

Network Working Group
Internet-Draft
Updates: [3550](#) (if approved)
Intended status: Standards Track
Expires: October 24, 2013

M.P.H. Petit-Huguenin
Impedance Mismatch
G. Zorn, Ed.
Network Zen
April 22, 2013

Support for Multiple Clock Rates in an RTP Session
draft-ietf-avtext-multiple-clock-rates-09

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](#).

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at <http://datatracker.ietf.org/drafts/current/>.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on October 24, 2013.

Copyright Notice

Copyright (c) 2013 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to [BCP 78](#) and the IETF Trust's Legal Provisions Relating to IETF Documents (<http://trustee.ietf.org/license-info>) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.

Internet-Draft

Multiple Clock Rates

April 2013

Table of Contents

1.	Introduction	2
2.	Terminology	4
3.	Legacy RTP	4
3.1.	Different SSRC	4
3.2.	Same SSRC	4
3.2.1.	Monotonic timestamps	5
3.2.2.	Non-monotonic timestamps	5
4.	Recommendations	6
4.1.	RTP Sender (with RTCP)	6
4.2.	RTP Sender (without RTCP)	6
4.3.	RTP Receiver	7
5.	Security Considerations	7
6.	IANA Considerations	7
7.	Acknowledgements	7
8.	References	8
8.1.	Normative References	8
8.2.	Informative References	8
Appendix A.	Example Values	9
Appendix B.	Using a Fixed Clock Rate	11
Appendix C.	Behavior of Legacy Implementations	11
C.1.	libccrtp 2.0.2	11
C.2.	libmediastreamer0 2.6.0	11
C.3.	libpjmedia 1.0	12
C.4.	Android RTP stack 4.0.3	12
	Authors' Addresses	12

[1.](#) Introduction

The clock rate is a parameter of the payload format as identified in RTP and RTCP by the payload type value. It is often defined as being the same as the sampling rate but that is not always the case (see e.g. the G722 and MPA audio codecs [[RFC3551](#)]).

An RTP sender can switch between different payloads during the lifetime of an RTP session and because clock rates are defined by payload format, it is possible that the clock rate will also vary during an RTP session. Schulzrinne, et al. [[RFC3550](#)] lists using multiple clock rates as one of the reasons to not use different payloads on the same SSRC but unfortunately this advice has not always been 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 3](#) and subsections all of the algorithms known to be used in existing RTP implementations at the time of writing. These sections are not normative.

[Section 4](#) and subsections then recommend a unique algorithm that modifies [RFC 3550](#). These sections are normative.

[2.](#) 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 [RFC 2119](#) [[RFC2119](#)]. In addition, this document uses the following terms:

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.

[3.](#) 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.

[3.1.](#) 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.

[3.2.](#) 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 subsections present the algorithms used in the field.

[3.2.1.](#) Monotonic timestamps

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

```
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 an 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.

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

1. $\text{current_capture_time} = (\text{current_timestamp} - \text{previous_timestamp}) / \text{current_clock_rate} + \text{previous_capture_time}$
2. $\text{transit} = \text{current_clock_rate} * (\text{arrival_time} - \text{current_capture_time})$
3. $\text{previous_capture_time} = \text{current_capture_time}$

The main problem with this method, in addition to the fact that the jitter calculation described in [RFC 3550](#) cannot be used, is that it is dependent on the previous RTP packets, packets that can be reordered or lost in the network.

[3.2.2.](#) Non-monotonic timestamps

An alternate way of generating the RTP timestamps is to use the following formula:

$$\text{timestamp} = \text{capture_time} * \text{clock_rate}$$

With this formula, the jitter calculation is correct but the RTP timestamp values are no longer increasing monotonically as shown in Table 3. [RFC 3550](#) states that "[t]he sampling instant MUST be derived from a clock that increments monotonically[...]" but nowhere says that the RTP timestamp must increment monotonically.

The advantage with this method is that it works with the jitter calculation described in [RFC 3550](#), as long as the correct clock rates are used. It seems that this is what most implementations are using.

[4.](#) Recommendations

The following subsections describe behavioral recommendations for RTP senders (with and without RTCP) and RTP receivers.

[4.1.](#) RTP Sender (with RTCP)

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.

The RTP extension defined in Perkins & Schierl [[RFC6051](#)] MAY be used to accelerate the synchronization.

[4.2.](#) RTP Sender (without RTCP)

An RTP Sender with RTCP turned off (i.e. by setting the RS and RR bandwidth modifiers [[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 is calculated as explained below:

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 values:

