                Support for multiple clock rates in an RTP session
                   draft-ietf-avtext-multiple-clock-rates-01

Abstract

   This document clarifies the RTP specification when different clock
   rates are used in an RTP session. It also provides guidance on how
   to interoperate with legacy RTP implementations that use multiple
   clock rates.

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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   This Internet-Draft will expire on January 12, 2012.

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Table of Contents

1. Introduction

   The clock rate is a parameter of the payload format. It is often
   defined as been the same as the sampling rate but it is not always
   the case (see e.g. the G722 and MPA audio codecs in [RFC3551]).

   An RTP sender can switch between different payloads during the
   lifetime of an RTP session and because clock rates are defined by
   payload types, it is possible that the clock rate also varies during
   an RTP session. RTP [RFC3550] lists using multiple clock rates as
   one of the reasons to not use different payloads on the same SSRC but
   unfortunately this advice was not always followed and some RTP
   implementations change the payload in the same SSRC even if the
   different payloads use different clock rates.

   This creates three problems:
   o The method used to calculate the RTP timestamp field in an RTP
      packet is underspecified.
   o When the same SSRC is used for different clock rates, it is
      difficult to know what clock rate was used for the RTP timestamp
      field in an RTCP SR packet.
   o When the same SSRC is used for different clock rates, it is
      difficult to know what clock rate was used for the interarrival
      jitter field in an RTCP RR packet.

   Table 1 contains a non-exhaustive list of fields in RTCP packets that
   uses a clock rate as unit:

Field name	RTCP packet type	Reference
RTP timestamp	SR	[RFC3550]
Interarrival jitter	RR	[RFC3550]
min_jitter	XR Summary Block	[RFC3611]
max_jitter	XR Summary Block	[RFC3611]
mean_jitter	XR Summary Block	[RFC3611]
dev_jitter	XR Summary Block	[RFC3611]
Interarrival jitter	IJ	[RFC5450]
RTP timestamp	SMPTETC	[RFC5484]
Jitter	RSI Jitter Block	[RFC5760]
Median jitter	RSI Stats Block	[RFC5760]

                                Table 1

   This document first tries to list in Section 2 and subsections all
   known algorithms used in existing RTP implementations at the time of
   writing. This sections are not normative.

Section 4 and subsections then recommend a unique algorithm that modifies [RFC3550]. This sections are normative.

Section 5 and subsections then analyze what happen when the legacy algorithms listed in Section 2 are used with the new algorithm listed in Section 4. This sections are not normative.

2. Legacy RTP

The following sections describe the various ways legacy RTP implementations behave when multiple clock rates are used. Legacy RTP refers to RFC 3550 without the modifications introduced by this document.

[[We need to list here all the methods used in the field. Please send them to the author. NDA can be arranged if needed]]

2.1. Different SSRC

One way of managing multiple clock rates is to use a different SSRC for each different clock rate, as in this case there is no ambiguity on the clock rate used by fields in the RTCP packets. This method also seems to be the original intent of RTP as can be deduced from points 2 and 3 of section 5.2 of RFC 3550.

On the other hand changing the SSRC can be a problem for some implementations designed to work only with unicast IP addresses, where having multiple SSRCs is considered a corner case. Lip synchronization can also be a problem in the interval between the beginning of the new stream and the first RTCP SR packet. This is not different than what happen at the beginning of the RTP session but it can be more annoying for the end-user.

2.2. Same SSRC

The simplest way of managing multiple clock rates is to use the same SSRC for all the payload types regardless of the clock rates.

Unfortunately there is no clear definition on how the RTP timestamp should be calculated in this case. The following subsection presents one algorithm used in the field.

2.2.1. Monotonic timestamps

The most common method of calculating the RTP timestamp ensures that the value increases monotonically. The formula used by this method is as follow:

```
timestamp = previous_timestamp + (current_capture_time -
previous_capture_time) * current_clock_rate
```

The problem with this method is that the jitter calculation on the
receiving side gives invalid result during the transition between two
clock rates, as shown in Table 2. The capture and arrival time are
in seconds, starting at the beginning of the capture of the first
packet; clock rate is in Hz; the RTP timestamp does not include the
random offset; the transit, jitter, and average jitter use the clock
rate as unit.

