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RTCP XR Report Block for TS Decodability Statistics Metric reporting
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Abstract

Transport Stream is a standard container format used in the transmission and storage of multimedia data. Unicast/Multicast/Broadcast MPEG-TS over RTP is widely adopted in the IPTV deployment. This document defines an RTCP XR Report Block that allows the reporting of decodability statistics metrics related to transmissions of MPEG-TS over RTP.

Status of this Memo

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1. Introduction

The European Telecommunications Standards Institute (ETSI) has defined a set of syntax and information consistency tests and resulting indicators [ETSI] recommended for the monitoring of MPEG-2 Transport Streams (TS) [ISO-IEC.13818-1.2007]. The tests and corresponding indicators are grouped according to priority:

- o First priority - Necessary for de-codability (basic monitoring)
- o Second priority - Recommended for continuous or periodic monitoring
- o Third priority - Recommended for application-dependant monitoring

This draft is based on information consistency tests and resulting indicators in the [ETSI] and defines a new block type to augment those defined in [RFC3611] for use with Transport Stream (TS) [ISO-IEC.13818-1.2007]. The new block type supports reporting of the number of each indicator in the first and second priority; third priority indicators are not supported. This new block type can be useful for measuring content stream or TS quality by checking TS header information [ETSI] and identifying the existence, and characterizing the severity, of bitstream packetization problem which may affect users' perception of a service delivered over RTP; it may also be useful for verifying the continued correct operation of an existing system management.

The new report block is in compliance with the monitoring architecture specified in [MONARCH] and the Performance Metrics Framework [RFC6390]. The metric is applicable to any type of RTP application that uses TS standard format for container of multimedia data, for example MPEG4 TS content over RTP.

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].

3. TR 101 290 Decodability Statistics Metric Report Block

This block reports decodability statistics metrics beyond the information carried in the standard RTCP packet format. It defines eight metrics based on ETSI TR 101 290. Information is reported about basic monitoring parameters necessary to ensure that the TS can

be decoded including:

- o Transport Stream Synchronization Losses
- o Sync byte errors
- o Continuity count errors

and continuous monitoring parameters including:

- o Transport errors
- o Program Clock Reference (PCR) errors
- o PCR repetition errors
- o PCR discontinuity indicator errors
- o Presentation Time Stamp (PTS) errors

The other parameters are ignored since they are not applied to all the MPEG implementations. For further information on these parameters, see [ETSI]

The Decodability Metrics Block has the following format:

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								
BT=TBD										rvd										block length																			
SSRC of source																																							
begin_seq															end_seq																								
Number of TSs																																							
TS_sync_loss_count																																							
Sync_byte_error_count																																							
Continuity_count_error_count																																							
Transport_error_count																																							
PCR_error_count																																							
PCR_repetition_error_count																																							
PCR_discontinuity_indicator_error_count																																							
PTS_error_count																																							

block type (BT): 8 bits

A TR 101 290 decodability metrics report block is identified by the constant <TDM>.

rvd: 8 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and SHOULD be ignored by the receiver.

block length: 16 bits

The constant 11, in accordance with the definition of this field in Section 3 of RFC 3611 [RFC3611].

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

begin_seq: 16 bits

As defined in Section 4.1 of [RFC3611].

end_seq: 16 bits

As defined in Section 4.1 of [RFC3611].

Number of TSs: 32 bits

Number of TS in the above sequence number interval.

TS_sync_loss_count: 32 bits

Number of TS_sync_loss errors in the above sequence number interval.

Sync_byte_error_count: 32 bits

Number of sync_byte_errors in the above sequence number interval.

Continuity_count_error_count: 32 bits

Number of Continuity_count_errors in the above sequence number interval.

Transport_error_count: 32 bits

Number of Transport_errors in the above sequence number interval.

PCR_error_count: 32 bits

Number of PCR_errors in the above sequence number interval.

PCR_repetition_error_count: 32 bits

Number of PCR_repetition_errors in the above sequence number interval.

PCR_discontinuity_indicator_error_count: 32 bits

Number of PCR_discontinuity_indicator_errors in the above sequence number interval.

PTS_error_count: 32 bits

Number of PTS_errors in the above sequence number interval.

4. SDP Signaling

One new parameter is defined for the report block defined in this document to be used with Session Description Protocol (SDP) [RFC4566] using the Augmented Backus-Naur Form (ABNF) [RFC5234]. It has the following syntax within the "rtcp-xr" attribute [RFC3611]:

```
rtcp-xr-attr = "a=rtcp-xr:"  
               [xr-format *(SP xr-format)] CRLF
```

```
xr-format = decodability-metrics
```

```
decodability-metrics = "decodability-metrics"
```

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

5. 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 one new block type value in the RTCP XR Block Type Registry:

Name: TDM
Long Name: TR 101 290 Decodability Metrics
Value <TDM>
Reference: section 3

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

* "decodability-metrics"

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6. Security Considerations

This proposed RTCP XR report block introduces no new security considerations beyond those described in [RFC3611].

7. References

7.1. Normative References

- [ETSI] ETSI, "Digital Video Broadcasting (DVB); Measurement guidelines for DVB systems", Technical Report TR 101 290, 2001.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
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- [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,

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[RFC6390] Clark, A. and B. Claise, "Guidelines for Considering New Performance Metric Development", BCP 170, RFC 6390, October 2011.

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RTP Control Protocol (RTCP) Extended Report (XR) Block for MPEG2
Transport Stream (TS) Program Specific Information (PSI) Independent
Decodability Statistics Metrics reporting
draft-ietf-xrblock-rtcp-xr-decodability-12

Abstract

An MPEG2 Transport Stream (TS) is a standard container format used in the transmission and storage of multimedia data. Unicast/Multicast MPEG2 TS over RTP is widely deployed in IPTV systems. This document defines an RTP Control Protocol (RTCP) Extended Report (XR) Block that allows the reporting of MPEG2 TS decodability statistics metrics related to transmissions of MPEG2 TS over RTP. The metrics specified in the RTCP XR Block are not dependent on Program specific information carried in MPEG TS.

Status of this Memo

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1. Introduction

1.1. MPEG2 Transport Stream Decodability Metrics

The European Telecommunications Standards Institute (ETSI) has defined a set of syntax and information consistency tests and corresponding indicators [ETSI] that are recommended for the monitoring of MPEG2 Transport Streams [ISO-IEC.13818-1.2007]. The tests and corresponding indicators are grouped according to priority:

- o First priority - Necessary for decodability (basic monitoring)
- o Second priority - Recommended for continuous or periodic monitoring
- o Third priority - Recommended for application-dependent monitoring

This memo is based on information consistency tests and resulting indicators defined by ETSI [ETSI] and defines a new block type to augment those defined in [RFC3611] for use with MPEG2 Transport Stream (TS) [ISO-IEC.13818-1.2007]. The new block type supports reporting of the number of occurrences of each Program Specific Information (PSI) Independent indicator in the first and second priorities; third priority indicators are not supported.

