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RTCP XR Blocks for Synchronization Delay and Offset Metrics Reporting
draft-asaeda-xrblock-rtcp-xr-synchronization-04

Abstract

This document defines two RTCP XR Report Blocks and associated SDP parameters that allow the reporting of synchronization delay and offset metrics for use in a range of RTP applications.

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 August 25, 2012.

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1. Introduction

This draft defines a new block type to augment those defined in [RFC3611], for use in a range of RTP applications.

This new block type supports reporting of Initial synchronization Delay to establish multimedia session. Information is recorded about time difference between the start of RTP sessions and the time the RTP receiver acquires all components of RTP sessions in the multimedia session [RFC6051].

It also supports reporting of the relative Synchronization offset time of two arbitrary streams (e.g., between audio and video streams), with the same RTCP CNAME included in RTCP SDES packets [RFC3550]. Information is recorded about the synchronization offset time of each RTP stream relative to the reference RTP stream with the same CNAME and General Synchronization Offset of zero.

These metrics belong to the class of terminal related transport level metrics defined in [MONARCH].

2. Terminology

2.1. Standards Language

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, the following terms are defined:

Synchronization offset:

> The time difference of the actual delay of the measured RTP stream relative to the expected delay of the reference RTP stream with the same CNAME.

3. Applicability

The report blocks defined in this document could be used by dedicated network monitoring applications.

When joining each session in layered video sessions [RFC6190] or the multimedia sessions, a receiver may not synchronize playout across the multimedia sessions or layered video sessions until RTCP SR

packets have been received on all the components of RTP sessions. The components of RTP sessions are referred to as each RTP stream for each media type in multimedia sessions or each RTP stream at each layer in the layered video sessions. For unicast session, the delay due to negotiation of NAT pinholes, firewall holes, quality-of-service, and media security keys is contributed to such initial synchronization playout. For multicast session, such initial synchronization delay varies with the session bandwidth,the number of members, and the number of senders in the session. The RTP flow Initial synchronization delay block can be used to report the initial synchronization delay to receive all the RTP streams belonging to the same multimedia session or layered video session. beyond the information carried in the standard RTCP packet format. In the absence of packet loss, the initial synchronization delay equals to the average time taken to receive the first RTCP packet in the RTP session with the longest RTCP reporting interval. In the presence of packet loss, the media synchronization needs to based on the in-band mapping of RTP and NTP- format timestamps [RFC6051] or wait until the reporting interval has passed, and the next RTCP SR packet is sent.

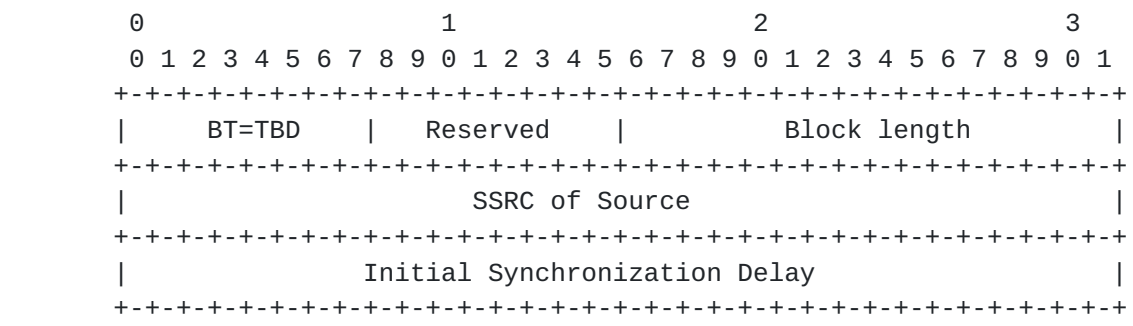
In an RTP multimedia session, there can be an arbitrary number of streams carried in different RTP streams, with the same RTCP CNAME. These streams may be not synchronized with each other. For example, one audio stream and one video stream belong to the same session and audio stream are transmitted lag behind video stream for multiple tens of milliseconds. The RTP Flows General Synchronization Offset block can be used to report such synchronization offset between video stream and audio stream.

4. RTP Flows Initial Synchronization Delay Report Block

This block is sent by RTP receivers and reports Initial synchronization delay beyond the information carried in the standard RTCP packet format. Information is recorded about time difference between the start of RTP sessions and the time the RTP receiver acquires all components of RTP sessions [RFC6051].

