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# Codec agnostic RTP payload format for video
# draft-gouaillard-avtcore-codec-agn-rtp-payload-01

Abstract

RTP Media Chains usually rely on piping encoder output directly to packetizers. Media packetization formats often support a specific codec format and optimize RTP packets generation accordingly.

With the development of Selective Forward Unit (SFU) solutions, that do not process media content server side, the need for media content processing at the origin and at the destination has arised.

RTP Media Chains used e.g. in WebRTC solutions are increasingly relying on application-specific transforms that sit in-between encoder and packetizer on one end and in-between depacketizer and decoder on the other end. This use case has become so important, that the W3C is standardizing the capacity to access encoded content with the [WebRTCInsertableStreams] API proposal. An extremely popular use case is application level end-to-end encryption of media content, using for instance [SFrame].

Whatever the modification applied to the media content, RTP packetizers can no longer expect to use packetization formats that mandate media content to be in a specific codec format.

In the extreme cases like encryption, where the RTP Payload is made completely opaque to the SFUs, some extra mechanism must also be added for them to be able to route the packets without depending on RTP payload or payload headers.

The traditionnal process of creating a new RTP Payload specification per content would not be practical as we would need to make a new one for each codec-transform pair.

This document describes a solution, which provides the following features in the case the encoded content has been modified before reaching the packetizer: - a paylaod agnostic RTP packetization format that can be used on any media content, - a negotiation

mechanism for the above format and the inner payload, Both of the above mechanism are backward compatible with most of (S)RTP/RTCP mechanisms used for bandwidth estimation and congestion control in RTP/SRTP/webrtc, including but not limited to SSRC, RED, FEC, RTX, NACK, SR/RR, REMB, transport-wide-CC, TMBR, .... It as illustrated by existing implementations in chrome, safari, and Medooze.

This document also describes a solution to allow SFUs to continue performing packet routing on top of this generic RTP packetization format.

This document complements the SFrame (media encryption), and Dependency Descriptor (AV1 payload annex) documents to provide an End-to-End-Encryption solution that would sit on top of SRTP/Webrtc, use SFUs on the media back-end, and leverage W3C APIs in the browser. A high level description of such system will be provided as an informational I-D in the SFrame WG and then cited here.

## Status of This Memo



## Copyright Notice

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## 1. Introduction

As per Figure 1 of [RFC7656], a Media Packetizer transforms a single Encoded Stream into one or several RTP packets. The Encoded Stream is coming straight from the Media Encoder and is expected to follow the format produced by the Media Encoder. A number of Media Packetizer formats have been designed to process a specific format produced by Media Encoder. For instance [RFC6184] is dedicated to the processing of content produced by H.264 Media Encoders, and generates packets following NALUs organization.

WebRTC applications are increasingly deploying end-to-end encryption solutions on top of RTP Media Chains. End-to-end encryption is implemented by inserting application-specific Media Transformers between Media Encoder and Media Packetizer on the sending side, and between Media Depacketizer and Media Decoder on the receiving side, as described in Figure 1 and Figure 2. To support end-to-end encryption, Media Transformers can use the [SFrame] format. In browsers, Media Transformers are implemented using

[WebRTCInsertableStreams], for instance by injecting JavaScript code provided by web pages.

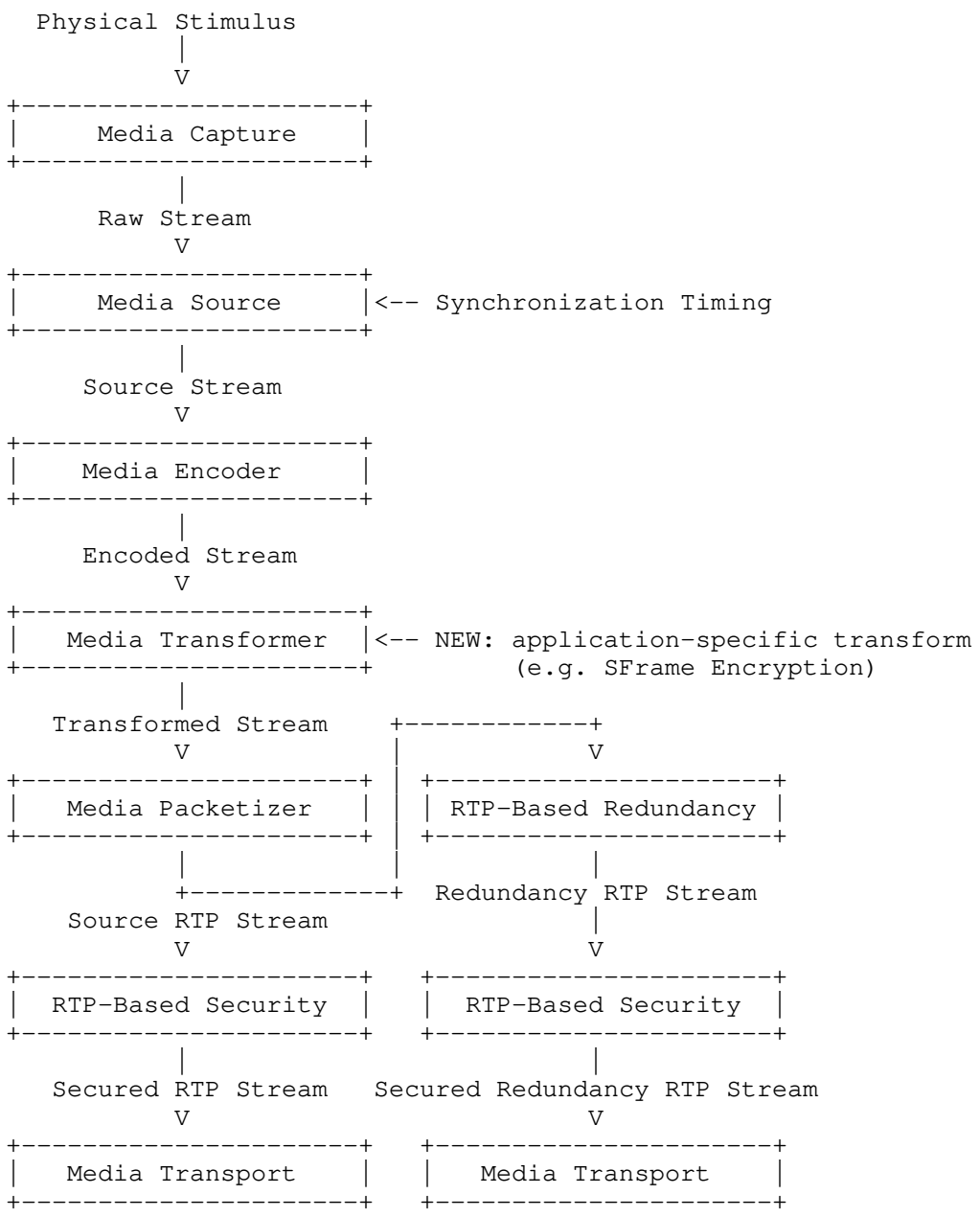
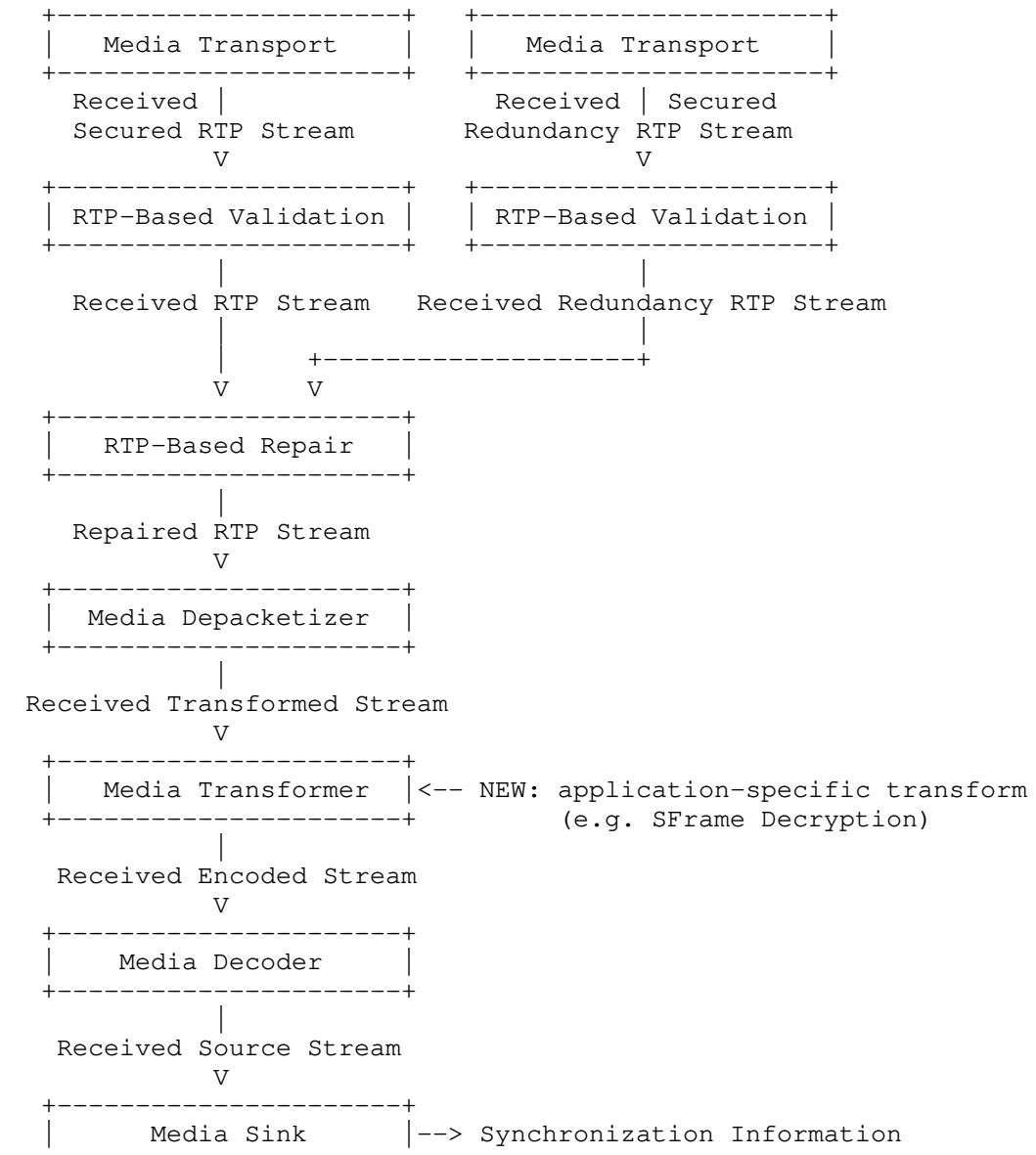


Figure 1: Sender Side Concepts in the Media Chain
With Application-level Media Transform

These RTP packets are sent over the wire to a receiver media chain matching the sender side, reaching the Media Depacketizer that will reconstruct the Encoded Stream before passing it to the Media Decoder.



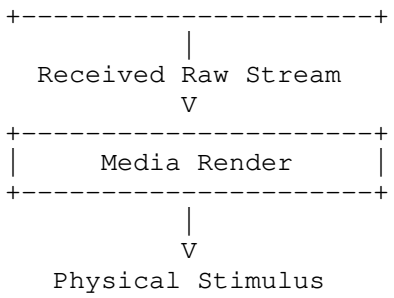



Figure 2: Receiver Side Concepts in the Media Chain With Application-level Media Transform

This generic packetization does not change how the mapping between one or several encoded or dependant streams are mapped to the RTP streams or how the synchronization sources(s) (SSRC) are assigned.

Given the use of post-encoder application-specific transforms, the whole Media Chain needs to be made aware of it. This includes the sender post-transform Media Chain, Media Transport intermediaries (SFUs typically) and receiver pre-transform Media Chain.

As these transforms can alter Encoded Streams in any possible way, the use of codec-specific Media Packetizers like [RFC6184] on Transformed Stream may be suboptimal on sender side. It may also be problematic on the receiving side in case codec-specific processing is done prior the Media Transformer. Media Transport intermediaries are often looking at the Media Content itself to fuel their packet selection algorithms.

## 2. Goals

The objective of this document is to support inserting any application-specific transform between encoders and packetizers in the Media Chain. For that purpose, this document will: 1. Provide a generic packetization format that supports any media content (compressed audio, compressed video, encrypted content...) that allows reuse of existing RTP mechanisms in place in WebRTC applications such as RTX, RED or FEC. 2. Provide a way to negotiate use of the generic packetization format between sender and receiver, with minimum impact on existing negotiation approaches. 3. Provide a side-channel information so that network intermediaries (SFU in particular) can do their existing packet routing strategies without inspecting the media content.

## 3. RTP Packetization

A generic packetizer, by design, is not expected to understand the format of the media to transmit. The unit used by the packetizer to do processing is called a frame in the remainder of the document.

It is the responsibility of the application using the packetizer to group media content in meaningful frames. In the common case of a video codec, the packetizer frame is the frame in byte format (h264 annex b for example) generated by the encoder.

If the application wants to transform encoded content, the application needs to split the encoded content into frames prior the transform. Each frame is then transformed independently, for instance encrypted using [SFrame]. The content of each transformed frame is then processed by the packetizer.

In the case of a video codec supporting spatial scalability, each spatial layer MUST be split in its own frame by the application before passing it to the packetizer.

When the packetizer receives a frame from the application, it MUST fragment the frame content in multiple RTP packets to ensure packets do not exceed the network maximum transmission unit. The content of the frame will be treated as a binary blob by the packetizer, so the decision about the boundaries of each fragment is decided arbitrarily by the packetizer. The packetizer or any relaying server MUST NOT modify the frame content and concatenating the RTP payload of the RTP packets for each frame MUST produce the exact binary content of the input frame content.

The marker bit of each RTP packet in a frame MUST be set according to the audio and video profiles specified in [RFC3551].

The spatial layer frames are sent in ascending order, with the same RTP timestamp, and only the last RTP packet of the last spatial layer frame will have the marker bit set to 1.

## 4. Payload Multiplexing

In order to reduce the number of payload type in the SDP exchange, a single payload type code for the generic packetization can be used for all negotiated media formats. That requires to identify the original payload type code of the frame negotiated media format, called the associated payload type (APT) hereunder. The APT value is the payload type code of the associated format passed to the generic Media Packetizer before any transformation is applied.

The APT value is sent in a dedicated header extension. The payload of this header extension can be encoded using either the one-byte or two-byte header defined in [RFC5285]. Figures 3 and 4 show examples with each one of these examples.

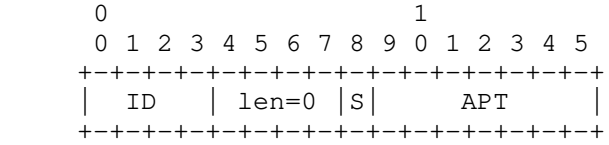



Figure 3: Frame Associated Payload Type Encoding Using the One-Byte Header Format

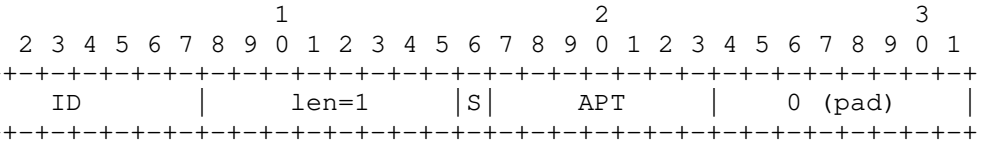



Figure 4: Frame Associated Payload Type Encoding Using the Two-Byte Header Format

The APT value is the associated payload type value. The S bit indicates if the media stream can be forwarded safely starting from this RTP packet. Typically, it will be set to 1 on the first RTP packet of an intra video frame and in all RTP audio packets.

Receivers MUST be ready to receive RTP packets with different associated payload types in the same way they would receive different payload type codes on the RTP packets.

The URI for declaring this header extension in an extmap attribute is "urn:ietf:params:rtp-hdrext:associated-payload-type".

## 5. SDP Negotiation

To use the RTP generic packetization, the SDP Offer/Answer exchange MUST negotiate: - The payload type of the negotiated codec format - The generic payload type - The associated payload type header extension

Only the negotiated payload types are allowed to be used as associated payload types. Figure 5 illustrates a SDP that negotiates exchange of video using either VP8 or VP9 codecs with the possibility to use the generic packetization. In this example, RTX is also negotiated and will be applied normally on each associated payload type.

```
m=video 9 UDP/TLS/RTP/SAVPF 96 97 98 99 100 101
c=IN IP4 0.0.0.0
a=rtcp:9 IN IP4 0.0.0.0
a=setup:actpass
a=mid:1
a=extmap:1 urn:ietf:params:rtp-hdrext:sdes:mid
a=extmap:2 urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
a=extmap:3 urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-stream-id
a=extmap:4 urn:ietf:params:rtp-hdrext:associated-payload-type
a=sendrecv
a=rtpmap:96 vp9/90000
a=rtpmap:97 vp8/90000
a=rtpmap:98 generic/90000
a=rtpmap:99 rtx/90000
a=fmtp:99 apt=96
a=rtpmap:100 rtx/90000
a=fmtp:100 apt=97
a=rtpmap:101 rtx/90000
a=fmtp:101 apt=98
```

Figure 5: SDP example negotiating the generic payload type and related header extension for video

## 6. SFU Packet Selection

SFUs need to have a basic understanding of each frame they receive so they can decide to forward it or not and to which endpoint. They might need similar information to support media content recording. This information is either generic to a group of frame (called a stream hereafter) or specific to each frame.

The information is transmitted as a RTP header extension as the RTP packet payload should be treated as opaque by the SFU. This is especially necessary if the payload is end-to-end encrypted. The amount of information should be limited to what is strictly necessary to the SFU task since it is not always as trusted as individual peers.

For audio, configuration information such as Opus TOC might be useful. For video, configuration information might include: - Stream configuration information: resolution, quality, frame rate... - Codec specific configuration information: codec profile like profile_idc... - Frame specific information: whether the stream is decodable when starting from this frame, whether the frame is skippable...

For video content, this information can be sent using a Dependency Descriptor header extension. In that case, the first RTP packet of

the frame will have its start_of_frame equal to 1 and the last packet will have its end_of_frame equal to 1.

## 7. Redundancy Techniques Considerations

The solution described in this document is expected to integrate well with the existing RTP ecosystem. This section describes how the generic packetizer can be used jointly with existing techniques that allow to mitigate unreliable transports.

### 7.1. Retransmission Techniques

[RFC4588] defines a retransmission payload format (RTX) that can be used in case of packet loss. As defined in [RFC4588], RTX is able to handle any payload format, including the format described in this document. Given RTX preserves both RTP packet payload and headers, the receiver will be able to identify the payload type of the recovered packet and whether generic packetization is used. RTX will also allow recovering RTP header extensions that convey information on the media content itself.

### 7.2. Forward Error Correction (FEC) Techniques

FEC is another technique used in RTP Media Chains to protect media content against packet loss. [RFC5109] defines such a payload format used to transmit FEC for specific packets protection.

FEC may protect some parts of the media content more than others. For instance, intra video frame encoded data or important network abstraction layer units (NALUs) like SPS/PPS may be more protected. With a post-encoder transform and the use of a generic packetization, the granularity of the recovery mechanism is no longer at the NALU level but at the level of the frame generated by the post-encoder transform. In case a SVC codec is used, each spatial layer will be processed as an independent frame. In that case, base layers can be protected more heavily than higher resolution layers.

### 7.3. Redundant Audio Data Techniques

As defined in [RFC7656] RTP-based redundancy is defined here as a transformation that generates redundant or repair packets sent out as a Redundancy RTP Stream to mitigate Network Transport impairments, like packet loss and delay.

[RFC2198] defines a payload format for sending the same audio data encoded multiple times at different quality levels. This allows to use a lower quality encoding of the audio data, should the higher quality encoding of the audio data is lost during the transmission.

If a Media Transformation is in use, both the primary and redundant encoding must be transformed independently and the redundant packet created normally. As the RTP headers present in the redundant packet are only applicable to the primary encoding, if the payload type for a redundant encoding block is mapped to the generic packetizer, the value of the associated payload type for the primary encoding is applied to the redundant encoding block as well.

# 8. Alternatives

Various alternatives can be used to implement and negotiate generic packetization. This section describes a few additional alternatives. This section is to be removed before finalization of the document.

## 8.1. Generic Packetization With In-Payload APT

Instead of using a RTP header extension to convey the APT value, it is prepended in the RTP payload itself. As the value cannot change for a whole frame, its value is prepended to the first packet generated of the frame only. This removes the need to negotiate a dedicated header extension, but may require the SFU to update the payload when sending or recording content.

## 8.2. A Payload Type for Generic Packetization AND Media Format

The payload type is negotiated in the SDP so as to identify both the negotiated codec format and the generic packetization use. There is no network cost but this increases the number of payload types used in the SDP.

```
m=video 9 UDP/TLS/RTP/SAVPF 96 97 98 99 100 101
c=IN IP4 0.0.0.0
a=rtcp:9 IN IP4 0.0.0.0
a=setup:actpass
a=mid:1
a=extmap:1 urn:ietf:params:rtp-hdrext:sdes:mid
a=extmap:2 urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
a=extmap:3 urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-stream-id
a=sendrecv
a=rtpmap:96 vp9/90000
a=rtpmap:97 generic/90000
a=fmtp:97 apt=96
a=rtpmap:98 vp8/90000
a=rtpmap:99 generic/90000
a=fmtp:99 apt=98
a=rtpmap:100 rtx/90000
a=fmtp:100 apt=96
a=rtpmap:101 rtx/90000
a=fmtp:101 apt=97
a=rtpmap:102 rtx/90000
a=fmtp:102 apt=98
a=rtpmap:103 rtx/90000
a=fmtp:103 apt=99
```

Figure 6: SDP example negotiating a payload type for format and generic packetization

A variation of this approach is to consider defining generic payload types, each of them having an identified codec format.

```
m=video 9 UDP/TLS/RTP/SAVPF 96 97 98 99 100 101
c=IN IP4 0.0.0.0
a=rtcp:9 IN IP4 0.0.0.0
a=setup:actpass
a=mid:1
a=extmap:1 urn:ietf:params:rtp-hdrext:sdes:mid
a=extmap:2 urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
a=extmap:3 urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-stream-id
a=sendrecv
a=rtpmap:96 generic/90000
a=fmtp:96 codec=vp9
a=rtpmap:97 generic/90000
a=fmtp:97 codec=vp8
a=rtpmap:98 rtx/90000
a=fmtp:98 apt=96
a=rtpmap:99 rtx/90000
a=fmtp:99 apt=97
```

Figure 7: SDP example negotiating a payload type for format and generic packetization

### 8.3. A RTP Header To Choose Packetization

A RTP header extension can be used to flag content as opaque so that the receiver knows whether to use or not the generic packetization. As for the API header extension, the RTP header extension may not need to be sent for every packet, it could for instance be sent for the first packet of every intra video frame. The main advantage of this approach is the reduced impact on SDP negotiation.

```
m=video 9 UDP/TLS/RTP/SAVPF 96 97 98 99 100 101
c=IN IP4 0.0.0.0
a=rtcp:9 IN IP4 0.0.0.0
a=setup:actpass
a=mid:1
a=extmap:1 urn:ietf:params:rtp-hdrext:sdes:mid
a=extmap:2 urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
a=extmap:3 urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-stream-id
a=extmap:4 urn:ietf:params:rtp-hdrext:generic-packetization-use
a=sendrecv
a=rtpmap:96 vp9/90000
a=rtpmap:97 vp8/90000
a=rtpmap:98 rtx/90000
a=fmtp:98 apt=96
a=rtpmap:99 rtx/90000
a=fmtp:99 apt=97
```

Figure 8: SDP example negotiating generic packetization as RTP header extension

## 9. Security Considerations

RTP packets using the payload format defined in this specification are subject to the general security considerations discussed in [RFC3550]. It is not expected that the proposed solutions (generic packetization and header extension) presented in this document can create new security threats. The use and implementation of RTP Media Chains containing Media Transformers needs to be done carerefully. It is important to refer to the security considerations discussed in [SFrame] and [WebRTCInsertableStreams]. In particular Media Transformers on the receiver side need to be prepared to receive arbitrary content, like decoders already do. Similarly, since Media Transformers can be implemented as JavaScript in browsers, RTP Packetizers should be prepared to receive arbitrary content.

## 10. IANA Considerations

Two new media subtypes have been registered with IANA, as described in this section.

### 10.1. Registration of audio/generic

Type name: audio

Subtype name: generic

Required parameters: none

Optional parameters: none

Encoding considerations: This format is framed (see Section 4.8 in the template document) and contains binary data.

Security considerations: TBD.

Interoperability considerations: TBD

Published specification: TBD.

Applications that use this media type: TBD.

Additional information: none

Intended usage: COMMON

Restrictions on usage: TBD

Author:

Change controller:

## 11. Registration of video/generic

Type name: video

Subtype name: generic

Required parameters: none

Optional parameters: none

Encoding considerations: This format is framed (see Section 4.8 in the template document) and contains binary data.

Security considerations: TBD.

Interoperability considerations: TBD

Published specification: TBD.

Applications that use this media type: TBD.

Additional information: none

Intended usage: COMMON

Restrictions on usage: TBD

Author:

Change controller:

## 12. References

### 12.1. Normative References

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Expires: 11 September 2021

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10 March 2021

# Completely Encrypting RTP Header Extensions and Contributing Sources
draft-ietf-avtcore-cryptex-01

## Abstract

While the Secure Real-time Transport Protocol (SRTP) provides confidentiality for the contents of a media packet, a significant amount of metadata is left unprotected, including RTP header extensions and contributing sources (CSRCs). However, this data can be moderately sensitive in many applications. While there have been previous attempts to protect this data, they have had limited deployment, due to complexity as well as technical limitations.

This document proposes a new mechanism to completely encrypt header extensions and CSRCs as well a simpler signaling mechanism intended to facilitate deployment.

## Discussion Venues

This note is to be removed before publishing as an RFC.

Source for this draft and an issue tracker can be found at https://github.com/juberti/cryptex.

## Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at https://datatracker.ietf.org/drafts/current/.

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This Internet-Draft will expire on 11 September 2021.



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## 1. Introduction

### 1.1. Problem Statement

The Secure Real-time Transport Protocol [RFC3711] mechanism provides message authentication for the entire RTP packet, but only encrypts the RTP payload. This has not historically been a problem, as much of the information carried in the header has minimal sensitivity (e.g., RTP timestamp); in addition, certain fields need to remain as cleartext because they are used for key scheduling (e.g., RTP SSRC and sequence number).

However, as noted in [RFC6904], the security requirements can be different for information carried in RTP header extensions, including the per-packet sound levels defined in [RFC6464] and [RFC6465], which are specifically noted as being sensitive in the Security Considerations section of those RFCs.

In addition to the contents of the header extensions, there are now enough header extensions in active use that the header extension identifiers themselves can provide meaningful information in terms of determining the identity of endpoint and/or application. Accordingly, these identifiers can be considered at least slightly sensitive.

Finally, the CSRCs included in RTP packets can also be sensitive, potentially allowing a network eavesdropper to determine who was speaking and when during an otherwise secure conference call.

### 1.2. Previous Solutions

[RFC6904] was proposed in 2013 as a solution to the problem of unprotected header extension values. However, it has not seen significant adoption, and has a few technical shortcomings.

First, the mechanism is complicated. Since it allows encryption to be negotiated on a per-extension basis, a fair amount of signaling logic is required. And in the SRTP layer, a somewhat complex transform is required to allow only the selected header extension values to be encrypted. One of the most popular SRTP implementations had a significant bug in this area that was not detected for five years.

Second, it only protects the header extension values, and not their ids or lengths. It also does not protect the CSRCs. As noted above, this leaves a fair amount of potentially sensitive information exposed.

Third, it bloats the header extension space. Because each extension must be offered in both unencrypted and encrypted forms, twice as many header extensions must be offered, which will in many cases push implementations past the 14-extension limit for the use of one-byte extension headers defined in [RFC8285]. Accordingly, implementations will need to use two-byte headers in many cases, which are not supported well by some existing implementations.

Finally, the header extension bloat combined with the need for backwards compatibility results in additional wire overhead. Because two-byte extension headers may not be handled well by existing implementations, one-byte extension identifiers will need to be used for the unencrypted (backwards compatible) forms, and two-byte for the encrypted forms. Thus, deployment of [RFC6904] encryption for header extensions will typically result in multiple extra bytes in each RTP packet, compared to the present situation.

### 1.3. Goals

From this analysis we can state the desired properties of a solution:

* Build on existing [RFC3711] SRTP framework (simple to understand)

* Build on existing [RFC8285] header extension framework (simple to implement)

* Protection of header extension ids, lengths, and values

* Protection of CSRCs when present

* Simple signaling

* Simple crypto transform and SRTP interactions

* Backward compatible with unencrypted endpoints, if desired

* Backward compatible with existing RTP tooling

The last point deserves further discussion. While we considered possible solutions that would have encrypted more of the RTP header (e.g., the number of CSRCs), we felt the inability to parse the resultant packets with current tools, as well as additional complexity incurred, outweighed the slight improvement in confidentiality.

## 2. Terminology

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].

## 3. Design

This specification proposes a mechanism to negotiate encryption of all RTP header extensions (ids, lengths, and values) as well as CSRC values. It reuses the existing SRTP framework, is accordingly simple to implement, and is backward compatible with existing RTP packet parsing code, even when support for this mechanism has been negotiated.

## 4. Signaling

In order to determine whether this mechanism defined in this specification is supported, this document defines a new "a=cryptex" Session Description Protocol (SDP) [RFC4566] attribute to indicate support. This attribute takes no value, and can be used at the session level or media level. Offering this attribute indicates that the endpoint is capable of receiving RTP packets encrypted as defined below.

The formal definition of this attribute is:

Name: cryptex

Value: None

Usage Level: session, media

Charset Dependent: No

Example:

```
a=cryptex
```

When used with BUNDLE, this attribute is assigned to the TRANSPORT category [RFC8859].

## 5. RTP Header Processing

[RFC8285] defines two values for the "defined by profile" field for carrying one-byte and two-byte header extensions. In order to allow a receiver to determine if an incoming RTP packet is using the encryption scheme in this specification, two new values are defined:

* 0xC0DE for the encrypted version of the one-byte header extensions (instead of 0xBEDE).

* 0xC2DE for the encrypted versions of the two-byte header extensions (instead of 0x100).

In the case of using two-byte header extensions, the extension id with value 256 MUST NOT be negotiated, as the value of this id is meant to be contained in the "appbits" of the "defined by profile" field, which are not available when using the values above.

If the "a=extmap-allow-mixed" attribute defined in [RFC8285] is negotiated, either one-byte or two-byte header ids can be used (with the values above), as in [RFC8285].

## 5.1. Sending

When the mechanism defined by this specification has been negotiated, sending a RTP packet that has any CSRCs or contains any {RFC8285}} header extensions follows the steps below. This mechanism MUST NOT be used with header extensions other than the [RFC8285] variety.

If the packet contains solely one-byte extension ids, the 16-bit RTP header extension tag MUST be set to 0xC0DE to indicate that the encryption has been applied, and the one-byte framing is being used. If the packet contains only two-byte extension ids, the header extension tag MUST be set to 0xC2DE to indicate encryption has been applied, and the two-byte framing is being used.

If the packet contains CSRCs but no header extensions, an empty extension block consisting of the 0xC0DE tag and a 16-bit length field set to zero (explicitly permitted by [RFC3550]) MUST be appended, and the X bit MUST be set to 1 to indicate an extension block is present. This is necessary to provide the receiver an indication that the CSRCs in the packet are encrypted.

The RTP packet MUST then be encrypted as described in Encryption Procedure.

## 5.2. Receiving

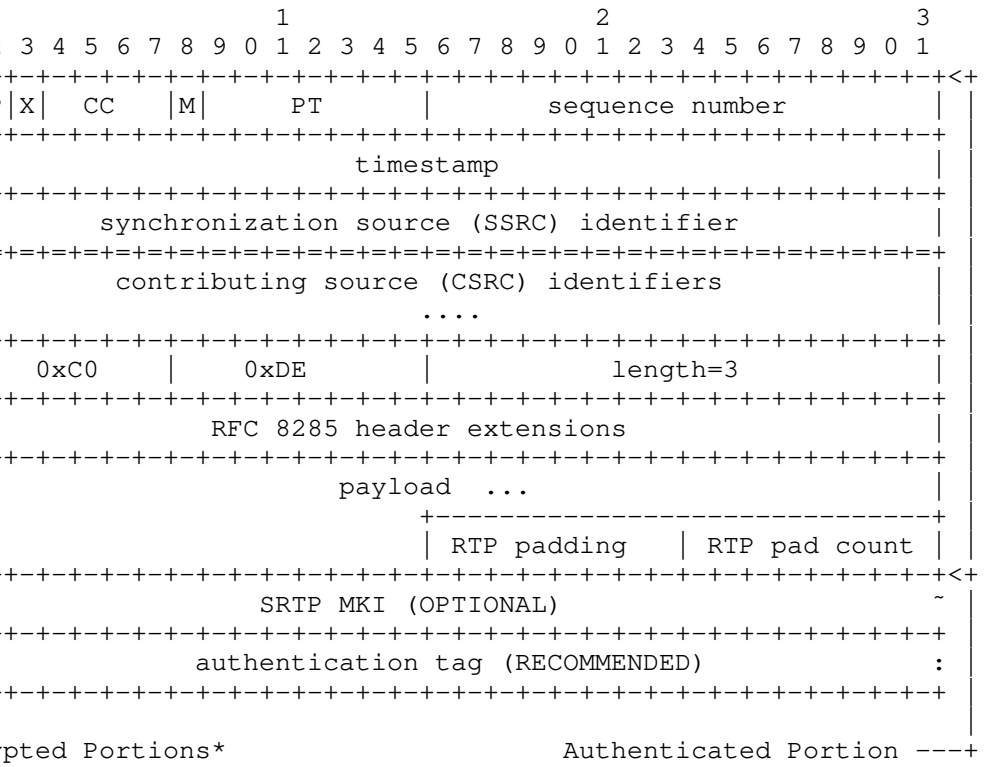
When receiving an RTP packet that contains header extensions, the "defined by profile" field MUST be checked to ensure the payload is formatted according to this specification. If the field does not match one of the values defined above, the implementation MUST instead handle it according to the specification that defines that value. The implemntation MAY stop and report an error if it considers use of this specification mandatory for the RTP stream.

If the RTP packet passes this check, it is then decrypted according to Decryption Procedure, and passed to the the next layer to process the packet and its extensions. In the event that a zero-length extension block was added as indicated above, it can be left as-is and will be processed normally.

# 6. Encryption and Decryption

## 6.1. Packet Structure

When this mechanism is active, the SRTP packet is protected as follows:




* Note that the 4 bytes at the start of the extension block are not encrypted, as required by [RFC8285].

Specifically, the encrypted portion MUST include any CSRC identifiers, any RTP header extension (except for the first 4 bytes), and the RTP payload.

### 6.2. Encryption Procedure

The encryption procedure is identical to that of [RFC3711] except for the region to encrypt, which is as shown in the section above.

To minimize changes to surrounding code, the encryption mechanism can choose to replace a "defined by profile" field from [RFC8285] with its counterpart defined in RTP Header Processing above and encrypt at the same time.

### 6.3. Decryption Procedure

The decryption procedure is identical to that of [RFC3711] except for the region to decrypt, which is as shown in the section above.

To minimize changes to surrounding code, the decryption mechanism can choose to replace the "defined by profile" field with its no-encryption counterpart from [RFC8285] and decrypt at the same time.

## 7. Backwards Compatibility

This specification attempts to encrypt as much as possible without interfering with backwards compatibility for systems that expect a certain structure from an RTPv2 packet, including systems that perform demultiplexing based on packet headers. Accordingly, the first two bytes of the RTP packet are not encrypted.

This specification also attempts to reuse the key scheduling from SRTP, which depends on the RTP packet sequence number and SSRC identifier. Accordingly these values are also not encrypted.

## 8. Security Considerations

This specification extends SRTP by expanding the portion of the packet that is encrypted, as shown in Packet Structure. It does not change how SRTP authentication works in any way. Given that more of the packet is being encrypted than before, this is necessarily an improvement.

The RTP fields that are left unencrypted (see rationale above) are as follows:

* RTP version

* padding bit

* extension bit

* number of CSRCs
* marker bit
* payload type
* sequence number
* timestamp
* SSRC identifier
* number of [RFC8285] header extensions

These values contain a fixed set (i.e., one that won't be changed by extensions) of information that, at present, is observed to have low sensitivity. In the event any of these values need to be encrypted, SRTP is likely the wrong protocol to use and a fully-encapsulating protocol such as DTLS is preferred (with its attendant per-packet overhead).

