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RTP/RTCP extension for RTP Splicing Notification
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Abstract

Content splicing is a process that replaces the content of a main multimedia stream with other multimedia content, and delivers the substitutive multimedia content to the receivers for a period of time. The RTP mixer is designed to handle RTP splicing in [RFC6828], but how the RTP mixer knows when to start and end the splicing is still unspecified.

This memo defines two RTP/RTCP extensions to indicate the splicing related information to the RTP mixer: an RTP header extension that conveys the information in-band and an RTCP packet that conveys the information out-of-band.

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1. Introduction

Splicing is a process that replaces some multimedia content with other multimedia content and delivers the substitutive multimedia content to the receivers during specific, pre-designated time slot. Certain timing information about when to start and end the splicing must be first acquired by the splicer to start the splicing. This document refers to this information as Splicing Metadata.

[SCTE35] provides a method that encapsulates the Splicing Metadata inside the MPEG2-TS layer in cable TV systems. But in RTP splicing scenario described in [RFC6828], the mixer has to decode the RTP packets, search and solve the Splicing Metadata inside the payloads. The need for such processing enhances the workload of the mixer and limits the size of RTP sessions the mixer can support.

The document defines an RTP header extension [RFC5285] through which the main RTP sender can provide the Splicing Metadata by including it in the RTP packets.

Nevertheless, the Splicing Metadata conveyed in the RTP header extension might not reach the mixer successfully, any splicing un-aware middlebox on the path between the RTP sender and the mixer might strip the RTP header extension.

To increase robustness against above case, the document also defines a new RTCP packet type in a complementary fashion to carry the Splicing Metadata to the mixer even though RTCP is inherently unreliable too.

2. Terminology

The keywords "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].

Most terminology defined in "Content Splicing for RTP Sessions" [RFC6828] applies to this document except the following one:

Splicing Metadata

A set of certain metadata that allows the mixer to know when to start and end the RTP splicing. The information consists of a couple of NTP-format timestamps on the splicing in point and on the splicing out point.

3. Overview of RTP Splicing Notification

According to RTP Splicing draft [RFC6828], a mixer is designed to do splicing on the RTP layer, but it cannot insert the substitutive content randomly but only do that at the reserved time slots set by the main RTP sender. This implies the mixer must first know the Splicing Metadata from the main RTP sender before splicing starts.

When a new splicing is forthcoming, the main RTP sender MUST send the Splicing Metadata to the mixer. Usually, the Splicing Metadata SHOULD be sent more than once to against the possible packet loss. To enable the mixer to get the substitutive content before the splicing starts, the main RTP sender MUST send the Splicing Metadata far enough in advance. Alternatively, the main RTP sender can estimate when to send the Splicing Metadata based on the round-trip time (RTT) following the mechanisms in section 6.4.1 of [RFC3550] when the mixer sends RTCP RR to the main sender.

The substitutive sender also needs to learn the Splicing Metadata from the main RTP sender in advance, and thus estimates when to transfer the substitutive content to the mixer. The Splicing Metadata could be transmitted from the main RTP sender to the substitutive content using some out-of-band mechanisms, the details how to achieve that are beyond the scope of this memo. To ensure the Splicing Metadata is valid to the main RTP sender and the substitutive RTP sender, the two senders MUST share a common reference clock, so the mixer can achieve accurate splicing.

In this document, the main RTP sender uses a couple of NTP-format timestamps, derived from the common reference clock, to indicate when to start and end the splicing to the mixer: the timestamp of the first substitutive RTP packet on the splicing in point, followed by the timestamp of the first main RTP packet on the splicing out point.

When the substitutive RTP sender gets the Splicing Metadata, it must prepare the substitutive stream. The RTP timestamp of the first substitutive RTP packet that would be presented on the receivers MUST correspond to the same time instant as the former NTP timestamp in the Splicing Metadata. To enable mixer to know the first substitutive RTP packet it begins to output, the substitutive RTP sender MUST enable the mixer to know above RTP timestamp in advance, e.g., from prior receipt of RTCP SR message.

When the splicing will end, the RTP timestamp of the first main RTP packet that would be presented on the receivers MUST correspond to the same time instant as the latter NTP timestamp in the Splicing Metadata.

4. Conveying Splicing Metadata in RTP/RTCP extensions

This memo defines two backwards compatible RTP extensions to convey the Splicing Metadata to the mixer: an RTP header extension and an RTCP splicing notification message.