4.1. Metric Block Structure

The RTP Flows Initial Synchronization Delay Report Block has the following format:



4.2. Definition of Fields in RTP Flow Initial Synchronization Delay Metrics Block

Block type (BT): 8 bits

The Statistics Summary Report Block is identified by the constant <RFISD>.

Block length: 16 bits

The constant 2, in accordance with the definition of this field in Section 3 of RFC 3611 [RFC3611].

SSRC of Source: 32 bits

The SSRC of the media source SHALL be set to the value of the SSRC identifier carried in the arbitrary RTP stream belonging to the same multimedia session.

Initial Synchronization Delay: 32 bits

The average delay, expressed in units of 1/65536 seconds, between the RTCP packets received on all of the components RTP sessions and the beginning of session [RFC6051]. The value is calculated based on the information contained in RTCP SR packets or the in-band mapping of RTP and NTP- format timestamps [RFC6051]. If there is no packet loss, the initial synchronisation delay is expected to be equal to the average time taken to receive the first RTCP packet in the RTP session with the longest RTCP reporting interval.

5. RTP Flows General Synchronization Offset Metrics Block

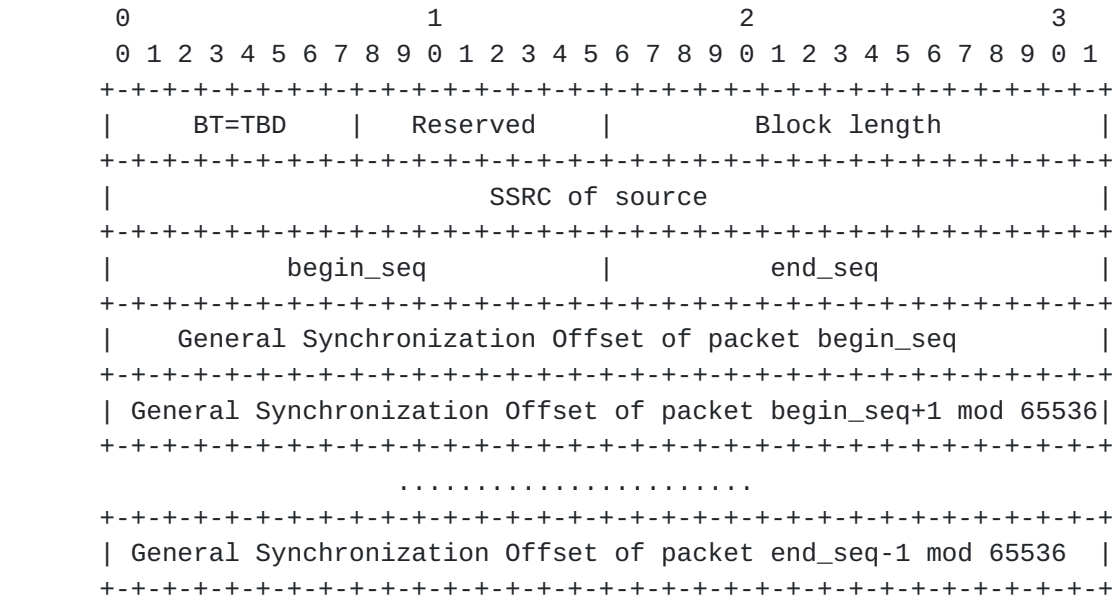
In the RTP multimedia sessions, there can be an arbitrary number of streams and each stream (e.g., audio stream or video stream) is sent in a separate RTP stream. The receiver associates RTP streams to be synchronised by means of RTCP CNAME contained in the RTCP Source

Description (SDES) packets [RFC3550].

This block is sent by RTP receivers and reports the general synchronization offset time of these RTP streams to be synchronized. Information is recorded about the time difference of the actual delay of the measured RTP stream relative to the expected delay of the reference RTP stream with the same CNAME. The expected delay of the reference RTP stream can be chosen as the first stream delay or the average delay of all the streams in the same session according to the similar rule defined in section 6.2.2.1 of [Y.1540].

5.1. Metric Block Structure

The RTP Flow General Synchronization Offset Report Block has the following format:



5.2. Definition of Fields in RTP Flow General Synchronization Offset Metrics Block

Block type (BT): 8 bits

The RTP Flow General Synchronization Offset Report Block is identified by the constant <RFGSO>.

Block length: 16 bits

The constant end_seq-begin_seq-1+2, in accordance with the definition of this field in Section 3 of RFC 3611 [RFC3611].

