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RTP Payload Format for Versatile Video Coding (VVC)  
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Abstract

This memo describes an RTP payload format for the video coding standard ITU-T Recommendation [H.266] and ISO/IEC International Standard [ISO23090-3], 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.

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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 [ISO23090-3], is currently in the ISO/IEC approval process and is planned for ratification in mid 2020. H.266 is reported to provide significant coding efficiency gains over H.265 and earlier video codec formats.

This memo describes 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) [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 so that fewer bits may be spent on motion vectors. Bi-directional optical flow (BDOF) is a similar method to DMVR but at 4x4 sub-block level. Another difference is that DMVR is based on block matching while BDOF derives MVs with equations. 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 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 weightings 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

[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), 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. Decoding capability informations can include profile, level, and sub-profile information to determine a maximum complexity interop point that is guaranteed to be never exceeded, even if splicing of video sequences occurs within a session. It further optionally includes constraint flags, which indicate that the video bitstream will be constraint in the use of certain features as indicated by the values of those flags. 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 Video Parameter Set (VPS) pertains to a Coded Video Sequences (CVS) of multiple layers covering the same range of picture units, and includes, among other information decoding dependency expressed as information for reference picture set 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 and the random access point picture. In MPEG-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. 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.

## 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 the lowest layer among the layers 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 semantic. 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 complexity 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.

#### Constraint Flags

The `profile_tier_level` structure optionally carries a considerable number of constraint 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

Editor notes: need will update along with VVC new draft in the future

[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 sub-layer cannot be used for inter prediction reference by pictures of a lower temporal sub-layer, 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.

#### Spatial, SNR, View Scalability

[VVC] includes support for spatial, SNR, and View 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 main profiles of video codecs and not wide deployed due to additional costs. In VVC, however, all those forms of scalability are supported 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. Scalability support can be implemented in a single decoding "loop" and is widely considered a comparatively lightweight operation.

### Spatial Scalability

With the existence of Reference Picture Resampling, in the "main" profile of VVC, the additional burden for scalability support is just a minor modification of the high-level syntax (HLS). In technical aspects, 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 resampled video data of the reconstructed reference picture from a reference layer to predict the current enhancement layer. Then, the resampling process for inter-layer prediction is performed at the block-level, by modifying the existing interpolation process for motion compensation. It means that no additional resampling process is needed to support 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.

### SEI Messages

Supplementary Enhancement Information (SEI) messages are codepoints 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. In the [VVC] environment, some of the SEI messages considered to be generally useful also in other video coding technologies have been moved out of the main specification into a

companion document (TO DO: add reference once ITU designation is known).

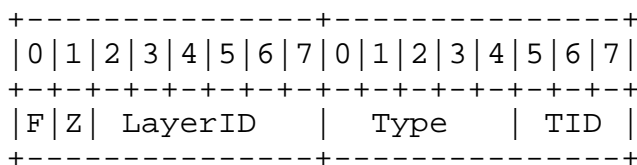
1.1.3. Parallel Processing Support (informative)

Compared to HEVC, the [VVC] design to support parallelization offers numerous improvements. Some of those improvements are still undergoing changes in JVET. Information, to the extent relevant for this memo, will be added in future versions of this memo as the standardization in JVET progresses and the technology stabilizes.

Editor notes: update on sub-picture/slice/tile is needed following new VVC draft

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 .

Type: 6 bits

nal\_unit\_type. This field specifies the NAL unit type as defined in Table 7-1 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]:

- o Usage of RTP header with this payload format
- o 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
- o Transmission of [VVC] NAL units of the same bitstream within a single RTP stream.
- o Media type parameters to be used with the Session Description Protocol (SDP) [RFC4566]
- o Frame-marking mapping [FrameMarking]

## 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

Editor notes:

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.

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.

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.

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.

Sub-layer: 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.

Sub-layer representation: A subset of the bitstream consisting of NAL units of a particular sub-layer and the lower sub-layers.

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.

Editor Notes: the following informative needs to be updated along with frame marking update

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.