## 9. IANA Considerations

This document defines two new 'defined by profile' attributes, as noted in RTP Header Processing.

## 10. Acknowledgements

The authors wish to thank Lennart Grahl for pointing out many of the issues with the existing header encryption mechanism, as well as suggestions for this proposal. Thanks also to Jonathan Lennox and Inaki Castillo for their review and suggestions.

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# RTP Payload Format for Essential Video Coding (EVC)
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## Abstract

This memo describes an RTP payload format for the video coding standard ISO/IEC International Standard 23094-1 [EVC], also known as Essential Video Coding [EVC] and developed by ISO/IEC JTC1/SC29/WG11 (MPEG). The RTP payload format allows for packetization of one or more Network Abstraction Layer (NAL) units in each RTP packet payload as well as fragmentation of a NAL unit into multiple RTP packets. The payload format has wide applicability in videoconferencing, Internet video streaming, and high-bitrate entertainment-quality video, among other applications.

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This Internet-Draft will expire on 8 August 2021.

## Copyright Notice

Table of Contents

# 1. Introduction

The [EVC] specification, which is formally designated as ISO/IEC International Standard 23094-1 [ISO23094-1] has been published in October 2020. One goal of MPEG is to keep [EVC]'s Baseline profile essentially royalty free by by using the technologies published more than 20 years or otherwise freely available for use, whereas more advanced profiles follow a reasonable and non-discriminatory licensing terms policy. Both Baseline profile and higher profiles of [EVC] are reported to provide coding efficiency gains over [HEVC] and [AVC] under certain configurations.

This memo describes an RTP payload format for [EVC]. It shares its basic design with the NAL unit-based RTP payload formats of H.264 Video Coding [RFC6184], Scalable Video Coding (SVC) [RFC6190], High Efficiency Video Coding (HEVC) [RFC7798], and Versatile Video Coding (VVC)[I-D.ietf-avtcore-rtp-vvc]. With respect to design philosophy, security, congestion control, and overall implementation complexity, it has similar properties to those earlier payload format specifications. This is a conscious choice, as at least RFC 6184 is widely deployed and generally known in the relevant implementer communities. Certain mechanisms known from [RFC6190] were incorporated as EVC supports temporal scalability. [EVC] currently does not offer higher forms of scalability.

## 1.1. Overview of the EVC Codec

[EVC], [AVC], [HEVC] and [VVC] share a similar hybrid video codec design. In this memo, we provide a very brief overview of those features of [EVC] that are, in some form, addressed by the payload format specified herein. Implementers have to read, understand, and apply the ISO/IEC specifications pertaining to [EVC] to arrive at interoperable, well-performing implementations. The EVC standard has a Baseline profile and on top of that, a Main profile, the latter including more advanced features. The syntax elements allow encoders to mark a bitstream as to what of the many independent coding tools are exercised in the bitstream, in a spirit similar to the general_constraint_flags of [VVC] is provided.

Conceptually, all [EVC], [AVC], [HEVC] and [VVC] include a Video Coding Layer (VCL), which is often used to refer to the coding-tool features, and a Network Abstraction Layer (NAL), which is often used to refer to the systems and transport interface aspects of the codecs.

#### 1.1.1. Coding-Tool Features (informative)

Coding blocks and transform structure

[EVC] uses a traditional quad-tree coding structure, which divides the encoded image into blocks of up to 128x128 luma samples, which can be recursively divided into smaller blocks. The Main profile adds two advanced coding structure tools: Binary Ternary Tree (BTT) that allows non-square coding units and segmentation that changes the processing order of the segmentation unit from traditional left-scanning order processing to right-scanning order processing Unit Coding Order (SUCO). In the Main profile, the picture can be divided into slices and tiles, and these slices can be independently encoded and/or decoded in parallel.

When predicting a data block using intra prediction or inter prediction, the remaining data is usually added to the prediction block. The residual data is added to the prediction block. The residual data is obtained by applying an inverse quantization process and an inverse transform. [EVC] includes integer discrete cosine transform (DCT2) and scalar quantization. For the Main profile, Improved Quantization and Transform (IQT) uses a different mapping/ clipping function for quantization. An inverse zig-zag scanning order is used for coefficient coding. Advanced Coefficient Coding (ADCC) in the Main profile can code coefficient values more efficiently, for example, indicated by the last non-zero coefficient. In Main profile, Adaptive Transformation Selection (ATS) is also available and can be applied to integer versions of DST7 or DCT8, and not just DCT2.

Entropy coding

[EVC] uses a similar binary arithmetic coding mechanism as [AVC]. The mechanism includes a binarization step and a probability update defined by a lookup table. In the Main profile, the derivation process of syntax elements based on adjacent blocks makes the context modeling and initialization process more efficient.

In-loop filtering

The Baseline profile of [EVC] uses the deblocking filter defined in H.263 Annex J. In the Main profile, compared to the deblocking filter in the Baseline profile, an Advanced Deblocking Filter (ADDB) can be used, which can further reduce artifacts. The Main profile also defines two additional in-loop filters that can be used to improve the quality of decoded pictures before output and/or for inter prediction. A Walsh-Hadamard Transform Domain Filter (HTDF) is applied to the luma samples before deblocking, and the scanning

process is used to determine 4 adjacent samples for filtering. An adaptive Loop Filter (ALF) allows to send signals of up to 25 different filters for the luma components, and the best filter can be selected through the classification process for each 4x4 block. The filter parameters of the ALF filter are signaled in the Adaptation Parameter Set (APS).

Inter-prediction

The basis of [EVC] inter prediction is motion compensation using interpolation filters with a quarter sample resolution. In Baseline profile, a motion vector signal is transmitted using one of three spatially neighboring motion vectors and a temporally collocated motion vector as a predictor. The motion vector difference may be signaled relative to the selected predictor, but for the case where no motion vector difference is signaled and there is no remaining data in the block, there is a specific mode called a skip mode. The Main profile includes six additional tools to provide improved inter prediction. With advanced Motion Interpolation and Signaling (AMIS), adjacent blocks can be conceptually merged to indicate that they use the same motion, but more advanced schemes can also be used to create predictions from the basic model list of candidate predictors. The Merge with Motion Vector Difference (MMVD) tool uses a process similar to the concept of merging neighboring blocks, but also allows the use of expressions that include a starting point, motion amplitude, and direction of motion to send a motion vector signal.

Using Advanced Motion Vector Prediction (AMVP), candidate motion vector predictions for the block can be derived from its neighboring blocks in the same picture and collocated blocks in the reference picture. The Adaptive Motion Vector Resolution (AMVR) tool provides a way to reduce the accuracy of a motion vector from a quarter sample to half sample, full sample, double sample, or quad sample, which provides the efficiency advantage, such as when sending large motion vector differences. The Main profile also includes the Decoder-side Motion Vector Refinement (DMVR), which uses a bilateral template matching process to refine the motion vectors in a bidirectional fashion.

Intra prediction and intra-coding

Intra prediction in [EVC] is performed on adjacent samples of coding units in a partitioned structure. For the Baseline profile, all coding units are square, and there are five different prediction modes: DC (mean value of the neighborhood), horizontal, vertical, and two different diagonal directions. In the Main profile, intra prediction can be applied to any rectangular coding unit, and there are 28 additional direction modes available in the so-called Enhanced

Intra Prediction Directions (EIPD). In the Main profile, an encoder can also use Intra Block Copy (IBC), where a previously decoded sample blocks of the same picture is used as a predictor. A displacement vector in integer sample precision is signaled to indicate where the prediction block in the current picture is used for this mode.

Decoded picture buffer management

In [EVC], decoded pictures can be stored in a decoded picture buffer (DPB) for predicting pictures that follow them in decoding order. In the Baseline profile, the management of the DPB (i.e. the process of adding and deleting reference pictures) is controlled by the information in the SPS. For the Main profile, if a Reference Picture List (RPL) scheme is used, DPB management can be controlled by information that is signaled at the picture level.

### 1.1.2. Systems and Transport Interfaces

[EVC] inherited the basic systems and transport interfaces designs from [AVC] and [HEVC]. These include the NAL-unit-based syntax structure, the hierarchical syntax and data unit structure and the Supplemental Enhancement Information (SEI) message mechanism. The hierarchical syntax and data unit structure consists of a sequence-level parameter set (SPS), two picture-level parameter sets (PPS and APS, each of which can apply to one or more pictures), slice-level header parameters, and lower-level parameters.

A number of key components that influenced the Network Abstraction Layer design of [EVC] as well as this memo are described below

Sequence parameter set

The Sequence Parameter Set (SPS) contains syntax elements pertaining to a coded video sequence (CVS), which is a group of pictures, starting with a random access point, and followed by pictures that may depend on each other and the random access point picture. In MPGEG-2, the equivalent of a CVS was a Group of Pictures (GOP), which normally started with an I frame and was followed by P and B frames. While more complex in its options of random access points, EVC retains this basic concept. In many TV-like applications, a CVS contains a few hundred milliseconds to a few seconds of video. In video conferencing (without switching MCUs involved), a CVS can be as long in duration as the whole session.

Picture and adaptation parameter set

The Picture Parameter Set and the Adaptation Parameter Set (PPS and APS, respectively) carry information pertaining to a single picture. The PPS contains information that is likely to stay constant from picture to picture-at least for pictures for a certain type-whereas the APS contains information, such as adaptive loop filter coefficients, that are likely to change from picture to picture.

Profile, level and toolsets

Profiles and levels follow the same design considerations ask known form [AVC], [HEVC], and in fact video codecs as old as MPEG-1 visual. A profile defines a set of tools (not to confuse with the "toolset" discussed below) that a decoder compliant with this profile has to support. In [EVC], profiles are defined in Annex A. Formally, they are defined as a set of constraints that a bitstream needs to conform to. In [EVC], the Baseline profile is much more severely constraint than Main profile, reducing implementation complexity. Levels relate to bitstream complexity in dimensions such as maximum sample decoding rate, maximum picture size, and similar parameters that are directly related to computational complexity.

Profiles and levels are signaled in the highest parameter set available, the SPS.

[EVC] contains another mechanism related to the use of coding tools, known as the toolset syntax element. This syntax element, toolset_idc_h and toolset_idc_l located in the SPS, is a bitmask that allows encoders to indicate which coding tools they are using, within the menu of profiles offered by the profile that is also signaled. No decoder conformance point is associated with the toolset, but a bitstream that were using a coding tool that is indicated as not used in the toolset syntax element would obviously be non-compliant. While MPEG specifically rules out the use of the toolset syntax element as a conformance point, walled garden implementations could do so without incurring the interoperability problems MPEG fears, and create bitstreams and decoders that do not support one or more given tools. That, in turn, may be useful to mitigate certain patent related risks.

Bitstream and elementary stream

Above the Coded Video Sequence (CVS), [EVC] defines a video bitstream that can be used in the MPEG systems context as an elementary stream. For the purpose of this memo, this is not relevant.

Random access support

[EVC] supports random access mechanism solely based on IDR access unit.

Temporal scalability support

[EVC] includes support for temporal scalability through the generalized reference picture selection approach known since [AVC]/SVC. Up to six temporal layers are supported. The temporal layer is signaled in the NAL unit header (which co-serves as the payload header in this memo), in the nuh_temporal_id field.

Reference picture management

placeholder

SEI Message

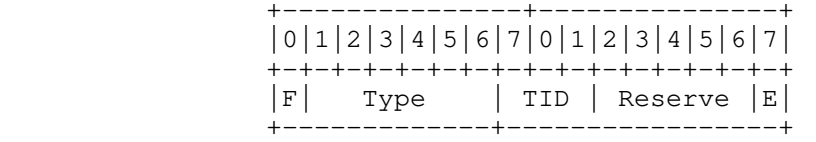
[EVC] inherits many of [HEVC]'s SEI Messages, occasionally with changes in syntax and/or semantics making them applicable to EVC.

### 1.1.3. Parallel Processing Support (informative)

Placeholder

### 1.1.4. NAL Unit Header

[EVC] maintains the NAL unit concept of [HEVC] with different parameter options. EVC also uses a two-byte NAL unit header, as shown in Figure 1. The payload of a NAL unit refers to the NAL unit excluding the NAL unit header.




The Structure of the EVC NAL Unit Header

Figure 1

The semantics of the fields in the NAL unit header are as specified in [EVC] and described briefly below for convenience. In addition to the name and size of each field, the corresponding syntax element name in [EVC] is also provided.

F: 1 bit

forbidden_zero_bit. Required to be zero in [EVC]. Note that the inclusion of this bit in the NAL unit header was included to enable transport of EVC video over MPEG-2 transport systems (avoidance of start code emulations) [MPEG2S]. In the context of this memo,the value 1 may be used to indicate a syntax violation, e.g., for a NAL unit resulted from aggregating a number of fragmented units of a NAL unit but missing the last fragment, as described in Section xxx. (section # placeholder)

Type: 6 bits

nal_unit_type_plus1. This field specifies the NAL unit type as defined in Table 4 of [EVC]. If the value of this field is less than and equal to 23, the NAL unit is a VCL NAL unit. Otherwise, the NAL unit is a non-VCL NAL unit. For a reference of all currently defined NAL unit types and their semantics, please refer to Section 7.4.2.2 in [EVC].

TID: 3 bits

nuh_temporal_id. This field specifies the temporal identifier of the NAL unit. The value of TemporalId is equal to TID. TemporalId shall be equal to 0 if it is a IDR NAL unit type (NAL unit type 1).

Reserve: 5 bits

nuh_reserved_zero_5bits. This field shall be equal to the version of the [EVC] specification. Values of nuh_reserved_zero_5bits greater than 0 are reserved for future use by ISO/IEC. Decoders conforming to a profile specified in [EVC] Annex A shall ignore (i.e., remove from the bitstream and discard) all NAL units with values of nuh_reserved_zero_5bits greater than 0.

E: 1 bit

nuh_extension_flag. This field shall be equal the version of the [EVC] specification. Value of nuh_extesion_flag equal to 1 is reserved for future use by ISO/IEC. Decoders conforming to a profile specified in Annex A shall ignore (i.e., remove from the bitstream and discard) all NAL units with values of nuh_extension_flag equal to 1.

## 1.2. Overview of the Payload Format

This payload format defines the following processes required for transport of [EVC] coded data over RTP [RFC3550]:

* Usage of RTP header with this payload format

* Packetization of [EVC] coded NAL units into RTP packets using three types of payload structures: a single NAL unit, aggregation, and fragment unit packet

* Transmission of [EVC] NAL units of the same bitstream within a single RTP stream.

* Media type parameters to be used with the Session Description Protocol (SDP) [RFC4566]

## 2. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown above.

## 3. Definitions and Abbreviations

### 3.1. Definitions

This document uses the terms and definitions of EVC. Section 3.1.1 lists relevant definitions from [EVC] for convenience. Section 3.1.2 provides definitions specific to this memo.

#### 3.1.1. Definitions from the EVC Specification

Access Unit: A set of NAL units that are associated with each other according to a specified classification rule, are consecutive in decoding order, and contain exactly one coded picture.

Bitstream: A sequence of bits, in the form of a NAL unit stream or a byte stream, that forms the representation of coded pictures and associated data forming one or more coded video sequences (CVSs).

Coded Picture: A coded representation of a picture containing all CTUs of the picture.

Coded Video Sequence (CVS): A sequence of access units that consists, in decoding order, of an IDR access unit, followed by zero or more access units that are not IDR access units, including all subsequent access units up to but not including any subsequent access unit that is an IDR access unit.

Coding Tree Block (CTB): An NxN block of samples for some value of N such that the division of a component into CTBs is a partitioning.

Coding Tree Unit (CTU): A CTB of luma samples, two corresponding CTBs of chroma samples of a picture that has three sample arrays, or a CTB of samples of a monochrome picture or a picture that is coded using three separate colour planes and syntax structures used to code the samples.

Decoded Picture: A decoded picture is derived by decoding a coded picture.

Decoded Picture Buffer (DPB): A buffer holding decoded pictures for reference, output reordering, or output delay specified for the hypothetical reference decoder in Annex C of [EVC] specification.

Dynamic Range Adjustment (DRA): A mapping process that is applied to decoded picture prior to cropping and output as part of the decoding process and is controlled by parameters conveyed in an Adaptation Parameter Set (APS).

Hypothetical Reference Decoder (HRD): A hypothetical decoder model that specifies constraints on the variability of conforming NAL unit streams or conforming byte streams that an encoding process may produce.

Instantaneous Decoding Refresh (IDR) access unit: An access unit in which the coded picture is an IDR picture.

Instantaneous Decoding Refresh (IDR) picture: A coded picture for which each VCL NAL unit has NalUnitType equal to IDR_NUT.

Level: A defined set of constraints on the values that may be taken by the syntax elements and variables of this document, or the value of a transform coefficient prior to scaling.

Network Abstraction Layer (NAL) unit: A syntax structure containing an indication of the type of data to follow and bytes containing that data in the form of an RBSP interspersed as necessary.

Network Abstraction Layer (NAL) Unit Stream: A sequence of NAL units.

Non-IDR Picture: A coded picture that is not an IDR picture.

Non-VCL NAL Unit: A NAL unit that is not a VCL NAL unit.

Picture Parameter Set (PPS): A syntax structure containing syntax elements that apply to zero or more entire coded pictures as determined by a syntax element found in each slice header.

Picture Order Count (POC): A variable that is associated with each picture, uniquely identifies the associated picture among all pictures in the CVS, and, when the associated picture is to be output from the decoded picture buffer, indicates the position of the associated picture in output order relative to the output order positions of the other pictures in the same CVS that are to be output from the decoded picture buffer.

Raw Byte Sequence Payload (RBSP): A syntax structure containing an integer number of bytes that is encapsulated in a NAL unit and that is either empty or has the form of a string of data bits containing syntax elements followed by an RBSP stop bit and zero or more subsequent bits equal to 0.

Sequence Parameter Set (SPS): A syntax structure containing syntax elements that apply to zero or more entire CVSs as determined by the content of a syntax element found in the PPS referred to by a syntax element found in each slice header.

Tile row: A rectangular region of CTUs having a height specified by syntax elements in the PPS and a width equal to the width of the picture.

Tile scan: A specific sequential ordering of CTUs partitioning a picture in which the CTUs are ordered consecutively in CTU raster scan in a tile whereas tiles in a picture are ordered consecutively in a raster scan of the tiles of the picture.

Video coding layer (VCL) NAL unit: A collective term for coded slice NAL units and the subset of NAL units that have reserved values of NalUnitType that are classified as VCL NAL units in this document.

### 3.1.2. Definitions Specific to This Memo

Media-Aware Network Element (MANE): A network element, such as a middlebox, selective forwarding unit, or application-layer gateway that is capable of parsing certain aspects of the RTP payload headers or the RTP payload and reacting to their contents.

> Informative note: The concept of a MANE goes beyond normal routers or gateways in that a MANE has to be aware of the signaling (e.g., to learn about the payload type mappings of the media streams), and in that it has to be trusted when working with Secure RTP (SRTP). The advantage of using MANEs is that they allow packets

to be dropped according to the needs of the media coding. For example, if a MANE has to drop packets due to congestion on a certain link, it can identify and remove those packets whose elimination produces the least adverse effect on the user experience. After dropping packets, MANEs must rewrite RTCP packets to match the changes to the RTP stream, as specified in Section 7 of [RFC3550].

NAL unit decoding order: A NAL unit order that conforms to the constraints on NAL unit order given in Section 8.2 and 8.3 in [EVC], follow the Order of NAL units in the bitstream.

NAL unit output order: A NAL unit order in which NAL units of different access units are in the output order of the decoded pictures corresponding to the access units, as specified in [EVC], and in which NAL units within an access unit are in their decoding order.

RTP stream: See [RFC7656]. Within the scope of this memo, one RTP stream is utilized to transport one or more temporal sub-layers.

Transmission order: The order of packets in ascending RTP sequence number order (in modulo arithmetic). Within an aggregation packet, the NAL unit transmission order is the same as the order of appearance of NAL units in the packet.

## 3.2. Abbreviations

| | |
|---|---|
| APS | Adaptation Parameter Set |
| ATS | Adaptive Transform Selection |
| B | Bi-predictive |
| CBR | Constant Bit Rate |
| CPB | Coded Picture Buffer |
| CTB | Coding Tree Block |
| CTU | Coding Tree Unit |
| CVS | Coded Video Sequence |
| DPB | Decoded Picture Buffer |
| HRD | Hypothetical Reference Decoder |

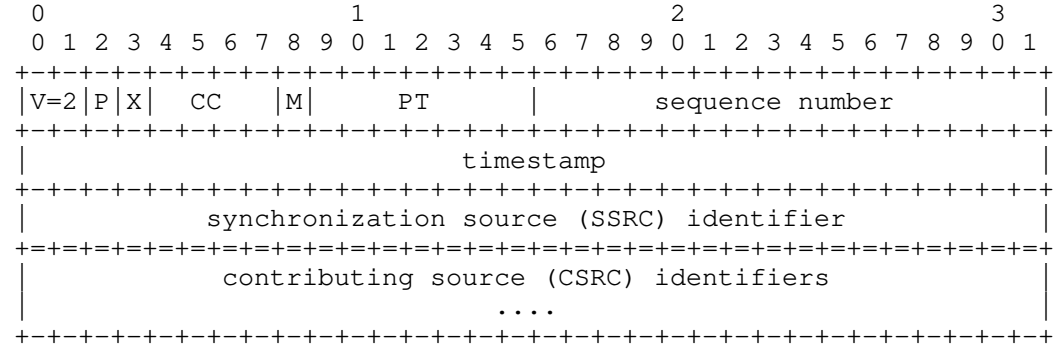
| | |
|---|---|
| HSS | Hypothetical Stream Scheduler |
| I | Intra |
| IDR | Instantaneous Decoding Refresh |
| LSB | Least Significant Bit |
| LTRP | Long-Term Reference Picture |
| MMVD | Merge with Motion Vector Difference |
| MSB | Most Significant Bit |
| NAL | Network Abstraction Layer |
| P | Predictive |
| POC | Picture Order Count |
| PPS | Picture Parameter Set |
| QP | Quantization Parameter |
| RBSP | Raw Byte Sequence Payload |
| RGB | Same as GBR |
| SAR | Sample Aspect Ratio |
| SEI | Supplemental Enhancement Information |
| SODB | String Of Data Bits |
| SPS | Sequence Parameter Set |
| STRP | Short-Term Reference Picture |
| VBR | Variable Bit Rate |
| VCL | Video Coding Layer |

# 4. RTP Payload Format

### 4.1. RTP Header Usage

The format of the RTP header is specified in [RFC3550] (reprinted as Figure 2 for convenience). This payload format uses the fields of the header in a manner consistent with that specification.

The RTP payload (and the settings for some RTP header bits) for aggregation packets and fragmentation units are specified in Section 4.3.2 and Section 4.3.3, respectively.




RTP Header According to {{RFC3550}}

Figure 2

The RTP header information to be set according to this RTP payload format is set as follows:

Marker bit (M): 1 bit

Set for the last packet of the access unit, carried in the current RTP stream. This is in line with the normal use of the M bit in video formats to allow an efficient playout buffer handling.

editor-note 4: The informative note below needs updating once the NAL unit type table is stable in the [EVC] spec.

Informative note: The content of a NAL unit does not tell whether or not the NAL unit is the last NAL unit, in decoding order, of an access unit. An RTP sender implementation may obtain this information from the video encoder. If, however, the implementation cannot obtain this information directly from the encoder, e.g., when the bitstream was pre-encoded, and also there is no timestamp allocated for each NAL unit, then the sender implementation can inspect subsequent NAL units in decoding order to determine whether or not the NAL unit is the

last NAL unit of an access unit as follows. A NAL unit is determined to be the last NAL unit of an access unit if it is the last NAL unit of the bitstream. A NAL unit naluX is also determined to be the last NAL unit of an access unit if both the following conditions are true: 1) the next VCL NAL unit naluY in decoding order has the high-order bit of the first byte after its NAL unit header equal to 1 or nal_unit_type equal to 27, and 2) all NAL units between naluX and naluY, when present, have nal_unit_type in the range of 24 to 26, inclusive, equal to 28 or in the range of 29 to 55.

Payload Type (PT): 7 bits

The assignment of an RTP payload type for this new payload format is outside the scope of this document and will not be specified here. The assignment of a payload type has to be performed either through the profile used or in a dynamic way.

Sequence Number (SN): 16 bits

Set and used in accordance with [RFC3550].

Timestamp: 32 bits

The RTP timestamp is set to the sampling timestamp of the content. A 90 kHz clock rate MUST be used. If the NAL unit has no timing properties of its own (e.g., parameter sets or certain SEI NAL units), the RTP timestamp MUST be set to the RTP timestamp of the coded picture of the access unit in which the NAL unit (according to Annex D of [EVC]) is included. Receivers MUST use the RTP timestamp for the display process, even when the bitstream contains picture timing SEI messages or decoding unit information SEI messages as specified in [EVC].

Synchronization source (SSRC): 32 bits

Used to identify the source of the RTP packets. When using SRST, by definition a single SSRC is used for all parts of a single bitstream.

## 4.2. Payload Header Usage

The first two bytes of the payload of an RTP packet are referred to as the payload header. The payload header consists of the same fields (F, TID, Reserve and E) as the NAL unit header as shown in Section 1.1.4, irrespective of the type of the payload structure.

The TID value indicates (among other things) the relative importance of an RTP packet, for example, because NAL units belonging to higher temporal sub-layers are not used for the decoding of lower temporal sub-layers. A lower value of TID indicates a higher importance. More-important NAL units MAY be better protected against transmission losses than less-important NAL units.

### 4.3. Payload Structures

Three different types of RTP packet payload structures are specified. A receiver can identify the type of an RTP packet payload through the Type field in the payload header.

The Three different payload structures are as follows:

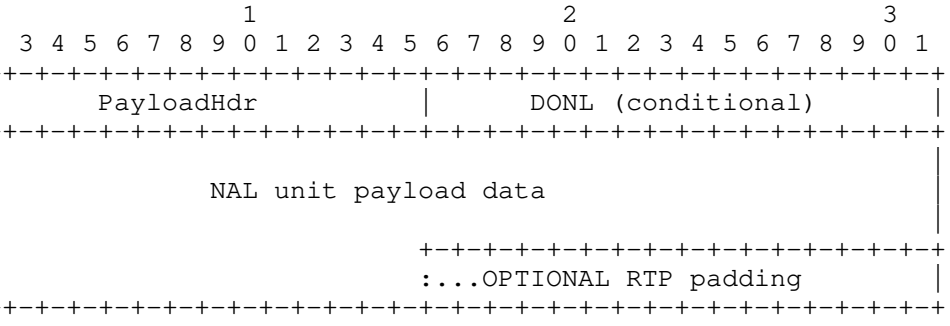
* Single NAL unit packet: Contains a single NAL unit in the payload, and the NAL unit header of the NAL unit also serves as the payload header. This payload structure is specified in Section 4.3.1.

* Aggregation Packet (AP): Contains more than one NAL unit within one access unit. This payload structure is specified in Section 4.3.2.

* Fragmentation Unit (FU): Contains a subset of a single NAL unit. This payload structure is specified in Section 4.3.3.

#### 4.3.1. Single NAL Unit Packets

A single NAL unit packet contains exactly one NAL unit, and consists of a payload header (denoted as PayloadHdr), a conditional 16-bit DONL field (in network byte order), and the NAL unit payload data (the NAL unit excluding its NAL unit header) of the contained NAL unit, as shown in Figure 3.




The Structure of a Single NAL Unit Packet

Figure 3

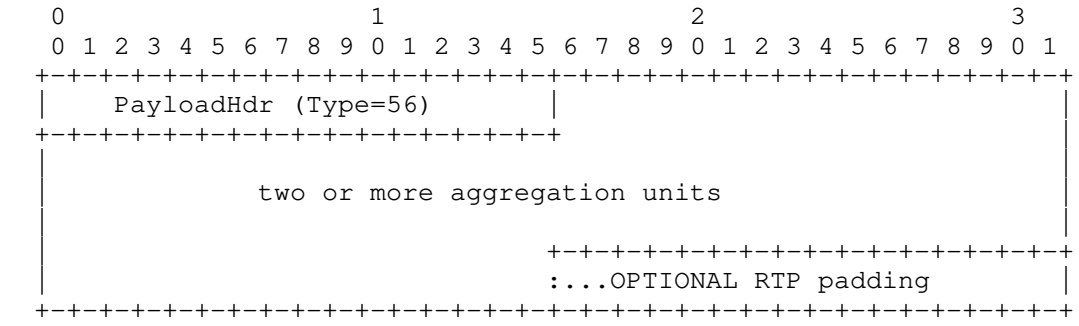
The DONL field, when present, specifies the value of the 16 least significant bits of the decoding order number of the contained NAL unit. If sprop-max-don-diff is greater than 0 for any of the RTP streams, the DONL field MUST be present, and the variable DON for the contained NAL unit is derived as equal to the value of the DONL field. Otherwise (sprop-max-don-diff is equal to 0 for all the RTP streams), the DONL field MUST NOT be present.

### 4.3.2. Aggregation Packets (APs)

Aggregation Packets (APs) enable the reduction of packetization overhead for small NAL units, such as most of the non-VCL NAL units, which are often only a few octets in size.

An AP aggregates NAL units within one access unit. Each NAL unit to be carried in an AP is encapsulated in an aggregation unit. NAL units aggregated in one AP are in NAL unit decoding order.

An AP consists of a payload header (denoted as PayloadHdr) followed by two or more aggregation units, as shown in Figure 4.




The Structure of an Aggregation Packet

Figure 4

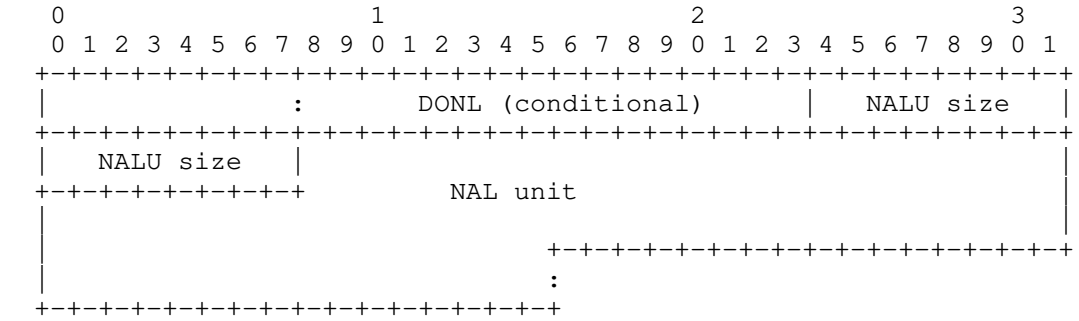
The fields in the payload header are set as follows. The F bit MUST be equal to 0 if the F bit of each aggregated NAL unit is equal to zero; otherwise, it MUST be equal to 1. The Type field MUST be equal to 56.

The value of TID MUST be the lowest value of TID of all the aggregated NAL units. The value of Reserve and E Must match the version of [EVC] specification.

Informative note: All VCL NAL units in an AP have the same TID value since they belong to the same access unit. However, an AP may contain non-VCL NAL units for which the TID value in the NAL unit header may be different than the TID value of the VCL NAL units in the same AP.

An AP MUST carry at least two aggregation units and can carry as many aggregation units as necessary; however, the total amount of data in an AP obviously MUST fit into an IP packet, and the size SHOULD be chosen so that the resulting IP packet is smaller than the path MTU size so to avoid IP layer fragmentation. An AP MUST NOT contain FUs specified in Section 4.3.3. APs MUST NOT be nested; i.e., an AP can not contain another AP.

The first aggregation unit in an AP consists of a conditional 16-bit DONL field (in network byte order) followed by a 16-bit unsigned size information (in network byte order) that indicates the size of the NAL unit in bytes (excluding these two octets, but including the NAL unit header), followed by the NAL unit itself, including its NAL unit header, as shown in Figure 5.



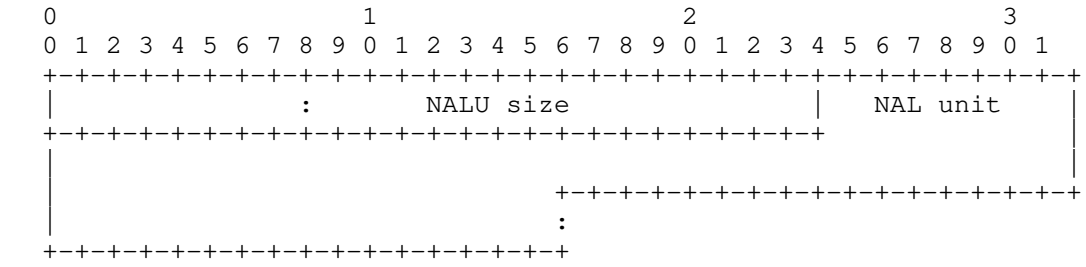

The Structure of the First Aggregation Unit in an AP

Figure 5

The DONL field, when present, specifies the value of the 16 least significant bits of the decoding order number of the aggregated NAL unit.

If sprop-max-don-diff is greater than 0 for any of the RTP streams, the DONL field MUST be present in an aggregation unit that is the first aggregation unit in an AP, and the variable DON for the aggregated NAL unit is derived as equal to the value of the DONL field. Otherwise (sprop-max-don-diff is equal to 0 for all the RTP streams), the DONL field MUST NOT be present in an aggregation unit that is the first aggregation unit in an AP.

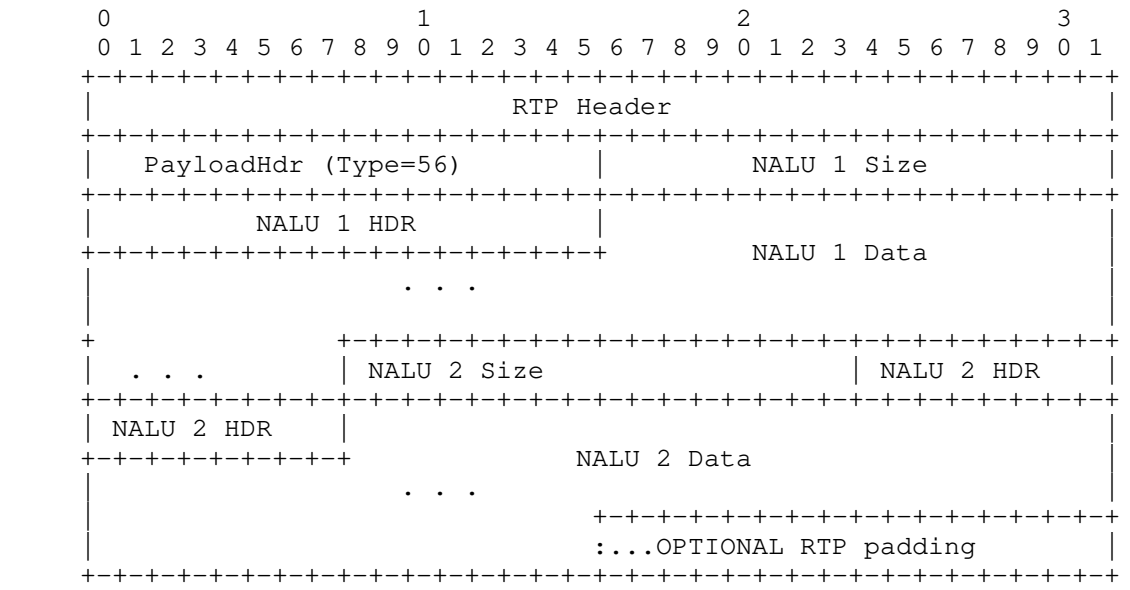
An aggregation unit that is not the first aggregation unit in an AP will be followed immediately by a 16-bit unsigned size information (in network byte order) that indicates the size of the NAL unit in bytes (excluding these two octets, but including the NAL unit header), followed by the NAL unit itself, including its NAL unit header, as shown in Figure 6.




