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Internet-Draft	Dolby Laboratories	
Obsoletes: 3016 (if approved)	F. de Bont	
Intended status: Standards Track	Philips Electronics	
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	January 11, 2011	

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RTP Payload Format for MPEG-4 Audio/Visual Streams
draft-ietf-avt-rfc3016bis-02.txt

Abstract

This document describes Real-Time Transport Protocol (RTP) payload formats for carrying each of MPEG-4 Audio and MPEG-4 Visual bitstreams without using MPEG-4 Systems. For the purpose of directly mapping MPEG-4 Audio/Visual bitstreams onto RTP packets, it provides specifications for the use of RTP header fields and also specifies fragmentation rules. It also provides specifications for Media Type registration and the use of Session Description Protocol (SDP). The audio payload format described in this document has some limitations. for new system designs [RFC3640] is preferred.

Status of this Memo

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1. Introduction

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The RTP payload formats described in this document specify how MPEG-4 Audio [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.) and MPEG-4 Visual streams [14496-2] (MPEG, “ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual,” 2003.) are to be fragmented and mapped directly onto RTP packets.
These RTP payload formats enable transport of MPEG-4 Audio/Visual streams without using the synchronization and stream management functionality of MPEG-4 Systems [14496-1] (MPEG, “ISO/IEC International Standard 14496-1 - Coding of audio-visual objects, Part 1 Systems,” 2004.). Such RTP payload formats will be used in systems that have intrinsic stream management functionality and thus require no such functionality from MPEG-4 Systems. H.323 terminals are an example of such systems, where MPEG-4 Audio/Visual streams are not managed by MPEG-4 Systems Object Descriptors but by H.245. The streams are directly mapped onto RTP packets without using the MPEG-4 Systems Sync Layer. Other examples are SIP and RTSP where Media Type and SDP are used. Media Type and SDP usages of the RTP payload formats described in this document are defined to directly specify the attribute of Audio/Visual streams (e.g., media type, packetization format and codec configuration) without using MPEG-4 Systems. The obvious benefit is that these MPEG-4 Audio/Visual RTP payload formats can be handled in an unified way together with those formats defined for non-MPEG-4 codecs. The disadvantage is that interoperability with environments using MPEG-4 Systems may be difficult, hence, other payload formats may be better suited to those applications.
The semantics of RTP headers in such cases need to be clearly defined, including the association with MPEG-4 Audio/Visual data elements. In addition, it is beneficial to define the fragmentation rules of RTP packets for MPEG-4 Video streams so as to enhance error resiliency by utilizing the error resiliency tools provided inside the MPEG-4 Video stream.

1.1. MPEG-4 Visual RTP Payload Format

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MPEG-4 Visual is a visual coding standard with many new features: high coding efficiency; high error resiliency; multiple, arbitrary shape object-based coding; etc. [14496-2] (MPEG, “ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual,” 2003.). It covers a wide range of bitrate from scores of Kbps to several Mbps. It also covers a wide variety of networks, ranging from those guaranteed to be almost error-free to mobile networks with high error rates.

With respect to the fragmentation rules for an MPEG-4 Visual bitstream defined in this document, since MPEG-4 Visual is used for a wide variety of networks, it is desirable not to apply too much restriction on fragmentation, and a fragmentation rule such as "a single video packet shall always be mapped on a single RTP packet" may be inappropriate. On the other hand, careless, media unaware fragmentation may cause degradation in error resiliency and bandwidth efficiency. The fragmentation rules described in this document are flexible but manage to define the minimum rules for preventing meaningless fragmentation while utilizing the error resiliency functionalities of MPEG-4 Visual. The fragmentation rule "Different VOPs SHOULD be fragmented into different RTP packets" is made so that the RTP timestamp uniquely indicates the VOP time framing. On the other hand, MPEG-4 video may generate VOPs of very small size, in cases with an empty VOP (vop_coded=0) containing only VOP header or an arbitrary shaped VOP with a small number of coding blocks. To reduce the overhead for such cases, the fragmentation rule permits concatenating multiple VOPs in an RTP packet. (See fragmentation rule (4) in Section 4.2 (Fragmentation of MPEG-4 Visual Bitstream) and marker bit and timestamp in Section 4.1 (Use of RTP Header Fields for MPEG-4 Visual).)

While the additional media specific RTP header defined for such video coding tools as H.261 or MPEG-1/2 is effective in helping to recover picture headers corrupted by packet losses, MPEG-4 Visual has already error resiliency functionalities for recovering corrupt headers, and these can be used on RTP/IP networks as well as on other networks (H.223/mobile, MPEG-2/TS, etc.). Therefore, no extra RTP header fields are defined in this MPEG-4 Visual RTP payload format.

1.2. MPEG-4 Audio RTP Payload Format

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MPEG-4 Audio is an audio standard that integrates many different types of audio coding tools. Low-overhead MPEG-4 Audio Transport Multiplex (LATM) manages the sequences of audio data with relatively small overhead. In audio-only applications, then, it is desirable for LATM-

based MPEG-4 Audio bitstreams to be directly mapped onto RTP packets without using MPEG-4 Systems.
For MPEG-4 Audio coding tools, as is true for other audio coders, if the payload is a single audio frame, packet loss will not impair the decodability of adjacent packets. Therefore, the additional media specific header for recovering errors will not be required for MPEG-4 Audio. Existing RTP protection mechanisms, such as Generic Forward Error Correction [RFC5109] (Li, A., “RTP Payload Format for Generic Forward Error Correction,” December 2007.) and Redundant Audio Data [RFC2198] (Perkins, C., Kouvelas, I., Hodson, O., Hardman, V., Handley, M., Bolot, J., Vega-Garcia, A., and S. Fosse-Parisis, “RTP Payload for Redundant Audio Data,” September 1997.), MAY be applied to improve error resiliency.

1.3. Interoperability with RFC 3016

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Although strictly speaking systems that support MPEG-4 Audio as specified in [RFC3016] (Kikuchi, Y., Nomura, T., Fukunaga, S., Matsui, Y., and H. Kimata, “RTP Payload Format for MPEG-4 Audio/Visual Streams,” November 2000.) will be incompatible with systems supporting this document, existing systems already comply with the specification in 3GPP PSS service (3GPP, “3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs (Release 9),” December 2010.) [3GPP] and therefore no incompatibility issues are foreseen.

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2. Definitions and Abbreviations

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This document makes use of terms, specified in [14496-2] (MPEG, “ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual,” 2003.), [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.), and [23003-1] (MPEG, “ISO/IEC International Standard 23003-1 - MPEG Surround (MPEG D),” 2007.). In addition, the following terms are used in this document and have specific meaning within the context of this document.
Core codec sampling rate:

> Audio codec sampling rate. When SBR (Spectral Band Replication) is used, typically the double value of this will be regarded as the definitive sampling rate (i.e., the decoder's output sampling rate)

> Note: The exception is downsampled SBR mode in which the SBR sampling rate equals the core codec sampling rate.

Core codec channel configuration:

> Audio codec channel configuration. When PS (Parametric Stereo) is used, the core codec channel configuration indicates one channel (i.e., mono) whereas the definitive channel configuration is two channels (i.e. stereo). When MPEG Surround is used, the definitive channel configuration depends on the output of the MPEG Surround decoder.

SBR sampling rate:

> When SBR is used, typically the sampling rate is the double value of the core codec sampling rate, with the exception of downsampled SBR mode, where the SBR sampling rate and core codec sampling rate are identical.