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. [RFC3611] defined an extensible structure for reporting using an RTCP Extended Report (XR). This document defines a new Extended Report block for use with [RFC3550] and [RFC3611].

1.3. Performance Metrics Framework

The Performance Metrics Framework [RFC6390] provides guidance on the definition and specification of performance metrics. The RTP Monitoring Architectures [RFC6792] provides guideline for reporting block format using RTCP XR. The new report block described in this memo is in compliance with the monitoring architecture specified in [RFC6792] and the Performance Metrics Framework [RFC6390].

1.4. Applicability

This block type allows a counts of MPEG Transport Stream quality metrics that are measured in accordance with ETSI TR 101290 [ETSI] to be reported by an endpoint. These metrics are useful for identifying bitstream packetization and transport stream encoding problems that may affect the user's perception of a video service delivered over RTP.

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].

3. MPEG2 TS PSI Independent Decodability Statistics Metrics Block

ETSI TR 101290 [ETSI] generally defines metrics related to error events while this document contains counts of those metrics defined in [ETSI]. The block defined in this document reports MPEG2 TS PSI Independent decodability statistics metrics beyond the information carried in the standard RTCP packet format, which are measured at the receiving end of the RTP stream. It contains counts of eight metrics defined in ETSI TR 101290 [ETSI]. Information is reported about basic monitoring parameters necessary to ensure that the TS can be decoded including:

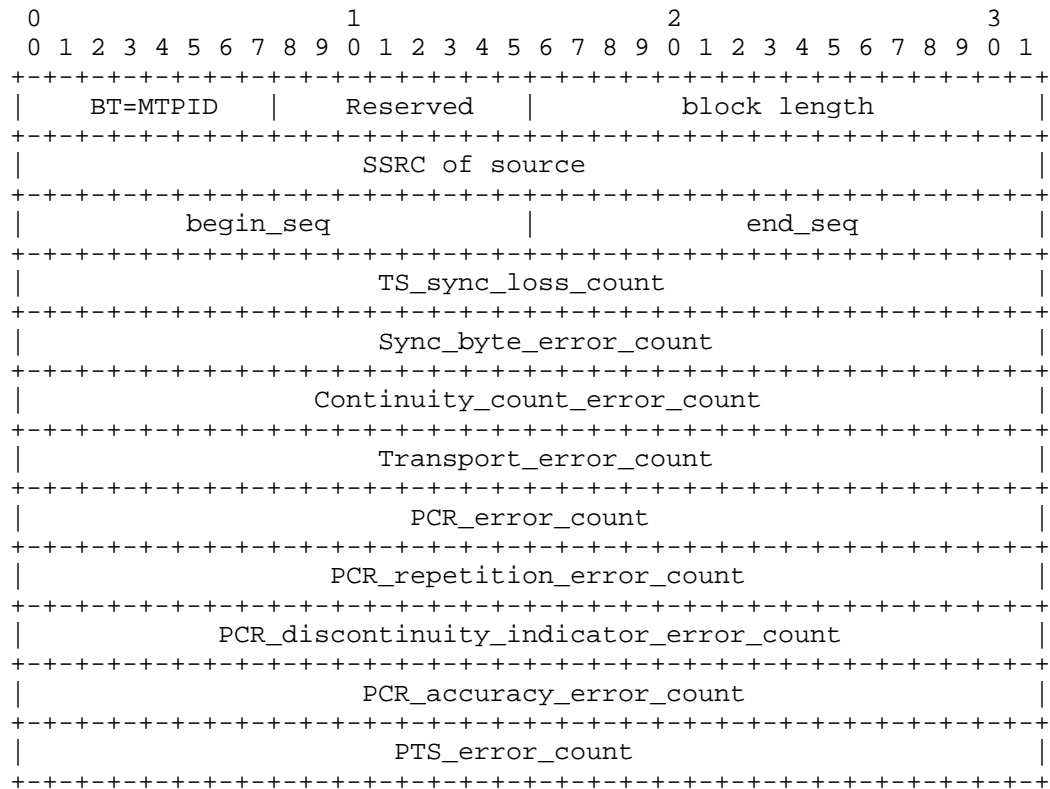
- o Transport Stream Synchronization Losses
- o Sync byte errors
- o Continuity count errors

and continuous monitoring parameters necessary to ensure the continuous decoding including:

- o Transport errors
- o Program Clock Reference (PCR) errors
- o PCR repetition errors
- o PCR discontinuity indicator errors
- o PCR accuracy errors
- o Presentation Time Stamp (PTS) errors

The other parameters are ignored since they do not apply to all MPEG2 implementations. For further information on these parameters, see [ETSI]. Note that when the report of this block spans across more than one measurement intervals [RFC6776], the count of the metrics (e.g., Sync byte errors, PCR errors)defined in [ETSI] may reflect a problem in the current or previous measurement interval.

The MPEG2 TS PSI Independent Decodability Metrics Block has the following format:



block type (BT): 8 bits

The MPEG2 TS PSI Independent Decodability Metrics Block is identified by the constant <MTPID>.

Reserved: 8 bits

These bits are reserved. They MUST be set to zero by senders and ignored by receivers (See [RFC6709] section 4.2).

block length: 16 bits

The constant 11, in accordance with the definition of this field in Section 3 of RFC 3611. The block MUST be discarded if the block length is set to a different value.

SSRC of source: 32 bits

As defined in Section 4.1 of RFC 3611.

begin_seq: 16 bits

The RTP sequence number corresponding to the start of the measurement period, as defined in Section 4.1 of RFC 3611.

end_seq: 16 bits

The RTP sequence number corresponding to the end of the measurement period, as defined in Section 4.1 of RFC 3611.

TS_sync_loss_count: 32 bits

A count of the number of TS_sync_loss errors that occurred in the above sequence number interval. A TS_sync_loss error occurs when there are two or more consecutive incorrect sync bytes within the MPEG TS stream, as defined in section 5.2.1 of [ETSI].

Sync_byte_error_count: 32 bits

A count of the number of Sync_byte_errors that occurred in the above sequence number interval. A sync byte error occurs when the sync byte is not equal to 0x47 in any TS packet contained in the MPEG TS stream, as defined in section 5.2.1 of [ETSI].

Continuity_count_error_count: 32 bits

A count of the number of Continuity_count_errors that occurred in the above sequence number interval. A Continuity_count_error occurs when any of the following faults happen within the MPEG TS stream - incorrect packet order, a packet occurs more than twice or a packet is lost, as defined in the section 5.2.1 of [ETSI].

