Using SSRC with WebRTC Simulcast
draft-alvestrand-mmusic-simulcast-ssrc-00

Abstract

   This document describes a convention for sending "a=ssrc" attributes
   in SDP together with "a=simulcast" attributes. This allows SFUs that
   need SSRC information to have this info easily accessible.

   Given that it is intended as an interim measure, it does not aim for
   being published as an RFC.

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1. Introduction

In developing the WebRTC specification, the IETF decided on a form of
simulcast that doesn't require fixed SSRC allocation, but rather used
a combination of SDP tags (a=rid) [I-D.ietf-mmusic-rid] and RTP
header extensions (RTPStreamId) [I-D.ietf-avtext-rid] to describe the
mapping between simulcast layers and RTP streams.

The SDP format is described in [I-D.ietf-mmusic-sdp-simulcast].

This posed a problem for some SFUs, which required information on
what SSRCs the incoming streams were going to appear on in order to
be configured correctly.

This document gives a convention for adding information about SSRCs
to the SDP produced by conformant WebRTC implementations in order to
make this information available.

This document does not specify an Internet standard. It is an
interim measure, intended to be useful in the time between the
introduction of RID-based simulcast in browsers and the full support
of RID-based simulcast by SFUs.

2. Conventions and Definitions

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT",
"SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and
"OPTIONAL" in this document are to be interpreted as described in BCP

14 [RFC2119] [RFC8174] when, and only when, they appear in all
capitals, as shown here.

3. How to represent SSRC information

The syntax for representing SSRC information is taken from [RFC5576].

Each media section in the SDP contains one a=ssrc attribute per
simulcast stream, formatted as "a=ssrc:<ssrc> cname:<cname>". The
cname is the sender's single cname as defined in
[I-D.ietf-rtcweb-rtp-usage]; carrying some attribute is required by
the "a=ssrc" syntax, and sending the cname is compatible with what
has been done in other instances.

The list of SSRCs used is declared in an attribute with the FID (Flow
Identification) semantic, as defined in [RFC5888].

The order of SSRCs in the a=ssrc-group attribute MUST match the order
of the rid attributes in the corresponding streams in the "send" part
of the a=simulcast: attribute.

It is RECOMMENDED that both the a=rid: attributes and the a=ssrc:
attributes appear in the same order as the order in the a=simulcast
and a=ssrc-group attributes.

It is RECOMMENDED not to use RTX with this configuration, since the
inclusion of the required declarations for associating RTX SSRCs with
their main SSRCs would make the SDP unwieldy and hard to interpret
correctly.

4. How to request that SSRC information be included

If an SFU wishes to request that a browser send SSRC information, it
should send an offer containing the line "a=x-please-send-ssrcs",
together with a line requesting simulcast:

    m=video
    a=simulcast:recv a,b,c
    a=please-send-ssrcs

The SFU can detect whether the request has been honored by looking
for a=ssrc attributes in the responding answer.

If a Javascript application wishes to request that the browser
generate offers containing SSRC, it can include the non-standard
attribute "showSsrcInSimulcastOffer" in the RTCPeerConnection
constructor:

```
pc = new RTCPeerConnection({showSsrcInSimulcastOffer: true})
```

It is possible to verify that the request is understood by checking
for the presence of this attribute in the RTCPeerConnection
parameters:

```
if (showSsrcInSimulcastOffer in pc.getConfiguration) {
   // the request has been understood correctly
}
```

Formally, this amounts to changing the API of a W3C specification,
but adding nonstandard attributes to an initialization dictionary has
been done before in other contexts; it seems like a relatively
harmless thing to do, but should be reviewed in the W3C WEBRTC WG
anyway.

[5]. Example

```
m=video
a=simulcast:send hi,mid,low
a=rid:hi send
a=rid:mid send
a=rid:low send
a=ssrc-group:FID 123 456 789
a=ssrc:123 cname:foo
a=ssrc:456 cname:foo
a=ssrc:789 cname:foo
```

[6]. Sunsetting the interim measure

This specification is intended to give SFU authors time to convert to
the new mechanism. Since the invocation of this mechanism is
explicit, it is easy to check on what the usage is, and emit
deprecation warnings; those should probably be emitted from day 1.

Once enough time has passed, this mechanism can be removed.

[7]. Open questions

NOTE IN DRAFT: The goal is to make this section empty.

The SSRC-group "FID" was picked because it seemed to have the right
semantic, but it's not clear what it's been used for elsewhere.
Chrome has been using the group "SIM" without registering it; this
might be a better choice.

It's been suggested that it's better to replace the a=ssrc-group:
line with new tag fields either on the a=ssrc: lines or the a=rid:

lines, thus giving explicit correlation. This, however, breaks the
standard format of those lines. Inventing new syntax for an interim
solution seems like a Bad Thing.

People have asked whether and how this document should be published.
If it makes sense to publish it as a historical record, it might make
sense to publish as an RFC; it does not make sense to the author to
ask for standards track publication. At the moment, it claims that
publication is not sought.

8. Security Considerations

This document describes two existing mechanisms: a=simulcast and
a=ssrc-group. Each of these is defined in an RFC with security
considerations.

The only added attack surface here is the ability to create
mismatches between the two lists of simulcast RTP streams, causing
different implementations to choose different streams to display.
This is a special instance of the general rule that "people who can
modify your SDP can mess things up"; normal precautions when passing
SDP around should be adequate.

9. IANA Considerations

This document has no IANA actions.

If it were to be published, this section would have to request IANA
to register the "please-send-ssrc" attribute, and if it mints a new
group semantic for a=ssrc-group, this will also have to be
registered.

If the document succeeds in being transitory in nature, registration
may not be needed.

10. References

10.1. Normative References

[I-D.ietf-avtext-rid]
          Roach, A., Nandakumar, S., and P. Thatcher, "RTP Stream
          Identifier Source Description (SDES)", draft-ietf-avtext-
          rid-09 (work in progress), October 2016.

[I-D.ietf-mmusic-rid]
          Roach, A., "RTP Payload Format Restrictions", draft-ietf-
          mmusic-rid-15 (work in progress), May 2018.

   [I-D.ietf-mmusic-sdp-simulcast]
              Burman, B., Westerlund, M., Nandakumar, S., and M. Zanaty,
              "Using Simulcast in SDP and RTP Sessions", draft-ietf-
              mmusic-sdp-simulcast-14 (work in progress), March 2019.

   [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>.

   [RFC5576] Lennox, J., Ott, J., and T. Schierl, "Source-Specific
              Media Attributes in the Session Description Protocol
              (SDP)", RFC 5576, DOI 10.17487/RFC5576, June 2009,
              <https://www.rfc-editor.org/info/rfc5576>.

   [RFC5888] Camarillo, G. and H. Schulzrinne, "The Session Description
              Protocol (SDP) Grouping Framework", RFC 5888,
              DOI 10.17487/RFC5888, June 2010,
              <https://www.rfc-editor.org/info/rfc5888>.

   [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>.

10.2. Informative References

   [I-D.ietf-rtcweb-rtp-usage]
              Perkins, C., Westerlund, M., and J. Ott, "Web Real-Time
              Communication (WebRTC): Media Transport and Use of RTP",
              draft-ietf-rtcweb-rtp-usage-26 (work in progress), March
              2016.

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