Abbreviations:

> AAC: Advanced Audio Coding
>
> ASC: AudioSpecificConfig
>
> HE AAC: High Efficiency AAC
>
> LATM: Low-overhead MPEG-4 Audio Transport Multiplex
>
> PS: Parametric Stereo
>
> SBR: Spectral Band Replication
>
> VOP: Video Object Plane

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] (Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels," March 1997.).

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3. LATM Restrictions for RTP Packetization of MPEG-4 Audio Bitstreams

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While LATM has several multiplexing features as follows;

*Carrying configuration information with audio data,

*Concatenation of multiple audio frames in one audio stream,

*Multiplexing multiple objects (programs),

*Multiplexing scalable layers,

in RTP transmission there is no need for the last two features. Therefore, these two features MUST NOT be used in applications based on RTP packetization specified by this document. Since LATM has been developed for only natural audio coding tools, i.e., not for synthesis tools, it seems difficult to transmit Structured Audio (SA) data and Text to Speech Interface (TTSI) data by LATM. Therefore, SA data and TTSI data MUST NOT be transported by the RTP packetization in this document.
For transmission of scalable streams, audio data of each layer SHOULD be packetized onto different RTP streams allowing for the different layers to be treated differently at the IP level, for example via some means of differentiated service. On the other hand, all configuration data of the scalable streams are contained in one LATM configuration data "StreamMuxConfig" and every scalable layer shares the StreamMuxConfig. The mapping between each layer and its configuration data is achieved by LATM header information attached to the audio data. In order to indicate the dependency information of the scalable streams, the signaling mechanism as specified in [RFC5583] (Schierl, T. and S. Wenger, “Signaling Media Decoding Dependency in the Session Description Protocol (SDP),” July 2009.) SHOULD be used (see Section 5.2 (Use of RTP Header Fields for MPEG-4 Audio)).

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4. RTP Packetization of MPEG-4 Visual Bitstreams

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This section specifies RTP packetization rules for MPEG-4 Visual content. An MPEG-4 Visual bitstream is mapped directly onto RTP packets without the addition of extra header fields or any removal of Visual syntax elements. The Combined Configuration/Elementary stream mode MUST be used so that configuration information will be carried to the same RTP port as the elementary stream. (see 6.2.1 "Start codes" of [14496-2] (MPEG, “ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual,” 2003.)) The configuration information MAY additionally be specified by some out-of-band means. If needed by systems using Media Type parameters and SDP parameters, "e.g., SIP and RTSP", the optional parameter "config" MUST be used to specify the configuration information (see Section 6.1 (Media Type Registration for MPEG-4 Visual) and Section 6.2 (Mapping to SDP for MPEG-4 Visual)).
When the short video header mode is used, the RTP payload format for H. 263 SHOULD be used (the format defined in [RFC4629] (Ott, H., Bormann, C., Sullivan, G., Wenger, S., and R. Even, “RTP Payload Format for ITU-

T Rec," January 2007.) is RECOMMENDED, but the [RFC4628] (Even, R., "RTP Payload Format for H.263 Moving RFC 2190 to Historic Status," January 2007.) format MAY be used for compatibility with older implementations).

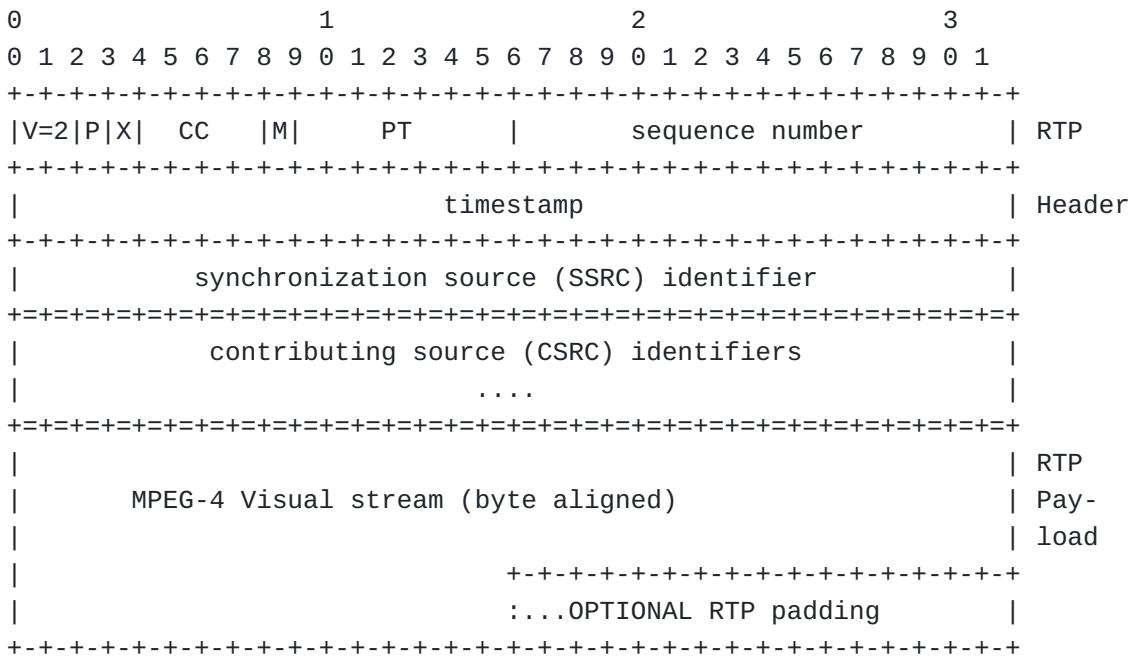

Figure 1 - An RTP packet for MPEG-4 Visual stream

4.1. Use of RTP Header Fields for MPEG-4 Visual

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Payload Type (PT): The assignment of an RTP payload type for this packet format is outside the scope of this document, and will not be specified here. It is expected that the RTP profile for a particular class of applications will assign a payload type for this encoding, or if that is not done then a payload type in the dynamic range SHALL be chosen by means of an out-of-band signaling protocol (e.g., H.245, SIP, etc).
Extension (X) bit: Defined by the RTP profile used.
Sequence Number: Incremented by one for each RTP data packet sent, starting, for security reasons, with a random initial value.
Marker (M) bit: The marker bit is set to one to indicate the last RTP packet (or only RTP packet) of a VOP. When multiple VOPs are carried in the same RTP packet, the marker bit is set to one.

Timestamp: The timestamp indicates the sampling instance of the VOP contained in the RTP packet. A constant offset, which is random, is added for security reasons.

- When multiple VOPs are carried in the same RTP packet, the timestamp indicates the earliest of the VOP times within the VOPs carried in the RTP packet. Timestamp information of the rest of the VOPs are derived from the timestamp fields in the VOP header (modulo_time_base and vop_time_increment).

- If the RTP packet contains only configuration information and/or Group_of_VideoObjectPlane() fields, the timestamp of the next VOP in the coding order is used.

- If the RTP packet contains only visual_object_sequence_end_code information, the timestamp of the immediately preceding VOP in the coding order is used.

The resolution of the timestamp is set to its default value of 90kHz, unless specified by an out-of-band means (e.g., SDP parameter or Media Type parameter as defined in Section 6 (Media Type Registration for MPEG-4 Audio/Visual Streams)).
Other header fields are used as described in [RFC3550] (Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," July 2003.).