Transport_error_count: 32 bits

A count of the number of Transport_errors that occurred in the above sequence number interval. A Transport_error occurs when erroneous TS packet can not be corrected within the MPEG TS stream. as defined in the section 5.2.2 of [ETSI].

PCR_error_count: 32 bits

A count of the number of PCR_errors that occurred in the above sequence number interval. A PCR_error occurs if the primary clock reference (PCR) is not seen for more than 100ms within the MPEG TS

stream, as defined in the section 5.2.2 of [ETSI]. The time interval between two consecutive PCR values should be no more than 40ms.

PCR_repetition_error_count: 32 bits

A count of the number of PCR_repetition_errors that occurred in the above sequence number interval. A PCR_repetition_error occurs when the time interval between two consecutive PCR values is more than 40ms within the MPEG TS stream, as defined in the section 5.2.2 of [ETSI].

PCR_discontinuity_indicator_error_count: 32 bits

A count of the number of PCR_discontinuity_indicator_errors that occurred in the above sequence number interval. A PCR_discontinuity_indicator_error occurs if the time interval between two consecutive PCR values is more than 100ms within the MPEG TS stream, as defined in the section 5.2.2 of [ETSI].

PCR_accuracy_error_count: 32 bits

A count of the Number of PCR_accuracy_errors that occurred in the above sequence number interval. A PCR_accuracy_error occurs when the PCR accuracy of the selected program is outside the range of +/-500ns within the MPEG TS stream, as defined in the section 5.2.2 of [ETSI].

PTS_error_count: 32 bits

A count of the number of PTS_errors that occurred in the above sequence number interval. A PTS_error occurs when the PTS repetition is more than 700ms within the MPEG TS stream, as defined in the section 5.2.2 of [ETSI]. Note that the PTS is contained in the MPEG-2 TS stream and is used to aid the decoder in presenting the program on time, at the correct speed and synchronized.

4. SDP Signaling

RFC 3611 defines the use of SDP (Session Description Protocol) [RFC4566] for signaling the use of RTCP XR blocks. However XR blocks MAY be used without prior signaling (See section 5 of RFC3611).

4.1. SDP rtcp-xr-attrib Attribute Extension

This session augments the SDP attribute "rtcp-xr" defined in Section 5.1 of RFC 3611 by providing an additional value of "xr-format" to signal the use of the report block defined in this document.

xr-format =/ xr-tpid-block

xr-tpid-block = "ts-psi-indep-decodability"

4.2. Offer/Answer Usage

When SDP is used in offer-answer context, the SDP Offer/Answer usage defined in [RFC3611] for unilateral "rtcp-xr" attribute parameters applies. For detailed usage of Offer/Answer for unilateral parameter, refer to section 5.2 of [RFC3611].

5. 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 RFC 3611.

5.1. New RTCP XR Block Type value

This document assigns the block type value MTPID in the IANA " RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry " to the "MPEG2 Transport Stream PSI Independent Decodability Statistics Metrics Block".

[Note to RFC Editor: please replace MPITD with the IANA provided RTCP XR block type for this block.]

5.2. New RTCP XR SDP Parameter

This document also registers a new parameter "ts-psi-indep-decodability" in the "RTP Control Protocol Extended Reports (RTCP XR) Session Description Protocol (SDP) Parameters Registry".

5.3. Contact information for registrations

The contact information for the registrations is:

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6. Security Considerations

There might be some relationship between reported error counters and contractual Service Level Agreements (SLA)s and hence an attack (e.g., RTP endpoints report fake information, or an attacker in the path modifies the data being reported) may deliberately corrupt these error counters fields and will result in financial implications for the network operator (either as a result of un-needed Performance metrics, or penalty charges for SLA failure).

A solution to prevent such attack is to apply an authentication and integrity protection framework for the RTCP XR report block. This can be accomplished using the RTP profile that combines Secure RTP [RFC3711] and AVPF into SAVPF [RFC5124].

Besides this, the proposed RTCP XR report block in this document introduces no other new security considerations beyond those described in [RFC3611].

7. Acknowledgements

Thanks to Ray van Brandenburg, Claire Bi, Colin Perkin, Roni Even, Dan Romascanu, Ali Begen and Alan Clark for useful review and suggestions.

8. References

8.1. Normative References

- [ETSI] ETSI, "Digital Video Broadcasting (DVB); Measurement guidelines for DVB systems", Technical Report TR 101 290, 2001.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
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pictures and associated audio information: Systems",
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- [RFC6792] Wu, Q., Hunt, G., and P. Arden, "Guidelines for Use of the RTP Monitoring Framework", RFC 6792, November 2012.

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RTCP XR Report Block for Discard Count metric Reporting
draft-ietf-xrblock-rtcp-xr-discard-05.txt

Abstract

This document defines an RTCP XR Report Block that allows the reporting of a simple discard count metric for use in a range of RTP applications.

Status of this Memo

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1. Introduction

1.1. Discard Count Report Block

This document defines a new block type to augment those defined in [RFC3611] for use in a range of RTP applications. The new block type supports the reporting of the number of packets which are received correctly but are never played out, typically because they arrive too late to be played out (buffer underflow) or too early (buffer overflow). The metric is applicable both to systems which use packet loss repair techniques (such as forward error correction [RFC5109] or retransmission [RFC4588]) and to those which do not.

This metric is useful for identifying the existence, and characterising the severity, of a packet transport problem which may affect users' perception of a service delivered over RTP.

The metric belongs to the class of transport-related terminal metrics defined in [MONARCH].

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. [RFC3611] defined an extensible structure for reporting using an RTCP Extended Report (XR). This document defines a new Extended Report block. The use of Extended Report blocks is defined by [RFC3611].

1.3. Performance Metrics Framework

The Performance Metrics Framework [RFC6390] provides guidance on the definition and specification of performance metrics. The RTP Monitoring Architectures [MONARCH] provides guideline for reporting block format using RTCP XR. The Metrics Block described in this document are in accordance with the guidelines in [RFC6390] and [MONARCH].

1.4. Applicability

This metric is believed to be applicable to a large class of RTP applications which use a jitter buffer.

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:

Received, Lost and Discarded

A packet shall be regarded as lost if it fails to arrive within an implementation-specific time window. A packet that arrives within this time window but is too early or late to be played out or thrown away before playout due to packet duplication or redundancy shall be regarded as discarded. A packet shall be classified as one of received (or OK), discarded or lost. The Discard Count Metric counts only discarded packets. The metric "cumulative number of packets lost" defined in [RFC3550] reports a count of packets lost from the media stream (single SSRC within single RTP session). Similarly the metric "number of packets discarded" reports a count of packets discarded from the media stream (single SSRC within single RTP session) arriving at the receiver. Another metric defined in [RFC5725] is available to report on packets which are not recovered by any repair techniques which may be in use.