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 [VVC], 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

AU	Access Unit
AP	Aggregation Packet
CTU	Coding Tree Unit
CVS	Coded Video Sequence
DPB	Decoded Picture Buffer
DCI	Decoding capability information
DON	Decoding Order Number
DONB	Decoding Order Number Base
FIR	Full Intra Request
FU	Fragmentation Unit
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

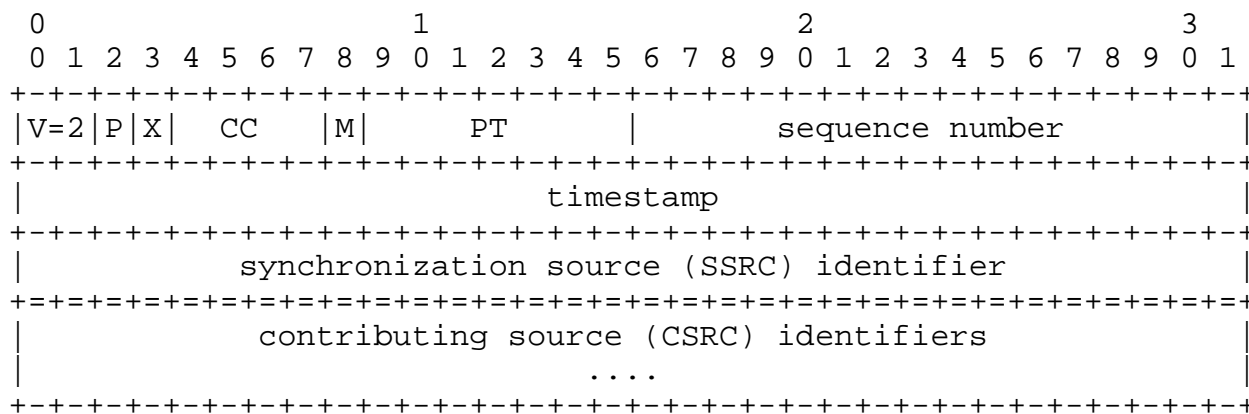
VPS            Video Parameter Set

#### 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 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 notes: The informative note below needs updating once the NAL unit type table is stable in the [VVC] 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 `nal_uX` 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 `nal_uY` 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 19, and 2) all NAL units between `nal_uX` and `nal_uY`, when present, have `nal_unit_type` in the range of 13 to 17, inclusive, equal to 20, equal to 23 or equal to 26.

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 picture of the access unit in which the NAL unit (according to Annex D 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 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.

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 four different payload structures are as follows:

- o 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.
- o Aggregation Packet (AP): Contains more than one NAL unit within one access unit. This payload structure is specified in Section 4.3.2.
- o Fragmentation Unit (FU): Contains a subset of a single NAL unit. This payload structure is specified in Section 4.3.3.

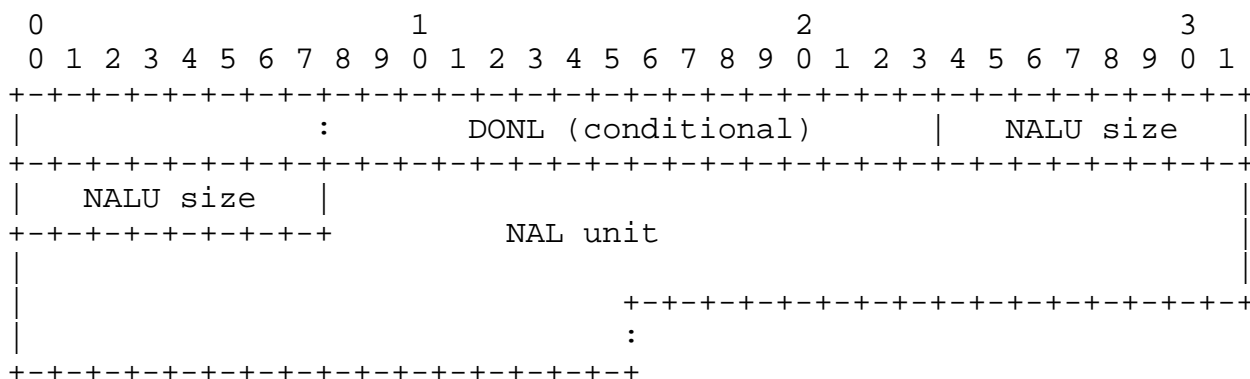
### 4.3.1. Single NAL Unit Packets

Editor notes: its better to add a section to describe DONL and sprop-max\_don\_diff

A single NAL unit packet contains exactly one NAL unit, and consists of a payload header (denoted as PayloadHdr), a conditional 16-bit