4.2. Fragmentation of MPEG-4 Visual Bitstream

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A fragmented MPEG-4 Visual bitstream is mapped directly onto the RTP payload without any addition of extra header fields or any removal of Visual syntax elements. The Combined Configuration/Elementary streams mode is used. The following rules apply for the fragmentation.
In the following, header means one of the following:

- Configuration information (Visual Object Sequence Header, Visual Object Header and Video Object Layer Header)

- visual_object_sequence_end_code

- The header of the entry point function for an elementary stream (Group_of_VideoObjectPlane() or the header of VideoObjectPlane(), video_plane_with_short_header(), MeshObject() or FaceObject())

- The video packet header (video_packet_header() excluding next_resync_marker())

- The header of gob_layer()

*See 6.2.1 "Start codes" of [14496-2] (MPEG, “ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual,” 2003.) for the definition of the configuration information and the entry point functions.

(1) Configuration information and Group_of_VideoObjectPlane() fields SHALL be placed at the beginning of the RTP payload (just after the RTP header) or just after the header of the syntactically upper layer function.
(2) If one or more headers exist in the RTP payload, the RTP payload SHALL begin with the header of the syntactically highest function.
Note: The visual_object_sequence_end_code is regarded as the lowest function.
(3) A header SHALL NOT be split into a plurality of RTP packets.
(4) Different VOPs SHOULD be fragmented into different RTP packets so that one RTP packet consists of the data bytes associated with a unique VOP time instance (that is indicated in the timestamp field in the RTP packet header), with the exception that multiple consecutive VOPs MAY be carried within one RTP packet in the decoding order if the size of the VOPs is small.
Note: When multiple VOPs are carried in one RTP payload, the timestamp of the VOPs after the first one may be calculated by the decoder. This operation is necessary only for RTP packets in which the marker bit equals to one and the beginning of RTP payload corresponds to a start code. (See timestamp and marker bit in Section 4.1 (Use of RTP Header Fields for MPEG-4 Visual).)
(5) It is RECOMMENDED that a single video packet is sent as a single RTP packet. The size of a video packet SHOULD be adjusted in such a way that the resulting RTP packet is not larger than the path-MTU. If the video packet is disabled by the coder configuration (by setting resync_marker_disable in the VOL header to 1), or in coding tools where the video packet is not supported, a VOP MAY be split at arbitrary byte-positions.
The video packet starts with the VOP header or the video packet header, followed by motion_shape_texture(), and ends with next_resync_marker() or next_start_code().

4.3. Examples of Packetized MPEG-4 Visual Bitstream

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Figure 2 shows examples of RTP packets generated based on the criteria described in Section 4.2 (Fragmentation of MPEG-4 Visual Bitstream)
(a) is an example of the first RTP packet or the random access point of an MPEG-4 Visual bitstream containing the configuration information. According to criterion (1), the Visual Object Sequence Header(VS header) is placed at the beginning of the RTP payload, preceding the Visual Object Header and the Video Object Layer Header(VO header, VOL

header). Since the fragmentation rule defined in Section 4.2 (Fragmentation of MPEG-4 Visual Bitstream) guarantees that the configuration information, starting with visual_object_sequence_start_code, is always placed at the beginning of the RTP payload, RTP receivers can detect the random access point by checking if the first 32-bit field of the RTP payload is visual_object_sequence_start_code.

(b) is another example of the RTP packet containing the configuration information. It differs from example (a) in that the RTP packet also contains a VOP header and a Video Packet in the VOP following the configuration information. Since the length of the configuration information is relatively short (typically scores of bytes) and an RTP packet containing only the configuration information may thus increase the overhead, the configuration information and the immediately following VOP can be packetized into a single RTP packet.

(c) is an example of an RTP packet that contains Group_of_VideoObjectPlane(GOV). Following criterion (1), the GOV is placed at the beginning of the RTP payload. It would be a waste of RTP/IP header overhead to generate an RTP packet containing only a GOV whose length is 7 bytes. Therefore, (a part of) the following VOP can be placed in the same RTP packet as shown in (c).

(d) is an example of the case where one video packet is packetized into one RTP packet. When the packet-loss rate of the underlying network is high, this kind of packetization is recommended. Even when the RTP packet containing the VOP header is discarded by a packet loss, the other RTP packets can be decoded by using the HEC(Header Extension Code) information in the video packet header. No extra RTP header field is necessary.

(e) is an example of the case where more than one video packet is packetized into one RTP packet. This kind of packetization is effective to save the overhead of RTP/IP headers when the bit-rate of the underlying network is low. However, it will decrease the packet-loss resiliency because multiple video packets are discarded by a single RTP packet loss. The optimal number of video packets in an RTP packet and the length of the RTP packet can be determined considering the packet-loss rate and the bit-rate of the underlying network.

(f) is an example of the case when the video packet is disabled by setting resync_marker_disable in the VOL header to 1. In this case, a VOP may be split into a plurality of RTP packets at arbitrary byte-positions. For example, it is possible to split a VOP into fixed-length packets. This kind of coder configuration and RTP packet fragmentation may be used when the underlying network is guaranteed to be error-free.

Figure 3 shows examples of RTP packets prohibited by the criteria of Section 4.2 (Fragmentation of MPEG-4 Visual Bitstream).

Fragmentation of a header into multiple RTP packets, as in (a), will not only increase the overhead of RTP/IP headers but also decrease the error resiliency. Therefore, it is prohibited by the criterion (3).

When concatenating more than one video packets into an RTP packet, VOP header or video_packet_header() are not allowed to be placed in the

middle of the RTP payload. The packetization as in (b) is not allowed by criterion (2) due to the aspect of the error resiliency. Comparing this example with Figure 2(d), although two video packets are mapped onto two RTP packets in both cases, the packet-loss resiliency is not identical. Namely, if the second RTP packet is lost, both video packets 1 and 2 are lost in the case of Figure 3(b) whereas only video packet 2 is lost in the case of Figure 2(d).