3. Discard Count Metric Report Block

3.1. Report Block Structure

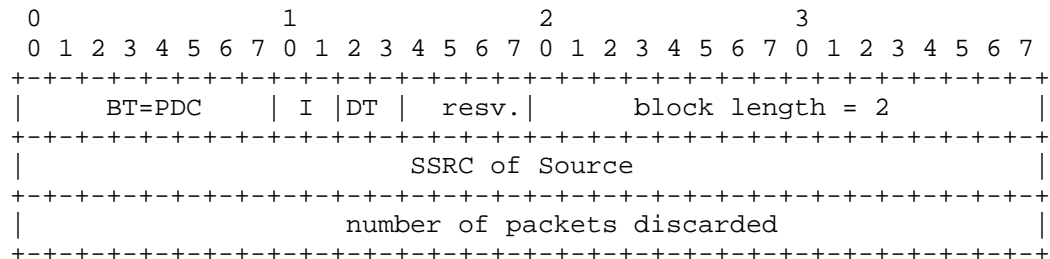


Figure 1: Report Block Structure

3.2. Definition of Fields in Discard Count Metric Report Block

Block type (BT): 8 bits

A Discard Count Metric Report Block is identified by the constant PDC.

[Note to RFC Editor: please replace PDC with the IANA provided RTCP XR block type for this block.]

Interval Metric flag (I): 2 bits

This field is used to indicate whether the Discard Count Metric is an Interval or Cumulative metric, Sample metric [MONARCH], that is, whether the reported values applies to the most recent measurement interval duration between successive metrics reports (I=10) (the Interval Duration) or to the accumulation period characteristic of cumulative measurements (I=11) (the Cumulative Duration) or is a sampled instantaneous value (I=01) (Sampled Value). In this document, Discard Count Metric is not measured at a particular time instant but over one or several reporting intervals. Therefore Discard Count Metric MUST not be chosen as Sampled Metric.

Discard Type (DT): 2bits

This field is used to identify the discard type used in this report block. The discard type is defined as follows:

00: Report packet discarded or being thrown away before playout due to packets duplication.

01: Report packet discarded due to too early to be played out.

10: Report packet discarded due to too late to be played out.

11: Report the total number of discarded packets due to both early and late to be played out.

An endpoint MAY send only one of the discard types (early, late, duplication packets discard) in one RTCP report or choose to report early (DT=1) and late (DT=2), duplication packets discard (DT=0) in separate block. It MAY also send the combined early and late discard type (DT=2) in one RTCP compound packet, but not any other combination of the three Discard Types. The endpoint MUST not report the the total number of discarded packets covering all three discard types. Instead, two separate report blocks should be used to carry duplication packets discard and the combined early and late discard respectively.

Reserved (resv): 4 bits

These bits are reserved. They SHOULD be set to zero by senders and MUST be ignored by receivers.

block length: 16 bits

The length of this report block in 32-bit words, minus one. For the Discard Count block, the block length is equal to 2.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

number of packets discarded: 32 bits

Number of packets discarded over the period (Interval or Cumulative) covered by this report.

If the measured value exceeds 0xFFFFFFFF, the value 0xFFFFFFFF MUST be reported to indicate an over-range measurement. If the measurement is unavailable, the value 0xFFFFFFFF MUST be reported.

Note that the number of packets expected in the period associated with this metric (whether interval or cumulative) is available from the difference between a pair of extended sequence numbers in the Measurement Information block [MEASI], so need not be repeated in this block.

4. SDP Signaling

[RFC3611] defines the use of SDP (Session Description Protocol) [RFC4566] for signaling the use of XR blocks. XR blocks MAY be used without prior signaling.

This section augments the SDP [RFC4566] attribute "rtcp-xr" defined in [RFC3611] by providing an additional value of "xr-format" to signal the use of the report block defined in this document.

```
rtcp-xr-attrib = "a=" "rtcp-xr" ":" [xr-format *(SP xr-format)] CRLF
```

(defined in [RFC3611])

```
xr-format =/ xr-pdc-block
```

```
xr-pdc-block = "pkt-dscrd-count"
```

5. IANA Considerations

New block types for RTCP XR are subject to IANA registration. For general guidelines on IANA considerations for RTCP XR, refer to [RFC3611].

5.1. New RTCP XR Block Type value

This document assigns the block type value PDC in the IANA "RTCP XR Block Type Registry" to the "Discard Count Metrics Block".

[Note to RFC Editor: please replace PDC with the IANA provided RTCP XR block type for this block.]

5.2. New RTCP XR SDP Parameter

This document also registers a new parameter "pkt-dscrd" in the "RTCP XR SDP Parameters Registry".

5.3. Contact information for registrations

The following contact information is provided for all registrations in this document:

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6. Security Considerations

It is believed that this proposed RTCP XR report block introduces no new security considerations beyond those described in [RFC3611]. This block does not provide per-packet statistics so the risk to confidentiality documented in Section 7, paragraph 3 of [RFC3611] does not apply.

7. Acknowledgments

The authors gratefully acknowledge the comments and contributions made by Bruce Adams, Philip Arden, Amit Arora, Bob Biskner, Kevin Connor, Claus Dahm, Randy Ethier, Roni Even, Jim Frauenthal, Albert Higashi, Tom Hock, Shane Holthaus, Paul Jones, Rajesh Kumar, Keith Lantz, Mohamed Mostafa, Amy Pendleton, Colin Perkins, Mike Ramalho, Ravi Raviraj, Albrecht Schwarz, Tom Taylor, and Hideaki Yamada, Kevin Gross, Varun Singh, Claire Bi, Roni Even, Dan Romascanu.

8. References

8.1. Normative References

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", March 1997.
- [RFC3550] Schulzrinne, H., "RTP: A Transport Protocol for Real-Time Applications", RFC 3550, July 2003.
- [RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", November 2003.
- [RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", July 2006.

8.2. Informative References

- [MEASI] Hunt, G., "Measurement Identity and information Reporting using SDES item and XR Block", ID draft-ietf-xrblock-rtcp-xr-meas-identity-06, April 2012.
- [MONARCH] Wu, Q., "Monitoring Architectures for RTP", ID draft-ietf-avtcore-monarch-12, April 2012.
- [RFC4588] Rey, J., "RTP Retransmission Payload Format", RFC 4588, July 2006.
- [RFC5109] Li, A., "RTP Payload Format for Generic Forward Error Correction", RFC 5109, July 2006.
- [RFC5725] Begen, A., "RTCP XR Report Block for Post-Repair Loss metric Reporting", RFC 5725, February 2010.
- [RFC6390] Clark, A. and B. Claise, "Framework for Performance Metric Development", RFC 6390, October 2011.