Capt. time	Clock rate	RTP timestamp	Arrival time	Transit	Jitter	Average jitter
0	8000	0	0.1	800		
0.02	8000	160	0.12	800	0	0
0.04	8000	320	0.14	800	0	0
0.06	8000	480	0.16	800	0	0
0.08	16000	800	0.18	2080	480	30
0.1	16000	1120	0.2	2080	0	28
0.12	16000	1440	0.22	2080	0	26
0.14	8000	1600	0.24	320	720	70
0.16	8000	1760	0.26	320	0	65

                             Table 2

Calculating the correct transit time on the receiving side can be
done by using the following formulas:

(1) current_time_capture = current_timestamp - previous_timestamp) /
     current_clock_rate + previous_time_capture
(2) transit = current_clock_rate * (time_arrival -
     current_time_capture)
(3) previous_time_capture = current_time_capture

The main problem with this method, in addition to the fact that the
jitter calculation described in RFC 3550 cannot be used, is that is
it dependent on the previous RTP packets, packets that can be
reordered or lost in the network. But it seems that this is what
most implementations are using.

3. Terminology

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].

   Clock rate: The multiplier used to convert from a wallclock value in
      seconds to an equivalent RTP timestamp value (without the fixed
      random offset). Note that RFC 3550 uses various terms like "clock
      frequency", "media clock rate", "timestamp unit", "timestamp
      frequency", and "RTP timestamp clock rate" as synonymous to clock
      rate.
   RTP Sender: A logical network element that sends RTP packets, sends
      RTCP SR packets, and receives RTCP RR packets.
   RTP Receiver: A logical network element that receives RTP packets,
      receives RTCP SR packets, and sends RTCP RR packets.

4. Recommendations

4.1. RTP Sender

   An RTP Sender with RTCP turned off (i.e. by setting the RS and RR
   bandwidth modifiers defined in [RFC3556] to 0) SHOULD use a different
   SSRC for each different clock rate but MAY use different clock rates
   on the same SSRC as long as the RTP timestamp without the random
   offset is calculated as explained below:

   [[This was designed to help VoIP implementations who anyway never
   cared about RTCP. Do we want to keep this?]]

   Each time the clock rate changes, the start_offset and capture_start
   values are calculated with the following formulas:

   start_offset = (capture_time - capture_start) * previous_clock_rate
   capture_start = capture_time

   For the first RTP packet, the values are initialized with the
   following formulas:

   start_offset = 0
   capture_start = capture_time

   After eventually updating this values, the RTP timestamp is
   calculated with the following formula:

   timestamp = (capture_time - capture_start) * clock_rate +
   start_offset

   Note that in all the formulas, capture_time is the first instant the
   new timestamp rate is used.

An RTP Sender with RTCP turned on MUST use a different SSRC for each
different clock rate. An RTCP BYE MUST be sent and a new SSRC MUST
be used if the clock rate switches back to a value already seen in
the RTP stream.

To accelerate lip synchronization, the next compound RTCP packet sent
by the RTP sender MUST contain multiple SR packets, the first one
containing the mapping for the current clock rate and the next SR
packets containing the mapping for the other clock rates seen during
the last period.

[[Some legacy implementations may dislike receiving multiple SR
packets. What should we do?]]

The RTP extension defined in [RFC6051] MAY be used to accelerate the
synchronization.

4.2. RTP Receiver

An RTP Receiver MUST calculate the jitter using the following
formula:

$$D(i,j) = (arrival\_time\_j * clock\_rate\_i - timestamp\_j) - (arrival\_time\_i * clock\_rate\_i - timestamp\_i)$$

An RTP Receiver MUST be able to handle a compound RTCP packet with
multiple SR packets.

For interoperability with legacy RTP implementations, an RTP receiver
MAY use the information in two consecutive SR packets to calculate
the clock rate used, i.e. if Ni is the NTP timestamp for the SR
packet i, Ri the RTP timestamp for the SR packet i and Nj and Rj the
NTP timestamp and RTP timestamp for the previous SR packet j, then
the clock rate can be guessed as the closest to (Ri - Rj) / (Ni -
Nj).

5. Interoperability Analysis

The next subsections analyze the various combinations between legacy
RTP implementations and RTP implementations that follow this document
specifications.

TBD

6. Security Considerations

   TBD

7. IANA Considerations

   No IANA considerations.