SSRC of Source: 32 bits

The SSRC of the media source SHALL be set to the value of the SSRC identifier of the reference RTP stream to which the XR relates.

begin_seq: 16 bits

The first sequence number that this block reports on.

end_seq: 16bits

The last sequence number that this block reports on plus one.

Packet i General synchronization offset: 32 bits

This field represents the synchronization offset time of one RTP stream in milliseconds relative to the reference RTP stream with the same CNAME and General Synchronisation Offset of zero [RFC6051] This value is calculated based on the interarrival time between an arbitrary RTP packet and the reference RTP packet with the same CNAME , and timestamps of this arbitrary RTP packet and the reference RTP packet with the same CNAME.

6. SDP Signaling

Two new parameters are defined for the two report blocks defined in this document to be used with Session Description Protocol (SDP) [RFC4566] using the Augmented Backus-Naur Form (ABNF) [RFC5234]. They have the following syntax within the "rtcp-xr" attribute [RFC3611]:

```
     rtcp-xr-attrib =  "a=rtcp-xr:"
                       [xr-format *(SP xr-format)] CRLF
          xr-format = RTP-flows-init-syn
                    / RTP-flows-general-syn
           RTP-flows-init-syn = "RTP-flows-init-syn"
                         ["=" max-size]
           RTP-flow-general-syn = "RTP-flows-general-syn"
                           ["=" max-size]
           max-size = 1*DIGIT ; maximum block size in octets
```

Refer to Section 5.1 of RFC 3611 [RFC3611] for a detailed description and the full syntax of the "rtcp-xr" attribute.

7. IANA Considerations

New report block types for RTCP XR are subject to IANA registration. For general guidelines on IANA allocations for RTCP XR, refer to Section 6.2 of [RFC3611].

This document assigns two new block type values in the RTCP XR Block Type Registry:

```
Name:       RFISD
Long Name:  RTP Flows Initial Synchronization Delay
Value       <RFISD>
Reference:  Section 4

Name:       RFGSO
Long Name:  RTP Flows General Synchronization Offset Metrics Block
Value       <RFGSO>
Reference:  Section 5
```

This document also registers two new SDP [RFC4566] parameters for the "rtcp-xr" attribute in the RTCP XR SDP Parameters Registry:

* "RTP-flows-init-syn"
* "RTP-flows-general-syn"

The contact information for the registrations is:

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8. Security Considerations

The new RTCP XR report blocks proposed in this document introduces no new security considerations beyond those described in [RFC3611].

9. Acknowledgements

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10. References

10.1. 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.

[RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 2003.

[RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", RFC 4566, July 2006.

[RFC5234] Crocker, D. and P. Overell, "Augmented BNF for Syntax Specifications: ABNF", STD 68, RFC 5234, January 2008.

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

[RFC6190] Wenger, S., Wang, Y., Schierl, T., and A. Eleftheriadis, "RTP Payload Format for Scalable Video Coding", RFC 6190, May 2011.

10.2. Informative References

[MONARCH] Wu, Q., "Monitoring Architectures for RTP", ID draft-ietf-avtcore-monarch-00, April 2011.

[Y.1540] ITU-T, "ITU-T Rec. Y.1540, IP packet transfer and availability performance parameters", November 2007.

Appendix A. Change Log

Note to the RFC-Editor: please remove this section prior to publication as an RFC.

A.1. draft-asaeda-xrblock-rtcp-xr-syncronization-00

This document is separated from draft-wu-xrblock-rtcp-xr-quality-monitoring-01 with some editorial changes and focuses on RTP Flow Initial Synchronization Delay and RTP Flows General Synchronization Offset.

A.2. draft-asaeda-xrblock-rtcp-xr-syncronization-01

Separate Synchronization Delay and Offset Metrics Block into two independent block based on comments on the list.

A.3. draft-asaeda-xrblock-rtcp-xr-syncronization-02

The following are the major changes compared to previous version 01:

- Clarify which synchronization is reported in section 4 and 5.
- Allow calculating the synchronization delay based on RTP header extension defined in RFC6051
- Explain what the components of RTP session are in section 3.

A.4. draft-asaeda-xrblock-rtcp-xr-syncronization-03

The following are the major changes compared to previous version 02:

- Support multiple general synchronization offset reporting.
- Other Editorial Changes.

A.5. draft-asaeda-xrblock-rtcp-xr-syncronization-04

The following are the major changes compared to previous version 03:

- Add a definition for synchronization offset.
- Use additional text in applicability section to clarify the difference between synchronization delay and offset.
- Add a reference to tell how to select the reference stream.
- Other Editorial Changes.

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