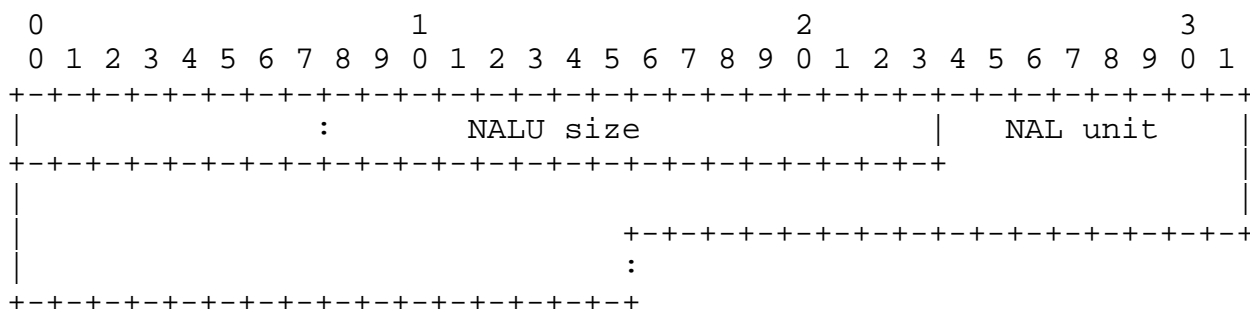
The Structure of the First Aggregation Unit in an AP

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.

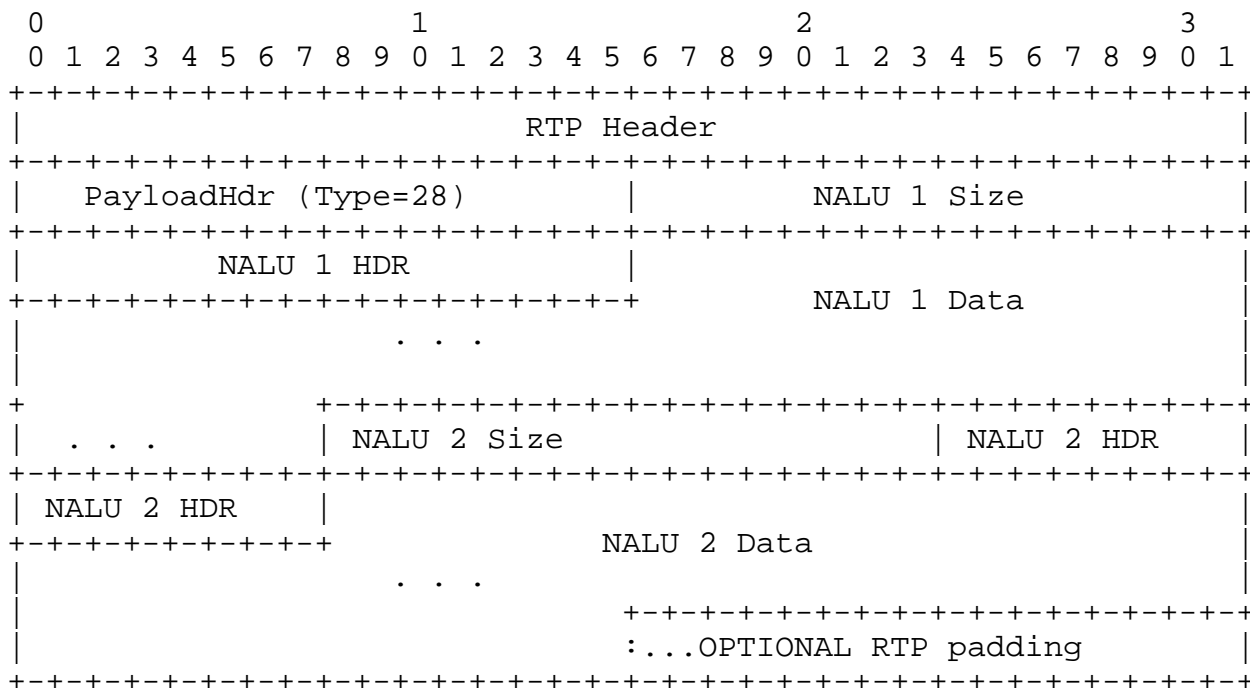
An aggregation unit that is not the first aggregation unit in an AP will be followed immediately by a 16-bit unsigned size information (in network byte order) that indicates the size of the NAL unit in bytes (excluding these two octets, but including the NAL unit header), followed by the NAL unit itself, 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 1 and 2 in the figure, without the DONL field being present.