The Structure of an Aggregation Unit That Is Not the First Aggregation Unit in an AP

Figure 6

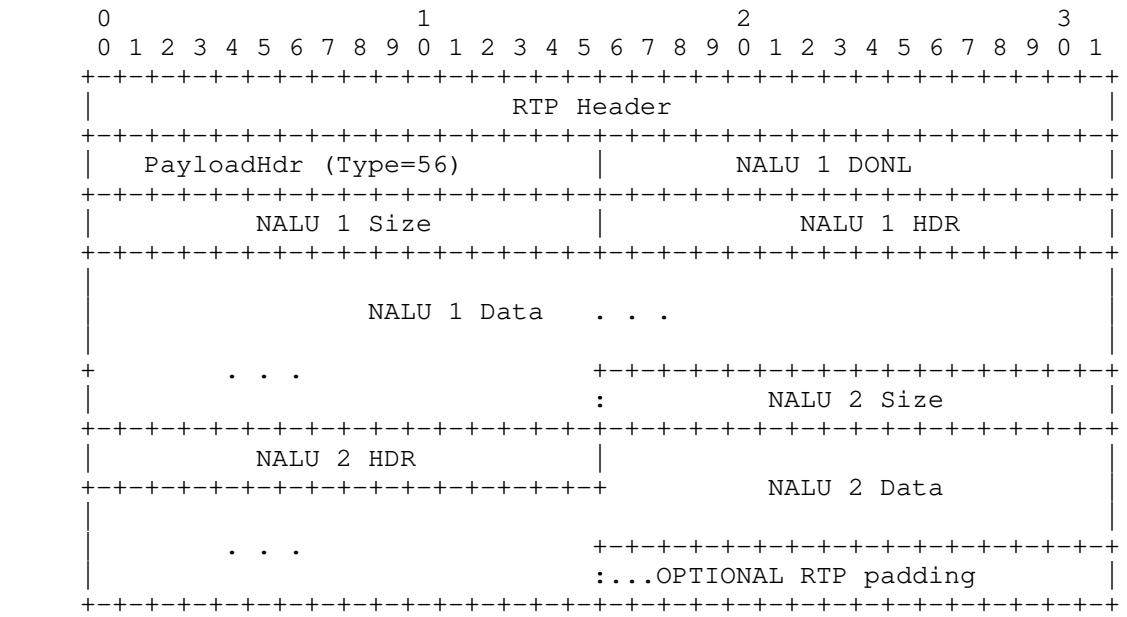
Figure 7 presents an example of an AP that contains two aggregation units, labeled as NALU 1 and NALU 2 in the figure, without the DONL field being present.




An Example of an AP Packet Containing
Two Aggregation Units without the DONL Field

Figure 7

Figure 8 presents an example of an AP that contains two aggregation units, labeled as NALU 1 and NALU 2 in the figure, with the DONL field being present.




An Example of an AP Containing
Two Aggregation Units with the DONL Field

Figure 8

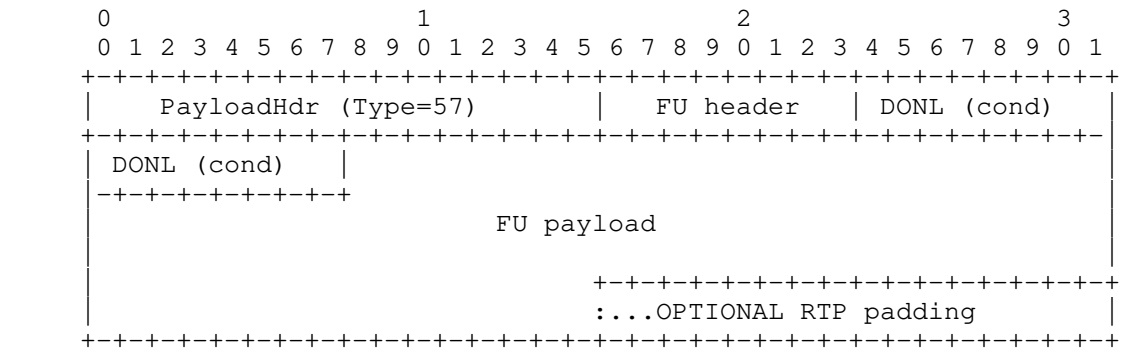
4.3.3. Fragmentation Units

Fragmentation Units (FUs) are introduced to enable fragmenting a single NAL unit into multiple RTP packets, possibly without cooperation or knowledge of the EVC [EVC] encoder. A fragment of a NAL unit consists of an integer number of consecutive octets of that NAL unit. Fragments of the same NAL unit MUST be sent in consecutive order with ascending RTP sequence numbers (with no other RTP packets within the same RTP stream being sent between the first and last fragment).

When a NAL unit is fragmented and conveyed within FUs, it is referred to as a fragmented NAL unit. APs MUST NOT be fragmented. FUs MUST NOT be nested; i.e., an FU must not contain a subset of another FU.

The RTP timestamp of an RTP packet carrying an FU is set to the NALU-time of the fragmented NAL unit.

An FU consists of a payload header (denoted as PayloadHdr), an FU header of one octet, a conditional 16-bit DONL field (in network byte order), and an FU payload, as shown in Figure 9.

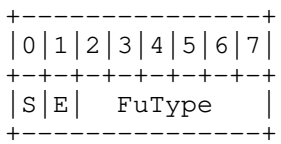



The Structure of an FU

Figure 9

The fields in the payload header are set as follows. The Type field MUST be equal to 57. The fields F, TID, Reserve and E MUST be equal to the fields F, TID, Reserve and E, respectively, of the fragmented NAL unit.

The FU header consists of an S bit, an E bit, and a 6-bit FuType field, as shown in Figure 10.




The Structure of FU Header

Figure 10

The semantics of the FU header fields are as follows:

S: 1 bit

When set to 1, the S bit indicates the start of a fragmented NAL unit, i.e., the first byte of the FU payload is also the first byte of the payload of the fragmented NAL unit. When the FU payload is not the start of the fragmented NAL unit payload, the S bit MUST be set to 0.

E: 1 bit

When set to 1, the E bit indicates the end of a fragmented NAL unit, i.e., the last byte of the payload is also the last byte of the fragmented NAL unit. When the FU payload is not the last fragment of a fragmented NAL unit, the E bit MUST be set to 0.

FuType: 6 bits

The field FuType MUST be equal to the field Type of the fragmented NAL unit.

The DONL field, when present, specifies the value of the 16 least significant bits of the decoding order number of the fragmented NAL unit.

If sprop-max-don-diff is greater than 0 for any of the RTP streams, and the S bit is equal to 1, the DONL field MUST be present in the FU, and the variable DON for the fragmented NAL unit is derived as equal to the value of the DONL field. Otherwise (sprop-max-don-diff is equal to 0 for all the RTP streams, or the S bit is equal to 0), the DONL field MUST NOT be present in the FU.

A non-fragmented NAL unit MUST NOT be transmitted in one FU; i.e., the Start bit and End bit must not both be set to 1 in the same FU header.

The FU payload consists of fragments of the payload of the fragmented NAL unit so that if the FU payloads of consecutive FUs, starting with an FU with the S bit equal to 1 and ending with an FU with the E bit equal to 1, are sequentially concatenated, the payload of the fragmented NAL unit can be reconstructed. The NAL unit header of the fragmented NAL unit is not included as such in the FU payload, but rather the information of the NAL unit header of the fragmented NAL unit is conveyed in F, TID, Reserve and E fields of the FU payload headers of the FUs and the FuType field of the FU header of the FUs. An FU payload MUST NOT be empty.

If an FU is lost, the receiver SHOULD discard all following fragmentation units in transmission order corresponding to the same fragmented NAL unit, unless the decoder in the receiver is known to gracefully handle incomplete NAL units.

A receiver in an endpoint or in a MANE MAY aggregate the first n-1 fragments of a NAL unit to an (incomplete) NAL unit, even if fragment n of that NAL unit is not received. In this case, the forbidden_zero_bit of the NAL unit MUST be set to 1 to indicate a syntax violation.

### 4.4. Decoding Order Number

For each NAL unit, the variable AbsDon is derived, representing the decoding order number that is indicative of the NAL unit decoding order.

Let NAL unit n be the n-th NAL unit in transmission order within an RTP stream.

If sprop-max-don-diff is equal to 0 for all the RTP streams carrying the HEVC bitstream, AbsDon[n], the value of AbsDon for NAL unit n, is derived as equal to n.

Otherwise (sprop-max-don-diff is greater than 0 for any of the RTP streams), AbsDon[n] is derived as follows, where DON[n] is the value of the variable DON for NAL unit n:

* If n is equal to 0 (i.e., NAL unit n is the very first NAL unit in transmission order), AbsDon[0] is set equal to DON[0].

* Otherwise (n is greater than 0), the following applies for derivation of AbsDon[n]:

```
If DON[n] == DON[n-1],
   AbsDon[n] = AbsDon[n-1]

If (DON[n] > DON[n-1] and DON[n] - DON[n-1] < 32768),
   AbsDon[n] = AbsDon[n-1] + DON[n] - DON[n-1]

If (DON[n] < DON[n-1] and DON[n-1] - DON[n] >= 32768),
   AbsDon[n] = AbsDon[n-1] + 65536 - DON[n-1] + DON[n]

If (DON[n] > DON[n-1] and DON[n] - DON[n-1] >= 32768),
   AbsDon[n] = AbsDon[n-1] - (DON[n-1] + 65536 -
   DON[n])

If (DON[n] < DON[n-1] and DON[n-1] - DON[n] < 32768),
   AbsDon[n] = AbsDon[n-1] - (DON[n-1] - DON[n])
```

For any two NAL units m and n, the following applies:

* AbsDon[n] greater than AbsDon[m] indicates that NAL unit n follows NAL unit m in NAL unit decoding order.

* When AbsDon[n] is equal to AbsDon[m], the NAL unit decoding order of the two NAL units can be in either order.

* AbsDon[n] less than AbsDon[m] indicates that NAL unit n precedes NAL unit m in decoding order.

  Informative note: When two consecutive NAL units in the NAL unit decoding order have different values of AbsDon, the absolute difference between the two AbsDon values may be greater than or equal to 1.

  Informative note: There are multiple reasons to allow for the absolute difference of the values of AbsDon for two consecutive NAL units in the NAL unit decoding order to be greater than one. An increment by one is not required, as at the time of associating values of AbsDon to NAL units, it may not be known whether all NAL units are to be delivered to the receiver. For example, a gateway might not forward VCL NAL units of higher sub-layers or some SEI NAL units when there is congestion in the network. In another example, the first intra-coded picture of a pre-encoded clip is transmitted in advance to ensure that it is readily available in the receiver, and when transmitting the first intra-coded picture, the originator does not exactly know how many NAL units will be encoded before the first intra-coded picture of the pre-encoded clip follows in decoding order. Thus, the values of AbsDon for the NAL units of the first intra-coded picture of the pre-encoded clip have to be estimated when they are transmitted, and gaps in values of AbsDon may occur.

## 5. Packetization Rules

The following packetization rules apply:

* If sprop-max-don-diff is greater than 0 for any of the RTP streams, the transmission order of NAL units carried in the RTP stream MAY be different than the NAL unit decoding order and the NAL unit output order.

* A NAL unit of a small size SHOULD be encapsulated in an aggregation packet together with one or more other NAL units in order to avoid unnecessary packetization overhead for small NAL units. For example, non-VCL NAL units such as access unit delimiters, parameter sets, or SEI NAL units are typically small and can often be aggregated with VCL NAL units without violating MTU size constraints.

* Each non-VCL NAL unit SHOULD, when possible from an MTU size match viewpoint, be encapsulated in an aggregation packet together with its associated VCL NAL unit, as typically a non-VCL NAL unit would be meaningless without the associated VCL NAL unit being available.

* For carrying exactly one NAL unit in an RTP packet, a single NAL unit packet MUST be used.

## 6. De-packetization Process

The general concept behind de-packetization is to get the NAL units out of the RTP packets in an RTP stream and pass them to the decoder in the NAL unit decoding order.

The de-packetization process is implementation dependent. Therefore, the following description should be seen as an example of a suitable implementation. Other schemes may be used as well, as long as the output for the same input is the same as the process described below. The output is the same when the set of output NAL units and their order are both identical. Optimizations relative to the described algorithms are possible.

All normal RTP mechanisms related to buffer management apply. In particular, duplicated or outdated RTP packets (as indicated by the RTP sequences number and the RTP timestamp) are removed. To determine the exact time for decoding, factors such as a possible intentional delay to allow for proper inter-stream synchronization must be factored in.

NAL units with NAL unit type values in the range of 0 to 55, inclusive, may be passed to the decoder. NAL-unit-like structures with NAL unit type values in the range of 56 to 63, inclusive, MUST NOT be passed to the decoder.

The receiver includes a receiver buffer, which is used to compensate for transmission delay jitter within individual RTP streams and across RTP streams, to reorder NAL units from transmission order to the NAL unit decoding order. In this section, the receiver operation is described under the assumption that there is no transmission delay jitter within an RTP stream. To make a difference from a practical receiver buffer that is also used for compensation of transmission delay jitter, the receiver buffer is hereafter called the de-packetization buffer in this section. Receivers should also prepare for transmission delay jitter; that is, either reserve separate buffers for transmission delay jitter buffering and de-packetization buffering or use a receiver buffer for both transmission delay jitter and de-packetization. Moreover, receivers should take transmission

delay jitter into account in the buffering operation, e.g., by additional initial buffering before starting of decoding and playback.

When sprop-max-don-diff is equal to 0 for the received RTP stream, the de-packetization buffer size is zero bytes, and the process described in the remainder of this paragraph applies. The NAL units carried in the RTP stream are directly passed to the decoder in their transmission order, which is identical to their decoding order. When there are several NAL units of the same RTP stream with the same NTP timestamp, the order to pass them to the decoder is their transmission order.

> Informative note: The mapping between RTP and NTP timestamps is conveyed in RTCP SR packets. In addition, the mechanisms for faster media timestamp synchronization discussed in [RFC6051] may be used to speed up the acquisition of the RTP-to-wall-clock mapping.

When sprop-max-don-diff is greater than 0 for the received RTP stream the process described in the remainder of this section applies.

There are two buffering states in the receiver: initial buffering and buffering while playing. Initial buffering starts when the reception is initialized. After initial buffering, decoding and playback are started, and the buffering-while-playing mode is used.

Regardless of the buffering state, the receiver stores incoming NAL units, in reception order, into the de-packetization buffer. NAL units carried in RTP packets are stored in the de-packetization buffer individually, and the value of AbsDon is calculated and stored for each NAL unit.

Initial buffering lasts until condition A (the difference between the greatest and smallest AbsDon values of the NAL units in the de-packetization buffer is greater than or equal to the value of sprop-max-don-diff) or condition B (the number of NAL units in the de-packetization buffer is greater than the value of sprop-depack-buf-nalus) is true.

After initial buffering, whenever condition A or condition B is true, the following operation is repeatedly applied until both condition A and condition B become false:

* The NAL unit in the de-packetization buffer with the smallest value of AbsDon is removed from the de-packetization buffer and passed to the decoder.

When no more NAL units are flowing into the de-packetization buffer, all NAL units remaining in the de-packetization buffer are removed from the buffer and passed to the decoder in the order of increasing AbsDon values.

## 7. Payload Format Parameters

This section specifies the optional parameters. A mapping of the parameters with Session Description Protocol (SDP) [RFC4556] is also provided for applications that use SDP.

### 7.1. Media Type Registration

The receiver MUST ignore any parameter unspecified in this memo.

Type name: video

Subtype name: evc

Required parameters: none

Optional parameters:

editor-note 5: To be updated

### 7.2. SDP Parameters

The receiver MUST ignore any parameter unspecified in this memo.

#### 7.2.1. Mapping of Payload Type Parameters to SDP

The media type video/evc string is mapped to fields in the Session Description Protocol (SDP) [RFC4566] as follows:

* The media name in the "m=" line of SDP MUST be video.
* The encoding name in the "a=rtpmap" line of SDP MUST be evc (the media subtype).
* The clock rate in the "a=rtpmap" line MUST be 90000.
* OPTIONAL PARAMETERS:

  editor-note 6: To be updated

#### 7.2.2. Usage with SDP Offer/Answer Model

When [EVC] is offered over RTP using SDP in an offer/answer model [RFC3264] for negotiation for unicast usage, the following limitations and rules apply:

editor-note 7: to be updated

#### 7.2.3. SDP Example

editor-note 8: to be updated

## 8. Use with Feedback Messages

Placeholder

### 8.1. Picture Loss Indication (PLI)

Placeholder

### 8.2. Full Intra Request (FIR)

Placeholder

## 9. Security Considerations

The scope of this Security Considerations section is limited to the payload format itself and to one feature of [EVC] that may pose a particularly serious security risk if implemented naively. The payload format, in isolation, does not form a complete system. Implementers are advised to read and understand relevant security-related documents, especially those pertaining to RTP (see the Security Considerations section in [RFC3550] ), and the security of the call-control stack chosen (that may make use of the media type registration of this memo). Implementers should also consider known security vulnerabilities of video coding and decoding implementations in general and avoid those.

Within this RTP payload format, neither the various media-plane-based mechanisms, nor the signaling part of this memo, seems to pose a security risk beyond those common to all RTP-based systems.

RTP packets using the payload format defined in this specification are subject to the security considerations discussed in the RTP specification [RFC3550], and in any applicable RTP profile such as RTP/AVP [RFC3551], RTP/AVPF [RFC4585], RTP/SAVP [RFC3711], or RTP/SAVPF [RFC5124]. However, as "Securing the RTP Framework: Why RTP Does Not Mandate a Single Media Security Solution" [RFC7202]

discusses, it is not an RTP payload format's responsibility to discuss or mandate what solutions are used to meet the basic security goals like confidentiality, integrity and source authenticity for RTP in general. This responsibility lays on anyone using RTP in an application. They can find guidance on available security mechanisms and important considerations in "Options for Securing RTP Sessions" [RFC7201]. Applications SHOULD use one or more appropriate strong security mechanisms. The rest of this section discusses the security impacting properties of the payload format itself.

Because the data compression used with this payload format is applied end-to-end, any encryption needs to be performed after compression. A potential denial-of-service threat exists for data encodings using compression techniques that have non-uniform receiver-end computational load. The attacker can inject pathological datagrams into the bitstream that are complex to decode and that cause the receiver to be overloaded. EVC is particularly vulnerable to such attacks, as it is extremely simple to generate datagrams containing NAL units that affect the decoding process of many future NAL units. Therefore, the usage of data origin authentication and data integrity protection of at least the RTP packet is RECOMMENDED, for example, with SRTP [RFC3711].

End-to-end security with authentication, integrity, or confidentiality protection will prevent a MANE from performing media-aware operations other than discarding complete packets. In the case of confidentiality protection, it will even be prevented from discarding packets in a media-aware way. To be allowed to perform such operations, a MANE is required to be a trusted entity that is included in the security context establishment.

## 10. Congestion Control

Congestion control for RTP SHALL be used in accordance with RTP [RFC3550] and with any applicable RTP profile, e.g., AVP [RFC3551]. If best-effort service is being used, an additional requirement is that users of this payload format MUST monitor packet loss to ensure that the packet loss rate is within an acceptable range. Packet loss is considered acceptable if a TCP flow across the same network path, and experiencing the same network conditions, would achieve an average throughput, measured on a reasonable timescale, that is not less than all RTP streams combined is achieving. This condition can be satisfied by implementing congestion-control mechanisms to adapt the transmission rate, the number of layers subscribed for a layered multicast session, or by arranging for a receiver to leave the session if the loss rate is unacceptably high.

The bitrate adaptation necessary for obeying the congestion control principle is easily achievable when real-time encoding is used, for example, by adequately tuning the quantization parameter. However, when pre-encoded content is being transmitted, bandwidth adaptation requires the pre-coded bitstream to be tailored for such adaptivity. The key mechanism available in [EVC] is temporal scalability. A media sender can remove NAL units belonging to higher temporal sub-layers (i.e., those NAL. units with a high value of TID) until the sending bitrate drops to an acceptable range.

The mechanisms mentioned above generally work within a defined profile and level and, therefore, no renegotiation of the channel is required. Only when non-downgradable parameters (such as profile) are required to be changed does it become necessary to terminate and restart the RTP stream(s). This may be accomplished by using different RTP payload types.

MANEs MAY remove certain unusable packets from the RTP stream when that RTP stream was damaged due to previous packet losses. This can help reduce the network load in certain special cases. For example, MANES can remove those FUs where the leading FUs belonging to the same NAL unit have been lost or those dependent slice segments when the leading slice segments belonging to the same slice have been lost, because the trailing FUs or dependent slice segments are meaningless to most decoders. MANES can also remove higher temporal scalable layers if the outbound transmission (from the MANE's viewpoint) experiences congestion.

## 11. IANA Considerations

Placeholder

## 12. Acknowledgements

Large parts of this specification share text with the RTP payload format for HEVC [RFC7798]. We thank the authors of that specification for their excellent work.

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# RTP Payload Format for Versatile Video Coding (VVC)
draft-ietf-avtcore-rtp-vvc-08

## Abstract

This memo describes an RTP payload format for the video coding standard ITU-T Recommendation H.266 and ISO/IEC International Standard 23090-3, both also known as Versatile Video Coding (VVC) and developed by the Joint Video Experts Team (JVET). The RTP payload format allows for packetization of one or more Network Abstraction Layer (NAL) units in each RTP packet payload as well as fragmentation of a NAL unit into multiple RTP packets. The payload format has wide applicability in videoconferencing, Internet video streaming, and high-bitrate entertainment-quality video, among other applications.

## Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on 8 September 2021.

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# 1. Introduction

The Versatile Video Coding [VVC] specification, formally published as both ITU-T Recommendation H.266 and ISO/IEC International Standard 23090-3, is currently in the ITU-T publication process and the ISO/IEC approval process. VVC is reported to provide significant coding efficiency gains over HEVC [HEVC] as known as H.265, and other earlier video codecs.

This memo specifies an RTP payload format for VVC. It shares its basic design with the NAL (Network Abstraction Layer) unit-based RTP payload formats of, H.264 Video Coding [RFC6184], Scalable Video Coding (SVC) [RFC6190], High Efficiency Video Coding (HEVC) [RFC7798] and their respective predecessors. With respect to design philosophy, security, congestion control, and overall implementation complexity, it has similar properties to those earlier payload format specifications. This is a conscious choice, as at least RFC 6184 is widely deployed and generally known in the relevant implementer communities. Certain mechanisms known from [RFC6190] were incorporated in VVC, as VVC version 1 supports temporal, spatial, and signal-to-noise ratio (SNR) scalability.

## 1.1. Overview of the VVC Codec

VVC and HEVC share a similar hybrid video codec design. In this memo, we provide a very brief overview of those features of VVC that are, in some form, addressed by the payload format specified herein. Implementers have to read, understand, and apply the ITU-T/ISO/IEC specifications pertaining to VVC to arrive at interoperable, well-performing implementations.

Conceptually, both VVC and HEVC include a Video Coding Layer (VCL), which is often used to refer to the coding-tool features, and a NAL, which is often used to refer to the systems and transport interface aspects of the codecs.

### 1.1.1. Coding-Tool Features (informative)

Coding tool features are described below with occasional reference to the coding tool set of HEVC, which is well known in the community.

Similar to earlier hybrid-video-coding-based standards, including HEVC, the following basic video coding design is employed by VVC. A prediction signal is first formed by either intra- or motion-

compensated prediction, and the residual (the difference between the original and the prediction) is then coded. The gains in coding efficiency are achieved by redesigning and improving almost all parts of the codec over earlier designs. In addition, VVC includes several tools to make the implementation on parallel architectures easier.

Finally, VVC includes temporal, spatial, and SNR scalability as well as multiview coding support.

Coding blocks and transform structure

Among major coding-tool differences between HEVC and VVC, one of the important improvements is the more flexible coding tree structure in VVC, i.e., multi-type tree. In addition to quadtree, binary and ternary trees are also supported, which contributes significant improvement in coding efficiency. Moreover, the maximum size of coding tree unit (CTU) is increased from 64x64 to 128x128. To improve the coding efficiency of chroma signal, luma chroma separated trees at CTU level may be employed for intra-slices. The square transforms in HEVC are extended to non-square transforms for rectangular blocks resulting from binary and ternary tree splits. Besides, VVC supports multiple transform sets (MTS), including DCT-2, DST-7, and DCT-8 as well as the non-separable secondary transform. The transforms used in VVC can have different sizes with support for larger transform sizes. For DCT-2, the transform sizes range from 2x2 to 64x64, and for DST-7 and DCT-8, the transform sizes range from 4x4 to 32x32. In addition, VVC also support sub-block transform for both intra and inter coded blocks. For intra coded blocks, intra sub-partitioning (ISP) may be used to allow sub-block based intra prediction and transform. For inter blocks, sub-block transform may be used assuming that only a part of an inter-block has non-zero transform coefficients.

Entropy coding

Similar to HEVC, VVC uses a single entropy-coding engine, which is based on context adaptive binary arithmetic coding [CABAC], but with the support of multi-window sizes. The window sizes can be initialized differently for different context models. Due to such a design, it has more efficient adaptation speed and better coding efficiency. A joint chroma residual coding scheme is applied to further exploit the correlation between the residuals of two color components. In VVC, different residual coding schemes are applied for regular transform coefficients and residual samples generated using transform-skip mode.

In-loop filtering

VVC has more feature support in loop filters than HEVC. The deblocking filter in VVC is similar to HEVC but operates at a smaller grid. After deblocking and sample adaptive offset (SAO), an adaptive loop filter (ALF) may be used. As a Wiener filter, ALF reduces distortion of decoded pictures. Besides, VVC introduces a new module before deblocking called luma mapping with chroma scaling to fully utilize the dynamic range of signal so that rate-distortion performance of both SDR and HDR content is improved.

Motion prediction and coding

Compared to HEVC, VVC introduces several improvements in this area. First, there is the adaptive motion vector resolution (AMVR), which can save bit cost for motion vectors by adaptively signaling motion vector resolution. Then the affine motion compensation is included to capture complicated motion like zooming and rotation. Meanwhile, prediction refinement with the optical flow with affine mode (PROF) is further deployed to mimic affine motion at the pixel level. Thirdly the decoder side motion vector refinement (DMVR) is a method to derive MV vector at decoder side based on block matching so that fewer bits may be spent on motion vectors. Bi-directional optical flow (BDOF) is a similar method to PROF. BDOF adds a sample wise offset at 4x4 sub-block level that is derived with equations based on gradients of the prediction samples and a motion difference relative to CU motion vectors. Furthermore, merge with motion vector difference (MMVD) is a special mode, which further signals a limited set of motion vector differences on top of merge mode. In addition to MMVD, there are another three types of special merge modes, i.e., sub-block merge, triangle, and combined intra-/inter-prediction (CIIP). Sub-block merge list includes one candidate of sub-block temporal motion vector prediction (SbTMVP) and up to four candidates of affine motion vectors. Triangle is based on triangular block motion compensation. CIIP combines intra- and inter- predictions with weighting. Adaptive weighting may be employed with a block-level tool called bi-prediction with CU based weighting (BCW) which provides more flexibility than in HEVC.

Intra prediction and intra-coding

To capture the diversified local image texture directions with finer granularity, VVC supports 65 angular directions instead of 33 directions in HEVC. The intra mode coding is based on a 6-most-probable-mode scheme, and the 6 most probable modes are derived using the neighboring intra prediction directions. In addition, to deal with the different distributions of intra prediction angles for different block aspect ratios, a wide-angle intra prediction (WAIP) scheme is applied in VVC by including intra prediction angles beyond those present in HEVC. Unlike HEVC which only allows using the most

adjacent line of reference samples for intra prediction, VVC also allows using two further reference lines, as known as multi-reference-line (MRL) intra prediction. The additional reference lines can be only used for the 6 most probable intra prediction modes. To capture the strong correlation between different colour components, in VVC, a cross-component linear mode (CCLM) is utilized which assumes a linear relationship between the luma sample values and their associated chroma samples. For intra prediction, VVC also applies a position-dependent prediction combination (PDPC) for refining the prediction samples closer to the intra prediction block boundary. Matrix-based intra prediction (MIP) modes are also used in VVC which generates an up to 8x8 intra prediction block using a weighted sum of downsampled neighboring reference samples, and the weights are hardcoded constants.

Other coding-tool feature

VVC introduces dependent quantization (DQ) to reduce quantization error by state-based switching between two quantizers.

### 1.1.2. Systems and Transport Interfaces (informative)

VVC inherits the basic systems and transport interfaces designs from HEVC and H.264. These include the NAL-unit-based syntax structure, the hierarchical syntax and data unit structure, the supplemental enhancement information (SEI) message mechanism, and the video buffering model based on the hypothetical reference decoder (HRD). The scalability features of VVC are conceptually similar to the scalable variant of HEVC known as SHVC. The hierarchical syntax and data unit structure consists of parameter sets at various levels (decoder, sequence (pertaining to all), sequence (pertaining to a single), picture), picture-level header parameters, slice-level header parameters, and lower-level parameters.

A number of key components that influenced the network abstraction layer design of VVC as well as this memo are described below

Decoding capability information

The decoding capability information includes parameters that stay constant for the lifetime of a Video Bitstream, which in IETF terms can translate to the lifetime of a session. Such information includes profile, level, and sub-profile information to determine a maximum capability interop point that is guaranteed to be never exceeded, even if splicing of video sequences occurs within a session. It further includes constraint fields (most of which are flags), which can optionally be set to indicate that the video bitstream will be constraint in the use of certain features as

indicated by the values of those fields. With this, a bitstream can be labelled as not using certain tools, which allows among other things for resource allocation in a decoder implementation.

Video parameter set

The ideo parameter set (VPS) pertains to a coded video sequences (CVS) of multiple layers covering the same range of access units, and includes, among other information decoding dependency expressed as information for reference picture list construction of enhancement layers. The VPS provides a "big picture" of a scalable sequence, including what types of operation points are provided, the profile, tier, and level of the operation points, and some other high-level properties of the bitstream that can be used as the basis for session negotiation and content selection, etc. One VPS may be referenced by one or more sequence parameter sets.

Sequence parameter set

The sequence parameter set (SPS) contains syntax elements pertaining to a coded layer video sequence (CLVS), which is a group of pictures belonging to the same layer, starting with a random access point, and followed by pictures that may depend on each other, until the next random access point picture. In MPGEG-2, the equivalent of a CVS was a group of pictures (GOP), which normally started with an I frame and was followed by P and B frames. While more complex in its options of random access points, VVC retains this basic concept. One remarkable difference of VVC is that a CLVS may start with a Gradual Decoding Refresh (GDR) picture, without requiring presence of traditional random access points in the bitstream, such as instantaneous decoding refresh (IDR) or clean random access (CRA) pictures. In many TV-like applications, a CVS contains a few hundred milliseconds to a few seconds of video. In video conferencing (without switching MCUs involved), a CVS can be as long in duration as the whole session.

Picture and adaptation parameter set

The picture parameter set and the adaptation parameter set (PPS and APS, respectively) carry information pertaining to zero or more pictures and zero or more slices, respectively. The PPS contains information that is likely to stay constant from picture to picture-at least for pictures for a certain type-whereas the APS contains information, such as adaptive loop filter coefficients, that are likely to change from picture to picture or even within a picture. A single APS is referenced by all slices of the same picture if that APS contains information about luma mapping with chroma scaling (LMCS) or scaling list. Different APSs containing ALF parameters can be referenced by slices of the same picture.

Picture header

A Picture Header contains information that is common to all slices that belong to the same picture. Being able to send that information as a separate NAL unit when pictures are split into several slices allows for saving bitrate, compared to repeating the same information in all slices. However, there might be scenarios where low-bitrate video is transmitted using a single slice per picture. Having a separate NAL unit to convey that information incurs in an overhead for such scenarios. For such scenarios, the picture header syntax structure is directly included in the slice header, instead of in its own NAL unit. The mode of the picture header syntax structure being included in its own NAL unit or not can only be switched on/off for an entire CLVS, and can only be switched off when in the entire CLVS each picture contains only one slice.

Profile, tier, and level

The profile, tier and level syntax structures in DCI, VPS and SPS contain profile, tier, level information for all layers that refer to the DCI, for layers associated with one or more output layer sets specified by the VPS, and for any layer that refers to the SPS, respectively.

Sub-profiles

Within the VVC specification, a sub-profile is a 32-bit number, coded according to ITU-T Rec. T.35, that does not carry a semantics. It is carried in the profile_tier_level structure and hence (potentially) present in the DCI, VPS, and SPS. External registration bodies can register a T.35 codepoint with ITU-T registration authorities and associate with their registration a description of bitstream restrictions beyond the profiles defined by ITU-T and ISO/IEC. This would allow encoder manufacturers to label the bitstreams generated by their encoder as complying with such sub-profile. It is expected that upstream standardization organizations (such as: DVB and ATSC), as well as walled-garden video services will take advantage of this labelling system. In contrast to "normal" profiles, it is expected that sub-profiles may indicate encoder choices traditionally left open in the (decoder- centric) video coding specs, such as GOP structures, minimum/maximum QP values, and the mandatory use of certain tools or SEI messages.

General constraint fields

The profile_tier_level structure carries a considerable number of constraint fields (most of which are flags), which an encoder can use to indicate to a decoder that it will not use a certain tool or

technology. They were included in reaction to a perceived market need for labelling a bitstream as not exercising a certain tool that has become commercially unviable.

Temporal scalability support

VVC includes support of temporal scalability, by inclusion of the signaling of TemporalId in the NAL unit header, the restriction that pictures of a particular temporal sublayer cannot be used for inter prediction reference by pictures of a lower temporal sublayer, the sub-bitstream extraction process, and the requirement that each sub-bitstream extraction output be a conforming bitstream. Media-Aware Network Elements (MANEs) can utilize the TemporalId in the NAL unit header for stream adaptation purposes based on temporal scalability.

Reference picture resampling (RPR)

In AVC and HEVC, the spatial resolution of pictures cannot change unless a new sequence using a new SPS starts, with an IRAP picture. VVC enables picture resolution change within a sequence at a position without encoding an IRAP picture, which is always intra-coded. This feature is sometimes referred to as reference picture resampling (RPR), as the feature needs resampling of a reference picture used for inter prediction when that reference picture has a different resolution than the current picture being decoded. RPR allows resolution change without the need of coding an IRAP picture, which causes a momentary bit rate spike in streaming or video conferencing scenarios, e.g., to cope with network condition changes. RPR can also be used in application scenarios wherein zooming of the entire video region or some region of interest is needed.

Spatial, SNR, and multiview scalability

VVC includes support for spatial, SNR, and multiview scalability. Scalable video coding is widely considered to have technical benefits and enrich services for various video applications. Until recently, however, the functionality has not been included in the first version of specifications of the video codecs. In VVC, however, all those forms of scalability are supported in the first version of VVC natively through the signaling of the layer_id in the NAL unit header, the VPS which associates layers with given layer_ids to each other, reference picture selection, reference picture resampling for spatial scalability, and a number of other mechanisms not relevant for this memo.

Spatial scalability

With the existence of Reference Picture Resampling (RPR), the additional burden for scalability support is just a modification of the high-level syntax (HLS). The inter-layer prediction is employed in a scalable system to improve the coding efficiency of the enhancement layers. In addition to the spatial and temporal motion-compensated predictions that are available in a single-layer codec, the inter-layer prediction in VVC uses the possibly resampled video data of the reconstructed reference picture from a reference layer to predict the current enhancement layer. The resampling process for inter-layer prediction, when used, is performed at the block-level, reusing the existing interpolation process for motion compensation in single-layer coding. It means that no additional resampling process is needed to support spatial scalability.

SNR scalability

SNR scalability is similar to spatial scalability except that the resampling factors are 1:1. In other words, there is no change in resolution, but there is inter-layer prediction.

Multiview scalability

The first version of VVC also supports multiview scalability, wherein a multi-layer bitstream carries layers representing multiple views, and one or more of the represented views can be output at the same time.

SEI messages

Supplementary enhancement information (SEI) messages are information in the bitstream that do not influence the decoding process as specified in the VVC spec, but address issues of representation/ rendering of the decoded bitstream, label the bitstream for certain applications, among other, similar tasks. The overall concept of SEI messages and many of the messages themselves has been inherited from the H.264 and HEVC specs. Except for the SEI messages that affect the specification of the hypothetical reference decoder (HRD), other SEI messages for use in the VVC environment, which are generally useful also in other video coding technologies, are not included in the main VVC specification but in a companion specification [VSEI].

### 1.1.3. High-Level Picture Partitioning (informative)

VVC inherited the concept of tiles and wavefront parallel processing (WPP) from HEVC, with some minor to moderate differences. The basic concept of slices was kept in VVC but designed in an essentially different form. VVC is the first video coding standard that includes subpictures as a feature, which provides the same functionality as HEVC motion-constrained tile sets (MCTSs) but designed differently to have better coding efficiency and to be friendlier for usage in application systems. More details of these differences are described below.