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RTP Control Protocol (RTCP) Extended Report (XR) Block for Discard Count
metric Reporting
draft-ietf-xrblock-rtcp-xr-discard-15.txt

Abstract

This document defines an RTP Control Protocol(RTCP) Extended Report (XR) Block that allows the reporting of a simple discard count metric 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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1. Introduction

1.1. Discard Count Report Block

This document defines a new block type to augment those defined in [RFC3611] for use in a range of RTP applications. The new block type supports the reporting of the number of packets which are received correctly but are never played out, typically because they arrive too late to be played out (buffer underflow) or too early (buffer overflow). The metric is applicable both to systems which use packet loss repair techniques (such as forward error correction [RFC5109] or retransmission [RFC4588]) and to those which do not.

This metric is useful for identifying the existence, and characterizing the severity, of a packet transport problem which may affect users' perception of a service delivered over RTP.

This block may be used in conjunction with [BGDISCARD] which provides additional information on the pattern of discarded packets. However the metric in [BGDISCARD] may be used independently of the metrics in this block.

When a Discard Count Metrics Block is sent together with a Burst Gap Discard Metrics Block (defined in [BGDISCARD]) to the media sender or RTP based network management system, the information carried in the Discard Count Metrics Block and the Burst Gap Discard Metrics Block allows systems receiving the blocks to calculate burst gap summary statistics (e.g., the gap discard rate).

The metric belongs to the class of transport-related end system metrics defined in [RFC6792].

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. [RFC3611] defined an extensible structure for reporting using an RTCP Extended Report (XR). This document defines a new Extended Report block for use with [RFC3550] and [RFC3611].

1.3. Performance Metrics Framework

The Performance Metrics Framework [RFC6390] provides guidance on the definition and specification of performance metrics. The RTP Monitoring Architectures [RFC6792] provides guideline for reporting block format using RTCP XR. The metrics block described in this document are in accordance with the guidelines in [RFC6390] and [RFC6792].

1.4. Applicability

This metric is believed to be applicable to a large class of RTP applications which use a de-jitter buffer [RFC5481].

Discards due to late or early arriving packets affects user experience. The reporting of discards alerts senders and other receivers to the need to adjust their transmission or reception strategies. The reports allow network managers to diagnose these user experience problems.

The ability to detect duplicate packets can be used by managers to detect network layer or sender behavior which may indicate network or device issues. Based on the reports, these issues may be addressed prior to any impact on user experience.

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:

Received, Lost and Discarded

A packet shall be regarded as lost if it fails to arrive within an implementation-specific time window. A packet that arrives within this time window but is either too early to be played out or too late to be played out or thrown away before playout due to packet duplication or redundancy shall be regarded as discarded. A packet shall not be regarded as discarded if it arrives within this time window but is dropped during decoding by some higher layer decoder, e.g., due to a decoding error. A packet shall be classified as one of received (or OK), discarded or lost. The Discard Count Metric counts only discarded packets. The metric "cumulative number of packets lost" defined in [RFC3550] reports a count of packets lost from the media stream (single Synchronization source (SSRC) within single RTP session). Similarly the metric "number of packets discarded" reports a count of packets discarded from the media stream (single SSRC within single RTP session) arriving at the receiver. Another metric defined in [RFC5725] is available to report on packets which are not recovered by any repair techniques which may be in use.

3. Discard Count Metric Report Block

Metrics in this block report on the number of packets discarded in the stream arriving at the RTP end system. The measurement of these metrics is made at the receiving end of the RTP stream. Instances of this metrics block refer by SSRC to the separate auxiliary Measurement Information Block [RFC6776] which describes measurement intervals in use. This metrics block relies on the measurement interval in the Measurement Information Block indicating the span of the report and MUST be sent in the same compound RTCP packet as the measurement information block. If the measurement interval is not received in the same compound RTCP packet as this metrics block, this metrics block MUST be discarded.

3.1. Report Block Structure

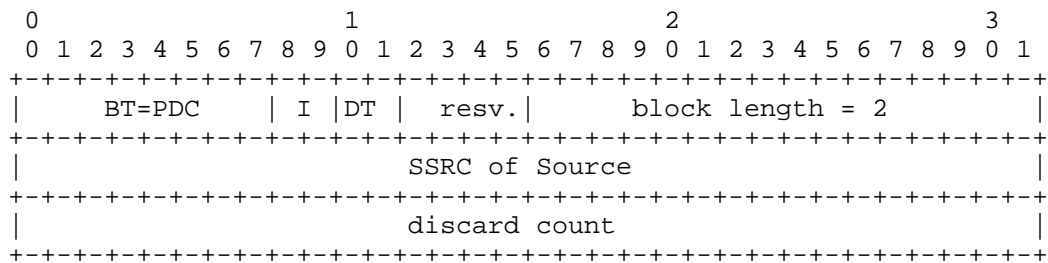


Figure 1: Report Block Structure

3.2. Definition of Fields in Discard Count Metric Report Block

Block type (BT): 8 bits

A Discard Count Metric Report Block is identified by the constant PDC.

[Note to RFC Editor: please replace PDC with the IANA provided RTCP XR block type for this block.]

Interval Metric Flag (I): 2 bits

This field indicates whether the reported metric is an interval, cumulative, or sampled metric [RFC6792]:

I=10: Interval Duration - the reported value applies to the most recent measurement interval duration between successive metrics reports.

I=11: Cumulative Duration - the reported value applies to the accumulation period characteristic of cumulative measurements.

I=01: Sampled Value - the reported value is a sampled instantaneous value.

In this document, the Discard Count Metric can only be measured over definite intervals, and cannot be sampled. Accordingly, the value I=01, indicating a sampled value, MUST NOT be sent, and MUST be discarded when received. In addition, the value I=00 is reserved and also MUST NOT be sent, and MUST be discarded when received.

Discard Type (DT): 2bits

This field is used to identify the discard type used in this report block. The discard type is defined as follows:

00: Report packet discarded or being thrown away before playout due to packets duplication.

01: Report packet discarded due to too early to be played out.

10: Report packet discarded due to too late to be played out.

The value DT=11 is reserved for future definition and MUST NOT be Sent, and MUST be discarded when received.

An endpoint MAY report any combination of discard types in each reporting interval by including several Discard Count Metric Report Blocks in a single RTCP XR packet.

Some systems send duplicate RTP packets for robustness or error resilience. This is NOT RECOMMENDED since it breaks RTCP packet statistics. If duplication is desired for error resilience, the mechanism described in [RTPDUP] can be used, since this will not cause breakage of RTP streams or RTCP statistics.

Reserved (resv): 4 bits

These bits are reserved. They MUST be set to zero by senders and ignored by receivers (See RFC6709 section 4.2).

block length: 16 bits

The length of this report block in 32-bit words, minus one, in accordance with the definition in [RFC3611]. This field MUST be set to 2 to match the fixed length of the report block. The block MUST be discarded if the block length is set to a different value.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

discard count

Number of packets discarded over the period (Interval or Cumulative) covered by this report.

