC-3 frame are always the synchronization word, which has the hex value 0x0B77.

Person & email address to contact for further information:

Brian Link <bdl@dolby.com> IETF AVT working group.

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.

Author/Change controller:

IETF Audio/Video Transport Working Group delegated from the IESG.

5.2. SDP Usage

The information carried in the 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 E-AC-3, the mapping is as follows:

- o The Media type ("audio") goes in SDP "m=" as the media name.

- o The Media subtype ("eac3") goes in SDP "a=rtpmap" as the encoding name.

- o The required parameter "rate" also goes in "a=rtpmap" as the clock rate. (The optional "channels" rtpmap encoding parameter is not used. Instead, the information is included in the optional parameter bitStreamConfig.)

- o The optional parameter "bitStreamConfig" goes in the SDP "a=fmtp" attribute.

The following is an example of the SDP data for E-AC-3:

```
m=audio 49111 RTP/AVP 100
a=rtpmap:100 eac3/48000
a=fmtp:100 bitStreamConfig i6d8d14i6d8
```

Certain considerations are needed when SDP is used to perform offer/answer exchanges [RFC3264].

- o The "rate" is a symmetric parameter, and the answer MUST use the same value or the answerer removes the payload type.

- o The "bitStreamConfig" parameter is declarative and indicates, for sendonly, the intended arrangement of substreams in the bit stream, along with the channel configuration, to transmit, and for recvonly or sendrecv, the desired bit stream arrangement and channel configuration to receive. The format of the bitStreamConfig value in an answer MAY differ from the offer value by replacing the number of channels for any undesired substreams with '0'. It is valid to zero out dependent substreams containing undesired channel configurations and to zero out all the substreams of an undesired program. Then the sender MAY reoffer the stream in the receiver's preferred configuration if it is capable of providing that configuration. Note that all receivers are capable of receiving, and all decoders are capable of decoding, any of the legal bit stream configurations, so the parameter exchange is not needed for interoperability. The parameter exchange might be used to help optimize the transmission to the number of programs or channels the receiver requests.

- o Since an AC-3 bit stream is a special case of an E-AC-3 bit stream, it is permissible for an AC-3 bit stream to be carried in the E-AC-3 payload format. To ensure interoperability with receivers that support the AC-3 payload format but not the E-AC-3 payload format, a sender that desires to send an AC-3 bit stream in the E-AC-3 payload format SHOULD also offer the session in the AC-3 payload format by including payload types for both media subtypes: 'ac3' and 'eac3'.

6. Security Considerations

The payload format described in this document is subject to the security considerations defined in RTP [RFC3550] and in any applicable RTP profile (e.g., [RFC3551]). To protect the user's privacy and any copyrighted material, confidentiality protection would have to be applied. To also protect against modification by intermediate entities and ensure the authenticity of the stream, integrity protection and authentication would be required. Confidentiality, integrity protection, and authentication have to be solved by a mechanism external to this payload format, for example, Secure Real-time Transport Protocol (SRTP) [RFC3711].

The E-AC-3 format is designed so that the validity of data frames can be determined by decoders. The required decoder response to a malformed frame is to discard the malformed data and conceal the errors in the audio output until a valid frame is detected and decoded. This is expected to prevent crashes and other abnormal decoder behavior in response to errors or attacks.

7. Congestion Control

The general congestion control considerations for transporting RTP data apply to E-AC-3 audio over RTP as well; see RTP [RFC3550], and any applicable RTP profile (e.g., [RFC3551]).

E-AC-3 is a variable bit rate coding system so it is possible to use a variety of techniques to adapt to network bandwidth.

8. IANA Considerations

The IANA has registered a new media subtype for E-AC-3 (see Section 5).

9. References

9.1. Normative References

[ETSI] ETSI, "Digital Audio Compression (AC-3, Enhanced AC-3) Standard", TS 102 366, February 2005.

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

[RFC4184] Link, B., Hager, T., and J. Flaks, "RTP Payload Format for AC-3 Audio", RFC 4184, October 2005.

[RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.

[RFC4288] Freed, N. and J. Klensin, "Media Type Specifications and Registration Procedures", BCP 13, RFC 4288, December 2005.

[RFC3555] Casner, S. and P. Hoschka, "MIME Type Registration of RTP Payload Formats", RFC 3555, July 2003.

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

[RFC3264] Rosenberg, J. and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)", RFC 3264, June 2002.

9.2. Informative References

[2004AES] Fielder, L., Andersen, R., Crockett, B., Davidson, G., Davis, M., Turner, S., Vinton, M., and P. Williams, "Introduction to Dolby Digital Plus, an Enhancement to the Dolby Digital Coding System", Preprint 6196, Presented at the 117th Convention of the Audio Engineering Society, October 2004.

[1994AES] Todd, C., Davidson, G., Davis, M., Fielder, L., Link, B., and S. Vernon, "AC-3: Flexible Perceptual Coding for Audio Transmission and Storage", Preprint 3796, Presented at the 96th Convention of the Audio Engineering Society, May 1994.

[RFC2736] Handley, M. and C. Perkins, "Guidelines for Writers of RTP Payload Format Specifications", BCP 36, RFC 2736, December 1999.

[RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, RFC 3551, July 2003.

[RFC3711] Baugher, M., McGrew, D., Naslund, M., Carrara, E., and K. Norrman, "The Secure Real-time Transport Protocol (SRTP)", RFC 3711, March 2004.

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Full Copyright Statement

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