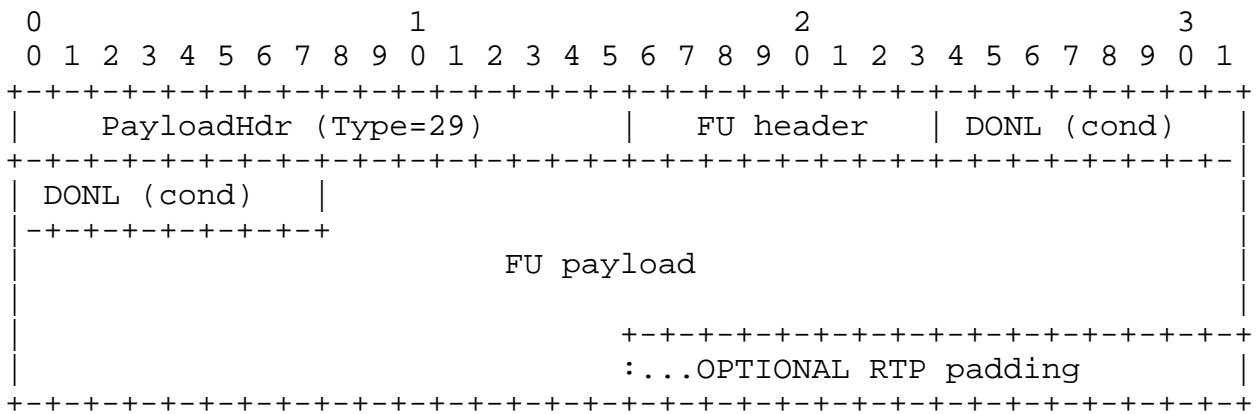
An Example of an AP Packet Containing Two Aggregation Units without the DONL Field

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.





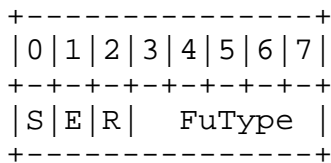


The Structure of an FU

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

Placeholder

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 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, 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 for all the RTP streams carrying the [VVC] 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  $n$ :

- o 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].
- o 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  $DON[n-1] - DON[n] \geq 32768$ ),  
AbsDon[ $n$ ] = AbsDon[ $n-1$ ] + 65536 -  $DON[n-1] + DON[n]$

If ( $DON[n] > DON[n-1]$  and  $DON[n] - DON[n-1] \geq 32768$ ),  
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:

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

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

- o 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.
- o 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.

- o 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.
- o 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 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 and across RTP streams. 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 all the received RTP streams, 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 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 any the received RTP streams, 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:

- o 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

Placeholder

## 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. Slice Loss Indication (SLI)

For further study. Maybe remove as there are no known implementations of SDLI in [HEVC] based systems

### 8.3. Reference Picture Selection Indication (RPSI)

Feedback-based reference picture selection has been shown as a powerful tool to stop temporal error propagation for improved error resilience [Girod99] [Wang05]. In one approach, the decoder side tracks errors in the decoded pictures and informs the encoder side that a particular picture that has been decoded relatively earlier is correct and still present in the decoded picture buffer; it requests the encoder to use that correct picture-availability information when encoding the next picture, so to stop further temporal error propagation. For this approach, the decoder side should use the RPSI feedback message.

Encoders can encode some long-term reference pictures as specified in [VVC] for purposes described in the previous paragraph without the need of a huge decoded picture buffer. As shown in [Wang05], with a flexible reference picture management scheme, as in VVC, even a decoded picture buffer size of two picture storage buffers would work for the approach described in the previous paragraph.

The text above is copy-paste from RFC 7798. If we keep the RPSI message, it needs adaptation to the [VVC] syntax. Doing so shouldn't be too hard as the [VVC] reference picture mechanism is not too different from the [HEVC] one.

### 8.4. 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. Frame marking

placeholder

## 10. 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.

## 11. 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 sub-layers (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.

## 12. IANA Considerations

Placeholder

## 13. 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.

## 14. References

### 14.1. Normative References

[H.266] "ITU-T, Versatile Video Coding", n.d..

[ISO23090-3]  
"ISO/IEC DIS Information technology --- Coded representation of immersive media --- Part 3 Versatile video codings", n.d.,  
<<https://www.iso.org/standard/73022.html>>.

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, DOI 10.17487/RFC2119, March 1997,  
<<https://www.rfc-editor.org/info/rfc2119>>.