Tiles and WPP

Same as in HEVC, a picture can be split into tile rows and tile columns in VVC, in-picture prediction across tile boundaries is disallowed, etc. However, the syntax for signaling of tile partitioning has been simplified, by using a unified syntax design for both the uniform and the non-uniform mode. In addition, signaling of entry point offsets for tiles in the slice header is optional in VVC while it is mandatory in HEVC. The WPP design in VVC has two differences compared to HEVC: i) The CTU row delay is reduced from two CTUs to one CTU; ii) Signaling of entry point offsets for WPP in the slice header is optional in VVC while it is mandatory in HEVC.

Slices

In VVC, the conventional slices based on CTUs (as in HEVC) or macroblocks (as in AVC) have been removed. The main reasoning behind this architectural change is as follows. The advances in video coding since 2003 (the publication year of AVC v1) have been such that slice-based error concealment has become practically impossible, due to the ever-increasing number and efficiency of in-picture and inter-picture prediction mechanisms. An error-concealed picture is the decoding result of a transmitted coded picture for which there is some data loss (e.g., loss of some slices) of the coded picture or a reference picture for at least some part of the coded picture is not error-free (e.g., that reference picture was an error-concealed picture). For example, when one of the multiple slices of a picture is lost, it may be error-concealed using an interpolation of the neighboring slices. While advanced video coding prediction mechanisms provide significantly higher coding efficiency, they also make it harder for machines to estimate the quality of an error-concealed picture, which was already a hard problem with the use of simpler prediction mechanisms. Advanced in-picture prediction mechanisms also cause the coding efficiency loss due to splitting a picture into multiple slices to be more significant. Furthermore,

network conditions become significantly better while at the same time techniques for dealing with packet losses have become significantly improved. As a result, very few implementations have recently used slices for maximum transmission unit size matching. Instead, substantially all applications where low-delay error resilience is required (e.g., video telephony and video conferencing) rely on system/transport-level error resilience (e.g., retransmission, forward error correction) and/or picture-based error resilience tools (feedback-based error resilience, insertion of IRAPs, scalability with higher protection level of the base layer, and so on). Considering all the above, nowadays it is very rare that a picture that cannot be correctly decoded is passed to the decoder, and when such a rare case occurs, the system can afford to wait for an error-free picture to be decoded and available for display without resulting in frequent and long periods of picture freezing seen by end users.

Slices in VVC have two modes: rectangular slices and raster-scan slices. The rectangular slice, as indicated by its name, covers a rectangular region of the picture. Typically, a rectangular slice consists of several complete tiles. However, it is also possible that a rectangular slice is a subset of a tile and consists of one or more consecutive, complete CTU rows within a tile. A raster-scan slice consists of one or more complete tiles in a tile raster scan order, hence the region covered by a raster-scan slices need not but could have a non-rectangular shape, but it may also happen to have the shape of a rectangle. The concept of slices in VVC is therefore strongly linked to or based on tiles instead of CTUs (as in HEVC) or macroblocks (as in AVC).

Subpictures

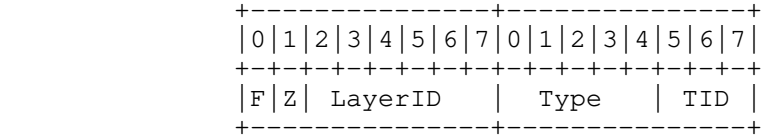
VVC is the first video coding standard that includes the support of subpictures as a feature. Each subpicture consists of one or more complete rectangular slices that collectively cover a rectangular region of the picture. A subpicture may be either specified to be extractable (i.e., coded independently of other subpictures of the same picture and of earlier pictures in decoding order) or not extractable. Regardless of whether a subpicture is extractable or not, the encoder can control whether in-loop filtering (including deblocking, SAO, and ALF) is applied across the subpicture boundaries individually for each subpicture.

Functionally, subpictures are similar to the motion-constrained tile sets (MCTSs) in HEVC. They both allow independent coding and extraction of a rectangular subset of a sequence of coded pictures, for use cases like viewport-dependent 360o video streaming optimization and region of interest (ROI) applications.

There are several important design differences between subpictures and MCTSs. First, the subpictures feature in VVC allows motion vectors of a coding block pointing outside of the subpicture even when the subpicture is extractable by applying sample padding at subpicture boundaries in this case, similarly as at picture boundaries. Second, additional changes were introduced for the selection and derivation of motion vectors in the merge mode and in the decoder side motion vector refinement process of VVC. This allows higher coding efficiency compared to the non-normative motion constraints applied at the encoder-side for MCTSs. Third, rewriting of SHs (and PH NAL units, when present) is not needed when extracting one or more extractable subpictures from a sequence of pictures to create a sub-bitstream that is a conforming bitstream. In sub-bitstream extractions based on HEVC MCTSs, rewriting of SHs is needed. Note that in both HEVC MCTSs extraction and VVC subpictures extraction, rewriting of SPSs and PPSs is needed. However, typically there are only a few parameter sets in a bitstream, while each picture has at least one slice, therefore rewriting of SHs can be a significant burden for application systems. Fourth, slices of different subpictures within a picture are allowed to have different NAL unit types. Fifth, VVC specifies HRD and level definitions for subpicture sequences, thus the conformance of the sub-bitstream of each extractable subpicture sequence can be ensured by encoders.

### 1.1.4. NAL Unit Header

VVC maintains the NAL unit concept of HEVC with modifications. VVC uses a two-byte NAL unit header, as shown in Figure 1. The payload of a NAL unit refers to the NAL unit excluding the NAL unit header.




The Structure of the VVC NAL Unit Header.

Figure 1

The semantics of the fields in the NAL unit header are as specified in VVC and described briefly below for convenience. In addition to the name and size of each field, the corresponding syntax element name in VVC is also provided.

F: 1 bit

forbidden_zero_bit. Required to be zero in VVC. Note that the inclusion of this bit in the NAL unit header was to enable transport of VVC video over MPEG-2 transport systems (avoidance of start code emulations) [MPEG2S]. In the context of this memo the value 1 may be used to indicate a syntax violation, e.g., for a NAL unit resulted from aggregating a number of fragmented units of a NAL unit but missing the last fragment, as described in Section TBD.

Z: 1 bit

nuh_reserved_zero_bit. Required to be zero in VVC, and reserved for future extensions by ITU-T and ISO/IEC.

This memo does not overload the "Z" bit for local extensions, as a) overloading the "F" bit is sufficient and b) to preserve the usefulness of this memo to possible future versions of [VVC].

LayerId: 6 bits

nuh_layer_id. Identifies the layer a NAL unit belongs to, wherein a layer may be, e.g., a spatial scalable layer, a quality scalable layer, a layer containing a different view, etc.

Type: 5 bits

nal_unit_type. This field specifies the NAL unit type as defined in Table 5 of [VVC]. For a reference of all currently defined NAL unit types and their semantics, please refer to Section 7.4.2.2 in [VVC].

TID: 3 bits

nuh_temporal_id_plus1. This field specifies the temporal identifier of the NAL unit plus 1. The value of TemporalId is equal to TID minus 1. A TID value of 0 is illegal to ensure that there is at least one bit in the NAL unit header equal to 1, so to enable independent considerations of start code emulations in the NAL unit header and in the NAL unit payload data.

## 1.2. Overview of the Payload Format

This payload format defines the following processes required for transport of VVC coded data over RTP [RFC3550]:

* Usage of RTP header with this payload format

* Packetization of VVC coded NAL units into RTP packets using three types of payload structures: a single NAL unit packet, aggregation packet, and fragment unit

* Transmission of VVC NAL units of the same bitstream within a single RTP stream

* Media type parameters to be used with the Session Description Protocol (SDP) [RFC4566]

* Usage of RTCP feedback messages

## 2. Conventions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown above.

## 3. Definitions and Abbreviations

### 3.1. Definitions

This document uses the terms and definitions of VVC. Section 3.1.1 lists relevant definitions from [VVC] for convenience. Section 3.1.2 provides definitions specific to this memo.

#### 3.1.1. Definitions from the VVC Specification

Access unit (AU): A set of PUs that belong to different layers and contain coded pictures associated with the same time for output from the DPB.

Adaptation parameter set (APS): A syntax structure containing syntax elements that apply to zero or more slices as determined by zero or more syntax elements found in slice headers.

Bitstream: A sequence of bits, in the form of a NAL unit stream or a byte stream, that forms the representation of a sequence of AUs forming one or more coded video sequences (CVSs).

Coded picture: A coded representation of a picture comprising VCL NAL units with a particular value of nuh_layer_id within an AU and containing all CTUs of the picture.

Clean random access (CRA) PU: A PU in which the coded picture is a CRA picture.

Clean random access (CRA) picture: An IRAP picture for which each VCL NAL unit has nal_unit_type equal to CRA_NUT.

Coded video sequence (CVS): A sequence of AUs that consists, in decoding order, of a CVSS AU, followed by zero or more AUs that are not CVSS AUs, including all subsequent AUs up to but not including any subsequent AU that is a CVSS AU.

Coded video sequence start (CVSS) AU: An AU in which there is a PU for each layer in the CVS and the coded picture in each PU is a CLVSS picture.

Coded layer video sequence (CLVS): A sequence of PUs with the same value of nuh_layer_id that consists, in decoding order, of a CLVSS PU, followed by zero or more PUs that are not CLVSS PUs, including all subsequent PUs up to but not including any subsequent PU that is a CLVSS PU.

Coded layer video sequence start (CLVSS) PU: A PU in which the coded picture is a CLVSS picture.

Coded layer video sequence start (CLVSS) picture: A coded picture that is an IRAP picture with NoOutputBeforeRecoveryFlag equal to 1 or a GDR picture with NoOutputBeforeRecoveryFlag equal to 1.

Coding tree unit (CTU): A CTB of luma samples, two corresponding CTBs of chroma samples of a picture that has three sample arrays, or a CTB of samples of a monochrome picture or a picture that is coded using three separate colour planes and syntax structures used to code the samples.

Decoding Capability Information (DCI): A syntax structure containing syntax elements that apply to the entire bitstream.

Decoded picture buffer (DPB): A buffer holding decoded pictures for reference, output reordering, or output delay specified for the hypothetical reference decoder.

Gradual decoding refresh (GDR) picture: A picture for which each VCL NAL unit has nal_unit_type equal to GDR_NUT.

Instantaneous decoding refresh (IDR) PU: A PU in which the coded picture is an IDR picture.

Instantaneous decoding refresh (IDR) picture: An IRAP picture for which each VCL NAL unit has nal_unit_type equal to IDR_W_RADL or IDR_N_LP.

Intra random access point (IRAP) AU: An AU in which there is a PU for each layer in the CVS and the coded picture in each PU is an IRAP picture.

Intra random access point (IRAP) PU: A PU in which the coded picture is an IRAP picture.

Intra random access point (IRAP) picture: A coded picture for which all VCL NAL units have the same value of nal_unit_type in the range of IDR_W_RADL to CRA_NUT, inclusive.

Layer: A set of VCL NAL units that all have a particular value of nuh_layer_id and the associated non-VCL NAL units.

Network abstraction layer (NAL) unit: A syntax structure containing an indication of the type of data to follow and bytes containing that data in the form of an RBSP interspersed as necessary with emulation prevention bytes.

Network abstraction layer (NAL) unit stream: A sequence of NAL units.

Operation point (OP): A temporal subset of an OLS, identified by an OLS index and a highest value of TemporalId.

Picture parameter set (PPS): A syntax structure containing syntax elements that apply to zero or more entire coded pictures as determined by a syntax element found in each slice header.

Picture unit (PU): A set of NAL units that are associated with each other according to a specified classification rule, are consecutive in decoding order, and contain exactly one coded picture.

Random access: The act of starting the decoding process for a bitstream at a point other than the beginning of the stream.

Sequence parameter set (SPS): A syntax structure containing syntax elements that apply to zero or more entire CLVSs as determined by the content of a syntax element found in the PPS referred to by a syntax element found in each picture header.

Slice: An integer number of complete tiles or an integer number of consecutive complete CTU rows within a tile of a picture that are exclusively contained in a single NAL unit.

Slice header (SH): A part of a coded slice containing the data elements pertaining to all tiles or CTU rows within a tile represented in the slice.

Sublayer: A temporal scalable layer of a temporal scalable bitstream consisting of VCL NAL units with a particular value of the TemporalId variable, and the associated non-VCL NAL units.

Subpicture: An rectangular region of one or more slices within a picture.

Sublayer representation: A subset of the bitstream consisting of NAL units of a particular sublayer and the lower sublayers.

Tile: A rectangular region of CTUs within a particular tile column and a particular tile row in a picture.

Tile column: A rectangular region of CTUs having a height equal to the height of the picture and a width specified by syntax elements in the picture parameter set.

Tile row: A rectangular region of CTUs having a height specified by syntax elements in the picture parameter set and a width equal to the width of the picture.

Video coding layer (VCL) NAL unit: A collective term for coded slice NAL units and the subset of NAL units that have reserved values of nal_unit_type that are classified as VCL NAL units in this Specification.

### 3.1.2. Definitions Specific to This Memo

Media-Aware Network Element (MANE): A network element, such as a middlebox, selective forwarding unit, or application-layer gateway that is capable of parsing certain aspects of the RTP payload headers or the RTP payload and reacting to their contents.

> Informative note: The concept of a MANE goes beyond normal routers or gateways in that a MANE has to be aware of the signaling (e.g., to learn about the payload type mappings of the media streams), and in that it has to be trusted when working with Secure RTP (SRTP). The advantage of using MANEs is that they allow packets to be dropped according to the needs of the media coding. For example, if a MANE has to drop packets due to congestion on a certain link, it can identify and remove those packets whose elimination produces the least adverse effect on the user experience. After dropping packets, MANEs must rewrite RTCP packets to match the changes to the RTP stream, as specified in Section 7 of [RFC3550].

NAL unit decoding order: A NAL unit order that conforms to the constraints on NAL unit order given in Section 7.4.2.4 in [VVC], follow the Order of NAL units in the bitstream.

RTP stream (See [RFC7656]): Within the scope of this memo, one RTP stream is utilized to transport a VVC bitstream, which may contain one or more layers, and each layer may contain one or more temporal sublayers.

Transmission order: The order of packets in ascending RTP sequence number order (in modulo arithmetic). Within an aggregation packet, the NAL unit transmission order is the same as the order of appearance of NAL units in the packet.

## 3.2. Abbreviations

| | |
|---|---|
| AU | Access Unit |
| AP | Aggregation Packet |
| APS | Adaptation Parameter Set |
| CTU | Coding Tree Unit |
| CVS | Coded Video Sequence |
| DPB | Decoded Picture Buffer |
| DCI | Decoding Capability Information |
| DON | Decoding Order Number |
| FIR | Full Intra Request |
| FU | Fragmentation Unit |
| GDR | Gradual Decoding Refresh |
| HRD | Hypothetical Reference Decoder |
| IDR | Instantaneous Decoding Refresh |
| MANE | Media-Aware Network Element |
| MTU | Maximum Transfer Unit |
| NAL | Network Abstraction Layer |

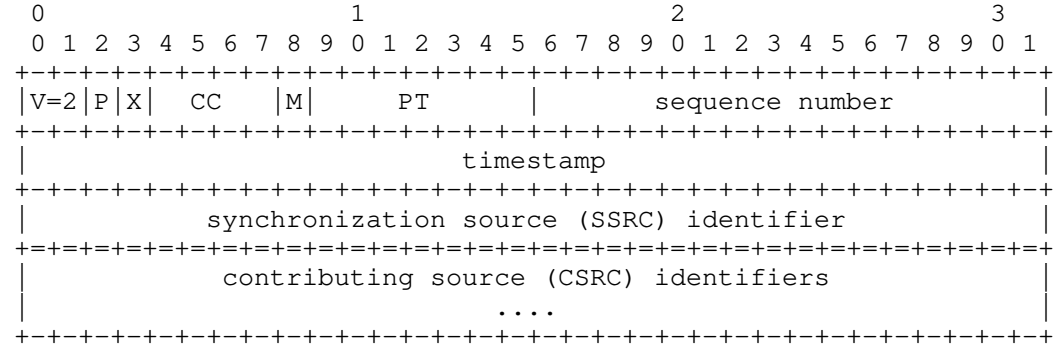
| | |
|---|---|
| NALU | Network Abstraction Layer Unit |
| PLI | Picture Loss Indication |
| PPS | Picture Parameter Set |
| RPS | Reference Picture Set |
| RPSI | Reference Picture Selection Indication |
| SEI | Supplemental Enhancement Information |
| SLI | Slice Loss Indication |
| SPS | Sequence Parameter Set |
| VCL | Video Coding Layer |
| VPS | Video Parameter Set |

# 4. RTP Payload Format

## 4.1. RTP Header Usage

The format of the RTP header is specified in [RFC3550] (reprinted as Figure 2 for convenience). This payload format uses the fields of the header in a manner consistent with that specification.

The RTP payload (and the settings for some RTP header bits) for aggregation packets and fragmentation units are specified in Section 4.3.2 and Section 4.3.3, respectively.




RTP Header According to {{RFC3550}}

Figure 2

The RTP header information to be set according to this RTP payload format is set as follows:

Marker bit (M): 1 bit

Set for the last packet, in transmission order, among each set of packets that contain NAL units of one access unit. This is in line with the normal use of the M bit in video formats to allow an efficient playout buffer handling.

Payload Type (PT): 7 bits

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. The assignment of a payload type has to be performed either through the profile used or in a dynamic way.

Sequence Number (SN): 16 bits

Set and used in accordance with [RFC3550].

Timestamp: 32 bits

The RTP timestamp is set to the sampling timestamp of the content. A 90 kHz clock rate MUST be used. If the NAL unit has no timing properties of its own (e.g., parameter set and SEI NAL units), the RTP timestamp MUST be set to the RTP timestamp of the coded pictures of the access unit in which the NAL unit (according to Section 7.4.2.4 of [VVC]) is included. Receivers MUST use the RTP timestamp for the display process, even when the bitstream contains picture timing SEI messages or decoding unit information SEI messages as specified in [VVC].

Synchronization source (SSRC): 32 bits

Used to identify the source of the RTP packets. A single SSRC is used for all parts of a single bitstream.

## 4.2. Payload Header Usage

The first two bytes of the payload of an RTP packet are referred to as the payload header. The payload header consists of the same fields (F, Z, LayerId, Type, and TID) as the NAL unit header as shown in Section 1.1.4, irrespective of the type of the payload structure.

The TID value indicates (among other things) the relative importance of an RTP packet, for example, because NAL units belonging to higher temporal sublayers are not used for the decoding of lower temporal

sublayers. A lower value of TID indicates a higher importance. More-important NAL units MAY be better protected against transmission losses than less-important NAL units.

For Discussion: quite possibly something similar can be said for the Layer_id in layered coding, but perhaps not in multiview coding. (The relevant part of the spec is relatively new, therefore the soft language). However, for serious layer pruning, interpretation of the VPS is required. We can add language about the need for stateful interpretation of LayerID vis-a-vis stateless interpretation of TID later.

### 4.3. Payload Structures

Three different types of RTP packet payload structures are specified. A receiver can identify the type of an RTP packet payload through the Type field in the payload header.

The three different payload structures are as follows:

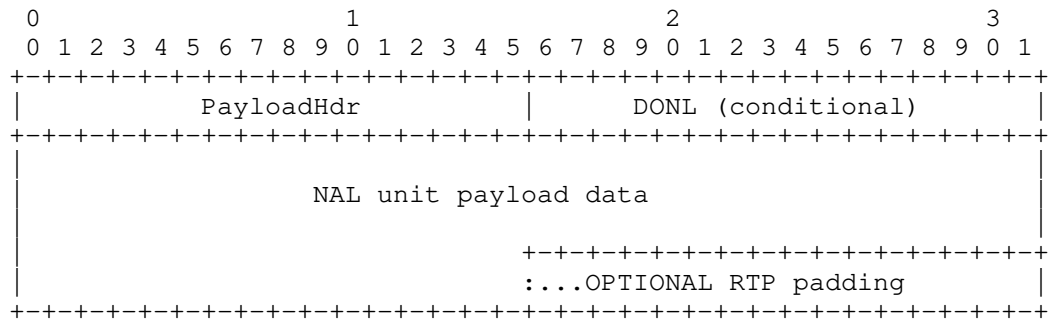
* Single NAL unit packet: Contains a single NAL unit in the payload, and the NAL unit header of the NAL unit also serves as the payload header. This payload structure is specified in Section 4.4.1.

* Aggregation Packet (AP): Contains more than one NAL unit within one access unit. This payload structure is specified in Section 4.3.2.

* Fragmentation Unit (FU): Contains a subset of a single NAL unit. This payload structure is specified in Section 4.3.3.

#### 4.3.1. Single NAL Unit Packets

A single NAL unit packet contains exactly one NAL unit, and consists of a payload header (denoted as PayloadHdr), a conditional 16-bit DONL field (in network byte order), and the NAL unit payload data (the NAL unit excluding its NAL unit header) of the contained NAL unit, as shown in Figure 3.




The Structure of a Single NAL Unit Packet

Figure 3

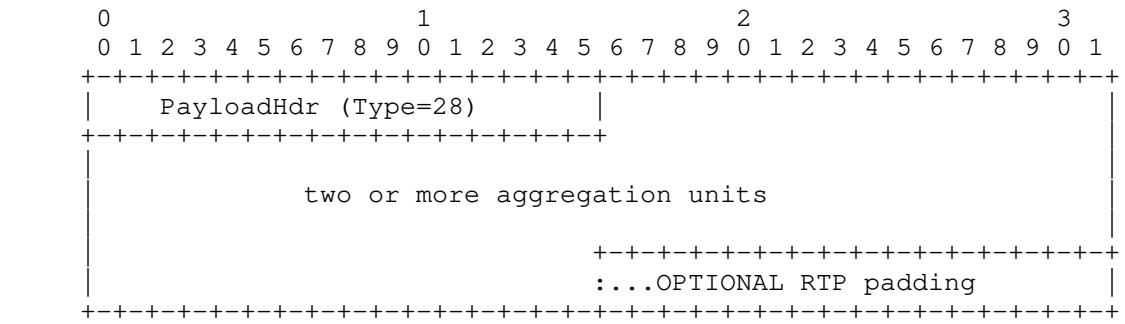
The DONL field, when present, specifies the value of the 16 least significant bits of the decoding order number of the contained NAL unit. If sprop-max-don-diff is greater than 0, the DONL field MUST be present, and the variable DON for the contained NAL unit is derived as equal to the value of the DONL field. Otherwise (sprop-max-don-diff is equal to 0), the DONL field MUST NOT be present.

#### 4.3.2. Aggregation Packets (APs)

Aggregation Packets (APs) can reduce packetization overhead for small NAL units, such as most of the non- VCL NAL units, which are often only a few octets in size.

An AP aggregates NAL units of one access unit. Each NAL unit to be carried in an AP is encapsulated in an aggregation unit. NAL units aggregated in one AP are included in NAL unit decoding order.

An AP consists of a payload header (denoted as PayloadHdr) followed by two or more aggregation units, as shown in Figure 4.




The Structure of an Aggregation Packet

Figure 4

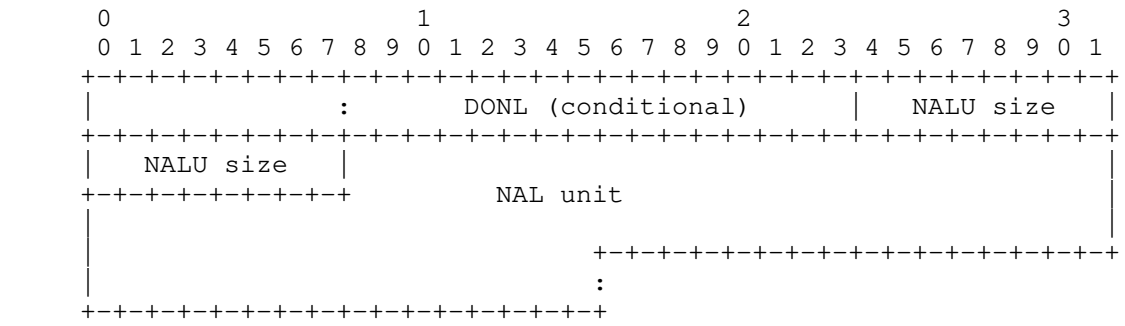
The fields in the payload header of an AP are set as follows. The F bit MUST be equal to 0 if the F bit of each aggregated NAL unit is equal to zero; otherwise, it MUST be equal to 1. The Type field MUST be equal to 28.

The value of LayerId MUST be equal to the lowest value of LayerId of all the aggregated NAL units. The value of TID MUST be the lowest value of TID of all the aggregated NAL units.

> Informative note: All VCL NAL units in an AP have the same TID value since they belong to the same access unit. However, an AP may contain non-VCL NAL units for which the TID value in the NAL unit header may be different than the TID value of the VCL NAL units in the same AP.

An AP MUST carry at least two aggregation units and can carry as many aggregation units as necessary; however, the total amount of data in an AP obviously MUST fit into an IP packet, and the size SHOULD be chosen so that the resulting IP packet is smaller than the MTU size so to avoid IP layer fragmentation. An AP MUST NOT contain FUs specified in Section 4.3.3. APs MUST NOT be nested; i.e., an AP can not contain another AP.

The first aggregation unit in an AP consists of a conditional 16-bit DONL field (in network byte order) followed by a 16-bit unsigned size information (in network byte order) that indicates the size of the NAL unit in bytes (excluding these two octets, but including the NAL unit header), followed by the NAL unit itself, including its NAL unit header, as shown in Figure 5.



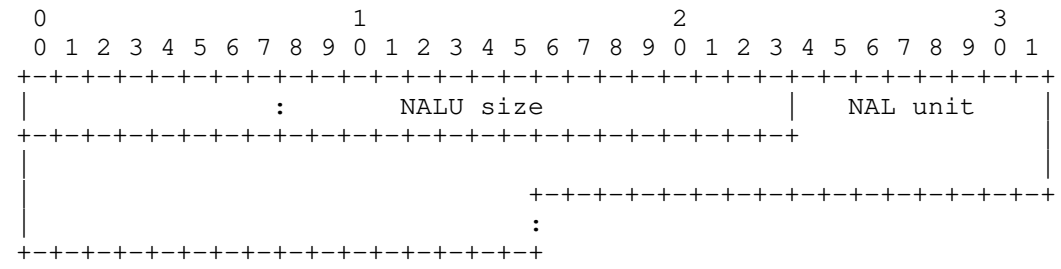

The Structure of the First Aggregation Unit in an AP

Figure 5

The DONL field, when present, specifies the value of the 16 least significant bits of the decoding order number of the aggregated NAL unit.

If sprop-max-don-diff is greater than 0, the DONL field MUST be present in an aggregation unit that is the first aggregation unit in an AP, and the variable DON for the aggregated NAL unit is derived as equal to the value of the DONL field, and the variable DON for an aggregation unit that is not the first aggregation unit in an AP aggregated NAL unit is derived as equal to the DON of the preceding aggregated NAL unit in the same AP plus 1 modulo 65536. Otherwise (sprop-max-don-diff is equal to 0), the DONL field MUST NOT be present in an aggregation unit that is the first aggregation unit in an AP.

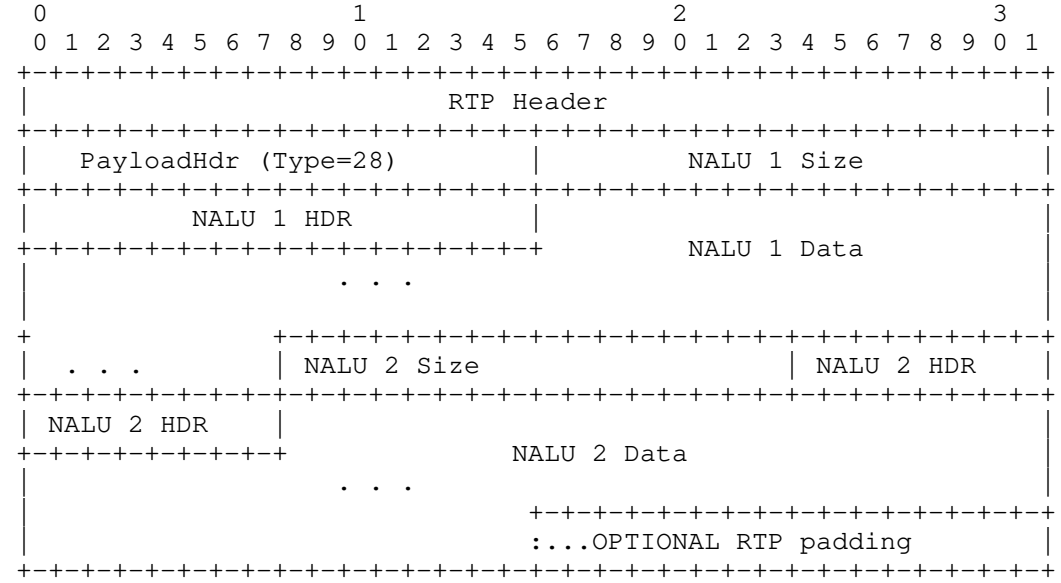
An aggregation unit that is not the first aggregation unit in an AP will be followed immediately by a 16-bit unsigned size information (in network byte order) that indicates the size of the NAL unit in bytes (excluding these two octets, but including the NAL unit header), followed by the NAL unit itself, including its NAL unit header, as shown in Figure 6.




The Structure of an Aggregation Unit That Is Not the First Aggregation Unit in an AP

Figure 6

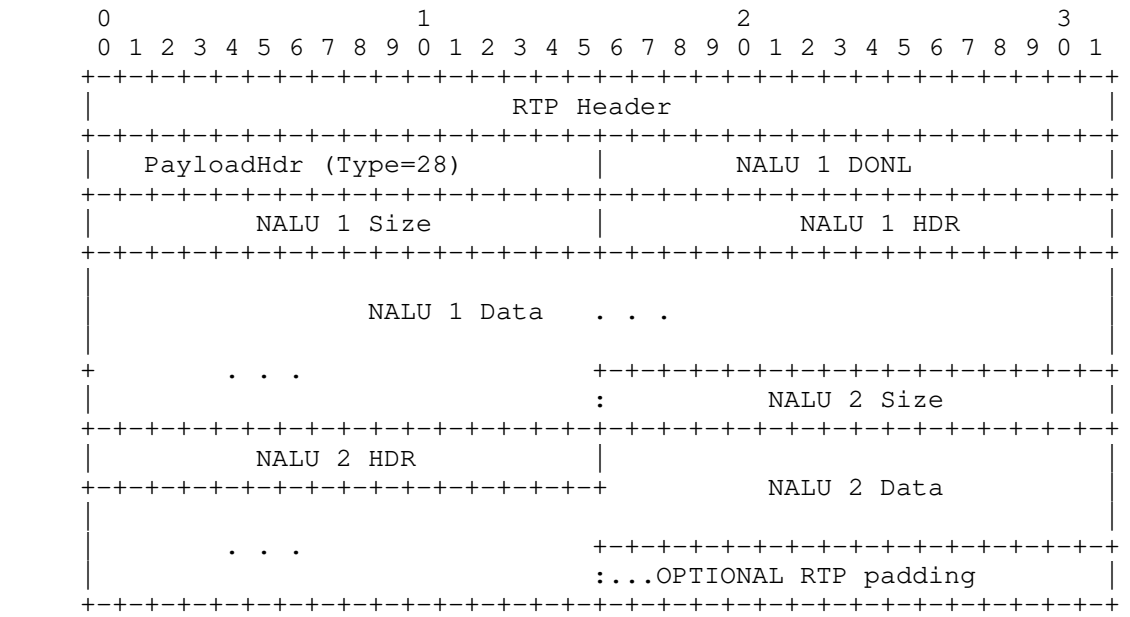
Figure 7 presents an example of an AP that contains two aggregation units, labeled as 1 and 2 in the figure, without the DONL field being present.




An Example of an AP Packet Containing Two Aggregation Units without the DONL Field

Figure 7

Figure 8 presents an example of an AP that contains two aggregation units, labeled as 1 and 2 in the figure, with the DONL field being present.




An Example of an AP Containing
Two Aggregation Units with the DONL Field

Figure 8

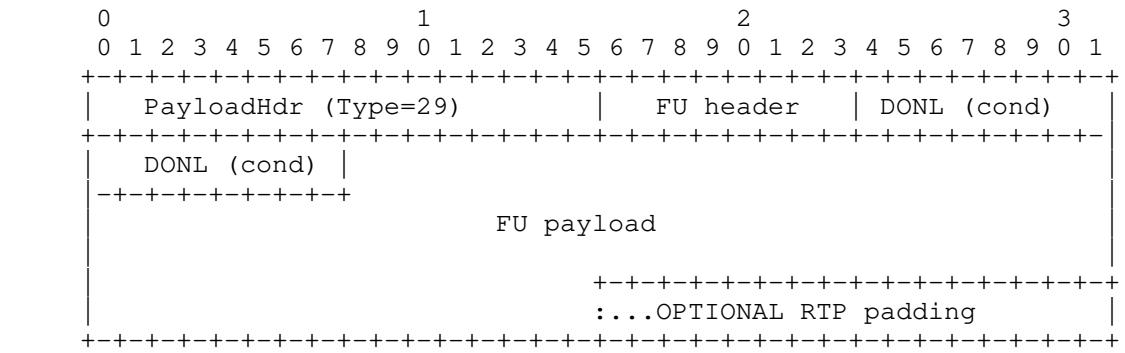
### 4.3.3. Fragmentation Units

Fragmentation Units (FUs) are introduced to enable fragmenting a single NAL unit into multiple RTP packets, possibly without cooperation or knowledge of the [VVC] encoder. A fragment of a NAL unit consists of an integer number of consecutive octets of that NAL unit. Fragments of the same NAL unit MUST be sent in consecutive order with ascending RTP sequence numbers (with no other RTP packets within the same RTP stream being sent between the first and last fragment).

When a NAL unit is fragmented and conveyed within FUs, it is referred to as a fragmented NAL unit. APs MUST NOT be fragmented. FUs MUST NOT be nested; i.e., an FU can not contain a subset of another FU.

The RTP timestamp of an RTP packet carrying an FU is set to the NALU-time of the fragmented NAL unit.

An FU consists of a payload header (denoted as PayloadHdr), an FU header of one octet, a conditional 16-bit DONL field (in network byte order), and an FU payload, as shown in Figure 9.




The Structure of an FU

Figure 9

The fields in the payload header are set as follows. The Type field MUST be equal to 29. The fields F, LayerId, and TID MUST be equal to the fields F, LayerId, and TID, respectively, of the fragmented NAL unit.

The FU header consists of an S bit, an E bit, an R bit and a 5-bit FuType field, as shown in Figure 10.


The Structure of FU Header

Figure 10

The semantics of the FU header fields are as follows:

S: 1 bit

When set to 1, the S bit indicates the start of a fragmented NAL unit, i.e., the first byte of the FU payload is also the first byte of the payload of the fragmented NAL unit. When the FU payload is not the start of the fragmented NAL unit payload, the S bit MUST be set to 0.

E: 1 bit

When set to 1, the E bit indicates the end of a fragmented NAL unit, i.e., the last byte of the payload is also the last byte of the fragmented NAL unit. When the FU payload is not the last fragment of a fragmented NAL unit, the E bit MUST be set to 0.

Reserved: 1 bit

editor-note 24: to be removed upon wg consensus

When set to 1, the R bit indicates the last NAL unit of a coded picture, i.e., the last byte of the FU payload is also the last byte of the coded picture. When the FU payload is not the last fragment of a coded picture, the R bit MUST be set to 0.

FuType: 5 bits

The field FuType MUST be equal to the field Type of the fragmented NAL unit.

The DONL field, when present, specifies the value of the 16 least significant bits of the decoding order number of the fragmented NAL unit.

If sprop-max-don-diff is greater than 0, and the S bit is equal to 1, the DONL field MUST be present in the FU, and the variable DON for the fragmented NAL unit is derived as equal to the value of the DONL field. Otherwise (sprop-max-don-diff is equal to 0, or the S bit is equal to 0), the DONL field MUST NOT be present in the FU.

A non-fragmented NAL unit MUST NOT be transmitted in one FU; i.e., the Start bit and End bit must not both be set to 1 in the same FU header.

The FU payload consists of fragments of the payload of the fragmented NAL unit so that if the FU payloads of consecutive FUs, starting with an FU with the S bit equal to 1 and ending with an FU with the E bit equal to 1, are sequentially concatenated, the payload of the fragmented NAL unit can be reconstructed. The NAL unit header of the fragmented NAL unit is not included as such in the FU payload, but rather the information of the NAL unit header of the fragmented NAL unit is conveyed in F, LayerId, and TID fields of the FU payload headers of the FUs and the FuType field of the FU header of the FUs. An FU payload MUST NOT be empty.