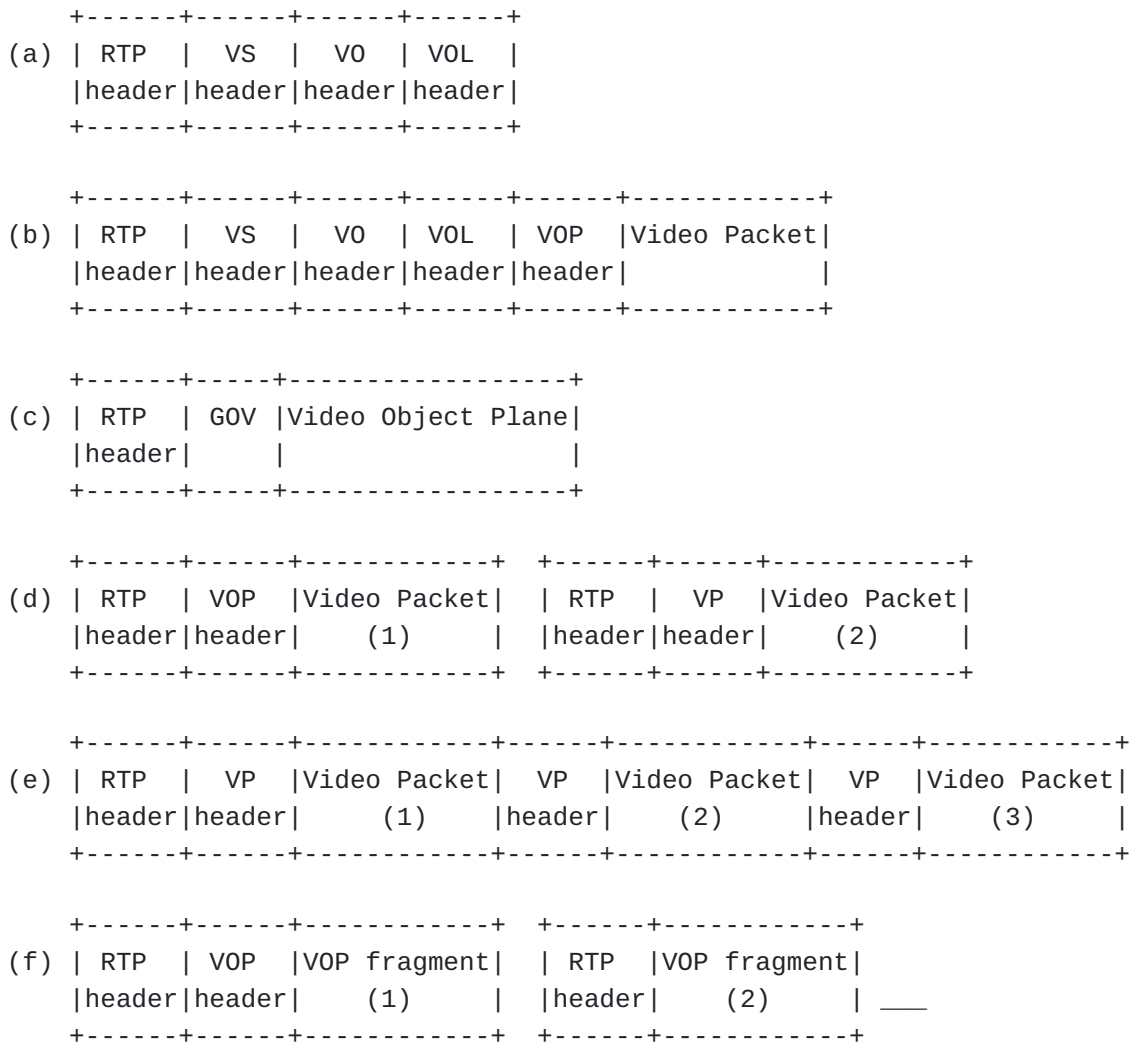

Figure 2 - Examples of RTP packetized MPEG-4 Visual bitstream

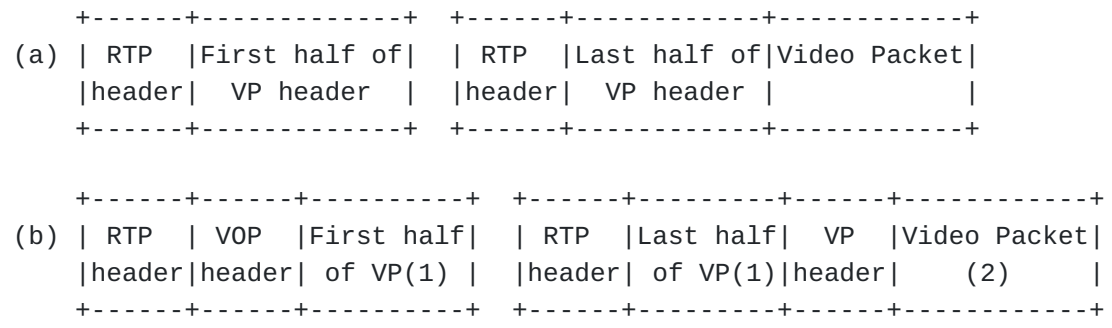

Figure 3 - Examples of prohibited RTP packetization for MPEG-4 Visual bitstream

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5. RTP Packetization of MPEG-4 Audio Bitstreams

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This section specifies RTP packetization rules for MPEG-4 Audio bitstreams. MPEG-4 Audio streams MUST be formatted LATM (Low-overhead MPEG-4 Audio Transport Multiplex) [14496-3] (MPEG, "ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio," 2009.) streams, and the LATM-based streams are then mapped onto RTP packets as described in the sections below.

5.1. RTP Packet Format

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LATM-based streams consist of a sequence of audioMuxElements that include one or more PayloadMux elements which carry the audio frames. A complete audioMuxElement or a part of one SHALL be mapped directly onto an RTP payload without any removal of audioMuxElement syntax elements (see Figure 4). The first byte of each audioMuxElement SHALL be located at the first payload location in an RTP packet.

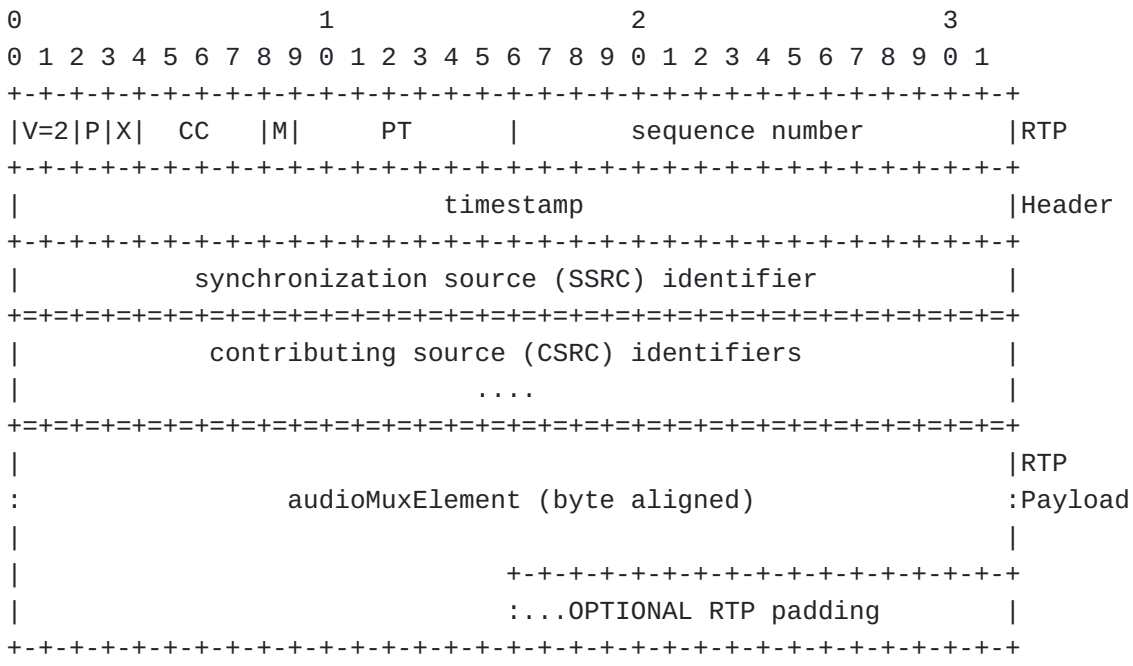

Figure 4 - An RTP packet for MPEG-4 Audio

In order to decode the audioMuxElement, the following muxConfigPresent information is required to be indicated by out-of-band means. When SDP is utilized for this indication, the Media Type parameter "cpresent" corresponds to the muxConfigPresent information (see Section 6.3 (Media Type Registration for MPEG-4 Audio)). The following restrictions apply:

*In the out-of-band configuration case the number of PayloadMux elements contained in each audioMuxElement can only be set once. If more than one PayloadMux elements are contained in each AudioMuxElement, special care is required to ensure that the last RTP packet remains decodable.

