                    Frame Marking RTP Header Extension
                    draft-berger-avtext-framemarking-01

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 encryption 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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   material or to cite them other than as "work in progress."

   This Internet-Draft will expire on January 7, 2016.

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1. Introduction

Many widely deployed RTP topologies 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 via end-to-
end encryption that excludes the switch. 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, 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 [I-D.ietf-avtcore-rtp-topologies-update].

```
       +---+      +------------+      +---+
       | A |<---->|            |<---->| B |
       +---+      |            |      +---+
                  |            |
                  |    RTP     |
       +---+      |   Switch   |      +---+
       | C |<---->|            |<---->| D |
       +---+      +------------+      +---+
```
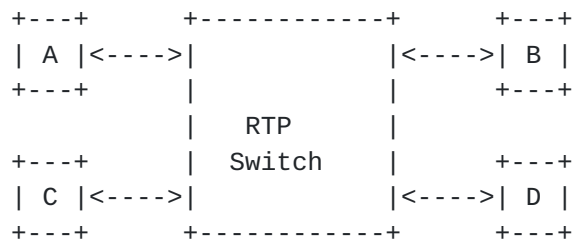
Figure 1: RTP switch

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.

o 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.

o 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.

o Furthermore, it is highly desirable to do this in a way which is
   not specific to the video codec. Nearly all modern video codecs
   share common concepts around frame types.

o 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.

By providing meta-information about the RTP streams outside the
encrypted media payload an RTP switch can do selective forwarding
without decrypting the payload. This document provides a solution to
this problem.

2. Solution

The solution uses RTP header extensions as defined in [RFC5285]. A
subset of meta-information from the video stream is provided as an
RTP header extension to allow a RTP switch to do generic video
switching handling of video streams encoded with different video
codecs.

2.1. Mandatory Extension

The following information are extracted from the media payload:

o S: Start of Frame (1 bit) - MUST be 1 in the first packet in a
   frame within a layer; otherwise MUST be 0.
o E: End of Frame (1 bit) - MUST be 1 in the last packet in a frame
   within a layer; otherwise MUST be 0.
o I: Independent Frame (1 bit) - MUST be 1 for frames that can be
   decoded independent of prior frames, e.g. intra-frame, VPx
   keyframe, H.264 IDR [RFC6184], H.265 CRA/BLA; otherwise MUST be 0.
o D: Discardable Frame (1 bit) - MUST be 1 for frames that can be
   dropped, and still provide a decodable media stream; otherwise
   MUST be 0.
o B: Base Layer Sync (1 bit) - MUST be 1 if this frame only depends
   on the base layer; otherwise MUST be 0.
o TID: Temporal ID (3 bits) - The base temporal quality starts with
   0, and increases with 1 for each temporal layer/sub-layer.
o LID: Layer ID (8 bits) - Identifies the spatial and quality layer
   encoded.

NOTE:Given the opaque nature of the LID, consider having the layer
structure information as RTCP SDES item (either in the RTCP SDES
message or as the RTP SDES Header extension) to map the LIDs to
specific resolutions and bitrates thus enabling the RTP Switch to
make informed decisions

The values of frame information can be carried as RTP header
extensions encoded using the one-byte header as described in
[RFC5285] as shown below.

```
 0                   1                   2
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  ID=2 |  L=1  |S|E|I|D|B| TID |    LID        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

2.2. Examples

The following example shows H265-LayerID (6 bits) mapped to the
generic LID field.

```
 0                   1                   2
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  ID=2 |  L=1  |S|E|I|D|B| TID |0|0|H265-LayerId|
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

The following example shows VP9 Layer encoding information (4 bits
for spatial and quality) mapped to the generic LID field.

```
 0                   1                   2
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|  ID=2 |  L=1  |S|E|I|D|B| TID |0|0|0|0| RS| RQ|
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

2.3. Signaling information

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

An example attribute line in SDP:

    a=extmap:3 urn:ietf:params:rtp-hdrext:framemarkinginfo

2.4. Considerations on use

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

When a 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 "discardable" bit.

When a 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 (I bit set) and forward the same. A RTP switch can
request a media source to generate a switching point for H.264 by
sending Full Intra Request (RTCP FIR) as defined in [RFC5104], for
example.

3. Security Considerations

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

[4](#). Acknowledgements

Many thanks to Bernard Aboba, Jonathan Lennox for their inputs.

[5](#). 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: espeberg@cisco.com
Reference: RFC XXXX

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

[6](#). References

[6.1](#). Normative References

[KEYWORDS]
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          Requirement Levels", [BCP 14](#), [RFC 2119](#), March 1997.

[6.2](#). Informative References

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[RFC5104] Wenger, S., Chandra, U., Westerlund, M., and B. Burman,
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[RFC5285] Singer, D. and H. Desineni, "A General Mechanism for RTP
          Header Extensions", [RFC 5285](#), July 2008.

   [RFC6184]   Wang, Y., Even, R., Kristensen, T., and R. Jesup, "RTP
               Payload Format for H.264 Video", RFC 6184, May 2011.

Authors' Addresses

   Espen Berger
   Cisco Systems

   Phone: +47 98228179
   Email: espeberg@cisco.com

   Suhas Nandakumar
   Cisco Systems
   170 West Tasman Drive
   San Jose, CA 95134
   US

   Email: snandaku@cisco.com

   Mo Zanaty
   Cisco Systems
   170 West Tasman Drive
   San Jose, CA 95134
   US

   Email: mzanaty@cisco.com