```
start_offset = random_initial_offset
capture_start = capture_time
```

After eventually updating these 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_start is the first instant that the new timestamp rate is used. The output of the above method is exemplified in Table 4.

[4.3.](#) RTP Receiver

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

$$D(i,j) = (\text{arrival_time_j} * \text{clock_rate_i} - \text{timestamp_j}) \\ - (\text{arrival_time_i} * \text{clock_rate_i} - \text{timestamp_i})$$

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

[5.](#) Security Considerations

This document is not believed to effect the security of the RTP sessions described here in any way.

[6.](#) IANA Considerations

This document requires no IANA actions.

[7.](#) Acknowledgements

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

Thanks to Bo Burman (who provided the values in Table 4).

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

[[RFC2629](#)].

8. References

8.1. Normative References

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate 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 Applications", STD 64, [RFC 3550](#), July 2003.

8.2. Informative References

- [I-D.ietf-avt-variable-rate-audio]
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.
- [RFC2629] Rose, M.T., "Writing I-Ds and RFCs using XML", [RFC 2629](#), June 1999.
- [RFC3551] Schulzrinne, H. and S. Casner, "RTP Profile for Audio and Video Conferences with Minimal Control", STD 65, [RFC 3551](#), July 2003.
- [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](#), November 2003.
- [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.

[Appendix A](#). Example Values

The following tables illustrate the timestamp and jitter values produced when the various methods discussed in the text are used.

The values shown are purely exemplary, illustrative and non-normative.

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: Monotonic Timestamps

Internet-Draft

Multiple Clock Rates

April 2013

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	1280	0.18	1600	0	0
0.1	16000	1600	0.2	1600	0	0
0.12	16000	1920	0.22	1600	0	0
0.14	8000	1120	0.24	800	0	0
0.16	8000	1280	0.26	800	0	0

Table 3: Non-monotonic Timestamps

Internet-Draft

Multiple Clock Rates

April 2013

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	640	0.18	1600	0	0
0.1	16000	960	0.2	1600	0	0
0.12	16000	1280	0.22	1600	0	0
0.14	8000	1600	0.24	320	0	0
0.16	8000	1760	0.26	320	0	0

Table 4: Recommended Method for RTP Sender (without RTCP)

[Appendix B](#). Using a Fixed Clock Rate

An alternate way of fixing the multiple clock rates issue was proposed by Wenger & Perkins [[I-D.ietf-avt-variable-rate-audio](#)]. This document proposed to define a unified clock rate, but the proposal was rejected at IETF 61.

[Appendix C](#). Behavior of Legacy Implementations

[C.1](#). libccrtp 2.0.2

This library uses the formula described in [Section 3.2.2](#).

Note that this library uses `gettimeofday(2)` which is not guaranteed to increment monotonically, like when the clock is adjusted by NTP.

[C.2](#). libmediastreamer0 2.6.0

This library (which uses the oRTP library) uses the formula described in [Section 3.2.2](#).

Note that in some environments this library uses `gettimeofday(2)` which is not guaranteed to increment monotonically.

Petit-Huguenin & Zorn Expires October 24, 2013

[Page 11]

Internet-Draft

Multiple Clock Rates

April 2013

[C.3](#). libpjmedia 1.0

This library uses the formula described in [Section 3.2.2](#).

[C.4](#). Android RTP stack 4.0.3

This library changes the SSRC each time the format changes, as described in [Section 3.1](#).

Authors' Addresses

Marc Petit-Huguenin
Impedance Mismatch

Email: petithug@acm.org

Glen Zorn (editor)
Network Zen
227/358 Thanon Sanphawut
Bang Na, Bangkok 10260
Thailand

Phone: +66 (0) 8-1000-4155

Email: glenzorn@gmail.com