*To construct the audioMuxElement in the in-band configuration case, non octet aligned configuration data is preceding the one or more PayloadMux elements. Since the generation of RTP payloads with non octet aligned data is not possible with RTP hint tracks, as defined by the MP4 file format [14496-12] (MPEG, “ISO/IEC International Standard 14496-12 - Coding of audio-visual objects, Part 12 ISO base media file format,” .) [14496-14] (MPEG, “ISO/IEC International Standard 14496-14 - Coding of audio-visual objects, Part 12 MP4 file format,” .), this document does not support RTP hint tracks for the in-band configuration case.

muxConfigPresent: If this value is set to 1 (in-band mode), the audioMuxElement SHALL include an indication bit "useSameStreamMux" and MAY include the configuration information for audio compression

"StreamMuxConfig". The useSameStreamMux bit indicates whether the StreamMuxConfig element in the previous frame is applied in the current frame. If the useSameStreamMux bit indicates to use the StreamMuxConfig from the previous frame, but if the previous frame has been lost, the current frame may not be decodable. Therefore, in case of in-band mode, the StreamMuxConfig element SHOULD be transmitted repeatedly depending on the network condition. On the other hand, if muxConfigPresent is set to 0 (out-band mode), the StreamMuxConfig element is required to be transmitted by an out-of-band means. In case of SDP, Media Type parameter "config" is utilized (see Section 6.3 (Media Type Registration for MPEG-4 Audio)).

5.2. Use of RTP Header Fields for MPEG-4 Audio

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Payload Type (PT): The assignment of an RTP payload type for this new packet format is outside the scope of this document, and will only be restricted here. It is expected that the RTP profile for a particular class of applications will assign a payload type for this encoding, or if that is not done then a payload type in the dynamic range shall be chosen by means of an out-of-band signaling protocol (e.g., H.245, SIP, etc). In the dynamic assignment of RTP payload types for scalable streams, the server SHALL assign a different value to each layer. The dependency relationships between the enhance layer and the base layer MUST be signaled as specified in [RFC5583] (Schierl, T. and S. Wenger, "Signaling Media Decoding Dependency in the Session Description Protocol (SDP)," July 2009.). An example of the use of such signaling for scalable audio streams can be found in [RFC5691] (de Bont, F., Doehla, S., Schmidt, M., and R. Sperschneider, "RTP Payload Format for Elementary Streams with MPEG Surround Multi-Channel Audio," October 2009.).

Marker (M) bit: The marker bit indicates audioMuxElement boundaries. It is set to one to indicate that the RTP packet contains a complete audioMuxElement or the last fragment of an audioMuxElement.

Timestamp: The timestamp indicates the sampling instance of the first audio frame contained in the RTP packet. Timestamps are RECOMMENDED to start at a random value for security reasons.

Unless specified by an out-of-band means, the resolution of the timestamp is set to its default value of 90 kHz.

Sequence Number: Incremented by one for each RTP packet sent, starting, for security reasons, with a random value.

Other header fields are used as described in [RFC3550] (Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," July 2003.).

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5.3. Fragmentation of MPEG-4 Audio Bitstream

It is RECOMMENDED to put one audioMuxElement in each RTP packet. If the size of an audioMuxElement can be kept small enough that the size of the RTP packet containing it does not exceed the size of the path-MTU, this will be no problem. If it cannot, the audioMuxElement SHALL be fragmented and spread across multiple packets.

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6. Media Type Registration for MPEG-4 Audio/Visual Streams

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The following sections describe the Media Type registrations for MPEG-4 Audio/Visual streams, which are registered in accordance with [RFC4855] (Casner, S., "Media Type Registration of RTP Payload Formats," February 2007.) and uses the template of [RFC4288] (Freed, N. and J. Klensin, "Media Type Specifications and Registration Procedures," December 2005.). Media Type registration and SDP usage for the MPEG-4 Visual stream are described in Section 6.1 (Media Type Registration for MPEG-4 Visual) and Section 6.2 (Mapping to SDP for MPEG-4 Visual), respectively, while Media Type registration and SDP usage for MPEG-4 Audio stream are described in Section 6.3 (Media Type Registration for MPEG-4 Audio) and Section 6.4 (Mapping to SDP for MPEG-4 Audio), respectively.

6.1. Media Type Registration for MPEG-4 Visual

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The receiver MUST ignore any unspecified parameter, to ensure that additional parameters can be added in any future revision of this specification.
Type name: video
Subtype name: MP4V-ES
Required parameters: none
Optional parameters:

> rate: This parameter is used only for RTP transport. It indicates the resolution of the timestamp field in the RTP header. If this parameter is not specified, its default value of 90000 (90kHz) is used.
>
> profile-level-id: A decimal representation of MPEG-4 Visual Profile and Level indication value (profile_and_level_indication) defined in Table G-1 of [14496-2] (MPEG, "ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual," 2003.). This parameter MAY be used in the capability exchange or session setup procedure to indicate MPEG-4 Visual Profile and Level

> combination of which the MPEG-4 Visual codec is capable. If this parameter is not specified by the procedure, its default value of 1 (Simple Profile/Level 1) is used.
>
> config: This parameter SHALL be used to indicate the configuration of the corresponding MPEG-4 Visual bitstream. It SHALL NOT be used to indicate the codec capability in the capability exchange procedure. It is a hexadecimal representation of an octet string that expresses the MPEG-4 Visual configuration information, as defined in subclause 6.2.1 Start codes of [14496-2] (MPEG, "ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual," 2003.). The configuration information is mapped onto the octet string in an MSB-first basis. The first bit of the configuration information SHALL be located at the MSB of the first octet. The configuration information indicated by this parameter SHALL be the same as the configuration information in the corresponding MPEG-4 Visual stream, except for first_half_vbv_occupancy and latter_half_vbv_occupancy, if exist, which may vary in the repeated configuration information inside an MPEG-4 Visual stream (See 6.2.1 Start codes of [14496-2] (MPEG, "ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual," 2003.)).

Published specification:

> The specifications for MPEG-4 Visual streams are presented in [14496-2] (MPEG, "ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual," 2003.). The RTP payload format is described in this document.

Encoding considerations:

> Video bitstreams MUST be generated according to MPEG-4 Visual specifications [14496-2] (MPEG, "ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual," 2003.). A video bitstream is binary data and MUST be encoded for non-binary transport (for Email, the Base64 encoding is sufficient). This type is also defined for transfer via RTP. The RTP packets MUST be packetized according to the MPEG-4 Visual RTP payload format defined in this document.

Security considerations:

> See Section 9 (Security Considerations) of this document.

Interoperability considerations:

> MPEG-4 Visual provides a large and rich set of tools for the coding of visual objects. For effective implementation of the standard, subsets of the MPEG-4 Visual tool sets have been provided for use in

specific applications. These subsets, called 'Profiles', limit the size of the tool set a decoder is required to implement. In order to restrict computational complexity, one or more Levels are set for each Profile. A Profile@Level combination allows:

*a codec builder to implement only the subset of the standard he needs, while maintaining interworking with other MPEG-4 devices included in the same combination, and

*checking whether MPEG-4 devices comply with the standard ('conformance testing').

The visual stream SHALL be compliant with the MPEG-4 Visual Profile@Level specified by the parameter "profile-level-id". Interoperability between a sender and a receiver may be achieved by specifying the parameter "profile-level-id", or by arranging a capability exchange/announcement procedure for this parameter.

Applications which use this Media Type:

Audio and visual streaming and conferencing tools

Additional information: none
Person and email address to contact for further information:

See Authors' Address section at the end of this document.

Intended usage: COMMON
Author:

See Authors' Address section at the end of this document.

Change controller:

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

6.2. Mapping to SDP for MPEG-4 Visual

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The Media Type video/MP4V-ES string is mapped to fields in the Session Description Protocol (SDP) [RFC4566] (Handley, M., Jacobson, V., and C. Perkins, “SDP: Session Description Protocol,” July 2006.), as follows:

*The Media Type (video) goes in SDP "m=" as the media name.