If an FU is lost, the receiver SHOULD discard all following fragmentation units in transmission order corresponding to the same fragmented NAL unit, unless the decoder in the receiver is known to be prepared to gracefully handle incomplete NAL units.

A receiver in an endpoint or in a MANE MAY aggregate the first n-1 fragments of a NAL unit to an (incomplete) NAL unit, even if fragment n of that NAL unit is not received. In this case, the forbidden_zero_bit of the NAL unit MUST be set to 1 to indicate a syntax violation.

### 4.4. Decoding Order Number

For each NAL unit, the variable AbsDon is derived, representing the decoding order number that is indicative of the NAL unit decoding order.

Let NAL unit n be the n-th NAL unit in transmission order within an RTP stream.

If sprop-max-don-diff is equal to 0, AbsDon[n], the value of AbsDon for NAL unit n, is derived as equal to n.

Otherwise (sprop-max-don-diff is greater than 0), AbsDon[n] is derived as follows, where DON[n] is the value of the variable DON for NAL unit n:

* If n is equal to 0 (i.e., NAL unit n is the very first NAL unit in transmission order), AbsDon[0] is set equal to DON[0].

* Otherwise (n is greater than 0), the following applies for derivation of AbsDon[n]:

```
If DON[n] == DON[n-1],
   AbsDon[n] = AbsDon[n-1]

If (DON[n] > DON[n-1] and DON[n] - DON[n-1] < 32768),
   AbsDon[n] = AbsDon[n-1] + DON[n] - DON[n-1]

If (DON[n] < DON[n-1] and DON[n-1] - DON[n] >= 32768),
   AbsDon[n] = AbsDon[n-1] + 65536 - DON[n-1] + DON[n]

If (DON[n] > DON[n-1] and DON[n] - DON[n-1] >= 32768),
   AbsDon[n] = AbsDon[n-1] - (DON[n-1] + 65536 -
   DON[n])

If (DON[n] < DON[n-1] and DON[n-1] - DON[n] < 32768),
   AbsDon[n] = AbsDon[n-1] - (DON[n-1] - DON[n])
```

For any two NAL units m and n, the following applies:

* AbsDon[n] greater than AbsDon[m] indicates that NAL unit n follows NAL unit m in NAL unit decoding order.

* When AbsDon[n] is equal to AbsDon[m], the NAL unit decoding order of the two NAL units can be in either order.

* AbsDon[n] less than AbsDon[m] indicates that NAL unit n precedes NAL unit m in decoding order.

  Informative note: When two consecutive NAL units in the NAL unit decoding order have different values of AbsDon, the absolute difference between the two AbsDon values may be greater than or equal to 1.

  Informative note: There are multiple reasons to allow for the absolute difference of the values of AbsDon for two consecutive NAL units in the NAL unit decoding order to be greater than one. An increment by one is not required, as at the time of associating values of AbsDon to NAL units, it may not be known whether all NAL units are to be delivered to the receiver. For example, a gateway might not forward VCL NAL units of higher sublayers or some SEI NAL units when there is congestion in the network. In another example, the first intra-coded picture of a pre-encoded clip is transmitted in advance to ensure that it is readily available in the receiver, and when transmitting the first intra-coded picture, the originator does not exactly know how many NAL units will be encoded before the first intra-coded picture of the pre-encoded clip follows in decoding order. Thus, the values of AbsDon for the NAL units of the first intra-coded picture of the pre-encoded clip have to be estimated when they are transmitted, and gaps in values of AbsDon may occur.

## 5. Packetization Rules

The following packetization rules apply:

* If sprop-max-don-diff is greater than 0, the transmission order of NAL units carried in the RTP stream MAY be different than the NAL unit decoding order. Otherwise (sprop-max-don-diff is equal to 0), the transmission order of NAL units carried in the RTP stream MUST be the same as the NAL unit decoding order.

* A NAL unit of a small size SHOULD be encapsulated in an aggregation packet together one or more other NAL units in order to avoid the unnecessary packetization overhead for small NAL units. For example, non-VCL NAL units such as access unit delimiters, parameter sets, or SEI NAL units are typically small and can often be aggregated with VCL NAL units without violating MTU size constraints.

* Each non-VCL NAL unit SHOULD, when possible from an MTU size match viewpoint, be encapsulated in an aggregation packet together with its associated VCL NAL unit, as typically a non-VCL NAL unit would be meaningless without the associated VCL NAL unit being available.

* For carrying exactly one NAL unit in an RTP packet, a single NAL unit packet MUST be used.

## 6. De-packetization Process

The general concept behind de-packetization is to get the NAL units out of the RTP packets in an RTP stream and pass them to the decoder in the NAL unit decoding order.

The de-packetization process is implementation dependent. Therefore, the following description should be seen as an example of a suitable implementation. Other schemes may be used as well, as long as the output for the same input is the same as the process described below. The output is the same when the set of output NAL units and their order are both identical. Optimizations relative to the described algorithms are possible.

All normal RTP mechanisms related to buffer management apply. In particular, duplicated or outdated RTP packets (as indicated by the RTP sequences number and the RTP timestamp) are removed. To determine the exact time for decoding, factors such as a possible intentional delay to allow for proper inter-stream synchronization MUST be factored in.

NAL units with NAL unit type values in the range of 0 to 27, inclusive, may be passed to the decoder. NAL-unit-like structures with NAL unit type values in the range of 28 to 31, inclusive, MUST NOT be passed to the decoder.

The receiver includes a receiver buffer, which is used to compensate for transmission delay jitter within individual RTP stream, to reorder NAL units from transmission order to the NAL unit decoding order. In this section, the receiver operation is described under the assumption that there is no transmission delay jitter within an RTP stream. To make a difference from a practical receiver buffer that is also used for compensation of transmission delay jitter, the receiver buffer is hereafter called the de-packetization buffer in this section. Receivers should also prepare for transmission delay jitter; that is, either reserve separate buffers for transmission delay jitter buffering and de-packetization buffering or use a receiver buffer for both transmission delay jitter and de-packetization. Moreover, receivers should take transmission delay jitter into account in the buffering operation, e.g., by additional initial buffering before starting of decoding and playback.

When sprop-max-don-diff is equal to 0, the de-packetization buffer size is zero bytes, and the process described in the remainder of this paragraph applies. The NAL units carried in the single RTP stream are directly passed to the decoder in their transmission order, which is identical to their decoding order.

When sprop-max-don-diff is greater than 0, the process described in the remainder of this section applies.

There are two buffering states in the receiver: initial buffering and buffering while playing. Initial buffering starts when the reception is initialized. After initial buffering, decoding and playback are started, and the buffering-while-playing mode is used.

Regardless of the buffering state, the receiver stores incoming NAL units in reception order into the de-packetization buffer. NAL units carried in RTP packets are stored in the de-packetization buffer individually, and the value of AbsDon is calculated and stored for each NAL unit.

Initial buffering lasts until condition A (the difference between the greatest and smallest AbsDon values of the NAL units in the de-packetization buffer is greater than or equal to the value of sprop-max-don-diff) or condition B (the number of NAL units in the de-packetization buffer is greater than the value of sprop-depack-buf-nalus) is true.

After initial buffering, whenever condition A or condition B is true, the following operation is repeatedly applied until both condition A and condition B become false:

* The NAL unit in the de-packetization buffer with the smallest value of AbsDon is removed from the de-packetization buffer and passed to the decoder.

When no more NAL units are flowing into the de-packetization buffer, all NAL units remaining in the de-packetization buffer are removed from the buffer and passed to the decoder in the order of increasing AbsDon values.

## 7. Payload Format Parameters

This section specifies the optional parameters. A mapping of the parameters with Session Description Protocol (SDP) [RFC4556] is also provided for applications that use SDP.

### 7.1. Media Type Registration

The receiver MUST ignore any parameter unspecified in this memo.

Type name: video

Subtype name: H266

Required parameters: none

Optional parameters:

profile-id, tier-flag, sub-profile-id, interop-constraints, and level-id:

These parameters indicate the profile, tier, default level, sub-profile, and some constraints of the bitstream carried by the RTP stream, or a specific set of the profile, tier, default level, sub-profile and some constraints the receiver supports.

The subset of coding tools that may have been used to generate the bitstream or that the receiver supports, as well as some additional constraints are indicated collectively by profile-id, sub-profile-id, and interop-constraints.

Informative note: There are 128 values of profile-id. The subset of coding tools identified by the profile-id can be further constrained with up to 255 instances of sub-profile-id. In addition, 68 bits included in interop-constraints, which can be extended up to 324 bits provide means to further restrict tools from existing profiles. To be able to support this fine-granular signalling of coding tool subsets with profile-id, sub-profile-id and interop-

constraints, it would be safe to require symmetric use of these parameters in SDP offer/answer unless recv-ols-id is included in the SDP answer for choosing one of the layers offered.

The tier is indicated by tier-flag. The default level is indicated by level-id. The tier and the default level specify the limits on values of syntax elements or arithmetic combinations of values of syntax elements that are followed when generating the bitstream or that the receiver supports.

In SDP offer/answer, when the SDP answer does not include the recv-ols-id parameter that is less than the sprop-ols-id parameter in the SDP offer, the following applies:

- The tier-flag, profile-id, sub-profile-id, and interop-constraints parameters MUST be used symmetrically, i.e., the value of each of these parameters in the offer MUST be the same as that in the answer, either explicitly signaled or implicitly inferred.

- The level-id parameter is changeable as long as the highest level indicated by the answer is either equal to or lower than that in the offer. Note that a highest level higher than level-id in the offer for receiving can be included as max-recv-level-id.

In SDP offer/answer, when the SDP answer does include the recv-ols-id parameter that is less than the sprop-ols-id parameter in the SDP offer, the set of tier- flag, profile-id, sub-profile-id, interop-constraints, and level-id parameters included in the answer MUST be consistent with that for the chosen output layer set as indicated in the SDP offer, with the exception that the level-id parameter in the SDP answer is changeable as long as the highest level indicated by the answer is either lower than or equal to that in the offer.

More specifications of these parameters, including how they relate to syntax elements specified in [VVC] are provided below.

profile-id:

When profile-id is not present, a value of 1 (i.e., the Main 10 profile) MUST be inferred.

When used to indicate properties of a bitstream, profile-id is derived from the general_profile_idc syntax element that applies to the bitstream in an instance of the profile_tier_level( ) syntax structure.

A profile_tier_level( ) syntax structure may be contained in an SPS, VPS, or DCI NAL units as specified in [VVC]. One of the following three cases applies to the container NAL unit of the profile_tier_level( ) syntax structure containing those PTL syntax elements used to derive the values of profile-id, tier-flag, level-id, sub-profile-id, or interop-constraints: 1) The container NAL unit is an SPS, the bitstream is a single-layer bitstream, and the profile_tier_level( ) syntax structures in all SPSs referenced by the CVSs in the bitstream has the same values respectively for those PTL syntax elements; 2) The container NAL unit is a VPS, the profile_tier_level( ) syntax structure is the one in the VPS that applies to the OLS corresponding to the bitstream, and the profile_tier_level( ) syntax structures applicable to the OLS corresponding to the bitstream in all VPSs referenced by the CVSs in the bitstream have the same values respectively for those PTL syntax elements; 3) The container NAL unit is a DCI NAL unit and the profile_tier_level( ) syntax structures in all DCI NAL units in the bitstream has the same values respectively for those PTL syntax elements.

tier-flag, level-id:

The value of tier-flag MUST be in the range of 0 to 1, inclusive. The value of level-id MUST be in the range of 0 to 255, inclusive.

If the tier-flag and level-id parameters are used to indicate properties of a bitstream, they indicate the tier and the highest level the bitstream complies with.

If the tier-flag and level-id parameters are used for capability exchange, the following applies. If max-recv-level-id is not present, the default level defined by level-id indicates the highest level the codec wishes to support. Otherwise, max-recv-level-id indicates the highest level the codec supports for receiving. For either receiving or sending, all levels that are lower than the highest level supported MUST also be supported.

If no tier-flag is present, a value of 0 MUST be inferred; if no level-id is present, a value of 51 (i.e., level 3.1) MUST be inferred.

Informative note: The level values currently defined in the VVC specification are in the form of "majorNum.minorNum", and the value of the level-id for each of the levels is equal to majorNum * 16 + minorNum * 3. It is expected that if any level are defined in the future, the same convention will be used, but this cannot be guaranteed.

When used to indicate properties of a bitstream, the tier-flag and level-id parameters are derived respectively from the syntax element general_tier_flag, and the syntax element general_level_idc or sub_layer_level_idc[j], that apply to the bitstream, in an instance of the profile_tier_level( ) syntax structure.

If the tier-flag and level-id are derived from the profile_tier_level( ) syntax structure in a DCI NAL unit, the following applies:

- tier-flag = general_tier_flag
- level-id = general_level_idc

Otherwise, if the tier-flag and level-id are derived from the profile_tier_level( ) syntax structure in an SPS or VPS NAL unit, and the bitstream contains the highest sub-layer representation in the OLS corresponding to the bitstream, the following applies:

- tier-flag = general_tier_flag
- level-id = general_level_idc

Otherwise, if the tier-flag and level-id are derived from the profile_tier_level( ) syntax structure in an SPS or VPS NAL unit, and the bitstream does not contains the highest sub-layer representation in the OLS corresponding to the bitstream, the following applies, with j being the value of the sprop-sub-layer-id parameter:

- tier-flag = general_tier_flag
- level-id = sub_layer_level_idc[j]

sub-profile-id:

The value of the parameter is a comma-separated (',') list of values.

editor-note 11: What is the value? integer, base32?

When used to indicate properties of a bitstream, sub-profile-id is derived from each of the ptl_num_sub_profiles general_sub_profile_idc[i] syntax elements that apply to the bitstream in an profile_tier_level( ) syntax structure.

interop-constraints:

A base16 [RFC4648] (hexadecimal) representation of the data that includes the syntax elements ptl_frame_only_constraint_flag and ptl_multilayer_enabled_flag and the general_constraints_info( ) syntax structure that apply to the bitstream in an instance of the profile_tier_level( ) syntax structure.

If the interop-constraints parameter is not present, the following MUST be inferred:

- ptl_frame_only_constraint_flag = 0
- ptl_multilayer_enabled_flag = 1
- gci_present_flag in the general_constraints_info( ) syntax structure = 1

editor-note 14: Double check the default values. Currently, no constraints, but actually, with the Main 10 profile as default multi-layer not possible.

Using interop-constraints for capability exchange results in a requirement on any bitstream to be compliant with the interop-constraints.

sprop-sub-layer-id:

This parameter MAY be used to indicate the highest allowed value of TID in the bitstream. When not present, the value of sprop-sub-layer-id is inferred to be equal to 6.

The value of sprop-sub-layer-id MUST be in the range of 0 to 6, inclusive.

sprop-ols-id:

This parameter MAY be used to indicate the OLS that the bitstream applies to. When not present, the value of sprop-ols-id is inferred to be equal to TargetOlsIdx as specified in

8.1.1 in [VVC]. If this optional parameter is present, sprop-vps MUST also be present or its content MUST be known a priori at the receiver.

The value of sprop-ols-id MUST be in the range of 0 to 257, inclusive.

recv-sub-layer-id:

This parameter MAY be used to signal a receiver's choice of the offered or declared sub-layer representations in the sprop-vps and sprop-sps. The value of recv-sub-layer-id indicates the TID of the highest sub-layer of the bitstream that a receiver supports. When not present, the value of recv-sub-layer-id is inferred to be equal to the value of the sprop-sub-layer-id parameter in the SDP offer.

The value of recv-sub-layer-id MUST be in the range of 0 to 6, inclusive.

recv-ols-id:

This parameter MAY be used to signal a receiver's choice of the offered or declared output layer sets in the sprop-vps. The value of recv-ols-id indicates the OLS index of the bitstream that a receiver supports. When not present, the value of recv-ols-id is inferred to be equal to the value of the sprop-ols-id parameter in the SDP offer. When present, the value of recv-ols-id must be included only when sprop-ols-id was received and must refer to an output layer set in the VPS that is in the same dependency tree as the OLS referred to by sprop-ols-id. If this optional parameter is present, sprop-vps must have been received or its content must be known a priori at the receiver.

The value of recv-ols-id MUST be in the range of 0 to 257, inclusive.

max-recv-level-id:

This parameter MAY be used to indicate the highest level a receiver supports.

The value of max-recv-level-id MUST be in the range of 0 to 255, inclusive.

When max-recv-level-id is not present, the value is inferred to be equal to level-id.

max-recv-level-id MUST NOT be present when the highest level the receiver supports is not higher than the default level.

sprop-dci:

This parameter MAY be used to convey a decoding capability information NAL unit of the bitstream for out-of-band transmission. The parameter MAY also be used for capability exchange. The value of the parameter a base64 [RFC4648] representations of the decoding capability information NAL unit as specified in Section 7.3.2.1 of [VVC].

sprop-vps:

This parameter MAY be used to convey any video parameter set NAL unit of the bitstream for out-of-band transmission of video parameter sets. The parameter MAY also be used for capability exchange and to indicate sub-stream characteristics (i.e., properties of output layer sets and sublayer representations as defined in [VVC]). The value of the parameter is a comma-separated (',') list of base64 [RFC4648] representations of the video parameter set NAL units as specified in Section 7.3.2.3 of [VVC].

The sprop-vps parameter MAY contain one or more than one video parameter set NAL unit. However, all other video parameter sets contained in the sprop-vps parameter MUST be consistent with the first video parameter set in the sprop-vps parameter. A video parameter set vpsB is said to be consistent with another video parameter set vpsA if any decoder that conforms to the profile, tier, level, and constraints indicated by the 12 bytes of data starting from the syntax element general_profile_space to the syntax element general_level_idc, inclusive, in the first profile_tier_level( ) syntax structure in vpsA can decode any bitstream that conforms to the profile, tier, level, and constraints indicated by the 12 bytes of data starting from the syntax element general_profile_space to the syntax element general_level_idc, inclusive, in the first profile_tier_level( ) syntax structure in vpsB.

sprop-sei:

This parameter MAY be used to convey one or more SEI messages that describe bitstream characteristics. When present, a decoder can rely on the bitstream characteristics that are described in the SEI messages for the entire duration of the session, independently from the persistence scopes of the SEI messages as specified in [VSEI].

The value of the parameter is a comma-separated (',') list of base64 [RFC4648] representations of SEI NAL units as specified in [VSEI].

Informative note: Intentionally, no list of applicable or inapplicable SEI messages is specified here. Conveying certain SEI messages in sprop-sei may be sensible in some application scenarios and meaningless in others. However, a few examples are described below:

1) In an environment where the bitstream was created from film-based source material, and no splicing is going to occur during the lifetime of the session, the film grain characteristics SEI message is likely meaningful, and sending it in sprop-sei rather than in the bitstream at each entry point may help with saving bits and allows one to configure the renderer only once, avoiding unwanted artifacts.

2) Examples for SEI messages that would be meaningless to be conveyed in sprop-sei include the decoded picture hash SEI message (it is close to impossible that all decoded pictures have the same hashtag), the display orientation SEI message when the device is a handheld device (as the display orientation may change when the handheld device is turned around), or the filler payload SEI message (as there is no point in just having more bits in SDP).

max-lsr:

The max-lsr MAY be used to signal the capabilities of a receiver implementation and MUST NOT be used for any other purpose. The value of max-lsr is an integer indicating the maximum processing rate in units of luma samples per second. The max-lsr parameter signals that the receiver is capable of decoding video at a higher rate than is required by the highest level.

Informative note: When the OPTIONAL media type parameters are used to signal the properties of a bitstream, and max-lsr is not present, the values of tier-flag, profile-id, sub-profile-id interop-constraints, and level-id must always be such that the bitstream complies fully with the specified profile, tier, and level.

When max-lsr is signaled, the receiver MUST be able to decode bitstreams that conform to the highest level, with the exception that the MaxLumaSr value in Table 136 of [VVC] for

the highest level is replaced with the value of max-lsr. Senders MAY use this knowledge to send pictures of a given size at a higher picture rate than is indicated in the highest level.

When not present, the value of max-lsr is inferred to be equal to the value of MaxLumaSr given in Table 136 of [VVC] for the highest level.

The value of max-lsr MUST be in the range of MaxLumaSr to 16 * MaxLumaSr, inclusive, where MaxLumaSr is given in Table 136 of [VVC] for the highest level.

max-fps:

The value of max-fps is an integer indicating the maximum picture rate in units of pictures per 100 seconds that can be effectively processed by the receiver. The max-fps parameter MAY be used to signal that the receiver has a constraint in that it is not capable of processing video effectively at the full picture rate that is implied by the highest level and, when present, max-lsr.

The value of max-fps is not necessarily the picture rate at which the maximum picture size can be sent, it constitutes a constraint on maximum picture rate for all resolutions.

Informative note: The max-fps parameter is semantically different from max-lsr in that max-fps is used to signal a constraint, lowering the maximum picture rate from what is implied by other parameters.

The encoder MUST use a picture rate equal to or less than this value. In cases where the max-fps parameter is absent, the encoder is free to choose any picture rate according to the highest level and any signaled optional parameters.

The value of max-fps MUST be smaller than or equal to the full picture rate that is implied by the highest level and, when present, max-lsr.

sprop-max-don-diff:

If there is no NAL unit naluA that is followed in transmission order by any NAL unit preceding naluA in decoding order (i.e., the transmission order of the NAL units is the same as the decoding order), the value of this parameter MUST be equal to 0.

Otherwise, this parameter specifies the maximum absolute difference between the decoding order number (i.e., AbsDon) values of any two NAL units naluA and naluB, where naluA follows naluB in decoding order and precedes naluB in transmission order.

The value of sprop-max-don-diff MUST be an integer in the range of 0 to 32767, inclusive.

When not present, the value of sprop-max-don-diff is inferred to be equal to 0.

sprop-depack-buf-bytes:

This parameter signals the required size of the de-packetization buffer in units of bytes. The value of the parameter MUST be greater than or equal to the maximum buffer occupancy (in units of bytes) of the de-packetization buffer as specified in Section 6.

The value of sprop-depack-buf-bytes MUST be an integer in the range of 0 to 4294967295, inclusive.

When sprop-max-don-diff is present and greater than 0, this parameter MUST be present and the value MUST be greater than 0. When not present, the value of sprop-depack-buf-bytes is inferred to be equal to 0.

Informative note: The value of sprop-depack-buf-bytes indicates the required size of the de-packetization buffer only. When network jitter can occur, an appropriately sized jitter buffer has to be available as well.

depack-buf-cap:

This parameter signals the capabilities of a receiver implementation and indicates the amount of de-packetization buffer space in units of bytes that the receiver has available for reconstructing the NAL unit decoding order from NAL units carried in the RTP stream. A receiver is able to handle any RTP stream for which the value of the sprop-depack-buf-bytes parameter is smaller than or equal to this parameter.

When not present, the value of depack-buf-cap is inferred to be equal to 4294967295. The value of depack-buf-cap MUST be an integer in the range of 1 to 4294967295, inclusive.

Informative note: depack-buf-cap indicates the maximum possible size of the de-packetization buffer of the receiver only, without allowing for network jitter.

editor-note 19: sprop-depack-buf-nalus not included but mentioned in section 6 for startup in de-packetization process. We should decide on whether it needs to be included or not.

### 7.2. SDP Parameters

The receiver MUST ignore any parameter unspecified in this memo.

#### 7.2.1. Mapping of Payload Type Parameters to SDP

The media type video/H266 string is mapped to fields in the Session Description Protocol (SDP) [RFC4566] as follows:

* The media name in the "m=" line of SDP MUST be video.

* The encoding name in the "a=rtpmap" line of SDP MUST be H266 (the media subtype).

* The clock rate in the "a=rtpmap" line MUST be 90000.

* The OPTIONAL parameters profile-id, tier-flag, sub-profile-id, interop-constraints, level-id, sprop-sub-layer-id, sprop-ols-id, recv-sub-layer-id, recv-ols-id, max-recv-level-id, max-lsr, max-fps, sprop-max-don-diff, sprop-depack-buf-bytes and depack-buf-cap, when present, MUST be included in the "a=fmtp" line of SDP. This parameter is expressed as a media type string, in the form of a semicolon-separated list of parameter=value pairs.

  editor-note 20: To Be updated

An example of media representation in SDP is as follows:

```
m=video 49170 RTP/AVP 98
a=rtpmap:98 H266/90000
a=fmtp:98 profile-id=1; sprop-vps=<video parameter sets data>
```

#### 7.2.2. Usage with SDP Offer/Answer Model

When [VVC] is offered over RTP using SDP in an offer/answer model [RFC3264] for negotiation for unicast usage, the following limitations and rules apply:

editor-note 21: the following needs to be updated

* Parameters to identify a media format configuration as VVC:

* Parameters as bitstream properties:

* SDP answer for media configurations.

* capability parameters:

* others:

# 8. Use with Feedback Messages

The following subsections define the use of the Picture Loss Indication (PLI), Slice Lost Indication (SLI), Reference Picture Selection Indication (RPSI), and Full Intra Request (FIR) feedback messages with HEVC. The PLI, SLI, and RPSI messages are defined in [RFC4585], and the FIR message is defined in [RFC5104].

## 8.1. Picture Loss Indication (PLI)

As specified in RFC 4585, Section 6.3.1, the reception of a PLI by a media sender indicates "the loss of an undefined amount of coded video data belonging to one or more pictures". Without having any specific knowledge of the setup of the bitstream (such as use and location of in-band parameter sets, non-IRAP decoder refresh points, picture structures, and so forth), a reaction to the reception of an PLI by a [VVC] sender SHOULD be to send an IRAP picture and relevant parameter sets; potentially with sufficient redundancy so to ensure correct reception. However, sometimes information about the bitstream structure is known. For example, state could have been established outside of the mechanisms defined in this document that parameter sets are conveyed out of band only, and stay static for the duration of the session. In that case, it is obviously unnecessary to send them in-band as a result of the reception of a PLI. Other examples could be devised based on a priori knowledge of different aspects of the bitstream structure. In all cases, the timing and congestion control mechanisms of RFC 4585 MUST be observed.

## 8.2. Full Intra Request (FIR)

The purpose of the FIR message is to force an encoder to send an independent decoder refresh point as soon as possible, while observing applicable congestion-control-related constraints, such as those set out in [RFC8082]).

Upon reception of a FIR, a sender MUST send an IDR picture. Parameter sets MUST also be sent, except when there is a priori knowledge that the parameter sets have been correctly established. A

typical example for that is an understanding between sender and receiver, established by means outside this document, that parameter sets are exclusively sent out-of-band.

## 9. Security Considerations

The scope of this Security Considerations section is limited to the payload format itself and to one feature of [VVC] that may pose a particularly serious security risk if implemented naively. The payload format, in isolation, does not form a complete system. Implementers are advised to read and understand relevant security-related documents, especially those pertaining to RTP (see the Security Considerations section in [RFC3550] ), and the security of the call-control stack chosen (that may make use of the media type registration of this memo). Implementers should also consider known security vulnerabilities of video coding and decoding implementations in general and avoid those.

Within this RTP payload format, and with the exception of the user data SEI message as described below, no security threats other than those common to RTP payload formats are known. In other words, neither the various media-plane-based mechanisms, nor the signaling part of this memo, seems to pose a security risk beyond those common to all RTP-based systems.

RTP packets using the payload format defined in this specification are subject to the security considerations discussed in the RTP specification [RFC3550] , and in any applicable RTP profile such as RTP/AVP [RFC3551] , RTP/AVPF [RFC4585] , RTP/SAVP [RFC3711] , or RTP/SAVPF [RFC5124] . However, as "Securing the RTP Framework: Why RTP Does Not Mandate a Single Media Security Solution" [RFC7202] discusses, it is not an RTP payload format's responsibility to discuss or mandate what solutions are used to meet the basic security goals like confidentiality, integrity and source authenticity for RTP in general. This responsibility lays on anyone using RTP in an application. They can find guidance on available security mechanisms and important considerations in "Options for Securing RTP Sessions" [RFC7201] . The rest of this section discusses the security impacting properties of the payload format itself.

Because the data compression used with this payload format is applied end-to-end, any encryption needs to be performed after compression. A potential denial-of-service threat exists for data encodings using compression techniques that have non-uniform receiver-end computational load. The attacker can inject pathological datagrams into the bitstream that are complex to decode and that cause the receiver to be overloaded. [VVC] is particularly vulnerable to such attacks, as it is extremely simple to generate datagrams containing

NAL units that affect the decoding process of many future NAL units. Therefore, the usage of data origin authentication and data integrity protection of at least the RTP packet is RECOMMENDED, for example, with SRTP [RFC3711] .

Like HEVC [RFC7798], [VVC] includes a user data Supplemental Enhancement Information (SEI) message. This SEI message allows inclusion of an arbitrary bitstring into the video bitstream. Such a bitstring could include JavaScript, machine code, and other active content. [VVC] leaves the handling of this SEI message to the receiving system. In order to avoid harmful side effects the user data SEI message, decoder implementations cannot naively trust its content. For example, it would be a bad and insecure implementation practice to forward any JavaScript a decoder implementation detects to a web browser. The safest way to deal with user data SEI messages is to simply discard them, but that can have negative side effects on the quality of experience by the user.

End-to-end security with authentication, integrity, or confidentiality protection will prevent a MANE from performing media-aware operations other than discarding complete packets. In the case of confidentiality protection, it will even be prevented from discarding packets in a media-aware way. To be allowed to perform such operations, a MANE is required to be a trusted entity that is included in the security context establishment.

## 10. Congestion Control

Congestion control for RTP SHALL be used in accordance with RTP [RFC3550] and with any applicable RTP profile, e.g., AVP [RFC3551]. If best-effort service is being used, an additional requirement is that users of this payload format MUST monitor packet loss to ensure that the packet loss rate is within an acceptable range. Packet loss is considered acceptable if a TCP flow across the same network path, and experiencing the same network conditions, would achieve an average throughput, measured on a reasonable timescale, that is not less than all RTP streams combined are achieving. This condition can be satisfied by implementing congestion-control mechanisms to adapt the transmission rate, the number of layers subscribed for a layered multicast session, or by arranging for a receiver to leave the session if the loss rate is unacceptably high.

The bitrate adaptation necessary for obeying the congestion control principle is easily achievable when real-time encoding is used, for example, by adequately tuning the quantization parameter. However, when pre-encoded content is being transmitted, bandwidth adaptation requires the pre-coded bitstream to be tailored for such adaptivity. The key mechanisms available in [VVC] are temporal scalability, and

spatial/SNR scalability. A media sender can remove NAL units belonging to higher temporal sublayers (i.e., those NAL units with a high value of TID) or higher spatio-SNR layers (as indicated by interpreting the VPS) until the sending bitrate drops to an acceptable range.

The mechanisms mentioned above generally work within a defined profile and level and, therefore, no renegotiation of the channel is required. Only when non-downgradable parameters (such as profile) are required to be changed does it become necessary to terminate and restart the RTP stream(s). This may be accomplished by using different RTP payload types.

MANEs MAY remove certain unusable packets from the RTP stream when that RTP stream was damaged due to previous packet losses. This can help reduce the network load in certain special cases. For example, MANES can remove those FUs where the leading FUs belonging to the same NAL unit have been lost or those dependent slice segments when the leading slice segments belonging to the same slice have been lost, because the trailing FUs or dependent slice segments are meaningless to most decoders. MANES can also remove higher temporal scalable layers if the outbound transmission (from the MANE's viewpoint) experiences congestion.

## 11. IANA Considerations

Placeholder

## 12. Acknowledgements

Dr. Byeongdoo Choi is thanked for the video codec related technical discussion and other aspects in this memo. Xin Zhao and Dr. Xiang Li are thanked for their contributions on [VVC] specification descriptive content. Spencer Dawkins is thanked for his valuable review comments that led to great improvements of this memo. Some parts of this specification share text with the RTP payload format for HEVC [RFC7798]. We thank the authors of that specification for their excellent work.

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## Appendix A. Change History

draft-zhao-payload-rtp-vvc-00 ..... initial version

draft-zhao-payload-rtp-vvc-01 ..... editorial clarifications and corrections

draft-ietf-payload-rtp-vvc-00 ..... initial WG draft

draft-ietf-payload-rtp-vvc-01 ..... VVC specification update

draft-ietf-payload-rtp-vvc-02 ..... VVC specification update

draft-ietf-payload-rtp-vvc-03 ..... VVC coding tool introduction update

draft-ietf-payload-rtp-vvc-04 ..... VVC coding tool introduction update

draft-ietf-payload-rtp-vvc-05 ..... reference udpate and adding placement for open issues

draft-ietf-payload-rtp-vvc-06 ..... address editor’s note

draft-ietf-payload-rtp-vvc-07 ..... address editor’s notes

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March 10, 2021

# Frame Marking RTP Header Extension
# draft-ietf-avtext-framemarking-12

## Abstract

This document describes a Frame Marking RTP header extension used to convey information about video frames that is critical for error recovery and packet forwarding in RTP middleboxes or network nodes. It is most useful when media is encrypted, and essential when the middlebox or node has no access to the media decryption keys. It is also useful for codec-agnostic processing of encrypted or unencrypted media, while it also supports extensions for codec-specific information.

## Status of This Memo

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This Internet-Draft will expire on September 11, 2021.

## Copyright Notice

Table of Contents



## 1. Introduction

Many widely deployed RTP [RFC3550] topologies [RFC7667] used in modern voice and video conferencing systems include a centralized component that acts as an RTP switch. It receives voice and video streams from each participant, which may be encrypted using SRTP [RFC3711], or extensions that provide participants with private media [RFC8871] via end-to-end encryption where the switch has no access to media decryption keys. The goal is to provide a set of streams back to the participants which enable them to render the right media content. In a simple video configuration, for example, the goal will be that each participant sees and hears just the active speaker. In that case, the goal of the switch is to receive the voice and video streams from each participant, determine the active speaker based on energy in the voice packets, possibly using the client-to-mixer audio level RTP header extension [RFC6464], and select the corresponding video stream for transmission to participants; see Figure 1.

In this document, an "RTP switch" is used as a common short term for the terms "switching RTP mixer", "source projecting middlebox", "source forwarding unit/middlebox" and "video switching MCU" as discussed in [RFC7667].

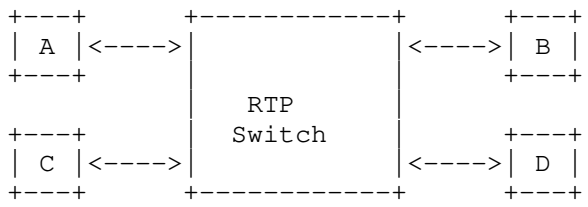



Figure 1: RTP switch

In order to properly support switching of video streams, the RTP switch typically needs some critical information about video frames in order to start and stop forwarding streams.

- Because of inter-frame dependencies, it should ideally switch video streams at a point where the first frame from the new speaker can be decoded by recipients without prior frames, e.g switch on an intra-frame.
- In many cases, the switch may need to drop frames in order to realize congestion control techniques, and needs to know which frames can be dropped with minimal impact to video quality.
- For scalable streams with dependent layers, the switch may need to selectively forward specific layers to specific recipients due to recipient bandwidth or decoder limits.
- Furthermore, it is highly desirable to do this in a payload format-agnostic way which is not specific to each different video codec. Most modern video codecs share common concepts around frame types and other critical information to make this codec-agnostic handling possible.
- It is also desirable to be able to do this for SRTP without requiring the video switch to decrypt the packets. SRTP will encrypt the RTP payload format contents and consequently this data is not usable for the switching function without decryption, which may not even be possible in the case of end-to-end encryption of private media [RFC8871].

By providing meta-information about the RTP streams outside the encrypted media payload, an RTP switch can do codec-agnostic selective forwarding without decrypting the payload. This document specifies the necessary meta-information in an RTP header extension.

## 2. Key Words for Normative Requirements

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].

## 3. Frame Marking RTP Header Extension

This specification uses RTP header extensions as defined in [RFC8285]. A subset of meta-information from the video stream is provided as an RTP header extension to allow an RTP switch to do generic selective forwarding of video streams encoded with potentially different video codecs.

The Frame Marking RTP header extension is encoded using the one-byte header or two-byte header as described in [RFC8285]. The one-byte header format is used for examples in this memo. The two-byte header format is used when other two-byte header extensions are present in the same RTP packet, since mixing one-byte and two-byte extensions is not possible in the same RTP packet.