- [RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, DOI 10.17487/RFC3550, July 2003, <<https://www.rfc-editor.org/info/rfc3550>>.
- [RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, RFC 3551, DOI 10.17487/RFC3551, July 2003, <<https://www.rfc-editor.org/info/rfc3551>>.
- [RFC3711] Baugher, M., McGrew, D., Naslund, M., Carrara, E., and K. Norrman, "The Secure Real-time Transport Protocol (SRTP)", RFC 3711, DOI 10.17487/RFC3711, March 2004, <<https://www.rfc-editor.org/info/rfc3711>>.
- [RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", RFC 4566, DOI 10.17487/RFC4566, July 2006, <<https://www.rfc-editor.org/info/rfc4566>>.
- [RFC4585] Ott, J., Wenger, S., Sato, N., Burmeister, C., and J. Rey, "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)", RFC 4585, DOI 10.17487/RFC4585, July 2006, <<https://www.rfc-editor.org/info/rfc4585>>.
- [RFC5104] Wenger, S., Chandra, U., Westerlund, M., and B. Burman, "Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)", RFC 5104, DOI 10.17487/RFC5104, February 2008, <<https://www.rfc-editor.org/info/rfc5104>>.
- [RFC5124] Ott, J. and E. Carrara, "Extended Secure RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/SAVPF)", RFC 5124, DOI 10.17487/RFC5124, February 2008, <<https://www.rfc-editor.org/info/rfc5124>>.
- [RFC7656] Lennox, J., Gross, K., Nandakumar, S., Salgueiro, G., and B. Burman, Ed., "A Taxonomy of Semantics and Mechanisms for Real-Time Transport Protocol (RTP) Sources", RFC 7656, DOI 10.17487/RFC7656, November 2015, <<https://www.rfc-editor.org/info/rfc7656>>.
- [RFC8082] Wenger, S., Lennox, J., Burman, B., and M. Westerlund, "Using Codec Control Messages in the RTP Audio-Visual Profile with Feedback with Layered Codecs", RFC 8082, DOI 10.17487/RFC8082, March 2017, <<https://www.rfc-editor.org/info/rfc8082>>.

[RFC8174] Leiba, B., "Ambiguity of Uppercase vs Lowercase in RFC 2119 Key Words", BCP 14, RFC 8174, DOI 10.17487/RFC8174, May 2017, <<https://www.rfc-editor.org/info/rfc8174>>.

[VVC] "Versatile Video Coding (Draft 8), Joint Video Experts Team (JVET)", January 2020.

#### 14.2. Informative References

[CABAC] Sole, J, . and . et al, "Transform coefficient coding in HEVC, IEEE Transactions on Circuits and Systems for Video Technology", DOI 10.1109/TCSVT.2012.2223055, December 2012.

##### [FrameMarking]

Berger, E, ., Nandakumar, S, ., and . Zanaty M, "Frame Marking RTP Header Extension", Work in Progress draft-berger-avtext-framemarking , 2015.

[Girod99] Girod, B, . and . et al, "Feedback-based error control for mobile video transmission, Proceedings of the IEEE", DOI 110.1109/5.790632, October 1999.

[HEVC] "High efficiency video coding, ITU-T Recommendation H.265", April 2013.

[MPEG2S] ISO/IEC, ., "Information technology - Generic coding of moving pictures and associated audio information - Part 1: Systems, ISO International Standard 13818-1", 2013.

[RFC6051] Perkins, C. and T. Schierl, "Rapid Synchronisation of RTP Flows", RFC 6051, DOI 10.17487/RFC6051, November 2010, <<https://www.rfc-editor.org/info/rfc6051>>.

[RFC6184] Wang, Y., Even, R., Kristensen, T., and R. Jesup, "RTP Payload Format for H.264 Video", RFC 6184, DOI 10.17487/RFC6184, May 2011, <<https://www.rfc-editor.org/info/rfc6184>>.

[RFC6190] Wenger, S., Wang, Y., Schierl, T., and A. Eleftheriadis, "RTP Payload Format for Scalable Video Coding", RFC 6190, DOI 10.17487/RFC6190, May 2011, <<https://www.rfc-editor.org/info/rfc6190>>.

[RFC7201] Westerlund, M. and C. Perkins, "Options for Securing RTP Sessions", RFC 7201, DOI 10.17487/RFC7201, April 2014, <<https://www.rfc-editor.org/info/rfc7201>>.

- [RFC7202] Perkins, C. and M. Westerlund, "Securing the RTP Framework: Why RTP Does Not Mandate a Single Media Security Solution", RFC 7202, DOI 10.17487/RFC7202, April 2014, <<https://www.rfc-editor.org/info/rfc7202>>.
- [RFC7798] Wang, Y., Sanchez, Y., Schierl, T., Wenger, S., and M. Hannuksela, "RTP Payload Format for High Efficiency Video Coding (HEVC)", RFC 7798, DOI 10.17487/RFC7798, March 2016, <<https://www.rfc-editor.org/info/rfc7798>>.
- [Wang05] Wang, YK, ., Zhu, C, ., and . Li, H, "Error resilient video coding using flexible reference frames", Visual Communications and Image Processing 2005 (VCIP 2005) , July 2005.

#### Appendix A. Change History

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

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