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Frame Marking RTP Header Extension
draft-ietf-avtext-framemarking-11

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

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

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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 [I-D.ietf-perc-private-media-framework] 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].

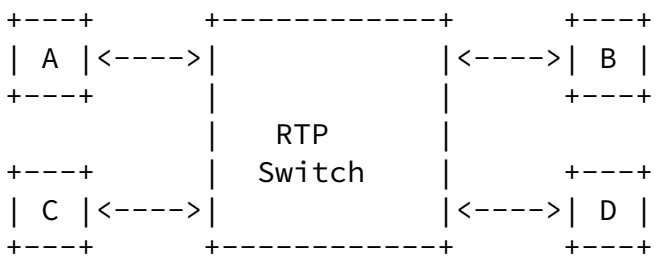


Figure 1: RTP switch

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 [I-D.ietf-perc-private-media-framework].

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

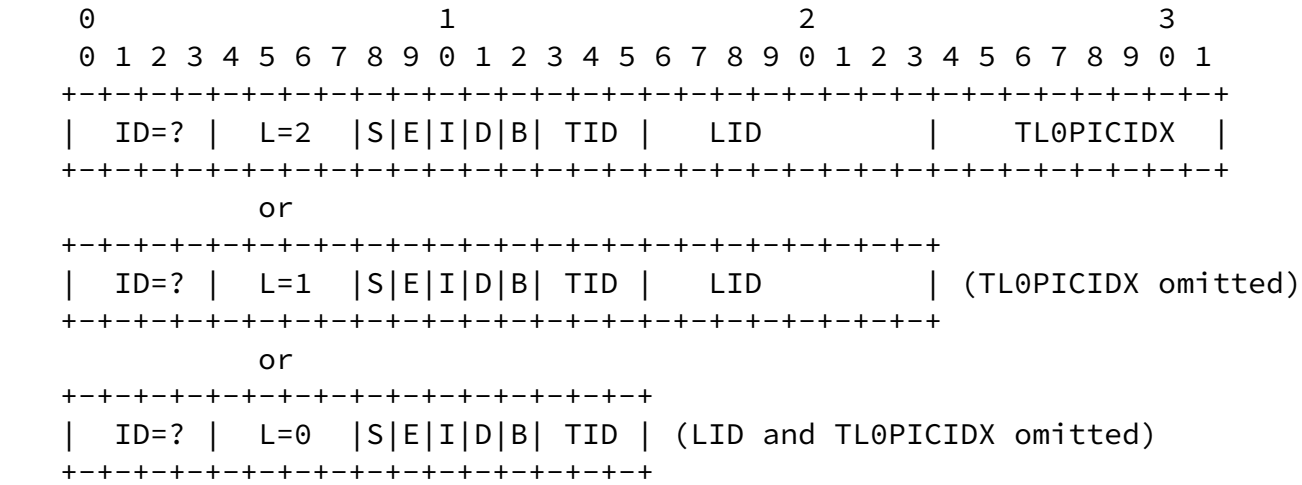
RTP Streams; for example, the I and D bits can be used to generate extra or no redundancy, respectively, and redundancy schemes with source blocks can align source block boundaries with Independent frame boundaries as marked by the I bit.

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.
- 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.

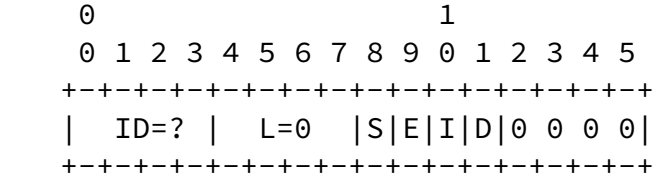
Without further information about the layer structure, these TID/LID

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.

- S: Start of Frame (1 bit) - MUST be 1 in the first packet in a frame; otherwise MUST be 0.
- 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.
- 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.
- 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.
- 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. 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.