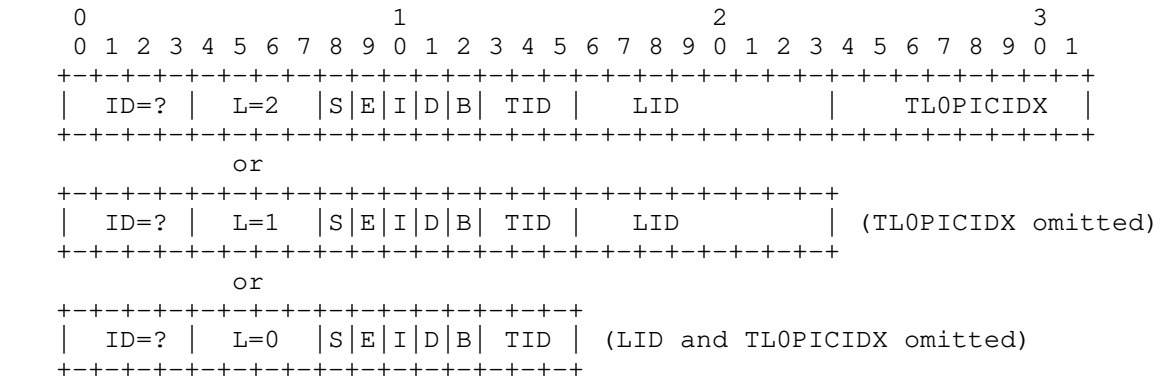
This extension is only specified for Source (not Redundancy) RTP Streams [RFC7656] that carry video payloads. It is not specified for audio payloads, nor is it specified for Redundancy RTP Streams. The (separate) specifications for Redundancy RTP Streams often include provisions for recovering any header extensions that were part of the original source packet. Such provisions SHALL be followed to recover the Frame Marking RTP header extension of the original source packet. Source packet frame markings may be useful when generating Redundancy RTP Streams; for example, the I and D bits can be used to generate extra or no redundancy, respectively, and redundancy schemes with source blocks can align source block boundaries with Independent frame boundaries as marked by the I bit.

A frame, in the context of this specification, is the set of RTP packets with the same RTP timestamp from a specific RTP synchronization source (SSRC). A frame within a layer is the set of RTP packets with the same RTP timestamp, SSRC, Temporal ID (TID), and Layer ID (LID).

### 3.1. Long Extension for Scalable Streams

The following RTP header extension is RECOMMENDED for scalable streams. It MAY also be used for non-scalable streams, in which case TID, LID and TL0PICIDX MUST be 0 or omitted. The ID is assigned per [RFC8285], and the length is encoded as L=2 which indicates 3 octets of data when nothing is omitted, or L=1 for 2 octets when TL0PICIDX

is omitted, or L=0 for 1 octet when both LID and TL0PICIDX are omitted.




The following information are extracted from the media payload and sent in the Frame Marking RTP header extension.

- S: Start of Frame (1 bit) - MUST be 1 in the first packet in a frame within a layer; otherwise MUST be 0.
- E: End of Frame (1 bit) - MUST be 1 in the last packet in a frame within a layer; otherwise MUST be 0. Note that the RTP header marker bit MAY be used to infer the last packet of the highest enhancement layer, in payload formats with such semantics.
- I: Independent Frame (1 bit) - MUST be 1 for a frame within a layer that can be decoded independent of temporally prior frames, e.g. intra-frame, VPX keyframe, H.264 IDR [RFC6184], H.265 IDR/CRA/BLA/RAP [RFC7798]; otherwise MUST be 0. Note that this bit only signals temporal independence, so it can be 1 in spatial or quality enhancement layers that depend on temporally co-located layers but not temporally prior frames.
- D: Discardable Frame (1 bit) - MUST be 1 for a frame within a layer the sender knows can be discarded, and still provide a decodable media stream; otherwise MUST be 0.
- B: Base Layer Sync (1 bit) - When TID is not 0, this MUST be 1 if the sender knows this frame within a layer only depends on the base temporal layer; otherwise MUST be 0. When TID is 0 or if no scalability is used, this MUST be 0.
- TID: Temporal ID (3 bits) - Identifies the temporal layer/sub-layer encoded, starting with 0 for the base layer, and increasing with higher temporal fidelity. If no scalability is used, this MUST be 0. It is implicitly 0 in the short extension format.
- LID: Layer ID (8 bits) - Identifies the spatial and quality layer encoded, starting with 0 for the base layer, and increasing with higher fidelity. If no scalability is used, this MUST be 0 or omitted to reduce length. When omitted, TL0PICIDX MUST also be

omitted. It is implicitly 0 in the short extension format or when omitted in the long extension format.

- o TL0PICIDX: Temporal Layer 0 Picture Index (8 bits) - When TID is 0 and LID is 0, this is a cyclic counter labeling base layer frames. When TID is not 0 or LID is not 0, this indicates a dependency on the given index, such that this frame within this layer depends on the frame with this label in the layer with TID 0 and LID 0. If no scalability is used, or the cyclic counter is unknown, this MUST be omitted to reduce length. Note that 0 is a valid index value for TL0PICIDX.

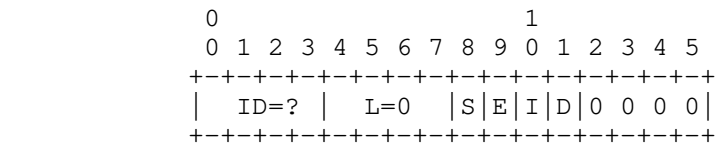
The layer information contained in TID and LID convey useful aspects of the layer structure that can be utilized in selective forwarding.

Without further information about the layer structure, these TID/LID identifiers can only be used for relative priority of layers and implicit dependencies between layers. They convey a layer hierarchy with TID=0 and LID=0 identifying the base layer. Higher values of TID identify higher temporal layers with higher frame rates. Higher values of LID identify higher spatial and/or quality layers with higher resolutions and/or bitrates. Implicit dependencies between layers assume that a layer with a given TID/LID MAY depend on layer(s) with the same or lower TID/LID, but MUST NOT depend on layer(s) with higher TID/LID.

With further information, for example, possible future RTCP SDES items that convey full layer structure information, it may be possible to map these TIDs and LIDs to specific absolute frame rates, resolutions and bitrates, as well as explicit dependencies between layers. Such additional layer information may be useful for forwarding decisions in the RTP switch, but is beyond the scope of this memo. The relative layer information is still useful for many selective forwarding decisions even without such additional layer information.

### 3.2. Short Extension for Non-Scalable Streams

The following RTP header extension is RECOMMENDED for non-scalable streams. It is identical to the shortest form of the extension for scalable streams, except the last four bits (B and TID) are replaced with zeros. It MAY also be used for scalable streams if the sender has limited or no information about stream scalability. The ID is assigned per [RFC8285], and the length is encoded as L=0 which indicates 1 octet of data.




The following information are extracted from the media payload and sent in the Frame Marking RTP header extension.

- o S: Start of Frame (1 bit) - MUST be 1 in the first packet in a frame; otherwise MUST be 0.
- o E: End of Frame (1 bit) - MUST be 1 in the last packet in a frame; otherwise MUST be 0. SHOULD match the RTP header marker bit in payload formats with such semantics for marking end of frame.
- o I: Independent Frame (1 bit) - MUST be 1 for frames that can be decoded independent of temporally prior frames, e.g. intra-frame, VPX keyframe, H.264 IDR [RFC6184], H.265 IDR/CRA/BLA/IRAP [RFC7798]; otherwise MUST be 0.
- o D: Discardable Frame (1 bit) - MUST be 1 for frames the sender knows can be discarded, and still provide a decodable media stream; otherwise MUST be 0.
- o The remaining (4 bits) - are reserved/fixed values and not used for non-scalable streams; they MUST be set to 0 upon transmission and ignored upon reception.

## 3.3. Layer ID Mappings for Scalable Streams

This section maps the specific Layer ID information contained in specific scalable codecs to the generic LID and TID fields.

Note that non-scalable streams have no Layer ID information and thus no mappings.

### 3.3.1. VP9 LID Mapping

The following shows the VP9 [I-D.ietf-payload-vp9] Spatial Layer ID (SID, 3 bits) and Temporal Layer ID (TID, 3 bits) from the VP9 payload descriptor mapped to the generic LID and TID fields.

The S bit MUST match the B bit in the VP9 payload descriptor.

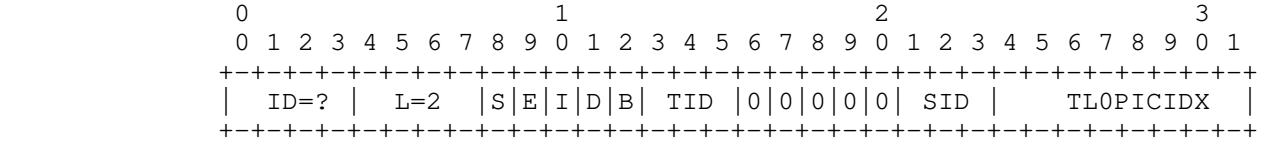
The E bit MUST match the E bit in the VP9 payload descriptor.

The I bit MUST match the inverse of the P bit in the VP9 payload descriptor.

The D bit MUST be 1 if the refresh_frame_flags in the VP9 payload uncompressed header are all 0, otherwise it MUST be 0.

The B bit MUST be 0 if TID is 0; otherwise, if TID is not 0, it MUST match the U bit in the VP9 payload descriptor. Note: When using temporally nested scalability structures as recommended in Section 3.5.2, the B bit and VP9 U bit will always be 1 if TID is not 0, since it is always possible to switch up to a higher temporal layer in such nested structures.

TID and TL0PICIDX MUST match the correspondingly named fields in the VP9 payload descriptor.




### 3.3.2. H265 LID Mapping

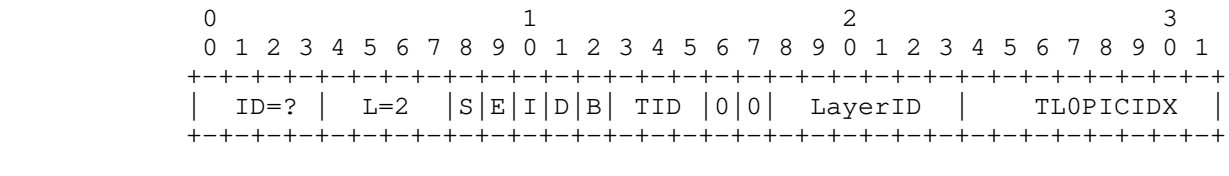
The following shows the H265 [RFC7798] LayerID (6 bits) and TID (3 bits) from the NAL unit header mapped to the generic LID and TID fields.

The S and E bits MUST match the correspondingly named bits in PACI:PHES:TSCI payload structures.

The I bit MUST be 1 when the NAL unit type is 16-23 (inclusive) or 32-34 (inclusive), or an aggregation packet or fragmentation unit encapsulating any of these types, otherwise it MUST be 0. These ranges cover intra (IRAP) frames as well as critical parameter sets (VPS, SPS, PPS).

The D bit MUST be 1 when the NAL unit type is 0, 2, 4, 6, 8, 10, 12, 14, or 38, or an aggregation packet or fragmentation unit encapsulating only these types, otherwise it MUST be 0. These ranges cover non-reference frames as well as filler data.

The B bit can not be determined reliably from simple inspection of payload headers, and therefore is determined by implementation-specific means. For example, internal codec interfaces may provide information to set this reliably.




### 3.3.3. H264-SVC LID Mapping

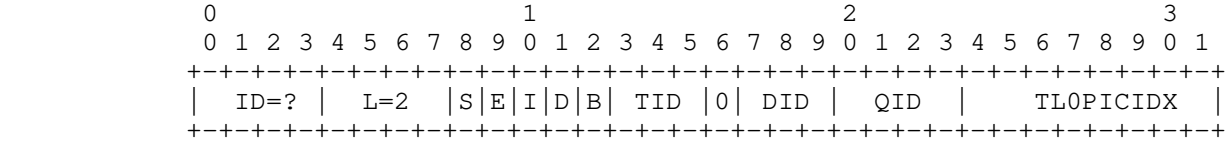
The following shows H264-SVC [RFC6190] Layer encoding information (3 bits for spatial/dependency layer, 4 bits for quality layer and 3 bits for temporal layer) mapped to the generic LID and TID fields.

The S, E, I and D bits MUST match the correspondingly named bits in PACSI payload structures.

The I bit MUST be 1 when the NAL unit type is 5, 7, 8, 13, or 15, or an aggregation packet or fragmentation unit encapsulating any of these types, otherwise it MUST be 0. These ranges cover intra (IDR) frames as well as critical parameter sets (SPS/PPS variants).

The D bit MUST be 1 when the NAL unit header NRI field is 0, or an aggregation packet or fragmentation unit encapsulating only NAL units with NRI=0, otherwise it MUST be 0. The NRI=0 condition signals non-reference frames.

The B bit can not be determined reliably from simple inspection of payload headers, and therefore is determined by implementation-specific means. For example, internal codec interfaces may provide information to set this reliably.




### 3.3.4. H264 (AVC) LID Mapping

The following shows the header extension for H264 (AVC) [RFC6184] that contains only temporal layer information.

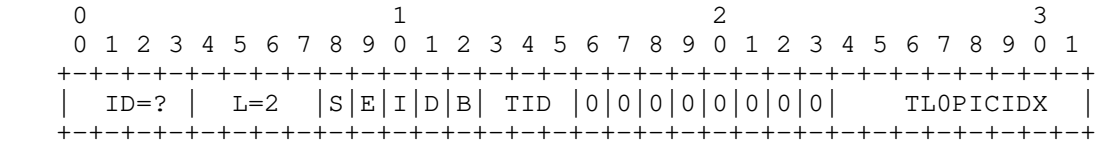
The S bit MUST be 1 when the timestamp in the RTP header differs from the timestamp in the prior RTP sequence number from the same SSRC, otherwise it MUST be 0.

The E bit MUST match the M bit in the RTP header.

The I bit MUST be 1 when the NAL unit type is 5, 7, or 8, or an aggregation packet or fragmentation unit encapsulating any of these types, otherwise it MUST be 0. These ranges cover intra (IDR) frames as well as critical parameter sets (SPS/PPS).

The D bit MUST be 1 when the NAL unit header NRI field is 0, or an aggregation packet or fragmentation unit encapsulating only NAL units with NRI=0, otherwise it MUST be 0. The NRI=0 condition signals non-reference frames.

The B bit can not be determined reliably from simple inspection of payload headers, and therefore is determined by implementation-specific means. For example, internal codec interfaces may provide information to set this reliably.




### 3.3.5. VP8 LID Mapping

The following shows the header extension for VP8 [RFC7741] that contains only temporal layer information.

The S bit MUST match the correspondingly named bit in the VP8 payload descriptor when PID=0, otherwise it MUST be 0.

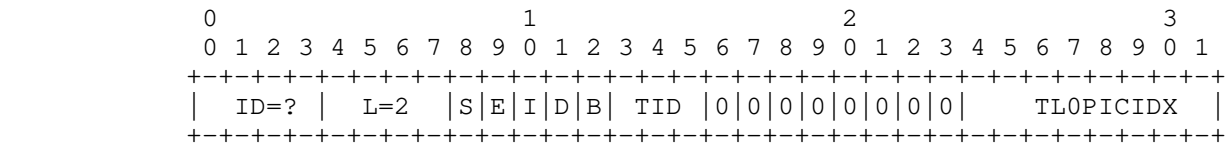
The E bit MUST match the M bit in the RTP header.

The I bit MUST match the inverse of the P bit in the VP8 payload header.

The D bit MUST match the N bit in the VP8 payload descriptor.

The B bit MUST match the Y bit in the VP8 payload descriptor. Note: When using temporally nested scalability structures as recommended in Section 3.5.2, the B bit and VP8 Y bit will always be 1 if TID is not 0, since it is always possible to switch up to a higher temporal layer in such nested structures.

TID and TL0PICIDX MUST match the correspondingly named fields in the VP8 payload descriptor.




### 3.3.6. Future Codec LID Mapping

The RTP payload format specification for future video codecs SHOULD include a section describing the LID mapping and TID mapping for the codec.

## 3.4. Signaling Information

The URI for declaring this header extension in an extmap attribute is "urn:ietf:params:rtp-hdrext:framemarking". It does not contain any extension attributes.

An example attribute line in SDP:

```
a=extmap:3 urn:ietf:params:rtp-hdrext:framemarking
```

## 3.5. Usage Considerations

The header extension values MUST represent what is already in the RTP payload.

When an RTP switch needs to discard a received video frame due to congestion control considerations, it is RECOMMENDED that it preferably drop frames marked with the D (Discardable) bit set, or the highest values of TID and LID, which indicate the highest temporal and spatial/quality enhancement layers, since those typically have fewer dependenices on them than lower layers.

When an RTP switch wants to forward a new video stream to a receiver, it is RECOMMENDED to select the new video stream from the first switching point with the I (Independent) bit set in all spatial layers and forward the same. An RTP switch can request a media source to generate a switching point by sending Full Intra Request (RTCP FIR) as defined in [RFC5104], for example.

### 3.5.1. Relation to Layer Refresh Request (LRR)

Receivers can use the Layer Refresh Request (LRR) [I-D.ietf-avtext-lrr] RTCP feedback message to upgrade to a higher layer in scalable encodings. The TID/LID values and formats used in LRR messages MUST correspond to the same values and formats specified in Section 3.1.

Because frame marking can only be used with temporally-nested streams, temporal-layer LRR refreshes are unnecessary for frame-marked streams. Other refreshes can be detected based on the I bit being set for the specific spatial layers.

### 3.5.2. Scalability Structures

The LID and TID information is most useful for fixed scalability structures, such as nested hierarchical temporal layering structures, where each temporal layer only references lower temporal layers or the base temporal layer. The LID and TID information is less useful, or even not useful at all, for complex, irregular scalability structures that do not conform to common, fixed patterns of inter-layer dependencies and referencing structures. Therefore it is RECOMMENDED to use LID and TID information for RTP switch forwarding decisions only in the case of temporally nested scalability structures, and it is NOT RECOMMENDED for other (more complex or irregular) scalability structures.

## 4. Security Considerations

In the Secure Real-Time Transport Protocol (SRTP) [RFC3711], RTP header extensions are authenticated but usually not encrypted. When header extensions are used some of the payload type information are exposed and visible to middle boxes. The encrypted media data is not exposed, so this is not seen as a high risk exposure.

## 5. Acknowledgements

Many thanks to Bernard Aboba, Jonathan Lennox, Stephan Wenger, Dale Worley, and Magnus Westerlund for their inputs.

## 6. IANA Considerations

This document defines a new extension URI to the RTP Compact HeaderExtensions sub-registry of the Real-Time Transport Protocol (RTP) Parameters registry, according to the following data:

Extension URI: urn:ietf:params:rtp-hdrext:framemarkinginfo
Description: Frame marking information for video streams
Contact: mzanaty@cisco.com
Reference: RFC XXXX

Note to RFC Editor: please replace RFC XXXX with the number of this RFC.

# 7. References

## 7.1. Normative References

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## 7.2. Informative References

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May 7, 2021

# RTP Payload Format for ISO/IEC 21122 (JPEG XS)
# draft-ietf-payload-rtp-jpegxs-13

## Abstract

This document specifies a Real-Time Transport Protocol (RTP) payload format to be used for transporting JPEG XS (ISO/IEC 21122) encoded video. JPEG XS is a low-latency, lightweight image coding system. Compared to an uncompressed video use case, it allows higher resolutions and frame rates, while offering visually lossless quality, reduced power consumption, and end-to-end latency confined to a fraction of a frame.

## Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on November 8, 2021.

## Copyright Notice

Table of Contents



## 1. Introduction

This document specifies a payload format for packetization of JPEG XS [ISO21122-1] encoded video signals into the Real-time Transport Protocol (RTP) [RFC3550].

The JPEG XS coding system offers compression and recompression of image sequences with very moderate computational resources while remaining robust under multiple compression and decompression cycles and mixing of content sources, e.g. embedding of subtitles, overlays or logos. Typical target compression ratios ensuring visually lossless quality are in the range of 2:1 to 10:1, depending on the nature of the source material. The end-to-end latency can be confined to a fraction of a frame, typically between a small number of lines down to below a single line.

## 2. Conventions, Definitions, and Abbreviations

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

Application Data Unit (ADU)
: The unit of source data provided as payload to the transport layer, and corresponding, in this RTP payload definition, to a single JPEG XS frame.

Colour specification box (CS box)
: A ISO colour specification box defined in JPEG XS Part 3 [ISO21122-3] that includes colour-related metadata required to correctly display JPEG XS frames, such as colour primaries, transfer characteristics and matrix coefficients.

EOC marker
: A marker that consists of the two bytes 0xff11 indicating the end of a JPEG XS codestream.

JPEG XS codestream
: A sequence of bytes representing a compressed image formatted according to JPEG XS Part 1 [ISO21122-1].

JPEG XS codestream header
: A sequence of bytes, starting with a SOC marker, at the beginning of each JPEG XS codestream encoded in multiple markers and marker segments that does not carry entropy coded data, but metadata such as the frame dimension and component precision.

JPEG XS frame
: A JPEG XS picture segment in the case of a progressive frame, or, in the case of an interlaced frame, the concatenation of two JPEG XS picture segments.

JPEG XS header segment
: The concatenation of a video support box [ISO21122-3], a colour specification box [ISO21122-3], and a JPEG XS codestream header.

JPEG XS picture segment
: The concatenation of a video support box [ISO21122-3], a colour specification box [ISO21122-3], and a JPEG XS codestream.

JPEG XS stream
: A sequence of JPEG XS frames.

Marker
: A two-byte functional sequence that is part of a JPEG XS codestream starting with a 0xff byte and a subsequent byte defining its function.

Marker segment
: A marker along with a 16-bit marker size and payload data following the size.

Packetization unit
: A portion of an Application Data Unit whose boundaries coincide with boundaries of RTP packet payloads (excluding payload header), i.e. the first (resp. last) byte of a packetization unit is the first (resp. last) byte of a RTP packet payload (excluding its payload header).

Slice
: The smallest independently decodable unit of a JPEG XS codestream, bearing in mind that it decodes to wavelet coefficients which still require inverse wavelet filtering to give an image.

SOC marker
: A marker that consists of the two bytes 0xff10 indicating the start of a JPEG XS codestream. The SOC marker is considered an integral part of the JPEG XS codestream header.

Video support box (VS box)
: An ISO video support box, as defined in [ISO21122-3], that includes metadata required to play back a JPEG XS stream, such as its maximum bitrate, its subsampling structure, its buffer model and its frame rate.

## 3. Media Format Description

### 3.1. Image Data Structures

JPEG XS is a low-latency lightweight image coding system for coding continuous-tone grayscale or continuous-tone colour digital images.

This coding system provides an efficient representation of image signals through the mathematical tool of wavelet analysis. The wavelet filter process separates each component into multiple bands, where each band consists of multiple coefficients describing the image signal of a given component within a frequency domain specific to the wavelet filter type, i.e. the particular filter corresponding to the band.

Wavelet coefficients are grouped into precincts, where each precinct includes all coefficients over all bands that contribute to a spatial region of the image.

One or multiple precincts are furthermore combined into slices consisting of an integer number of precincts. Precincts do not cross slice boundaries, and wavelet coefficients in precincts that are part of different slices can be decoded independently from each other. Note, however, that the wavelet transformation runs across slice boundaries. A slice always extends over the full width of the image, but may only cover parts of its height.

### 3.2. Codestream

A JPEG XS codestream header, starting with an SOC marker, followed by one or more slices, and terminated by an EOC marker form a JPEG XS codestream.

The JPEG XS codestream format, including the definition of all markers, is further defined in [ISO21122-1]. It represents sample values of a single image, without any interpretation relative to a colour space.

### 3.3. Video support box and colour specification box

While the information defined in the codestream is sufficient to reconstruct the sample values of one image, the interpretation of the samples remains undefined by the codestream itself. This interpretation is given by the video support box and the colour specification box which contain significant information to correctly play the JPEG XS stream. The layout and syntax of these boxes, together with their content, are defined in [ISO21122-3].

The video support box provides information on the maximum bitrate, the frame rate, the interlaced mode (progressive or interlaced), the

subsampling image format, the informative timecode of the current JPEG XS frame, the profile, level/sublevel used, and optionally on the buffer model and the mastering display metadata.

Note that the profile and level/sublevel, specified by respectively the Ppih and Plev fields, specify limits on the capabilities needed to decode the codestream and handle the output. Profiles represent a limit on the required algorithmic features and parameter ranges used in the codestream. The combination of level and sublevel defines a lower bound on the required throughput for a decoder in respectively the image (or decoded) domain and the codestream (or coded) domain. The actual defined profiles and level/sublevels, along with the associated values for the Ppih and Plev fields, are defined in [ISO21122-2].

The colour specification box indicates the colour primaries, transfer characteristics, matrix coefficients and video full range flag needed to specify the colour space of the video stream.

### 3.4. JPEG XS Frame

The concatenation of a video support box, a colour specification box, and a JPEG XS codestream forms a JPEG XS picture segment.

In the case of a progressive video stream, each JPEG XS frame consists of one single JPEG XS picture segment.

In the case of an interlaced video stream, each JPEG XS frame is made of two concatenated JPEG XS picture segments. The codestream of each picture segment corresponds exclusively to one of the two fields of the interlaced frame. Both picture segments SHALL contain identical boxes (i.e. concatenation of the video support box and the colour specification box is byte exact the same for both picture segments of the frame).

Note that the interlaced mode, as signaled by the frat field [ISO21122-3] in the video support box, indicates either progressive, interlaced top-field first, or interlaced bottom-field first mode. Thus, in the case of interlaced content, its value SHALL also be identical in both picture segments.

## 4. RTP Payload Format

This section specifies the payload format for JPEG XS streams over the Real-time Transport Protocol (RTP) [RFC3550].

In order to be transported over RTP, each JPEG XS stream is transported in a distinct RTP stream, identified by a distinct Synchronization source (SSRC) [RFC3550].

A JPEG XS stream is divided into Application Data Units (ADUs), each ADU corresponding to a single JPEG XS frame.

### 4.1. RTP packetization

An ADU is made of several packetization units. If a packetization unit is bigger than the maximum size of a RTP packet payload, the unit is split into multiple RTP packet payloads, as illustrated in Figure 1. As seen there, each packet SHALL contain (part of) one and only one packetization unit. A packetization unit may extend over multiple packets. The payload of every packet SHALL have the same size (based e.g. on the Maximum Transfer Unit of the network), except (possibly) the last packet of a packetization unit. The boundaries of a packetization unit SHALL coincide with the boundaries of the payload of a packet (excluding the payload header), i.e. the first (resp. last) byte of the packetization unit SHALL be the first (resp. last) byte of the payload (excluding its header).

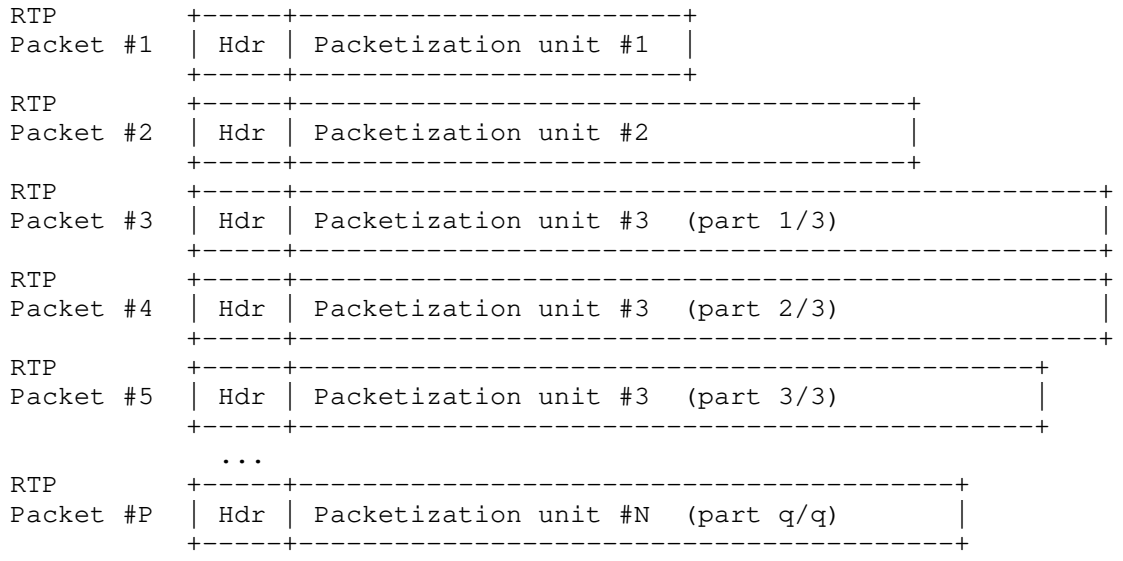



Figure 1: Example of ADU packetization

There are two different packetization modes defined for this RTP payload format.

1. Codestream packetization mode: in this mode, the packetization unit SHALL be the entire JPEG XS picture segment (i.e. codestream preceeded by boxes). This means that a progressive frame will have a single packetization unit, while an interlaced frame will have two. The progressive case is illustrated in Figure 2.

2. Slice packetization mode: in this mode, the packetization unit SHALL be the slice, i.e. there SHALL be data from no more than one slice per RTP packet. The first packetization unit SHALL be made of the JPEG XS header segment (i.e. the concatenation of the VS box, the CS box and the JPEG XS codestream header). This first unit is then followed by successive units, each containing one and only one slice. The packetization unit containing the last slice of a JPEG XS codestream SHALL also contain the EOC marker immediately following this last slice. This is illustrated in Figure 3. In the case of an interlaced frame, the JPEG XS header segment of the second field SHALL be in its own packetization unit.

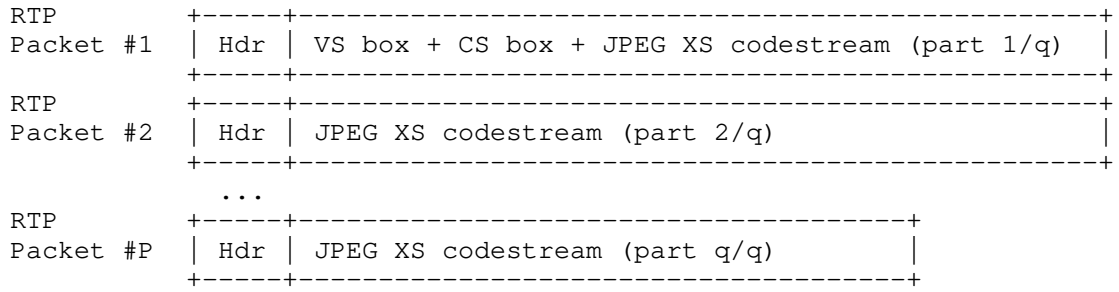



Figure 2: Example of codestream packetization mode

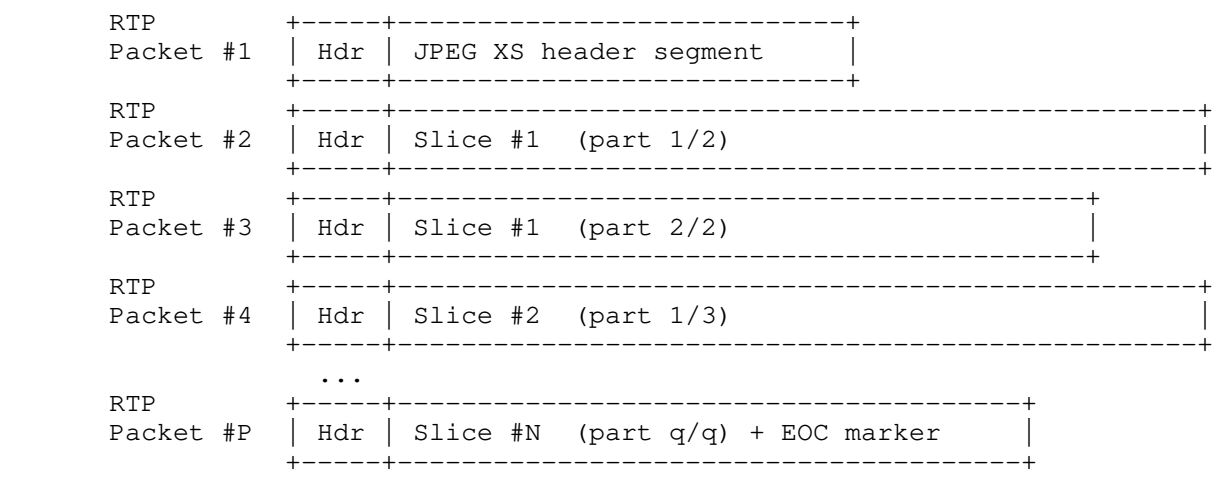



Figure 3: Example of slice packetization mode

Due to the constant bit-rate of JPEG XS, the codestream packetization mode guarantees that a JPEG XS RTP stream will produce a constant number of bytes per frame, and a constant number of RTP packets per frame. To reach the same guarantee with the slice packetization mode, an additional mechanism is required. This can involve a constraint at the rate allocation stage in the JPEG XS encoder to impose a constant bit-rate at the slice level, the usage of padding data, or the insertion of empty RTP packets (i.e. a RTP packet whose payload data is empty).

## 4.2. RTP Header Usage

The format of the RTP header is specified in [RFC3550] and reprinted in Figure 4 for convenience. This RTP payload format uses the fields of the header in a manner consistent with that specification.

The RTP payload (and the settings for some RTP header bits) for packetization units are specified in Section 4.3.

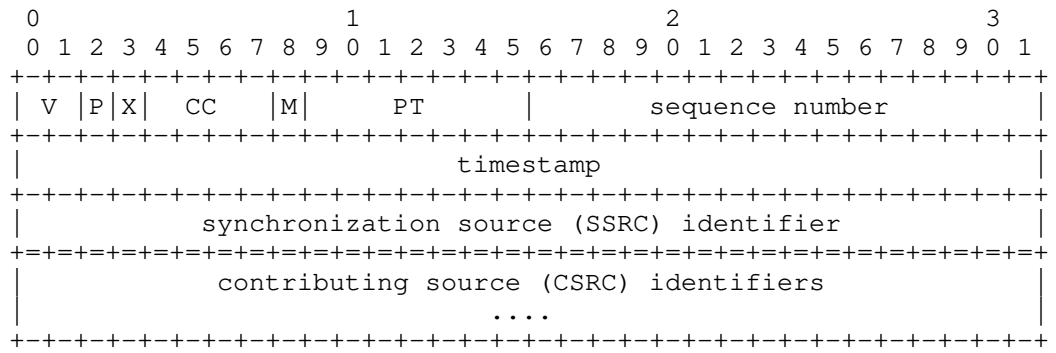



Figure 4: RTP header according to RFC 3550

The version (V), padding (P), extension (X), CSRC count (CC), sequence number, synchronization source (SSRC) and contributing source (CSRC) fields follow their respective definitions in [RFC3550].

The remaining RTP header information to be set according to this RTP payload format is set as follows:

Marker (M) [1 bit]:

If progressive scan video is being transmitted, the marker bit denotes the end of a video frame. If interlaced video is being transmitted, it denotes the end of the field. The marker bit SHALL be set to 1 for the last packet of the video frame/field. It SHALL be set to 0 for all other packets.

Payload Type (PT) [7 bits]:

A dynamically allocated payload type field that designates the payload as JPEG XS video.

Timestamp [32 bits]:

The RTP timestamp is set to the sampling timestamp of the content. A 90 kHz clock rate SHALL be used.

As specified in [RFC3550] and [RFC4175], the RTP timestamp designates the sampling instant of the first octet of the frame to which the RTP packet belongs. Packets SHALL NOT include data from multiple frames, and all packets belonging to the same frame SHALL have the same timestamp. Several successive RTP packets will consequently have equal timestamps if they belong to the same frame (that is until the marker bit is set to 1, marking the last

packet of the frame), and the timestamp is only increased when a new frame begins.

If the sampling instant does not correspond to an integer value of the clock, the value SHALL be truncated to the next lowest integer, with no ambiguity.

## 4.3. Payload Header Usage

The first four bytes of the payload of an RTP packet in this RTP payload format are referred to as the payload header. Figure 5 illustrates the structure of this payload header.

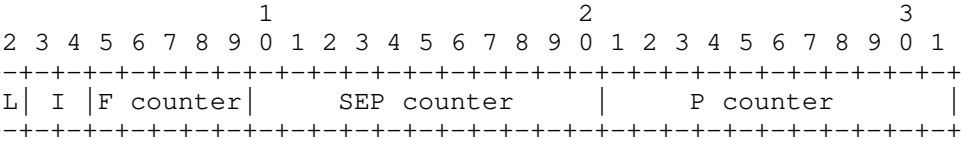



Figure 5: Payload header

The payload header consists of the following fields:

Transmission mode (T) [1 bit]:

The T bit is set to indicate that packets are sent sequentially by the transmitter. This information allows a receiver to dimension its input buffer(s) accordingly. If T=0, nothing can be assumed about the transmission order and packets may be sent out-of-order by the transmitter. If T=1, packets SHALL be sent sequentially by the transmitter.

pacKetization mode (K) [1 bit]:

The K bit is set to indicate which packetization mode is used. K=0 indicates codestream packetization mode, while K=1 indicates slice packetization mode. In the case that the Transmission mode (T) is set to 0, the slice packetization mode SHALL be used and K SHALL be set to 1.