4.1. RTP Header Extention

The RTP header extension mechanism defined in [RFC5285] can be adapted to carry the Splicing Metadata consisting of a couple of NTP-format timestamps.

One variant is defined for this header extension. The variant carries the lower 24-bit part of the Seconds of a NTP-format timestamp and the 32 bits of the Fraction of a NTP-format timestamp as defined in [RFC5905]. The format is shown in Figures 1.

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|       0xBE    |    0xDE       |           length=4            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+E
|   ID  | L=6   |   NTP timestamp format - Seconds (bit 8-31)   |x
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+t
|           NTP timestamp format - Fraction (bit 0-31)          |e
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+n
|   ID  | L=6   |   NTP timestamp format - Seconds (bit 8-31)   |s
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+i
|           NTP timestamp format - Fraction (bit 0-31)          |o
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+n
```

Figure 1: Sample 56-Bit NTP Encoding
Using the One-Byte Header Format

Note that the inclusion of an RTP header extension will reduce the efficiency of RTP header compression. It is RECOMMENDED that the main sender begins to insert the RTP header extensions into a number of RTP packets in advance of the splicing starting, while leaving the remain RTP packets unmarked.

After the mixer intercepts the RTP header extension and derives the Splicing Metadata, it will generate its own stream and could not include the RTP header extension in outgoing packets to reduce header overhead.

Furthermore, whether the in-band NTP-format timestamps are included or not, RTCP splicing notification message in next section MUST be

sent to provide robustness in the case of any splicing-unaware middlebox that might strip RTP header extensions.

When SDP is used, the use of the RTP header extensions defined above MUST be indicated as specified in [RFC5285]. Therefore, the following URI MUST be used:

o l The URI used for signalling the use of Variant B/56-bit NTP RTP header extension in SDP is "urn:ietf:params:rtp-hdrext: splicing-metadata-56".

4.2. RTCP Splicing Notification Message

Besides the RTP header extension, the main RTP sender includes the Splicing Metadata in an RTCP splicing notification message.

The RTCP splicing notification message is a new RTCP packet type defined with PT = 211. It has a fix header followed by a couple of NTP-format timestamps:

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|reserved |    PT=211     |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                              SSRC                             |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|             IN NTP Timestamp (most significant word)          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|             IN NTP Timestamp (least significant word)         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|             OUT NTP Timestamp (most significant word)         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|             OUT NTP Timestamp (least significant word)        |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 2: RTCP Splicing Notification Message

The RSI packet includes the following fields:

Length: 16 bits

As defined in [RFC3550], the length of the RTCP packet in 32-bit words minus one, including the header and any padding.

SSRC: 32 bits

The SSRC of the Main RTP Sender.

Timestamp: 64 bits

Indicates the wallclock time when this splicing starts and ends. The full-resolution NTP timestamp is used, which is a 64-bit, unsigned, fixed-point number with the integer part in the first 32 bits and the fractional part in the last 32 bits. This format is similar to RTCP Sender Report (Section 6.4.1 of [RFC3550]).

A basic assumption is that the main RTP sender and the mixers are in an SSM group with the source being the main RTP sender. RTCP splicing notification message from the main RTP sender is sent via multicast to the mixers.

The RTCP splicing notification message can be appended to RTCP SR the main RTP sender generates in compound RTCP packets, and hence follows the compound RTCP rules defined in Section 6.1 in [RFC3550].

If the use of non-compound RTCP [RFC5506] was previously negotiated between the sender and the mixer, the RTCP splicing notification message may be sent as non-compound RTCP packets.

When the mixer intercepts the RTCP splicing notification message, it MAY NOT forward the message to the receivers in order to reduce RTCP bandwidth consumption or to avoid downstream receivers from detecting splicing defined in Section 4.5 in [RFC6828].

5. Reduing Splicing Latency

When splicing starts or ends, the mixer outputs the multimedia content from another sender to the receivers. Given that the receivers must first acquire certain information ([RFC6285] refers to this information as Reference Information) to start processing the multimedia data, either the main RTP sender or the substitutive sender SHOULD provide the Reference Information align with its multimedia content to reduce the delay caused by acquiring the Reference Information. The means by which the Reference Information is distributed to the receivers is out of scope of this memo.

Another latency element is synchronization caused delay. The receivers must receive enough synchronization metadata prior to synchronizing the separate components of the multimedia streams when splicing starts or ends. Either the main RTP sender or the

substitutive sender SHOULD send the synchronization metadata early enough so that the receivers can play out the multimedia in a synchronized fashion. The mechanisms defined in [RFC6051] are RECOMMENDED to be adopted to reduce the possible synchronization delay.

6. Failure Cases

This section examines the implications of losing RTCP splicing notification message and other failure case, e.g., the RTP header extension is stripped on the path.

Given there may be splicing un-aware middlebox on the path between the main RTP sender and the mixer, one heuristics will be used to verify whether or not the Splicing Metadata reaches the mixers.

If the mixer does not get the Splicing Metadata when the splicing starts, it will still output the main content to the downstream receivers and forward the RTCP RR packets sent from downstream receivers to the main RTP sender. In such case, the main RTP sender can learn the splicing failed.

In a similar manner, the substitutive sender can learn the splicing failed if it does not receive any RTCP RR packets from downstream receivers when the splicing starts.

Upon the detection of a failure, the main RTP sender or the substitutive sender SHOULD check the path to the failed mixer, or fallback to the payload specific mechanisms, e.g., MPEG-TS splicing solution defined in [SCTE35].

7. SDP Signaling

This document defines the "rtp-splicing" media attribute, which is used for indicating whether the main RTP sender can provide time-slots within its RTP flows for RTP splicing. This attribute should be used in a declarative manner. If this attribute is included in the SDP description, the mixer can receive the Splicing Metadata and implement RTP splicing.

This document also reuses the Flow Identification (FID) semantics defined in SDP Grouping Framework [RFC5888] to represent the relationship between the main RTP stream and the substitutive RTP stream.

The next example shows how the "group" attribute used with FID

semantics can indicate RTP splicing support on RTP sender.