*The Media subtype (MP4V-ES) goes in SDP "a=rtpmap" as the encoding name.

*The optional parameter "rate" goes in "a=rtpmap" as the clock rate.

*The optional parameter "profile-level-id" and "config" go in the "a=fmtp" line to indicate the coder capability and configuration, respectively. These parameters are expressed as a string, in the form of as a semicolon separated list of parameter=value pairs.

Example usages for the profile-level-id parameter are:
1 : MPEG-4 Visual Simple Profile/Level 1
34 : MPEG-4 Visual Core Profile/Level 2
145: MPEG-4 Visual Advanced Real Time Simple Profile/Level 1

6.2.1. Declarative SDP Usage for MPEG-4 Visual

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The following are some examples of media representation in SDP:

Simple Profile/Level 1, rate=90000(90kHz), "profile-level-id" and "config" are present in "a=fmtp" line:

```
m=video 49170/2 RTP/AVP 98
a=rtpmap:98 MP4V-ES/90000
a=fmtp:98 profile-level-id=1;config=000001B001000001B509000001000000012
   0008440FA282C2090A21F
```

Core Profile/Level 2, rate=90000(90kHz), "profile-level-id" is present in "a=fmtp" line:

```
m=video 49170/2 RTP/AVP 98
a=rtpmap:98 MP4V-ES/90000
a=fmtp:98 profile-level-id=34
```

Advance Real Time Simple Profile/Level 1, rate=90000(90kHz), "profile-level-id" is present in "a=fmtp" line:

```
m=video 49170/2 RTP/AVP 98
a=rtpmap:98 MP4V-ES/90000
a=fmtp:98 profile-level-id=145
```

6.3. Media Type Registration for MPEG-4 Audio

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The receiver MUST ignore any unspecified parameter, to ensure that additional parameters can be added in any future revision of this specification.

Type name: audio
Subtype name: MP4A-LATM
Required parameters:

> rate: the rate parameter indicates the RTP time stamp clock rate. The default value is 90000. Other rates MAY be indicated only if they are set to the same value as the audio sampling rate (number of samples per second).
>
> In the presence of SBR, the sampling rates for the core en-/decoder and the SBR tool are different in most cases. This parameter SHALL therefore NOT be considered as the definitive sampling rate. If this parameter is used, the server must following the rules below:
>
> *When the presence of SBR is not explicitly signaled by the optional SDP parameters such as object parameter, profile-level-id or config string, this parameter SHALL be set to the core codec sampling rate.
>
> *When the presence of SBR is explicitly signaled by the optional SDP parameters such as object parameter, profile-level-id or config string this parameter SHALL be set to the SBR sampling rate.
>
> NOTE: The optional parameter SBR-enabled in SDP a=fmtp is useful for implicit HE AAC / HE AAC v2 signaling. But the SBR-enabled parameter can also be used in the case of explicit HE AAC / HE AAC v2 signaling. Therefore, its existence itself is not the criteria to determine whether HE AAC / HE AAC v2 signaling is explicit or not.

Optional parameters:

> profile-level-id: a decimal representation of MPEG-4 Audio Profile Level indication value defined in [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.). This parameter indicates which MPEG-4 Audio tool subsets the decoder is capable of using. If this parameter is not specified in the capability exchange or session setup procedure, its default value of 30 (Natural Audio Profile/Level 1) is used.
>
> MPS-profile-level-id: a decimal representation of the MPEG Surround Profile Level indication as defined in [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.). This parameter indicates the support of the MPEG Surround profile and level by the decoder to be capable to decode the stream.
>
> object: a decimal representation of the MPEG-4 Audio Object Type value defined in [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.).

This parameter specifies the tool to be used by the decoder. It CAN be used to limit the capability within the specified "profile-level-id".

bitrate: the data rate for the audio bit stream.

cpresent: a boolean parameter indicates whether audio payload configuration data has been multiplexed into an RTP payload (see Section 5.1 (RTP Packet Format)). A 0 indicates the configuration data has not been multiplexed into an RTP payload and in this case the "config" parameter MUST be present, a 1 indicates that it has. The default if the parameter is omitted is 1. If this parameter is set to 1 and the "config" parameter is present, the multiplexed configuration data and the value of the "config" parameter SHALL be consistent.

config: a hexadecimal representation of an octet string that expresses the audio payload configuration data "StreamMuxConfig", as defined in [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.). Configuration data is mapped onto the octet string in an MSB-first basis. The first bit of the configuration data SHALL be located at the MSB of the first octet. In the last octet, zero-padding bits, if necessary, SHALL follow the configuration data. Senders MUST set the StreamMuxConfig elements taraBufferFullness and latmBufferFullness to their largest respective value, indicating that buffer fullness measures are not used in SDP. Receivers MUST ignore the value of these two elements contained in the config parameter.

MPS-asc: a hexadecimal representation of an octet string that expresses audio payload configuration data "AudioSpecificConfig", as defined in [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.). If this parameter is not present the relevant signaling is performed by other means (e.g. in-band or contained in the config string).

The same mapping rules as for the config parameter apply.

ptime: duration of each packet in milliseconds.

SBR-enabled: a boolean parameter which indicates whether SBR-data can be expected in the RTP-payload of a stream. This parameter is relevant for an SBR-capable decoder if the presence of SBR can not be detected from an out-of-band decoder configuration (e.g. contained in the config string).

If this parameter is set to 0, a decoder MAY expect that SBR is not used. If this parameter is set to 1, a decoder CAN upsample the audio data with the SBR tool, regardless whether SBR data is present in the stream or not.

> If the presence of SBR can not be detected from out-of-band configuration and the SBR-enabled parameter is not present, the parameter defaults to 1 for an SBR-capable decoder. If the resulting output sampling rate or the computational complexity is not supported, the SBR tool can be disabled or run in downsampled mode.
>
> The timestamp resolution at RTP layer is determined by the rate parameter.

Published specification:

> Encoding specifications are provided in [14496-3] (MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.). The RTP payload format specification is described in this document.

Encoding considerations:

> This type is only defined for transfer via RTP.

Security considerations:

> See Section 9 (Security Considerations) of this document.

Interoperability considerations:

> MPEG-4 Audio provides a large and rich set of tools for the coding of audio objects. For effective implementation of the standard, subsets of the MPEG-4 Audio tool sets similar to those used in MPEG-4 Visual have been provided (see Section 6.1 (Media Type Registration for MPEG-4 Visual)).
>
> The audio stream SHALL be compliant with the MPEG-4 Audio Profile@Level specified by the parameters "profile-level-id" and "MPS-profile-level-id". Interoperability between a sender and a receiver may be achieved by specifying the parameters "profile-level-id" and "MPS-profile-level-id", or by arranging in the capability exchange procedure to set this parameter mutually to the same value. Furthermore, the "object" parameter can be used to limit the capability within the specified Profile@Level in capability exchange.

Applications which use this media type:

> Audio and video streaming and conferencing tools.

Additional information: none
Personal and email address to contact for further information:

> See Authors' Address section at the end of this document.

Intended usage: COMMON
Author:

> See Authors' Address section at the end of this document.

Change controller:

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

6.4. Mapping to SDP for MPEG-4 Audio

TOC

The Media Type audio/MP4A-LATM string is mapped to fields in the Session Description Protocol (SDP) [RFC4566] (Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol," July 2006.), as follows:

- The Media Type (audio) goes in SDP "m=" as the media name.
- The Media subtype (MP4A-LATM) goes in SDP "a=rtpmap" as the encoding name.
- The required parameter "rate" goes in "a=rtpmap" as the clock rate.
- The optional parameter "ptime" goes in SDP "a=ptime" attribute.
- The optional parameters "profile-level-id", "MPS-profile-level-id" and "object" goes in the "a=fmtp" line to indicate the coder capability.