Last (L) [1 bit]:

The L bit is set to indicate the last packet of a packetization unit. As the end of the frame also ends the packet containing the last unit of the frame, the L bit is set whenever the M bit is set. If codestream packetization mode is used, L bit and M bit are equivalent.

Interlaced information (I) [2 bit]:

These 2 bits are used to indicate how the JPEG XS frame is scanned (progressive or interlaced). In case of an interlaced frame, they also indicate which JPEG XS picture segment the payload is part of (first or second).

00: The payload is progressively scanned.

01: Reserved for future use.

10: The payload is part of the first JPEG XS picture segment of an interlaced video frame. The height specified in the included JPEG XS codestream header is half of the height of the entire displayed image.

11: The payload is part of the second JPEG XS picture segment of an interlaced video frame. The height specified in the included JPEG XS codestream header is half of the height of the entire displayed image.

F counter [5 bits]:

The frame (F) counter identifies the frame number modulo 32 to which a packet belongs. Frame numbers are incremented by 1 for each frame transmitted. The frame number, in addition to the timestamp, may help the decoder manage its input buffer and bring packets back into their natural order.

SEP counter [11 bits]:

The Slice and Extended Packet (SEP) counter is used differently depending on the packetization mode.

* In the case of codestream packetization mode (K=0), this counter resets whenever the Packet counter resets (see hereunder), and increments by 1 whenever the Packet counter overruns.

* In the case of slice packetization mode (K=1), this counter identifies the slice modulo 2047 to which the packet contributes. If the data belongs to the JPEG XS header segment, this field SHALL have its maximal value, namely 2047 = 0x07ff. Otherwise, it is the slice index modulo 2047. Slice indices are counted from 0 (corresponding to the top of the frame).

P counter [11 bits]:

The packet (P) counter identifies the packet number modulo 2048 within the current packetization unit. It is set to 0 at the start of the packetization unit and incremented by 1 for every subsequent packet (if any) belonging to the same unit. Practically, if codestream packetization mode is enabled, this field counts the packets within a JPEG XS picture segment and is extended by the SEP counter when it overruns. If slice packetization mode is enabled, this field counts the packets within a slice or within the JPEG XS header segment.

### 4.4. Payload Data

The payload data of a JPEG XS RTP stream consists of a concatenation of multiple JPEG XS frames. Within the RTP stream, all of the video support boxes and all of the colour specification boxes SHALL retain their respective layouts for each JPEG XS frame. Thus, each video support box in the RTP stream SHALL define the same sub boxes. The effective values in the boxes are allowed to change under the condition that their relative byte offsets SHALL NOT change.

Each JPEG XS frame is the concatenation of one or more packetization unit(s), as explained in Section 4.1. Figure 6 depicts this layout for a progressive frame in the codestream packetization mode, Figure 7 depicts this layout for an interlaced frame in the codestream packetization mode, Figure 8 depicts this layout for a progressive frame in the slice packetization mode and Figure 9 depicts this layout for an interlaced frame in the slice packetization mode. The Frame counter value is not indicated because the value is constant for all packetization units of a given frame.

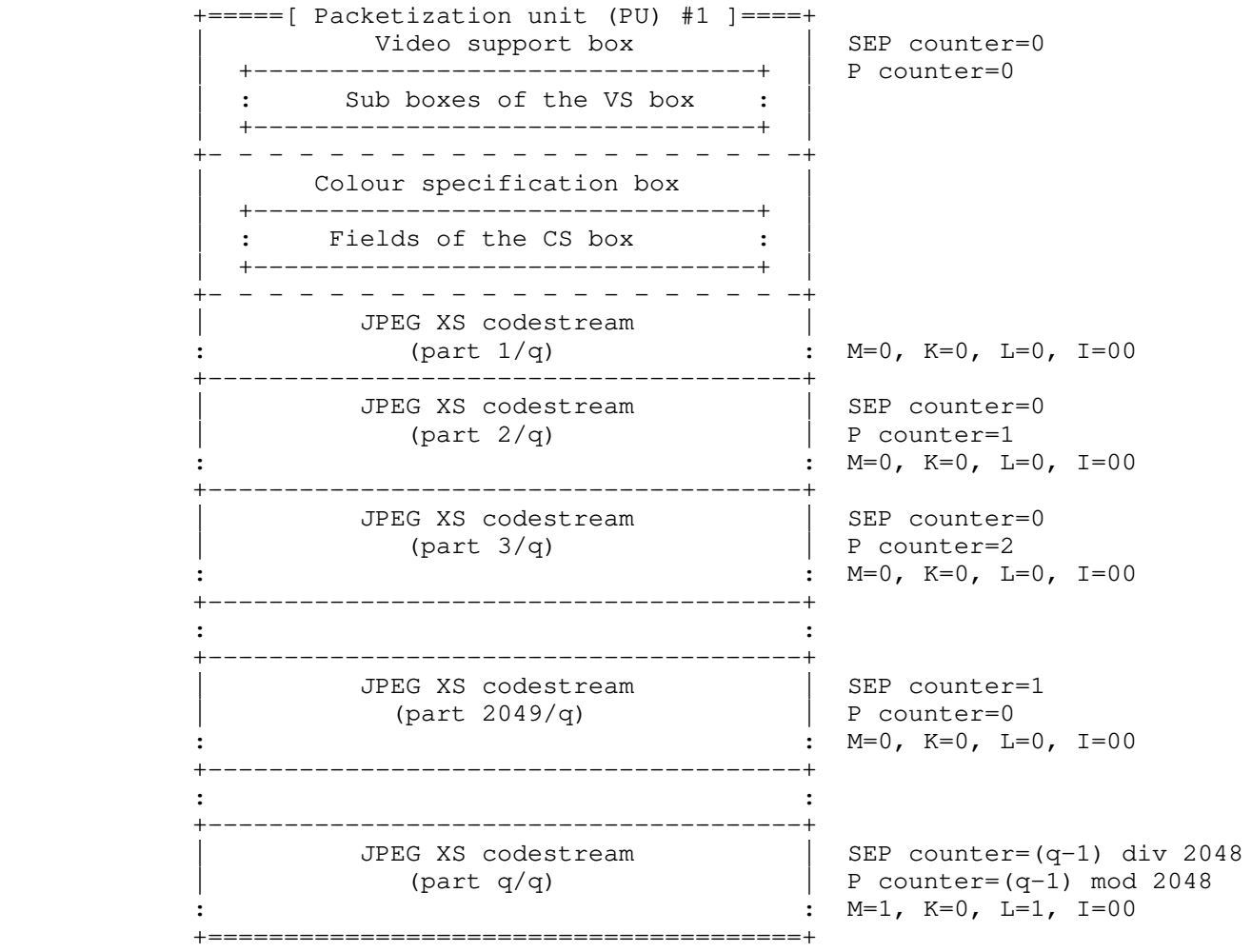



Figure 6: Example of JPEG XS Payload Data (codestream packetization mode, progressive frame)

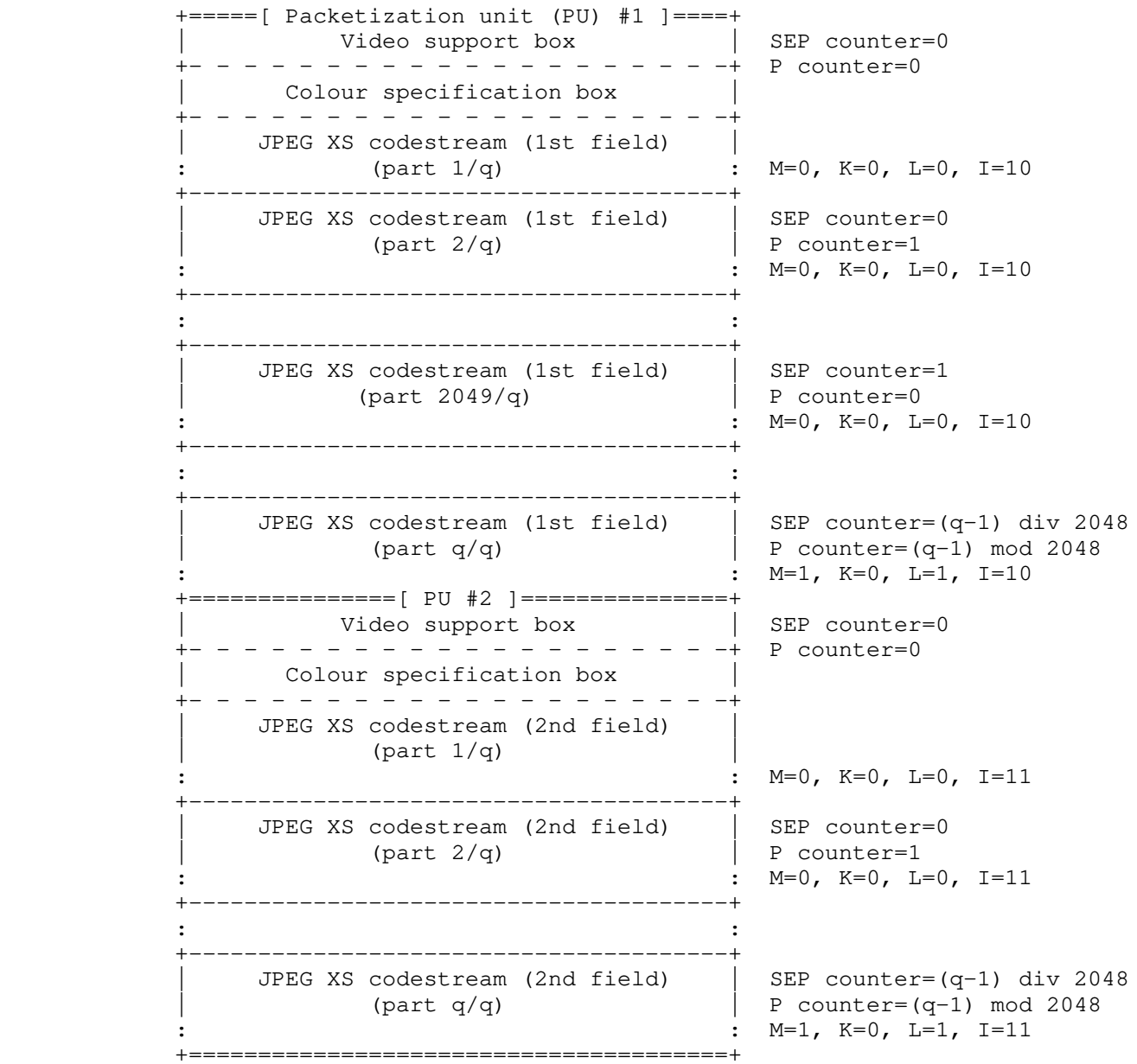



Figure 7: Example of JPEG XS Payload Data (codestream packetization mode, interlaced frame)

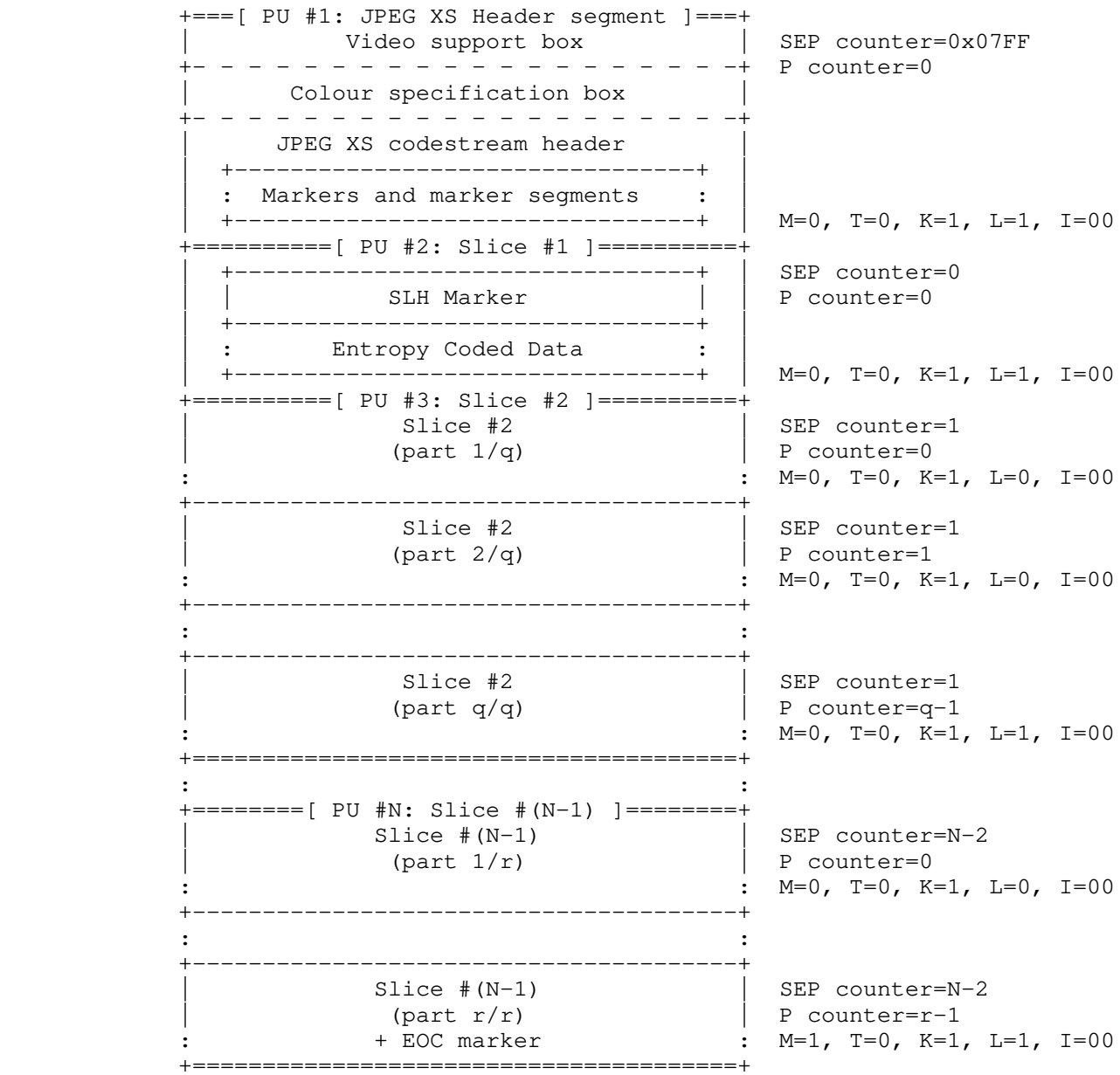
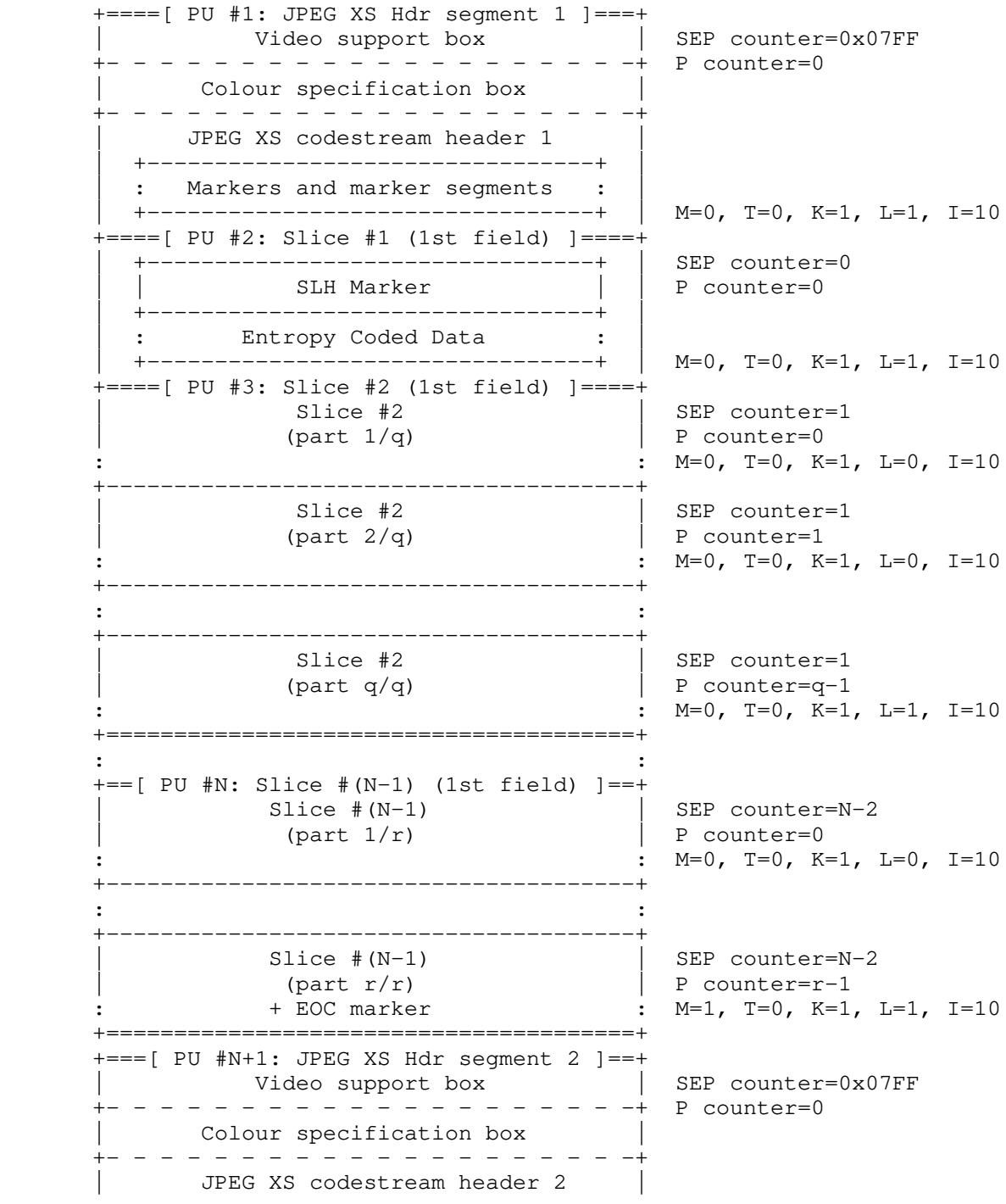



Figure 8: Example of JPEG XS Payload Data (slice packetization mode, progressive frame)


PU #1: JPEG XS Hdr segment 1
Video support box
SEP counter=0x07FF
P counter=0
Colour specification box
JPEG XS codestream header 1
Markers and marker segments
M=0, T=0, K=1, L=1, I=10
PU #2: Slice #1 (1st field)
SEP counter=0
SLH Marker
P counter=0
Entropy Coded Data
M=0, T=0, K=1, L=1, I=10
PU #3: Slice #2 (1st field)
Slice #2
(part 1/q)
SEP counter=1
P counter=0
M=0, T=0, K=1, L=0, I=10
Slice #2
(part 2/q)
SEP counter=1
P counter=1
M=0, T=0, K=1, L=0, I=10
Slice #2
(part q/q)
SEP counter=1
P counter=q-1
M=0, T=0, K=1, L=1, I=10
PU #N: Slice #(N-1) (1st field)
Slice #(N-1)
(part 1/r)
SEP counter=N-2
P counter=0
M=0, T=0, K=1, L=0, I=10
Slice #(N-1)
(part r/r)
SEP counter=N-2
P counter=r-1
+ EOC marker
M=1, T=0, K=1, L=1, I=10
PU #N+1: JPEG XS Hdr segment 2
Video support box
SEP counter=0x07FF
P counter=0
Colour specification box
JPEG XS codestream header 2

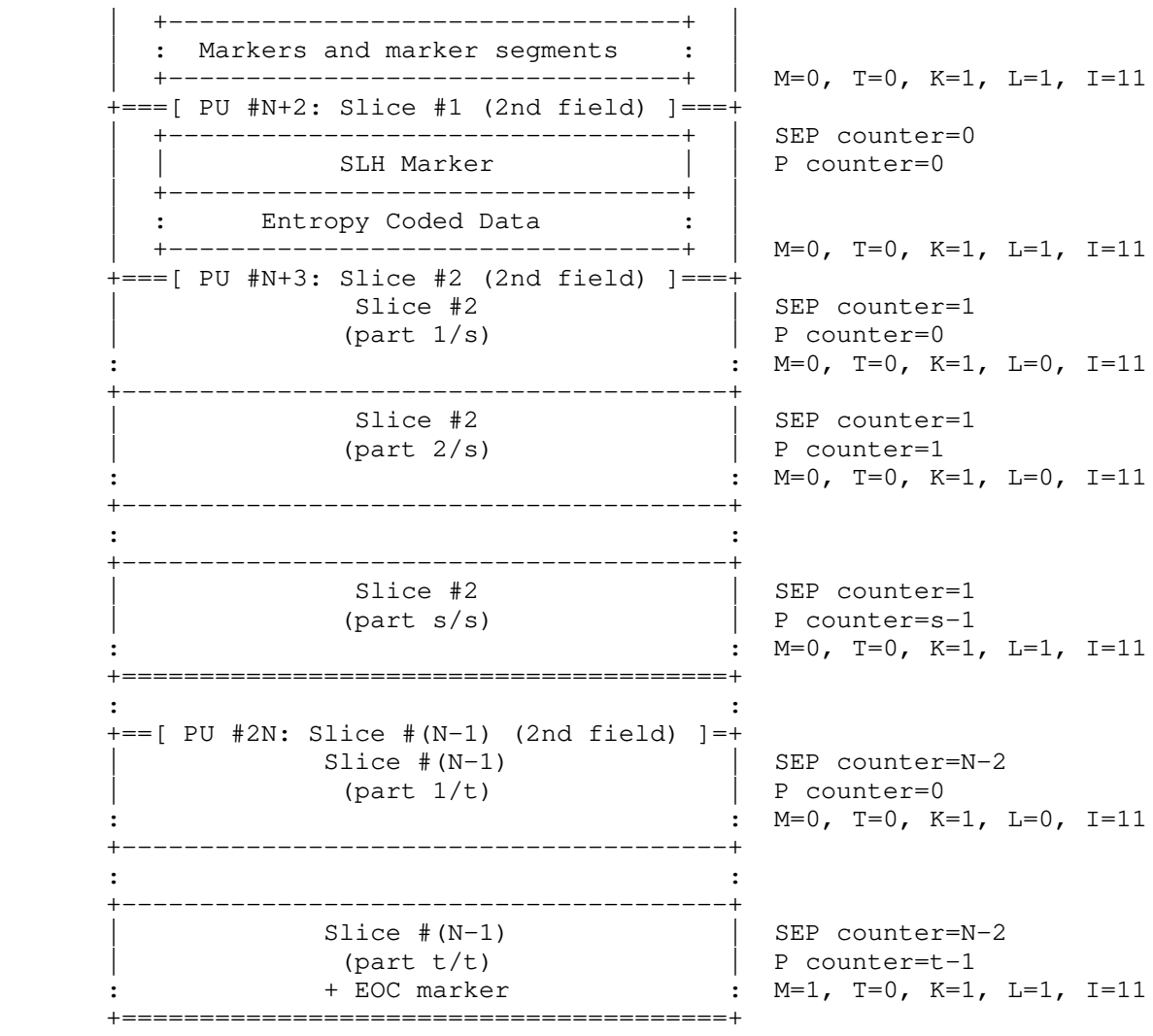



Figure 9: Example of JPEG XS Payload Data (slice packetization mode, interlaced frame)

## 5. Traffic Shaping and Delivery Timing

In order to facilitate proper synchronization between senders and receivers it is RECOMMENDED to implement traffic shaping and delivery timing in accordance with the Network Compatibility Model compliance definitions specified in [SMPTE-ST2110-21] for either Narrow Linear Senders (Type NL) or Wide Senders (Type W). In such case, the session description SHALL include a format-specific parameter of

either TP=2110TPNL or TP=2110TPW to indicate compliance with Type NL or Type W respectively. The actual applied traffic shaping and timing delivery mechanism is outside the scope of this memo and does not influence the payload packetization.

NOTE: The Virtual Receiver Buffer Model compliance definitions of [SMPTE-ST2110-21] do not apply.

## 6. Congestion Control Considerations

Congestion control for RTP SHALL be used in accordance with [RFC3550], and with any applicable RTP profile: e.g., [RFC3551]. An additional requirement if best-effort service is being used is users of this payload format SHALL monitor packet loss to ensure that the packet loss rate is within acceptable parameters. Circuit Breakers [RFC8083] is an update to RTP [RFC3550] that defines criteria for when one is required to stop sending RTP Packet Streams and applications implementing this standard SHALL comply with it. [RFC8085] provides additional information on the best practices for applying congestion control to UDP streams.

## 7. Payload Format Parameters

This section specifies the required and optional parameters of the payload format or the RTP stream. The information signaled by the any of the optional parameters is also present in the payload data, namely in the payload header or in the JPEG XS header segment [ISO21122-1] [ISO21122-3]. When provided, their respective values SHALL be consistent with the payload. The sole purpose of the optional parameters is to facilitate access to the RTP stream metadata.

### 7.1. Media Type Registration

This registration is done using the template defined in [RFC6838] and following [RFC4855].

The receiver SHALL ignore any unrecognized parameter.

Type name: video

Subtype name: jxsv

Required parameters:

rate: The RTP timestamp clock rate. Applications using this payload format SHALL use a value of 90000.

transmode: This parameter specifies the configured transmission mode as defined by the Transmission mode (T) bit in the payload header of Section 4.3. This value SHALL be equal to the T bit value configured in the RTP stream (i.e. 0 for out-of-order-allowed or 1 for sequential).

Optional parameters:

packetmode: This parameter specifies the configured packetization mode as defined by the pacKetization mode (K) bit in the payload header of Section 4.3. If specified, this value SHALL be equal to the K bit value configured in the RTP stream (i.e. 0 for codestream or 1 for slice).

profile: The JPEG XS profile [ISO21122-2] in use. Any white space in the profile name SHALL be replaced by a dash (-). Examples are 'Main-444.12' or 'High-444.12'.

level: The JPEG XS level [ISO21122-2] in use. Any white space in the level name SHALL be replaced by a dash (-). Examples are '2k-1' or '4k-2'.

sublevel: The JPEG XS sublevel [ISO21122-2] in use. Any white space in the sublevel name SHALL be replaced by a dash (-). Examples are 'Sublev3bpp' or 'Sublev6bpp'.

depth: Determines the number of bits per sample. This is an integer with typical values including 8, 10, 12, and 16.

width: Determines the number of pixels per line. This is an integer between 1 and 32767.

height: Determines the number of lines per frame. This is an integer between 1 and 32767.

exactframerate: Signals the frame rate in frames per second. Integer frame rates SHALL be signaled as a single decimal number (e.g. "25") whilst non-integer frame rates SHALL be signaled as a ratio of two integer decimal numbers separated by a "forward-slash" character (e.g. "30000/1001"), utilizing the numerically smallest numerator value possible.

interlace: If this parameter name is present, it indicates that the video is interlaced, or that the video is Progressive segmented Frame (PsF). If this parameter name is not present, the progressive video format SHALL be assumed.

segmented: If this parameter name is present, and the interlace parameter name is also present, then the video is a Progressive segmented Frame (PsF). Signaling of this parameter without the interlace parameter is forbidden.

sampling: Signals the colour difference signal sub-sampling structure.

Signals utilizing the non-constant luminance Y'C'B C'R signal format of Recommendation ITU-R BT.601-7, Recommendation ITU-R BT.709-6, Recommendation ITU-R BT.2020-2, or Recommendation ITU-R BT.2100 SHALL use the appropriate one of the following values for the Media Type Parameter "sampling":

```
YCbCr-4:4:4    (4:4:4 sampling)
YCbCr-4:2:2    (4:2:2 sampling)
YCbCr-4:2:0    (4:2:0 sampling)
```

Signals utilizing the Constant Luminance Y'C C'BC C'RC signal format of Recommendation ITU-R BT.2020-2 SHALL use the appropriate one of the following values for the Media Type Parameter "sampling":

```
CLYCbCr-4:4:4  (4:4:4 sampling)
CLYCbCr-4:2:2  (4:2:2 sampling)
CLYCbCr-4:2:0  (4:2:0 sampling)
```

Signals utilizing the constant intensity I CT CP signal format of Recommendation ITU-R BT.2100 SHALL use the appropriate one of the following values for the Media Type Parameter "sampling":

```
ICtCp-4:4:4    (4:4:4 sampling)
ICtCp-4:2:2    (4:2:2 sampling)
ICtCp-4:2:0    (4:2:0 sampling)
```

Signals utilizing the 4:4:4 R' G' B' or RGB signal format (such as that of Recommendation ITU-R BT.601, Recommendation ITU-R BT.709, Recommendation ITU-R BT.2020, Recommendation ITU-R BT.2100, SMPTE ST 2065-1 or ST 2065-3) SHALL use the following value for the Media Type Parameter sampling.

```
RGB            (RGB or R' G' B' samples)
```

Signals utilizing the 4:4:4 X' Y' Z' signal format (such as defined in SMPTE ST 428-1) SHALL use the following value for the Media Type Parameter sampling.

XYZ (X' Y' Z' samples)

Key signals as defined in SMPTE RP 157 SHALL use the value key for the Media Type Parameter sampling. The Key signal is represented as a single component.

KEY (Samples of the key signal)

Signals utilizing a colour sub-sampling other than what is defined here SHALL use the following value for the Media Type Parameter sampling.

UNSPECIFIED (Sampling signaled by the payload.)

colorimetry: Specifies the system colorimetry used by the image samples. Valid values and their specification are:

| | |
|---|---|
| BT601-5 | ITU-R Recommendation BT.601-5. |
| BT709-2 | ITU-R Recommendation BT.709-2. |
| SMPTE240M | SMPTE ST 240M. |
| BT601 | ITU-R Recommendation BT.601-7. |
| BT709 | ITU-R Recommendation BT.709-6. |
| BT2020 | ITU-R Recommendation BT.2020-2. |
| BT2100 | ITU-R Recommendation BT.2100 Table 2 titled "System colorimetry". |
| ST2065-1 | SMPTE ST 2065-1 Academy Color Encoding Specification (ACES). |
| ST2065-3 | SMPTE ST 2065-3 Academy Density Exchange Encoding (ADX). |
| XYZ | ISO/IEC 11664-1, section titled "1931 Observer". |
| UNSPECIFIED | Colorimetry is signaled in the payload by the color specification box of [ISO21122-3], or it must be manually coordinated between sender and receiver. |

Signals utilizing the Recommendation ITU-R BT.2100 colorimetry SHOULD also signal the representational range using the optional parameter RANGE defined below. Signals utilizing the UNSPECIFIED colorimetry might require manual coordination between the sender and the receiver.

TCS: Transfer Characteristic System. This parameter specifies the transfer characteristic system of the image samples. Valid values and their specification are:

| | |
|---|---|
| SDR | Standard Dynamic Range video streams that utilize the OETF of ITU-R Recommendation BT.709 or ITU-R Recommendation BT.2020. Such streams SHALL be assumed to target the EOTF specified in ITU-R Recommendation BT.1886. |
| PQ | High dynamic range video streams that utilize the Perceptual Quantization system of ITU-R Recommendation BT.2100. |
| HLG | High dynamic range video streams that utilize the Hybrid Log-Gamma system of ITU-R Recommendation BT.2100. |
| UNSPECIFIED | Video streams whose transfer characteristics are signaled by the payload as specified in [ISO21122-3], or must be manually coordinated between sender and receiver. |

RANGE: This parameter SHOULD be used to signal the encoding range of the sample values within the stream. When paired with ITU Rec BT.2100 colorimetry, this parameter has two allowed values NARROW and FULL, corresponding to the ranges specified in table 9 of ITU Rec BT.2100. In any other context, this parameter has three allowed values: NARROW, FULLPROTECT, and FULL, which correspond to the ranges specified in SMPTE RP 2077. In the absence of this parameter, and for all but the UNSPECIFIED colorimetry, NARROW SHALL be the assumed value. When paired with the UNSPECIFIED colorimetry, FULL SHALL be the default assumed value.

Encoding considerations:
This media type is framed in RTP and contains binary data; see Section 4.8 in [RFC6838].

Security considerations:
Please see the Security Considerations (Section 9) of RFC XXXX.

Interoperability considerations:
None.

Published specification:
See RFC XXXX and its References section.

Applications that use this media type:
For example: SMPTE ST 2110, Video over IP, Video conferencing, Broadcast applications.

Fragment identifier considerations:
N/A.

Additional information:
None.

Person & email address to contact for further information:
S. Lugan <rtp@intopix.com> and Th. Richter <jpeg-xs-techsupport@iis.fraunhofer.de>.

Intended usage:
COMMON

Restrictions on usage:
This media type depends on RTP framing, and hence is only defined for transfer via RTP [RFC3550].

Author:
See the Authors' Addresses section of RFC XXXX.

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

### 7.2. Mapping to SDP

#### 7.2.1. General

A Session Description Protocol (SDP) [RFC8866] media description SHALL be created for each RTP stream.

The information carried in the media type specification of Section 7.1 has a specific mapping to the SDP fields, used to describe RTP sessions. This information is redundant with the information found in the payload data (namely, in the JPEG XS header segment) and SHALL be consistent with it. In case of discrepancy between parameter values found in the payload data and in the SDP fields, the values from the payload data SHALL prevail.

The receiver SHALL ignore any unrecognized parameter.

#### 7.2.2. Media type and subtype

The media type ("video") goes in SDP "m=" as the media name.

The media subtype ("jxsv") goes in SDP "a=rtpmap" as the encoding name, followed by a slash ("/") and the required parameter "rate" corresponding to the RTP timestamp clock rate (which for the payload format defined in this document SHALL be 90000). The required parameter "transmode" and the additional optional parameters go in the SDP "a=fmtp" attribute by copying them directly from the MIME

media type string as a semicolon-separated list of parameter=value pairs.

An example SDP mapping for JPEG XS video is as follows:

```
m=video 30000 RTP/AVP 112
a=rtpmap:112 jxsv/90000
a=fmtp:112 transmode=1;sampling=YCbCr-4:2:2;width=1920;
           height=1080;depth=10;colorimetry=BT709;TCS=SDR;
           RANGE=FULL;TP=2110TPNL
```

In this example, a JPEG XS RTP stream is being sent to UDP destination port 30000, with an RTP dynamic payload type of 112 and a media clock rate of 90000 Hz. Note that the "a=fmtp:" line has been wrapped to fit this page, and will be a single long line in the SDP file. This example includes the TP parameter (as specified in Section 5).

### 7.2.3. Offer/Answer Considerations

When XS is offered using An Offer/Answer Model with Session Description Protocol (SDP) [RFC3264] for negotiation for unicast usage, the following limitations and rules apply:

All parameters are declarative, i.e. apply only to media sent by the entity that generated the SDP. Thus, a declarative parameter in an offer applies to media sent by the offeror, whereas a declarative parameter in an answer applies to media sent by the answerer. All parameters must be supported by both sides, i.e. the answerer SHALL either maintain all parameters or remove the media format (payload type) completely if one or more of the parameter values are not supported.

## 8. IANA Considerations

The IANA is requested to register the media type registration "video/jxsv" as specified in Section 7.1. The media type is also requested to be added to the IANA registry for "RTP Payload Format MIME types" <https://www.iana.org/assignments/rtp-parameters>.

## 9. Security Considerations

RTP packets using the payload format defined in this memo are subject to the security considerations discussed in [RFC3550] and in any applicable RTP profile such as RTP/AVP [RFC3551], RTP/AVPF [RFC4585], RTP/SAVP [RFC3711], or RTP/SAVPF [RFC5124]. This implies that confidentiality of the media streams is achieved by encryption.

However, as "Securing the RTP Framework: Why RTP Does Not Mandate a Single Media Security Solution" [RFC7202] discusses, it is not an RTP payload format's responsibility to discuss or mandate what solutions are used to meet the basic security goals like confidentiality, integrity, and source authenticity for RTP in general. This responsibility lies on anyone using RTP in an application. They can find guidance on available security mechanisms and important considerations in "Options for Securing RTP Sessions" [RFC7201]. Applications SHOULD use one or more appropriate strong security mechanisms.