```
v=0
o=xia 1122334455 1122334466 IN IP4 splicing.example.com
s=RTP Splicing Example
t=0 0
a=group:FID 1 2
m=video 30000 RTP/AVP 100
i=Main RTP Stream
c=IN IP4 233.252.0.1/127
a=rtpmap:100 MP2T/90000
a=rtp-splicing
a=mid: 1
m= video 30001 RTP/AVP 100
i=Substitutive RTP Stream
c=IN IP4 233.252.0.2/127
a=sendonly
a=mid: 2
```

Figure 3: Example SDP for a single-channel splicing scenario

The mixer receiving the SDP message above receives one MPEG2-TS stream (payload 100) from the main RTP sender (with multicast destination address of 233.252.0.1) on port 30000, and/or receives another MPEG2-TS stream from the substitutive RTP sender (with multicast destination address of 233.252.0.2) on port 30001. But at a particular point in time, the mixer only selects one stream and output the content from the chosen stream to the downstream receivers.

8. Security Considerations

The security considerations of the RTP specification [RFC3550], the general mechanism for RTP header extensions [RFC5285] and the security considerations of the RTP splicing specification [RFC6828] apply.

The RTP header extension defined in Section 4.1 include two NTP-format timestamps. In the Secure Real-time Transport Protocol (SRTP)[RFC3711], RTP header extensions are authenticated but not encrypted. A malicious endpoint could choose to set the values in this header extension falsely, so as to falsely claim the splicing time.

In scenarios where this is a concern, additional mechanisms MUST be used to protect the confidentiality of the header extension. This

mechanism could be header extension encryption [SRTP-ENCR-HDR], or a lower-level security and authentication mechanism such as IPsec [RFC4301].

9. IANA considerations

9.1. SDP Attribute Registration

Following the guidelines in [RFC4566], the IANA is requested to register one new SDP attribute:

- o Contact name, email address and telephone number: Authors of RFCXXXX
- o Attribute-name: rtp-splicing
- o Long-form: Support RTP Splicing
- o Type of attribute: media-level
- o Subject to charset: no

This attribute is used to signal that the main RTP stream can provide time slots for RTP splicing. It is a property attribute, which does not take a value.

9.2. RTCP Control Packet Types

Based on the guidelines suggested in [RFC5226], a new RTCP packet format has been registered with the RTCP Control Packet Type (PT) Registry:

Name: SNM

Long name: Splicing Notification Message

Value: TBA1

Reference: This document

9.3. RTP Compact Header Extensions

The IANA has also registered a new RTP Compact Header Extension [RFC5285], according to the following:

Extension URI: urn:ietf:params:rtp-hdrext:splicing-metadata-56

Description: Splicing metadata: 56-bit timestamp format

Contact: Jinwei Xia <xiajinwei@huawei.com>

Reference: This document

10. Acknowledgments

TBD

11. References

11.1. Normative References

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