The measured value is unsigned value. If the measured value exceeds 0xFFFFFFFF, the value 0xFFFFFFFF MUST be reported to indicate an over-range measurement. If the measurement is unavailable, the value 0xFFFFFFFF MUST be reported.

Note that the number of packets expected in the period associated with this metric (whether interval or cumulative) is available from the difference between a pair of extended sequence numbers in the Measurement Information block [RFC6776], so need not be repeated in this block.

4. SDP Signaling

[RFC3611] defines the use of SDP (Session Description Protocol) [RFC4566] for signaling the use of XR blocks. However XR blocks MAY be used without prior signaling (see section 5 of RFC3611).

4.1. SDP rtcp-xr-attrib Attribute Extension

This section augments the SDP [RFC4566] attribute "rtcp-xr" defined in [RFC3611] by providing an additional value of "xr-format" to signal the use of the report block defined in this document.

```
xr-format =/ xr-pdc-block  
xr-pdc-block = "pkt-discard-count"
```

4.2. Offer/Answer Usage

When SDP is used in offer-answer context, the SDP Offer/Answer usage defined in [RFC3611] for unilateral "rtcp-xr" attribute parameters applies. For detailed usage of Offer/Answer for unilateral parameter, refer to section 5.2 of [RFC3611].

5. IANA Considerations

New block types for RTCP XR are subject to IANA registration. For general guidelines on IANA considerations for RTCP XR, refer to [RFC3611].

5.1. New RTCP XR Block Type value

This document assigns the block type value PDC in the IANA " RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry " to the "Discard Count Metrics Block".

[Note to RFC Editor: please replace PDC with the IANA provided RTCP XR block type for this block.]

5.2. New RTCP XR SDP Parameter

This document also registers a new parameter "pkt-discard-count" in the " RTP Control Protocol Extended Reports (RTCP XR) Session Description Protocol (SDP) Parameters Registry ".

5.3. Contact information for registrations

The following contact information is provided for all registrations in this document:

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China

6. Security Considerations

In some situations, returning very detailed error information (e.g., over-range measurement or measurement unavailable) using this report block can provide an attacker with insight into the security processing. Where this is a concern, the implementation should apply authentication to this report block. This can be achieved by using the AVPF profile together with the Secure RTP profile as defined in [RFC3711]; as a prerequisite, an appropriate combination of those two profiles (an "SAVPF") is being specified [RFC5124].

Besides this, it is believed that this proposed RTCP XR report block introduces no new security considerations beyond those described in [RFC3611]. This block does not provide per-packet statistics so the risk to confidentiality documented in Section 7, paragraph 3 of [RFC3611] does not apply.

7. Contributors

Geoff Hunt wrote the initial draft of this document.

8. Acknowledgments

The authors gratefully acknowledge reviews and feedback provided by Bruce Adams, Philip Arden, Amit Arora, Bob Biskner, Kevin Connor, Claus Dahm, Randy Ethier, Roni Even, Jim Frauenthal, Albert Higashi, Tom Hock, Shane Holthaus, Paul Jones, Rajesh Kumar, Keith Lantz, Mohamed Mostafa, Amy Pendleton, Colin Perkins, Mike Ramalho, Ravi Raviraj, Albrecht Schwarz, Tom Taylor, and Hideaki Yamada, Kevin Gross, Varun Singh, Claire Bi, Roni Even, Dan Romascanu and Jonathan Lennox.

9. References

9.1. Normative References

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", March 1997.
- [RFC3550] Schulzrinne, H., "RTP: A Transport Protocol for Real-Time Applications", RFC 3550, July 2003.
- [RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", November 2003.
- [RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", July 2006.
- [RFC6709] Carpenter, B., Aboba, B., and S. Cheshire, "Design Considerations for Protocol Extensions", RFC 6709, September 2012.
- [RFC6776] Hunt, G., "Measurement Identity and information Reporting using SDES item and XR Block", RFC 6776, October 2012.

9.2. Informative References

- [BGDISCARD] Hunt, G., "RTCP XR Report Block for Burst Gap Discard metric Reporting", ID draft-ietf-xrblock-rtcp-xr-burst-gap-discard-14, April 2013.
- [RFC4588] Rey, J., "RTP Retransmission Payload Format", RFC 4588, July 2006.
- [RFC5109] Li, A., "RTP Payload Format for Generic Forward Error Correction", RFC 5109, July 2006.
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- [RFC6792] Wu, Q., "Monitoring Architectures for RTP", RFC 6792, November 2012.
- [RTPDUP] Begin, A. and C. Perkins, "Duplicating RTP Streams", ID draft-ietf-avtext-rtp-duplication-02, March 2013.

Appendix A. Metrics represented using RFC6390 Template

RFC EDITOR NOTE: please change XXXX in [RFCXXXX] by the new RFC number, when assigned.

a. Number of packets discarded Metric

- * Metric Name: Number of RTP packets discarded Metric
- * Metric Description: Number of RTP packets discarded over the period covered by this report.
- * Method of Measurement or Calculation: See section 3.2, number of packets discarded definition [RFCXXXX].
- * Units of Measurement: See section 3.2, number of packets discarded definition [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 3, 1st paragraph [RFCXXXX].
- * Measurement Timing: See section 3, 1st paragraph [RFCXXXX] for measurement timing and section 3.2 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

Appendix B. Change Log

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

B.1. draft-ietf-xrblock-rtcp-xr-discard-14

The following are the major changes compared to previous version:

- o Editorial changes to paragraph 4 of section 1.1.

B.2. draft-ietf-xrblock-rtcp-xr-discard-13

The following are the major changes compared to previous version:

- o Some editorial changes to get in line with burst gap drafts.

B.3. draft-ietf-xrblock-rtcp-xr-discard-12

The following are the major changes compared to previous version:

- o Incorporate some changes to burst gap draft that applies to this document.
- o Use RFC6390 template to the metrics in the appendix.

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RTCP XR Blocks for QoE Metric Reporting
draft-ietf-xrblock-rtcp-xr-qoe-02

Abstract

This document defines an RTCP XR Report Block including two new segment types and associated SDP parameters that allow the reporting of QoE 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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1. Introduction

1.1. QoE Metrics Report Block

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

The new block type provides information on multimedia quality using one of several standard metrics.

The metrics belong to the class of application level metrics defined in [MONARCH].

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. [RFC3611] defined an extensible structure for reporting using an RTCP Extended Report (XR). This document defines a new Extended Report block. The use of Extended Report blocks is defined by [RFC3611].

1.3. Performance Metrics Framework

The Performance Metrics Framework [RFC6390] provides guidance on the definition and specification of performance metrics. Metrics described in this draft either reference external definitions or define metrics generally in accordance with the guidelines in [RFC6390].