8. Acknowledgements

   Thanks to Colin Perkins, Ali C. Begen and Magnus Westerlund for their
   comments, suggestions and questions that helped to improve this
   document.

   Thanks to Robert Sparks and the attendees of SIPit 26 for the survey
   on multiple clock rates interoperability.

   This document was written with the xml2rfc tool described in
   [RFC2629].

9. References

9.1. Normative References

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              Requirement Levels", BCP 14, RFC 2119, March 1997.

   [RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V.
              Jacobson, "RTP: A Transport Protocol for Real-Time
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9.2. Informative References

   [RFC2629] Rose, M., "Writing I-Ds and RFCs using XML", RFC 2629,
              June 1999.

   [RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and
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   [RFC3556] Casner, S., "Session Description Protocol (SDP) Bandwidth
              Modifiers for RTP Control Protocol (RTCP) Bandwidth",
              RFC 3556, July 2003.

   [RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control

                    Protocol Extended Reports (RTCP XR)", RFC 3611,
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   [RFC5450]   Singer, D. and H. Desineni, "Transmission Time Offsets in
               RTP Streams", RFC 5450, March 2009.

   [RFC5484]   Singer, D., "Associating Time-Codes with RTP Streams",
               RFC 5484, March 2009.

   [RFC5760]   Ott, J., Chesterfield, J., and E. Schooler, "RTP Control
               Protocol (RTCP) Extensions for Single-Source Multicast
               Sessions with Unicast Feedback", RFC 5760, February 2010.

   [RFC6051]   Perkins, C. and T. Schierl, "Rapid Synchronisation of RTP
               Flows", RFC 6051, November 2010.

   [uRTR]   Wenger, S. and C. Perkins, "RTP Timestamp Frequency for
               Variable Rate Audio Codecs",
               draft-ietf-avt-variable-rate-audio-00 (work in progress),
               October 2004.

Appendix A. Using a fixed clock rate

   An alternate way of fixing the multiple clock rates issue was
   proposed in [uRTR]. This document proposed to define a unified clock
   rate, but the proposal was rejected at IETF 61.

Appendix B. Release notes

   This section must be removed before publication as an RFC.

B.1. Modifications between draft-ietf-avtext-multiple-clock-rates-01
      and draft-ietf-avtext-multiple-clock-rates-00

   o New text says that the algorithms listed are the one currently
      known.
   o Explains capture_time.
   o Nits.

B.2. Modifications between draft-ietf-avtext-multiple-clock-rates-00
      and draft-petithuguenin-avtext-multiple-clock-rates-01

   o Changed capture_state to capture_start.

B.3. Modifications between
      draft-petithuguenin-avtext-multiple-clock-rates-01 and
      draft-petithuguenin-avtext-multiple-clock-rates-00

   o Clarified the goals for this documents
   o Removed the non-monotonic method (replaced by Magnus formula).
   o Moved the "RTP Sender and RTP Receiver section inside a new
      "Recommendations" section.
   o Inserted the new Sender formula inside the Recommendation section.
   o Inserted the new jitter formula in the RTP Receiver section.
   o Emptied the Analysis sections.

B.4. Modifications between
      draft-petithuguenin-avtext-multiple-clock-rates-00 and
      draft-petithuguenin-avt-multiple-clock-rates-03

   o Initial release for avtext WG.

B.5. Modifications between
      draft-petithuguenin-avt-multiple-clock-rates-03 and
      draft-petithuguenin-avt-multiple-clock-rates-02

   o Updated RFC reference.

B.6. Modifications between
      draft-petithuguenin-avt-multiple-clock-rates-02 and
      draft-petithuguenin-avt-multiple-clock-rates-01

   o Having multiple SRs in a compound RTCP packet is OK.
   o If RTCP is used, must send a BYE and not reuse the SSRC.
   o Removed resolved notes.
   o Acknowledged SIPit 26 survey.
   o Fixed some nits.

B.7. Modifications between
      draft-petithuguenin-avt-multiple-clock-rates-01 and
      draft-petithuguenin-avt-multiple-clock-rates-00

   o Complete rewrite as a Standard Track I-D modifying RFC 3550.

Author's Address

   Marc Petit-Huguenin
   Stonyfish, Inc.

   Email: petithug@acm.org