This payload format and the JPEG XS encoding do not exhibit any substantial non-uniformity, either in output or in complexity to perform the decoding operation and thus are unlikely to pose a denial-of-service threat due to the receipt of pathological datagrams.

This payload format and the JPEG XS encoding do not contain code that is executable.

It is important to note that HD or UHDTV JPEG XS-encoded video can have significant bandwidth requirements (typically more than 1 Gbps for ultra high-definition video, especially if using high framerate). This is sufficient to cause potential for denial-of-service if transmitted onto most currently available Internet paths.

Accordingly, if best-effort service is being used, users of this payload format SHALL monitor packet loss to ensure that the packet loss rate is within acceptable parameters. Packet loss is considered acceptable if a TCP flow across the same network path, and experiencing the same network conditions, would achieve an average throughput, measured on a reasonable timescale, that is not less than the RTP flow is achieving. This condition can be satisfied by implementing congestion control mechanisms to adapt the transmission rate (or the number of layers subscribed for a layered multicast session), or by arranging for a receiver to leave the session if the loss rate is unacceptably high.

This payload format may also be used in networks that provide quality-of-service guarantees. If enhanced service is being used, receivers SHOULD monitor packet loss to ensure that the service that was requested is actually being delivered. If it is not, then they SHOULD assume that they are receiving best-effort service and behave accordingly.

## 10. Acknowledgments

The authors would like to thank the following people for their valuable contributions to this memo: Arnaud Germain, Alexandre Willeme, Gael Rouvroy, Siegfried Foessel, and Jean-Baptise Lorent.

## 11. RFC Editor Considerations

Note to RFC Editor: This section may be removed after carrying out all the instructions of this section.

RFC XXXX is to be replaced by the RFC number this specification receives when published.

## 12. References

### 12.1. Normative References

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International Organization for Standardization (ISO) - International Electrotechnical Commission (IEC), "Information technology - JPEG XS low-latency lightweight image coding system - Part 2: Profiles and buffer models", ISO/IEC IS 21122-2.

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### 12.2. Informative References

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# RTP Payload Format for VP9 Video
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## Abstract

This specification describes an RTP payload format for the VP9 video codec. The payload format has wide applicability, as it supports applications from low bit-rate peer-to-peer usage, to high bit-rate video conferences. It includes provisions for temporal and spatial scalability.

## Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

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This Internet-Draft will expire on 8 November 2021.

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## 1. Introduction

This specification describes an RTP payload specification applicable to the transmission of video streams encoded using the VP9 video codec [VP9-BITSTREAM]. The format described in this document can be used both in peer-to-peer and video conferencing applications.

The VP9 video codec was developed by Google, and is the successor to its earlier VP8 [RFC6386] codec. Above the compression improvements and other general enhancements above VP8, VP9 is also designed in a way that allows spatially-scalable video encoding.

## 2. Conventions, Definitions and Acronyms

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in BCP 14 [RFC2119] [RFC8174] when, and only when, they appear in all capitals, as shown here.

## 3. Media Format Description

The VP9 codec can maintain up to eight reference frames, of which up to three can be referenced by any new frame.

VP9 also allows a frame to use another frame of a different resolution as a reference frame. (Specifically, a frame may use any references whose width and height are between 1/16th that of the current frame and twice that of the current frame, inclusive.) This allows internal resolution changes without requiring the use of key frames.

These features together enable an encoder to implement various forms of coarse-grained scalability, including temporal, spatial and quality scalability modes, as well as combinations of these, without the need for explicit scalable coding tools.

Temporal layers define different frame rates of video; spatial and quality layers define different and possibly dependent representations of a single input frame. Spatial layers allow a frame to be encoded at different resolutions, whereas quality layers allow a frame to be encoded at the same resolution but at different qualities (and thus with different amounts of coding error). VP9 supports quality layers as spatial layers without any resolution changes; hereinafter, the term "spatial layer" is used to represent both spatial and quality layers.

This payload format specification defines how such temporal and spatial scalability layers can be described and communicated.

Temporal and spatial scalability layers are associated with non-negative integer IDs. The lowest layer of either type has an ID of 0, and is sometimes referred to as the "base" temporal or spatial layer.

Layers are designed (and MUST be encoded) such that if any layer, and all higher layers, are removed from the bitstream along either of the two dimensions, the remaining bitstream is still correctly decodable.

For terminology, this document uses the term "frame" to refer to a single encoded VP9 frame for a particular resolution/quality, and "picture" to refer to all the representations (frames) at a single instant in time. A picture thus consists of one or more frames, encoding different spatial layers.

Within a picture, a frame with spatial layer ID equal to SID, where SID > 0, can depend on a frame of the same picture with a lower spatial layer ID. This "inter-layer" dependency can result in additional coding gain compared to the case where only traditional "inter-picture" dependency is used, where a frame depends on previously coded frame in time. For simplicity, this payload format assumes that, within a picture and if inter-layer dependency is used, a spatial layer SID frame can depend only on the immediately previous spatial layer SID-1 frame, when S > 0. Additionally, if inter-picture dependency is used, a spatial layer SID frame is assumed to only depend on a previously coded spatial layer SID frame.

Given above simplifications for inter-layer and inter-picture dependencies, a flag (the D bit described below) is used to indicate whether a spatial layer SID frame depends on the spatial layer SID-1 frame. Given the D bit, a receiver only needs to additionally know the inter-picture dependency structure for a given spatial layer frame in order to determine its decodability. Two modes of describing the inter-picture dependency structure are possible: "flexible mode" and "non-flexible mode". An encoder can only switch between the two on the first packet of a key frame with temporal layer ID equal to 0.

In flexible mode, each packet can contain up to 3 reference indices, which identify all frames referenced by the frame transmitted in the current packet for inter-picture prediction. This (along with the D bit) enables a receiver to identify if a frame is decodable or not and helps it understand the temporal layer structure. Since this is signaled in each packet it makes it possible to have very flexible temporal layer hierarchies and patterns which are changing dynamically.

In non-flexible mode, the inter-picture dependency (the reference indices) of a Picture Group (PG) MUST be pre-specified as part of the scalability structure (SS) data. In this mode, each packet has an index to refer to one of the described pictures in the PG, from which the pictures referenced by the picture transmitted in the current packet for inter-picture prediction can be identified.

(Note: A "Picture Group", as used in this document, is not the same thing as a the term "Group of Pictures" as it is traditionally used in video coding, i.e. to mean an independently-decoadable run of pictures beginning with a keyframe.)

The SS data can also be used to specify the resolution of each spatial layer present in the VP9 stream for both flexible and non-flexible modes.

## 4. Payload Format

This section describes how the encoded VP9 bitstream is encapsulated in RTP. To handle network losses usage of RTP/AVPF [RFC4585] is RECOMMENDED. All integer fields in the specifications are encoded as unsigned integers in network octet order.

### 4.1. RTP Header Usage

The general RTP payload format for VP9 is depicted below.

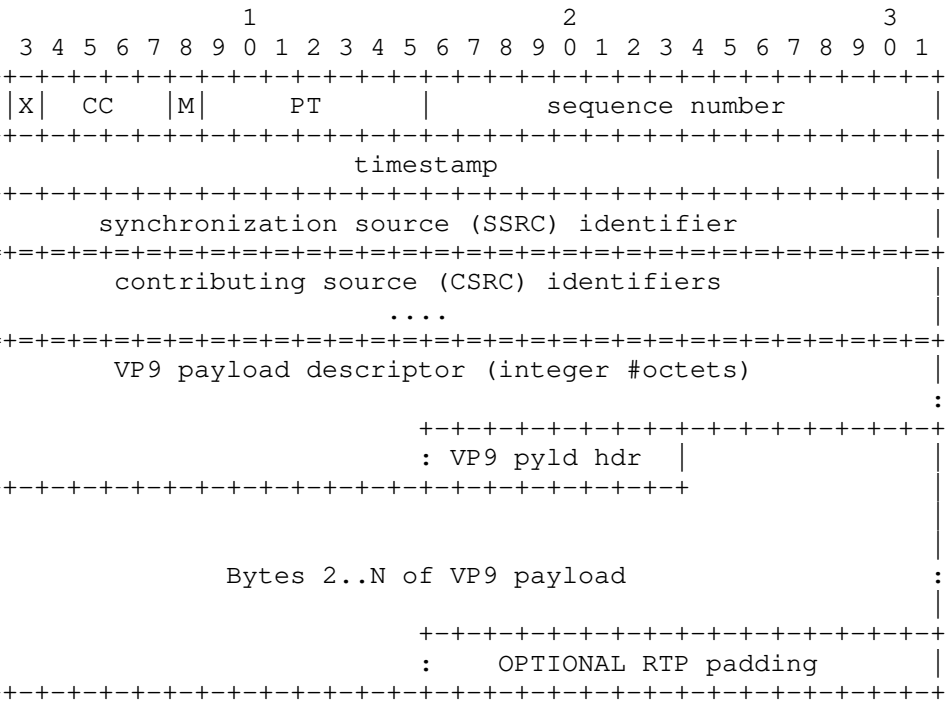

Figure 1

The VP9 payload descriptor will be described in Section 4.2; the VP9 payload header is described in [VP9-BITSTREAM]. OPTIONAL RTP padding MUST NOT be included unless the P bit is set. The figure specifically shows the format for the first packet in a frame. Subsequent packets will not contain the VP9 payload header, and will have later octets in the frame payload.

Marker bit (M): MUST be set to 1 for the final packet of the highest spatial layer frame (the final packet of the picture), and 0 otherwise. Unless spatial scalability is in use for this picture, this will have the same value as the E bit described below. Note this bit MUST be set to 1 for the target spatial layer frame if a stream is being rewritten to remove higher spatial layers.

Payload Type (PT): In line with the policy in Section 3 of [RFC3551], applications using the VP9 RTP payload profile MUST assign a dynamic payload type number to be used in each RTP session and provide a mechanism to indicate the mapping. See Section 6.1 for the mechanism to be used with the Session Description Protocol (SDP) [RFC8866].

Timestamp: The RTP timestamp indicates the time when the input frame was sampled, at a clock rate of 90 kHz. If the input picture is encoded with multiple layer frames, all of the frames of the picture MUST have the same timestamp.

If a frame has the VP9 show_frame field set to 0 (i.e., it is meant only to populate a reference buffer, without being output) its timestamp MAY alternatively be set to be the same as the subsequent frame with show_frame equal to 1. (This will be convenient for playing out pre-encoded content packaged with VP9 "superframes", which typically bundle show_frame==0 frames with a subsequent show_frame==1 frame.) Every frame with show_frame==1, however, MUST have a unique timestamp modulo the 2^32 wrap of the field.

The remaining RTP Fixed Header Fields (V, P, X, CC, sequence number, SSRC and CSRC identifiers) are used as specified in Section 5.1 of [RFC3550].

## 4.2. VP9 Payload Descriptor

In flexible mode (with the F bit below set to 1), the first octets after the RTP header are the VP9 payload descriptor, with the following structure.

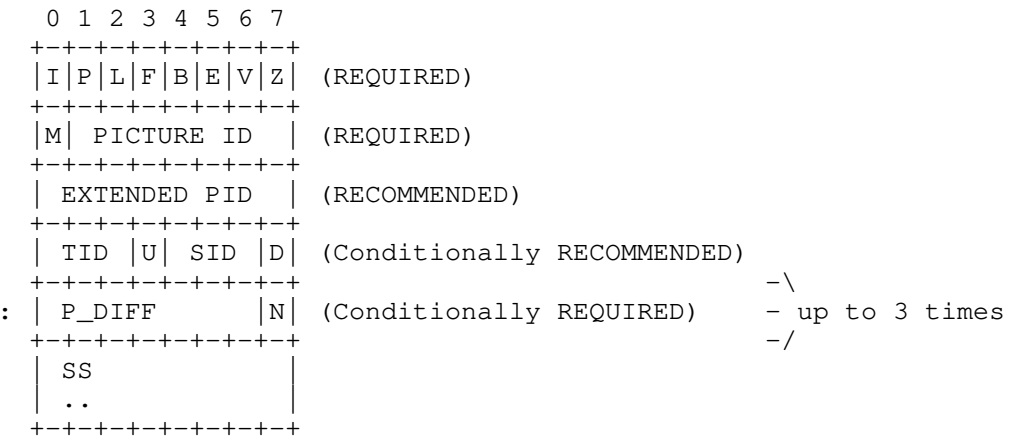


Figure 2

In non-flexible mode (with the F bit below set to 0), The first octets after the RTP header are the VP9 payload descriptor, with the following structure.

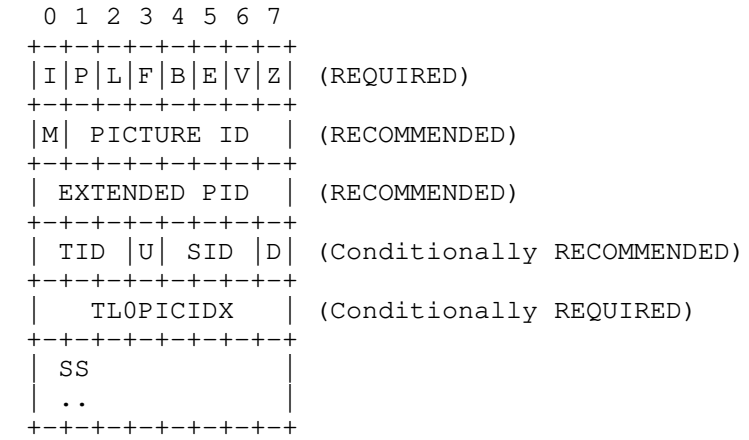


Figure 3

I: Picture ID (PID) present. When set to one, the OPTIONAL PID MUST be present after the mandatory first octet and specified as below. Otherwise, PID MUST NOT be present. If the SS field was present in the stream's most recent start of a keyframe (i.e., non-flexible scalability mode is in use), then the PID MUST also be present in every packet.

P: Inter-picture predicted frame. When set to zero, the frame does not utilize inter-picture prediction. In this case, up-switching to a current spatial layer's frame is possible from directly lower spatial layer frame. P SHOULD also be set to zero when encoding a layer synchronization frame in response to an LRR [I-D.ietf-avtext-lrr] message (see Section 5.3). When P is set to zero, the TID field (described below) MUST also be set to 0 (if present). Note that the P bit does not forbid intra-picture, inter-layer prediction from earlier frames of the same picture, if any.

L: Layer indices present. When set to one, the one or two octets following the mandatory first octet and the PID (if present) is as described by "Layer indices" below. If the F bit (described below) is set to 1 (indicating flexible mode), then only one octet is present for the layer indices. Otherwise if the F bit is set to 0 (indicating non-flexible mode), then two octets are present for the layer indices.

F: Flexible mode. F set to one indicates flexible mode and if the P bit is also set to one, then the octets following the mandatory first octet, the PID, and layer indices (if present) are as described by "Reference indices" below. This MUST only be set to 1 if the I bit is also set to one; if the I bit is set to zero, then this MUST also be set to zero and ignored by receivers. The value of this F bit MUST only change on the first packet of a key picture. A key picture is a picture whose base spatial layer frame is a key frame, and which thus completely resets the encoder state. This packet will have its P bit equal to zero, SID or D bit (described below) equal to zero, and B bit (described below) equal to 1.

B: Start of a frame. MUST be set to 1 if the first payload octet of the RTP packet is the beginning of a new VP9 frame, and MUST NOT be 1 otherwise. Note that this frame might not be the first frame of a picture.

E: End of a frame. MUST be set to 1 for the final RTP packet of a VP9 frame, and 0 otherwise. This enables a decoder to finish decoding the frame, where it otherwise may need to wait for the next packet to explicitly know that the frame is complete. Note that, if spatial scalability is in use, more frames from the same picture may follow; see the description of the M bit above.

V: Scalability structure (SS) data present. When set to one, the OPTIONAL SS data MUST be present in the payload descriptor. Otherwise, the SS data MUST NOT be present.

Z: Not a reference frame for upper spatial layers. If set to 1, indicates that frames with higher spatial layers SID+1 of the current and following pictures do not depend on the current spatial layer SID frame. This enables a decoder which is targeting a higher spatial layer to know that it can safely discard this packet's frame without processing it, without having to wait for the "D" bit in the higher-layer frame (see below).

The mandatory first octet is followed by the extension data fields that are enabled:

M: The most significant bit of the first octet is an extension flag. The field MUST be present if the I bit is equal to one. If set, the PID field MUST contain 15 bits; otherwise, it MUST contain 7 bits. See PID below.

Picture ID (PID): Picture ID represented in 7 or 15 bits, depending on the M bit. This is a running index of the pictures. The field MUST be present if the I bit is equal to one. If M is set to zero, 7 bits carry the PID; else if M is set to one, 15 bits carry the PID in network byte order. The sender may choose between a 7- or 15-bit index. The PID SHOULD start on a random number, and MUST wrap after reaching the maximum ID (0x7f or 0x7fff depending on the index size chosen). The receiver MUST NOT assume that the number of bits in PID stay the same through the session.

In the non-flexible mode (when the F bit is set to 0), this PID is used as an index to the picture group (PG) specified in the SS data below. In this mode, the PID of the key frame corresponds to the first specified frame in the PG. Then subsequent PIDs are mapped to subsequently specified frames in the PG (modulo N_G, specified in the SS data below), respectively.

All frames of the same picture MUST have the same PID value.

Frames (and their corresponding pictures) with the VP9 show_frame field equal to 0 MUST have distinct PID values from subsequent pictures with show_frame equal to 1. Thus, a Picture as defined in this specification is different than a VP9 Superframe.

All frames of the same picture MUST have the same value for show_frame.

Layer indices: This information is optional but RECOMMENDED whenever encoding with layers. For both flexible and non-flexible modes, one octet is used to specify a layer frame's temporal layer ID (TID) and spatial layer ID (SID) as shown both in Figure 2 and Figure 3. Additionally, a bit (U) is used to indicate that the

current frame is a "switching up point" frame. Another bit (D) is used to indicate whether inter-layer prediction is used for the current frame.

In the non-flexible mode (when the F bit is set to 0), another octet is used to represent temporal layer 0 index (TL0PICIDX), as depicted in Figure 3. The TL0PICIDX is present so that all minimally required frames - the base temporal layer frames - can be tracked.

The TID and SID fields indicate the temporal and spatial layers and can help middleboxes and endpoints quickly identify which layer a packet belongs to.

TID: The temporal layer ID of current frame. In the case of non-flexible mode, if PID is mapped to a picture in a specified PG, then the value of TID MUST match the corresponding TID value of the mapped picture in the PG.

U: Switching up point. If this bit is set to 1 for the current picture with temporal layer ID equal to TID, then "switch up" to a higher frame rate is possible as subsequent higher temporal layer pictures will not depend on any picture before the current picture (in coding order) with temporal layer ID greater than TID.

SID: The spatial layer ID of current frame. Note that frames with spatial layer SID > 0 may be dependent on decoded spatial layer SID-1 frame within the same picture. Different frames of the same picture MUST have distinct spatial layer IDs, and frames' spatial layers MUST appear in increasing order within the frame.

D: Inter-layer dependency used. MUST be set to one if and only if the current spatial layer SID frame depends on spatial layer SID-1 frame of the same picture, otherwise MUST set to zero. For the base layer frame (with SID equal to 0), this D bit MUST be set to zero.

TL0PICIDX: 8 bits temporal layer zero index. TL0PICIDX is only present in the non-flexible mode (F = 0). This is a running index for the temporal base layer pictures, i.e., the pictures with TID set to 0. If TID is larger than 0, TL0PICIDX indicates which temporal base layer picture the current picture depends on. TL0PICIDX MUST be incremented when TID is equal to 0. The index SHOULD start on a random number, and MUST restart at 0 after reaching the maximum number 255.

Reference indices: When P and F are both set to one, indicating a non-key frame in flexible mode, then at least one reference index MUST be specified as below. Additional reference indices (total of up to 3 reference indices are allowed) may be specified using the N bit below. When either P or F is set to zero, then no reference index is specified.

P_DIFF: The reference index (in 7 bits) specified as the relative PID from the current picture. For example, when P_DIFF=3 on a packet containing the picture with PID 112 means that the picture refers back to the picture with PID 109. This calculation is done modulo the size of the PID field, i.e., either 7 or 15 bits.

N: 1 if there is additional P_DIFF following the current P_DIFF.

4.2.1. Scalability Structure (SS):

The scalability structure (SS) data describes the resolution of each frame within a picture as well as the inter-picture dependencies for a picture group (PG). If the VP9 payload descriptor's "V" bit is set, the SS data is present in the position indicated in Figure 2 and Figure 3.

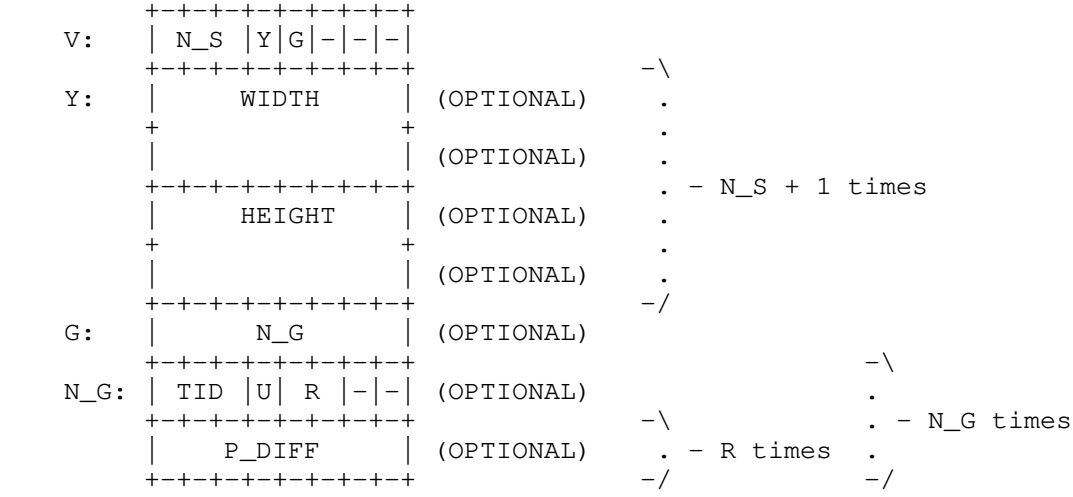


Figure 4

N_S: N_S + 1 indicates the number of spatial layers present in the VP9 stream.

Y: Each spatial layer's frame resolution present. When set to one,

the OPTIONAL WIDTH (2 octets) and HEIGHT (2 octets) MUST be present for each layer frame. Otherwise, the resolution MUST NOT be present.

G: PG description present flag.

-: Bit reserved for future use. MUST be set to zero and MUST be ignored by the receiver.

N_G: N_G indicates the number of pictures in a Picture Group (PG). If N_G is greater than 0, then the SS data allows the inter-picture dependency structure of the VP9 stream to be pre-declared, rather than indicating it on the fly with every packet. If N_G is greater than 0, then for N_G pictures in the PG, each picture's temporal layer ID (TID), switch up point (U), and the R reference indices (P_DIFFs) are specified.

The first picture specified in the PG MUST have TID set to 0.

G set to 0 or N_G set to 0 indicates that either there is only one temporal layer or no fixed inter-picture dependency information is present going forward in the bitstream.

Note that for a given picture, all frames follow the same inter-picture dependency structure. However, the frame rate of each spatial layer can be different from each other and this can be controlled with the use of the D bit described above. The specified dependency structure in the SS data MUST be for the highest frame rate layer.

In a scalable stream sent with a fixed pattern, the SS data SHOULD be included in the first packet of every key frame. This is a packet with P bit equal to zero, SID or D bit equal to zero, and B bit equal to 1. The SS data MUST only be changed on the picture that corresponds to the first picture specified in the previous SS data's PG (if the previous SS data's N_G was greater than 0).

### 4.3. Frame Fragmentation

VP9 frames are fragmented into packets, in RTP sequence number order, beginning with a packet with the B bit set, and ending with a packet with the E bit set. There is no mechanism for finer-grained access to parts of a VP9 frame.

### 4.4. Scalable encoding considerations

In addition to the use of reference frames, VP9 has several additional forms of inter-frame dependencies, largely involving probability tables for the entropy and tree encoders. In VP9 syntax, the syntax element "error_resilient_mode" resets this additional inter-frame data, allowing a frame's syntax to be decoded independently.

Due to the requirements of scalable streams, a VP9 encoder producing a scalable stream needs to ensure that a frame does not depend on a previous frame (of the same or a previous picture) that can legitimately be removed from the stream. Thus, a frame that follows a removable frame (in full decode order) MUST be encoded with "error_resilient_mode" set to true.

For spatially-scalable streams, this means that "error_resilient_mode" needs to be turned on for the base spatial layer; it can however be turned off for higher spatial layers, assuming they are sent with inter-layer dependency (i.e. with the "D" bit set). For streams that are only temporally-scalable without spatial scalability, "error_resilient_mode" can additionally be turned off for any picture that immediately follows a temporal layer 0 frame.

### 4.5. Examples of VP9 RTP Stream

#### 4.5.1. Reference picture use for scalable structure

As discussed in Section 3, the VP9 codec can maintain up to eight reference frames, of which up to three can be referenced or updated by any new frame. This section illustrates one way that a scalable structure (with three spatial layers and three temporal layers) can be constructed using these reference frames.

| Temporal | Spatial | References | Updates |
|---|---|---|---|
| 0 | 0 | 0 | 0 |
| 0 | 1 | 0,1 | 1 |
| 0 | 2 | 1,2 | 2 |
| 2 | 0 | 0 | 6 |
| 2 | 1 | 1,6 | 7 |
| 2 | 2 | 2,7 | - |
| 1 | 0 | 0 | 3 |
| 1 | 1 | 1,3 | 4 |
| 1 | 2 | 2,4 | 5 |
| 2 | 0 | 3 | 6 |
| 2 | 1 | 4,6 | 7 |
| 2 | 2 | 5,7 | - |

Table 1: Example scalability structure

This structure is constructed such that the "U" bit can always be set.

## 5. Feedback Messages and Header Extensions

### 5.1. Reference Picture Selection Indication (RPSI)

The reference picture selection index is a payload-specific feedback message defined within the RTCP-based feedback format. The RPSI message is generated by a receiver and can be used in two ways. Either it can signal a preferred reference picture when a loss has been detected by the decoder -- preferably then a reference that the decoder knows is perfect -- or, it can be used as positive feedback information to acknowledge correct decoding of certain reference pictures. The positive feedback method is useful for VP9 used for point to point (unicast) communication. The use of RPSI for VP9 is preferably combined with a special update pattern of the codec's two special reference frames -- the golden frame and the altref frame --

in which they are updated in an alternating leapfrog fashion. When a receiver has received and correctly decoded a golden or altref frame, and that frame had a PictureID in the payload descriptor, the receiver can acknowledge this simply by sending an RPSI message back to the sender. The message body (i.e., the "native RPSI bit string" in [RFC4585]) is simply the PictureID of the received frame.

Note: because all frames of the same picture must have the same inter-picture reference structure, there is no need for a message to specify which frame is being selected.

### 5.2. Full Intra Request (FIR)

The Full Intra Request (FIR) [RFC5104] RTCP feedback message allows a receiver to request a full state refresh of an encoded stream.

Upon receipt of an FIR request, a VP9 sender MUST send a picture with a keyframe for its spatial layer 0 layer frame, and then send frames without inter-picture prediction (P=0) for any higher layer frames.

### 5.3. Layer Refresh Request (LRR)

The Layer Refresh Request [I-D.ietf-avtext-lrr] allows a receiver to request a single layer of a spatially or temporally encoded stream to be refreshed, without necessarily affecting the stream's other layers.

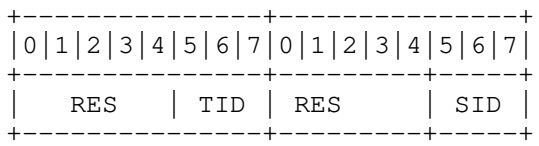


Figure 5

Figure 5 shows the format of LRR's layer index fields for VP9 streams. The two "RES" fields MUST be set to 0 on transmission and ingnored on reception. See Section 4.2 for details on the TID and SID fields.

Identification of a layer refresh frame can be derived from the reference IDs of each frame by backtracking the dependency chain until reaching a point where only decodable frames are being referenced. Therefore it's recommended for both the flexible and the non-flexible mode that, when upgrade frames are being encoded in response to a LRR, those packets should contain layer indices and the reference fields so that the decoder or an MCU can make this derivation.

Example:

LRR {1,0}, {2,1} is sent by an MCU when it is currently relaying {1,0} to a receiver and which wants to upgrade to {2,1}. In response the encoder should encode the next frames in layers {1,1} and {2,1} by only referring to frames in {1,0}, or {0,0}.

In the non-flexible mode, periodic upgrade frames can be defined by the layer structure of the SS, thus periodic upgrade frames can be automatically identified by the picture ID.

## 6. Payload Format Parameters

This payload format has three optional parameters, "max-fr", "max-fs", and "profile-id".

The max-fr and max-fs parameters are used to signal the capabilities of a receiver implementation. If the implementation is willing to receive media, both parameters MUST be provided. These parameters MUST NOT be used for any other purpose. A media sender SHOULD NOT send media with a frame rate or frame size exceeding the max-fr and max-fs values signaled. (There may be scenarios, such as pre-encoded media or selective forwarding middleboxes [RFC7667], where a media sender does not have media available that fits within a receivers max-fs and max-fr value; in such scenarios, a sender MAY exceed the signaled values.)

max-fr: The value of max-fr is an integer indicating the maximum frame rate in units of frames per second that the decoder is capable of decoding.

max-fs: The value of max-fs is an integer indicating the maximum frame size in units of macroblocks that the decoder is capable of decoding.

The decoder is capable of decoding this frame size as long as the width and height of the frame in macroblocks are less than int(sqrt(max-fs * 8)) - for instance, a max-fs of 1200 (capable of supporting 640x480 resolution) will support widths and heights up to 1552 pixels (97 macroblocks).

profile-id: The value of profile-id is an integer indicating the default coding profile, the subset of coding tools that may have been used to generate the stream or that the receiver supports). Table 2 lists all of the profiles defined in section 7.2 of [VP9-BITSTREAM] and the corresponding integer values to be used.

If no profile-id is present, Profile 0 MUST be inferred. (The

profile-id parameter was added relatively late in the development of this specification, so some existing implementations may not send it.)

Informative note: See Table 3 for capabilities of coding profiles defined in section 7.2 of [VP9-BITSTREAM].

A receiver MUST ignore any parameter unspecified in this specification.

| Profile | profile-id |
|---|---|
| 0 | 0 |
| 1 | 1 |
| 2 | 2 |
| 3 | 3 |

Table 2: Table of profile-id integer values representing the VP9 profile corresponding to the set of coding tools supported.

| Profile | Bit Depth | SRGB Colorspace | Chroma Subsampling |
|---|---|---|---|
| 0 | 8 | No | YUV 4:2:0 |
| 1 | 8 | Yes | YUV 4:2:0,4:4:0 or 4:4:4 |
| 2 | 10 or 12 | No | YUV 4:2:0 |
| 3 | 10 or 12 | Yes | YUV 4:2:0,4:4:0 or 4:4:4 |

Table 3: Table of profile capabilities.

## 6.1. SDP Parameters

#### 6.1.1. Mapping of Media Subtype Parameters to SDP

The media type video/VP9 string is mapped to fields in the Session Description Protocol (SDP) [RFC8866] as follows:

* The media name in the "m=" line of SDP MUST be video.

* The encoding name in the "a=rtpmap" line of SDP MUST be VP9 (the media subtype).

* The clock rate in the "a=rtpmap" line MUST be 90000.

* The parameters "max-fr" and "max-fs" MUST be included in the "a=fmtp" line of SDP if the receiver wishes to declare its receiver capabilities. These parameters are expressed as a media subtype string, in the form of a semicolon separated list of parameter=value pairs.

* The OPTIONAL parameter profile-id, when present, SHOULD be included in the "a=fmtp" line of SDP. This parameter is expressed as a media subtype string, in the form of a parameter=value pair. When the parameter is not present, a value of 0 MUST be inferred for profile-id.

##### 6.1.1.1. Example

An example of media representation in SDP is as follows:

```
m=video 49170 RTP/AVPF 98
a=rtpmap:98 VP9/90000
a=fmtp:98 max-fr=30;max-fs=3600;profile-id=0
```

#### 6.1.2. Offer/Answer Considerations

When VP9 is offered over RTP using SDP in an Offer/Answer model [RFC3264] for negotiation for unicast usage, the following limitations and rules apply:

* The parameter identifying a media format configuration for VP9 is profile-id. This media format configuration parameter MUST be used symmetrically; that is, the answerer MUST either maintain this configuration parameter or remove the media format (payload type) completely if it is not supported.

* The max-fr and max-fs parameters are used declaratively to describe receiver capabilities, even in the Offer/Answer model. The values in an answer are used to describe the answerer's capabilities, and thus their values are set independently of the values in the offer.

* To simplify the handling and matching of these configurations, the same RTP payload type number used in the offer SHOULD also be used in the answer and in a subsequent offer, as specified in [RFC3264]. An answer or subsequent offer MUST NOT contain the payload type number used in the offer unless the profile-id value is exactly the same as in the original offer. However, max-fr and max-fs parameters MAY be changed in subsequent offers and answers, with the same payload type number, if an endpoint wishes to change its declared receiver capabilties.

## 7. Media Type Definition

This registration is done using the template defined in [RFC6838] and following [RFC4855].

Type name:
video

Subtype name:
VP9

Required parameters:
N/A.

Optional parameters:
There are three optional parameters, "max-fr", "max-fs", and "profile-id". See Section 6 for their definition.

Encoding considerations:
This media type is framed in RTP and contains binary data; see Section 4.8 of [RFC6838].

Security considerations:
See Section 8 of RFC xxxx.

[RFC Editor: Upon publication as an RFC, please replace "XXXX" with the number assigned to this document and remove this note.]

Interoperability considerations:
None.

Published specification:

VP9 bitstream format [VP9-BITSTREAM] and RFC XXXX.

[RFC Editor: Upon publication as an RFC, please replace "XXXX" with the number assigned to this document and remove this note.]

Applications which use this media type:
For example: Video over IP, video conferencing.

Fragment identifier considerations:
N/A.

Additional information:
None.

Person & email address to contact for further information:
Jonathan Lennox <jonathan.lennox@8x8.com>

Intended usage:
COMMON

Restrictions on usage:
This media type depends on RTP framing, and hence is only defined for transfer via RTP [RFC3550].

Author:
Jonathan Lennox <jonathan.lennox@8x8.com>

Change controller:
IETF AVTCore Working Group delegated from the IESG.

## 8. Security Considerations

RTP packets using the payload format defined in this specification are subject to the security considerations discussed in the RTP specification [RFC3550], and in any applicable RTP profile such as RTP/AVP [RFC3551], RTP/AVPF [RFC4585], RTP/SAVP [RFC3711], or RTP/SAVPF [RFC5124]. SAVPF [RFC5124]. However, as "Securing the RTP Protocol Framework: Why RTP Does Not Mandate a Single Media Security Solution" [RFC7202] discusses, it is not an RTP payload format's responsibility to discuss or mandate what solutions are used to meet the basic security goals like confidentiality, integrity and source authenticity for RTP in general. This responsibility lays on anyone using RTP in an application. They can find guidance on available security mechanisms in Options for Securing RTP Sessions [RFC7201]. Applications SHOULD use one or more appropriate strong security mechanisms. The rest of this security consideration section discusses the security impacting properties of the payload format itself.

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

## 9. Congestion Control

Congestion control for RTP SHALL be used in accordance with RFC 3550 [RFC3550], and with any applicable RTP profile; e.g., RFC 3551 [RFC3551]. The congestion control mechanism can, in a real-time encoding scenario, adapt the transmission rate by instructing the encoder to encode at a certain target rate. Media aware network elements MAY use the information in the VP9 payload descriptor in Section 4.2 to identify non-reference frames and discard them in order to reduce network congestion. Note that discarding of non-reference frames cannot be done if the stream is encrypted (because the non-reference marker is encrypted).

## 10. IANA Considerations

The IANA is requested to register the media type registration "video/vp9" as specified in Section 7. The media type is also requested to be added to the IANA registry for "RTP Payload Format MIME types" <http://www.iana.org/assignments/rtp-parameters>.

## 11. Acknowledgments

Alex Eleftheriadis, Yuki Ito, Won Kap Jang, Sergio Garcia Murillo, Roi Sasson, Timothy Terriberry, Emircan Uysaler, and Thomas Volkert commented on the development of this document and provided helpful comments and feedback.

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