1.4. Applicability

The QoE Metrics Report Block can be used in any application of RTP for which QoE measurement algorithms are defined.

The factors that affect real-time AV application quality can be split into two categories. The first category consists of transport-dependent factors such as packet loss, delay and jitter (which also translates into losses in the playback buffer). The factors in the second category are application-specific factors that affect real time application (e.g., video) quality and are sensitivity to network errors. These factors can be but not limited to video codec and loss recovery technique, coding bit rate, packetization scheme, and content characteristics.

Compared with application-specific factors, the transport-dependent factors sometimes are not sufficient to measure real time data quality, since the ability to analyze the real time data in the application layer provides quantifiable measurements for subscriber Quality of Experience (QoE) that may not be captured in the

transmission layers or from the RTP layer down. In a typical scenario, monitoring of the transmission layers can produce statistics suggesting that quality is not an issue, such as the fact that network jitter is not excessive. However, problems may occur in the service layers leading to poor subscriber QoE. Therefore monitoring using only network-level measurements may be insufficient when application layer content quality is required.

In order to provide accurate measures of real time application quality when transporting real time contents across a network, the synthetical multimedia quality Metrics is highly required which can be conveyed in the RTCP XR packets[RFC3611] and may have the following three benefits:

- o Tuning the content encoder algorithm to satisfy real time data quality requirements.
- o Determining which system techniques to use in a given situation and when to switch from one technique to another as system parameters change.
- o Verifying the continued correct operation of an existing system.

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].

The terminology used is

Numeric formats S X:Y

where S indicates a two's complement signed representation, X the number of bits prior to the decimal place and Y the number of bits after the decimal place.

Hence 8:8 represents an unsigned number in the range 0.0 to 255.996 with a granularity of 0.0039. S7:8 would represent the range -127.996 to +127.996. 0:16 represents a proper binary fraction with range

0.0 to $1 - 1/65536 = 0.9999847$

though note that use of flag values at the top of the numeric range slightly reduces this upper limit. For example, if the 16-bit values 0xffffe and 0xfffff are used as flags for "over-range" and "unavailable" conditions, a 0:16 quantity has range

$$0.0 \text{ to } 1 - 3/65536 = 0.9999542$$

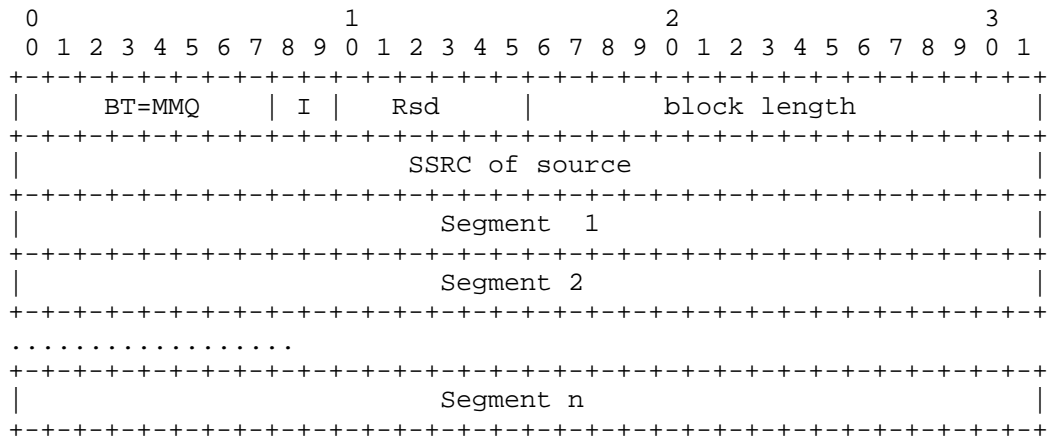
3. QoE Metrics Block

This block reports the multimedia application performance or quality beyond the information carried in the standard RTCP packet format. Information is recorded about multimedia application QoE metric which provides a measure that is indicative of the user's view of a service. Multimedia application QoE metric is commonly expressed as a MOS ("Mean Opinion Score"), MOS is on a scale from 1 to 5, in which 5 represents excellent and 1 represents unacceptable. MOS scores are usually obtained using subjective testing or using objective algorithm. However Subjective testing to estimate the multimedia quality may be not suitable for measuring the multimedia quality since the results may vary from test to test. Therefore using objective algorithm to calculate MOS scores is recommended. ITU-T recommendations define the methodologies for assessment of the performance of multimedia stream [G.107][P.564][G.1082][P.NAMS][P.NBAMS] and provides a method to evaluate QoE estimation algorithms and objective model for video and audio. Hence this document recommends vendors and implementers to use these International Telecommunication Union (ITU)-specified methodologies to measure parameters when possible.

3.1. Metric Block Structure

The report block contents are dependent upon a series of flag bits carried in the first part of the header. Not all parameters need to be reported in each block. Flags indicate which are and which are not reported. The fields corresponding to unreported parameters MUST be present, but are set to zero. The receiver MUST ignore any QoE Metrics Block with a non-zero value in any field flagged as unreported. The encoding of QoE metrics block payload consists of a series of 32 bit units called segments that describe MOS Type, MoS algorithm and MoS value.

The QoE Metrics Block has the following format:



3.2. Definition of Fields in QoE Metrics Block

Block type (BT): 8 bits

The QoE Metrics Block is identified by the constant <SMQ>.

Interval Metric flag (I): 2 bits

This field is used to indicate whether the QoE metrics are Interval or Cumulative metrics, that is, whether the reported values applies to the most recent measurement interval duration between successive metrics reports (I=10) (the Interval Duration) or to the accumulation period characteristic of cumulative measurements (I=11) (the Cumulative Duration) or is a sampled instantaneous value (I=01) (Sampled Value).

Rsd.: 6 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and MUST be ignored by the receiver.

Block Length: 16 bits

The length of this report block in 32-bit words, minus one. For the QoE Metrics Block, the block length is variable length.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

Segment i: 32 bits

There are two segment types defined in this document: single stream per SSRC segment, multi-channel audio per SSRC segment. Multi-channel audio per SSRC segment is used to deal with the case where Multi-channel audios are carried in one RTP stream while single stream per SSRC segment is used to deal with the case where each media stream is identified by SSRC and sent in separate RTP stream. The left two bits of the section determine its type. If the leftmost bit of the segment is zero, then it is single stream segment. If the leftmost bit is one, then it is multi-channel audio segment. Note that two segment types can not be present in the same metric block.