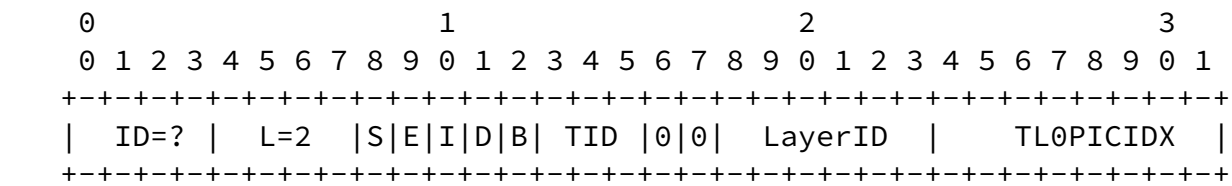
The I bit MUST be 1 when the NAL unit type is 16-23 (inclusive) or 32-34 (inclusive), or an aggregation packet or fragmentation unit encapsulating any of these types, otherwise it MUST be 0. These

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.



3.3.2. H264-SVC LID Mapping

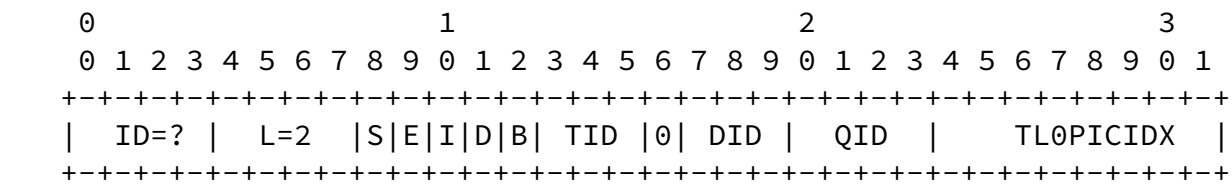
The following shows H264-SVC [RFC6190] Layer encoding information (3 bits for spatial/dependency layer, 4 bits for quality layer and 3 bits for temporal layer) mapped to the generic LID and TID fields.

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.



3.3.3. H264 (AVC) LID Mapping

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

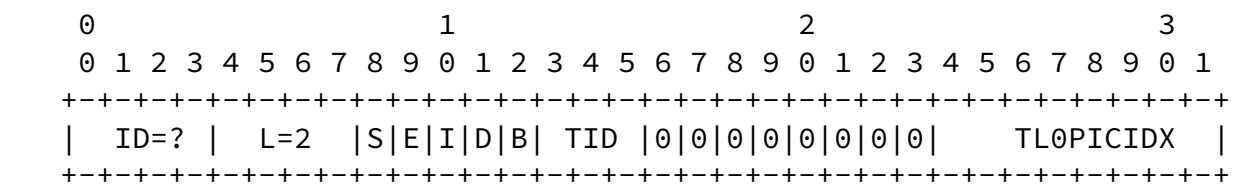
The S bit MUST be 1 when the timestamp in the RTP header differs from the timestamp in the prior RTP sequence number from the same SSRC, otherwise it MUST be 0.

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.



3.3.4. VP8 LID Mapping

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

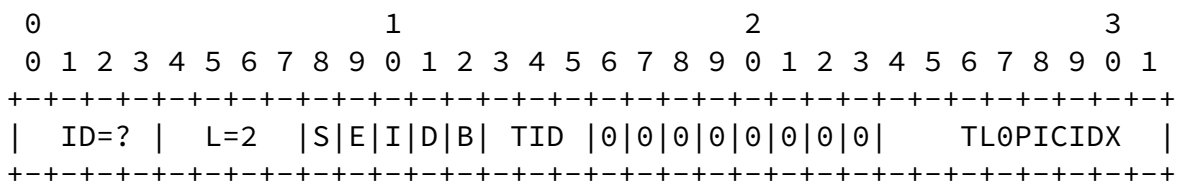
The S bit MUST match the correspondingly named bit in the VP8 payload descriptor when PID=0, otherwise it MUST be 0.

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.



3.3.5. 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. For example, the LID/TID mapping for the VP9 codec is described in the VP9 RTP Payload Format [I-D.ietf-payload-vp9].

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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