  Followings are some examples of the profile-level-id value:
  1 : Main Audio Profile Level 1
  9 : Speech Audio Profile Level 1
  15: High Quality Audio Profile Level 2
  30: Natural Audio Profile Level 1
  44: High Efficiency AAC Profile Level 2
  48: High Efficiency AAC v2 Profile Level 2
  55: Baseline MPEG Surround Profile (see ISO/IEC 23003-1) Level 3

  The optional payload-format-specific parameters "bitrate", "cpresent", "config", "MPS-asc" and "SBR-enabled" go also in the "a=fmtp" line. These parameters are expressed as a string, in the form of as a semicolon separated list of parameter=value pairs.

6.4.1. Declarative SDP Usage for MPEG-4 Audio

TOC

The following sections contain some examples of the media representation in SDP.
Note that the a=fmtp line in some of the examples has been wrapped to fit the page; they would comprise a single line in the SDP file.

6.4.1.1. Example: In-band Configuration

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In this example the audio configuration data appears in the RTP payload exclusively (i.e., the MPEG-4 audio configuration is known when a StreamMuxConfig element appears within the RTP payload).

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/90000
a=fmtp:96 object=2; cpresent=1
```

The "clock rate" is set to 90kHz. This is the default value and the real audio sampling rate is known when the audio configuration data is received.

6.4.1.2. Example: 6kb/s CELP

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6 kb/s CELP bitstreams (with an audio sampling rate of 8 kHz)

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/8000
a=fmtp:96 profile-level-id=9; object=8; cpresent=0;
  config=40008B18388380
a=ptime:20
```

In this example audio configuration data is not multiplexed into the RTP payload and is described only in SDP. Furthermore, the "clock rate" is set to the audio sampling rate.

6.4.1.3. Example: 64 kb/s AAC LC Stereo

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64 kb/s AAC LC stereo bitstream (with an audio sampling rate of 24 kHz)

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/24000/2
a=fmtp:96 profile-level-id=1; bitrate=64000; cpresent=0;
  object=2; config=400026203fc0
```

In this example audio configuration data is not multiplexed into the RTP payload and is described only in SDP. Furthermore, the "clock rate" is set to the audio sampling rate.
In this example, the presence of SBR can not be determined by the SDP parameter set. The clock rate represents the core codec sampling rate. An SBR enabled decoder can use the SBR tool to upsample the audio data if complexity and resulting output sampling rate permits.

6.4.1.4. Example: Use of the SBR-enabled Parameter

TOC

These two examples are identical to the example above with the exception of the SBR-enabled parameter. The presence of SBR is not signaled by the SDP parameters object, profile-level-id and config, but instead the SBR-enabled parameter is present. The rate parameter and the StreamMuxConfig contain the core codec sampling rate.
Example with "SBR-enabled=0", definitive and core codec sampling rate 24kHz:

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/24000/2
a=fmtp:96 profile-level-id=1; bitrate=64000; cpresent=0;
  SBR-enabled=0; config=400026203fc0
```

Example with "SBR-enabled=1", core codec sampling rate 24kHz, definitive and SBR sampling rate 48kHz:

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/24000/2
a=fmtp:96 profile-level-id=1; bitrate=64000; cpresent=0;
  SBR-enabled=1; config=400026203fc0
```

In this example, the clock rate is still 24000 and this information is used for RTP timestamp calculation. The value of 24000 is used to support old AAC decoders. This makes the decoder supporting only AAC understand the HE AAC coded data, although only plain AAC is supported. A HE AAC decoder is able to generate output data with the SBR sampling rate.

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6.4.1.5. Example: Hierarchical Signaling of SBR

When the presence of SBR is explicitly signaled by the SDP parameters object, profile-level-id or the config string as in the example below, the StreamMuxConfig contains both the core codec sampling rate and the SBR sampling rate.

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/48000/2
a=fmtp:96 profile-level-id=44; bitrate=64000; cpresent=0;
  config=40005623101fe0; SBR-enabled=1
```

This config string uses the explicit signaling mode 2.A (hierarchical signaling; See [14496-3] (MPEG, "ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio," 2009.). This means that the AOT(Audio Object Type) is SBR(5) and SFI(Sampling Frequency Index) is 6(24000 Hz) which refers to the underlying core codec sampling frequency. CC(Channel Configuration) is stereo(2), and the ESFI(Extension Sampling Frequency Index)=3 (48000) is referring to the sampling frequency of the extension tool(SBR).

6.4.1.6. Example: HE AAC v2 Signaling

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HE AAC v2 decoders are required to always produce a stereo signal from a mono signal. Hence, there is no parameter necessary to signal the presence of PS.
Example with "SBR-enabled=1" and 1 channel signaled in the a=rtpmap line and within the config parameter. Core codec sampling rate is 24kHz, definitive and SBR sampling rate is 48kHz. Core codec channel configuration is mono, PS channel configuration is stereo.

```
m=audio 49230 RTP/AVP 110
a=rtpmap:110 MP4A-LATM/24000/1
a=fmtp:110 profile-level-id=15; object=2; cpresent=0;
  config=400026103fc0; SBR-enabled=1
```

6.4.1.7. Example: Hierarchical Signaling of PS

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Example: 48khz stereo audio input:

```
m=audio 49230 RTP/AVP 110
a=rtpmap:110 MP4A-LATM/48000/2
a=fmtp:110 profile-level-id=48; cpresent=0; config=4001d613101fe0
```

The config parameter indicates explicit hierarchical signaling of PS and SBR. This configuration method is not supported by legacy AAC an HE AAC decoders and these are therefore unable to decode the the coded data.

6.4.1.8. Example: MPEG Surround

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The following examples show how MPEG Surround configuration data can be signaled using SDP. The configuration is carried within the config string in the first example by using two different layers. The general parameters in this example are: AudioMuxVersion=1; allStreamsSameTimeFraming=1; numSubFrames=0; numProgram=0; numLayer=1. The first layer describes the HE AAC payload and signals the following parameters: ascLen=25; audioObjectType=2 (AAC LC); extensionAudioObjectType=5 (SBR); samplingFrequencyIndex=6 (24kHz); extensionSamplingFrequencyIndex=3 (48kHz); channelConfiguration=2 (2.0 channels). The second layer describes the MPEG surround payload and specifies the following parameters: ascLen=110; AudioObjectType=30 (MPEG Surround); samplingFrequencyIndex=3 (48kHz); channelConfiguration=6 (5.1 channels); sacPayloadEmbedding=1; SpatialSpecificConfig=(48 kHz; 32 slots; 525 tree; ResCoding=1; ResBands=[7,7,7,7]).
In this example the signaling is carried by using two different LATM layers. The MPEG surround payload is carried together with the AAC payload in a single layer as indicated by the sacPayloadEmbedding Flag.

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/48000
a=fmtp:96 profile-level-id=1; bitrate=64000; cpresent=0;
  SBR-enabled=1;
  config=8FF8004192B11880FF0DDE3699F2408C00536C02313CF3CE0FF0
```

6.4.1.9. Example: MPEG Surround with Extended SDP Parameters

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The following example is an extension of the configuration given above by the MPEG Surround specific parameters. The MPS-asc parameter specifies the MPEG Surround Baseline Profile at Level 3 (PLI55) and the MPS-asc string contains the hexadecimal representation of the MPEG

Surround ASC [audioObjectType=30 (MPEG Surround); samplingFrequencyIndex=0x3 (48kHz); channelConfiguration=6 (5.1 channels); sacPayloadEmbedding=1; SpatialSpecificConfig=(48 kHz; 32 slots; 525 tree; ResCoding=1; ResBands=[0,13,13,13])].

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/48000
a=fmtp:96 profile-level-id=44; bitrate=64000; cpresent=0;
  config=40005623101fe0; MPS-profile-level-id=55;
  MPS-asc=F1B4CF920442029B501185B6DA00;
```

6.4.1.10. Example: MPEG Surround with Single Layer Configuration

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The following example shows how MPEG Surround configuration data can be signaled using the SDP config parameter. The configuration is carried within the config string using a single layer. The general parameters in this example are: AudioMuxVersion=1; allStreamsSameTimeFraming=1; numSubFrames=0; numProgram=0; numLayer=0. The single layer describes the combination of HE AAC and MPEG Surround payload and signals the following parameters: ascLen=101; audioObjectType=2 (AAC LC); extensionAudioObjectType=5 (SBR); samplingFrequencyIndex=7 (22.05kHz); extensionSamplingFrequencyIndex=7 (44.1kHz); channelConfiguration=2 (2.0 channels). A backward compatible extension according to [14496-3/Amd.1] (MPEG, "ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3: Audio, Amendment 1: HD-AAC profile and MPEG Surround signaling," 2009.) signals the presence of MPEG surround payload data and specifies the following parameters: SpatialSpecificConfig=(44.1 kHz; 32 slots; 525 tree; ResCoding=0). In this example the signaling is carried by using a single LATM layer. The MPEG surround payload is carried together with the HE AAC payload in a single layer.

```
m=audio 49230 RTP/AVP 96
a=rtpmap:96 MP4A-LATM/44100
a=fmtp:96 profile-level-id=44; bitrate=64000; cpresent=0;
  SBR-enabled=1; config=8FF8000652B920876A83A1F440884053620FF0;
  MPS-profile-level-id=55
```

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7. IANA Considerations

This document updates the media subtypes "MP4A-LATM" and "MP4V-ES" from RFC 3016. The new registrations are in Section 6.1 (Media Type Registration for MPEG-4 Visual) and Section 6.3 (Media Type Registration for MPEG-4 Audio) of this document.

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8. Acknowledgements

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The authors would like to thank Yoshihiro Kikuchi, Yoshinori Matsui, Toshiyuki Nomura, Shigeru Fukunaga and Hideaki Kimata for their work on RFC 3016, and Ali Begen, Keith Drage, Roni Even and Qin Wu for their valuable input and comments on this document.

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9. Security Considerations

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RTP packets using the payload format defined in this specification are subject to the security considerations discussed in the RTP specification [RFC3550] (Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, “RTP: A Transport Protocol for Real-Time Applications,” July 2003.), and in any applicable RTP profile. The main security considerations for the RTP packet carrying the RTP payload format defined within this document 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 most MPEG-4 codecs define an extension mechanism to transmit extra data within a stream that is gracefully skipped by decoders that do not support this extra data. This covert channel may be used to transmit unwanted data in an otherwise valid stream. 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] (Baugher, M., McGrew, D., Naslund, M., Carrara, E., and K. Norrman, “The Secure Real-time Transport Protocol (SRTP),” March 2004.) is recommended. Other mechanisms that may be used are IPsec [RFC4301] (Kent, S. and K. Seo, “Security Architecture for the Internet Protocol,” December 2005.) and Transport

Layer Security (TLS) [RFC5246] (Dierks, T. and E. Rescorla, "The Transport Layer Security (TLS) Protocol Version 1.2," August 2008.) (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. The complete MPEG-4 system allows for transport of a wide range of content, including Java applets (MPEG-J) and scripts. Since this payload format is restricted to audio and video streams, it is not possible to transport such active content in this format.

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10. Differences to RFC 3016

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The RTP payload format for MPEG-4 Audio as specified in RFC 3016 is used by the 3GPP PSS service (3GPP, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs (Release 9)," December 2010.) [3GPP]. However, there are some misalignments between RFC 3016 and the 3GPP PSS specification that are addressed by this update:

- The audio payload format (LATM) referenced in this document is binary compatible to the format used in [3GPP] (3GPP, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs (Release 9)," December 2010.).

- The audio signaling format (StreamMuxConfig) referenced in this document is binary compatible to the format used in [3GPP] (3GPP, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs (Release 9)," December 2010.).

- The use of an audio parameter "SBR-enabled" is now defined in this document, which is used by 3GPP implementations [3GPP] (3GPP, "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Transparent end-to-end Packet-switched Streaming Service (PSS); Protocols and codecs (Release 9)," December 2010.).

- The rate parameter is defined unambiguously in this document for the case of presence of SBR (Spectral Band Replication)

*The number of audio channels parameter is defined unambiguously in this document for the case of presence of PS (Parametric Stereo)

Furthermore some comments have been addressed and signaling support for MPEG surround [23003-1] (MPEG, “ISO/IEC International Standard 23003-1 - MPEG Surround (MPEG D),” 2007.) was added.

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11. References

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11.1. Normative References

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[14496-2]	MPEG, “ISO/IEC International Standard 14496-2 - Coding of audio-visual objects, Part 2: Visual,” 2003.
[14496-3]	MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3 Audio,” 2009.
[14496-3/ Amd.1]	MPEG, “ISO/IEC International Standard 14496-3 - Coding of audio-visual objects, Part 3: Audio, Amendment 1: HD-AAC profile and MPEG Surround signaling,” 2009.
[23003-1]	MPEG, “ISO/IEC International Standard 23003-1 - MPEG Surround (MPEG D),” 2007.
[RFC2119]	Bradner, S., “Key words for use in RFCs to Indicate Requirement Levels,” BCP 14, RFC 2119, March 1997 (TXT, HTML, XML).
[RFC3016]	Kikuchi, Y., Nomura, T., Fukunaga, S., Matsui, Y., and H. Kimata, “RTP Payload Format for MPEG-4 Audio/Visual Streams,” RFC 3016, November 2000 (TXT).
[RFC3550]	Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, “RTP: A Transport Protocol for Real-Time Applications,” STD 64, RFC 3550, July 2003 (TXT, PS, PDF).
[RFC4288]	Freed, N. and J. Klensin, “Media Type Specifications and Registration Procedures,” BCP 13, RFC 4288, December 2005 (TXT).
[RFC4566]	Handley, M., Jacobson, V., and C. Perkins, “SDP: Session Description Protocol,” RFC 4566, July 2006 (TXT).
[RFC4629]	Ott, H., Bormann, C., Sullivan, G., Wenger, S., and R. Even, “RTP Payload Format for ITU-T Rec,” RFC 4629, January 2007 (TXT).
[RFC4855]	Casner, S., “Media Type Registration of RTP Payload Formats,” RFC 4855, February 2007 (TXT).
[RFC5583]	

	Schierl, T. and S. Wenger, “Signaling Media Decoding Dependency in the Session Description Protocol (SDP),” RFC 5583, July 2009 (TXT).

11.2. Informative References

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[14496-1]	MPEG, “ISO/IEC International Standard 14496-1 - Coding of audio-visual objects, Part 1 Systems,” 2004.
[14496-12]	MPEG, “ISO/IEC International Standard 14496-12 - Coding of audio-visual objects, Part 12 ISO base media file format.”
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