3.2.1. Single Stream per SSRC Segment

```

+++++
|0| MT |CAlg| PT |Rsv. | MOS Value |
+++++

```

Segment Type (S): 1 bit

A zero identifies this as a single stream segment. Single stream means there is only one media stream carried in one RTP stream. The single stream segment can be used to report the MoS value associated with this media stream identified by SSRC. If there are multiple media streams and they want to use the single stream per SSRC segment to report the MOS value, they should be carried in the separate RTP streams with different SSRC. In this case, multiple QoE Metrics Blocks are required to report the MOS value corresponding to each media stream using single stream segment.

Reserved (R): 1bit

The bit in this field is reserved. It MUST be set to zero and MUST be ignored by the receiver if the leftmost bit of Single Stream Per SSRC Segment is set to 0.

MoS Type (MT): 4 bits

This field is used to indicate the MOS type to be reported. The MOS type is defined as follows:

0000 MOS-LQ - Listening Quality MoS.
0001 MOS-CQ - Conversation Quality MoS.
0010 MOS-A - Audio Quality MOS.
0010 MOS-V - Video Quality MOS.
0011 MOS-AV - Audio-Video Quality MOS.
0100~1111 - Reserved for future definitions.

MoS-LQ measures the quality of audio for listening purposes only while MoS-CQ measures the quality of audio for conversation purpose only. MoS-A, MoS-V and MoS-AV measures the quality of audio application, the quality of video application and Audio-Video application respectively. Both MoS-LQ and MoS-CQ are commonly used in VoIP applications. MOS-LQ uses either wideband audio codec or narrowband audio codec, or both and does not take into account any of bidirectional effects, such as delay and echo. MOS-CQ uses narrowband codec and takes into account listening quality in each direction, as well as the bidirectional effects.

Calculation Algorithm (CALg): 3 bits

000 - ITU-T P.564 Compliant Algorithm [P.564] (Voice)
001 - G.107 [G.107] (Voice)
010 - ETSI TS 101 329-5 Annex E [ETSI] (Voice)
011 - ITU-T P.NAMS [P.NAMS] (Multimedia)
100 - ITU-T P.NBAMS [P.NBAMS] (Multimedia)
101~111 - Reserved for future extension.

G.107 and P.564 and ETSI TS101 329-5 specify three Calculation algorithms or MoS algorithms that are used to estimate speech quality or conversation quality. P.NAMS and P.NBAMS specify two MoS algorithms that are used to estimate multimedia quality including video quality, audio quality and audio-video quality. If MoS type is MoS-LQ and MoS-CQ, the MoS value can be calculated based on ITU-T G.107[G.107], ITU-T P.564 [P.564] or ETSI TS 101 329-5 [ETSI], if the MoS type is MoS-V or MoS-AV, the MoS value can be calculated based on ITU-T P.NAMS [P.NAMS] or ITU-T P.NBAMS [P.NBAMS]. If new MOS types are defined, they can be added by an update to this document. If the receiver does not understand the MOS type defined in this document it should discard this report. If MoS Type does not match the MoS algorithm in the report (e.g., specify a voice MOS algorithm for a video quality MOS), the receiver should also discard this report.

Payload Type (PT): 7 bits

QoE metrics reporting depends on the payload format in use. This field identifies the format of the RTP payload. For RTP sessions where multiple payload formats can be negotiated or the payload format changes during the mid-session, the value of this field

will be used to indicate what payload format was in use for the reporting interval.

Rsd.:3 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and MUST be ignored by the receiver.

MOS Value: 14 bits

The estimated mean opinion score for multimedia application quality is defined as including the effects of delay, loss, discard, jitter and other effects that would affect multimedia quality. It is expressed in numeric format 6:8 with the value in the range 0.0 to 63.996. The valid the measured value ranges from 0.0 to 50.0, corresponding to MoS x 10 as for MoS. If the measured value is over ranged, the value 0xFFFFE SHOULD be reported to indicate an over-range measurement. If the measurement is unavailable, the value 0xFFFF SHOULD be reported. Values other than 0xFFFFE, 0xFFFF and the valid range defined above MUST NOT be sent and MUST be ignored by the receiving system.

3.2.2. Multi-Channel audio per SSRC Segment

```

+-----+-----+-----+-----+-----+-----+-----+-----+-----+
|1|  MT  |CAlg|      PT      |CHID|      MOS Value      |
+-----+-----+-----+-----+-----+-----+-----+-----+

```

Segment Type (S): 1 bit

A one identifies this as either a multi-channel segment or multi-layer segment.

Media Type (M): 1bit

A zero identifies this as a multi-channel per SSRC segment.

MoS Type (MT): 4 bits

As defined in Section 3.2.1 of this document. If the value of this field is not corresponding to MoS-CQ or MoS-LQ, the receiver using multi-channel segment should discard this invalid segment with the wrong MoS Type.

Calculation Algorithm (CALg): 3 bits

- 000 - ITU-T P.564 Compliant Algorithm [P.564] (Voice)
- 001 - G.107 [G.107] (Voice)
- 010 - ETSI TS 101 329-5 Annex E, [ETSI] (Voice)
- 011~100 - Reserved.
- 101~111 - Reserved for future extension.

Payload Type (PT): 7 bits

As defined in Section 3.2.1 of this document.

Channel Identifier (CHID): 3 bits

If multiple channels of audio are carried in one RTP stream, each channel of audio will be viewed as a independent channel(e.g., left channel audio, right channel audio). This field is used to identify each channel carried in the same media stream. The default Channel mapping follows static ordering rule described in the section 4.1 of [RFC3551]. However there are some payload formats that use different channel mappings, e.g., AC-3 audio over RTP [RFC4184] only follow AC-3 channel order scheme defined in [ATSC]. Enhanced AC-3 Audio over RTP [RFC4598] uses dynamic channel transform mechanism. In order that the appropriate channel mapping can be determined, QoE reports need to be tied to an RTP payload format, i.e., including the payload type of the reported media according to [MONARCH] and using Payload Type to determine the appropriate channel mapping.

Rsd.: 3 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and MUST be ignored by the receiver.

MOS Value: 14 bits

As defined in Section 3.2.1 of this document.

4. SDP Signaling

One new parameter is defined for the report block defined in this document to be used with Session Description Protocol (SDP) [RFC4566] using the Augmented Backus-Naur Form (ABNF) [RFC5234]. It has the following syntax within the "rtcp-xr" attribute [RFC3611]:


```
rtcp-xr-attrb = "a=rtcp-xr:"  
                [xr-format *(SP xr-format)] CRLF  
xr-format = qoe-metrics  
qoe-metrics = "multimedia-quality-metrics"
```

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

5. IANA Considerations

New block types for RTCP XR are subject to IANA registration. For general guidelines on IANA considerations for RTCP XR, refer to [RFC3611].

5.1. New RTCP XR Block Type value

This document assigns the block type value MMQ in the IANA "RTCP XR Block Type Registry" to the "QoE Metrics Block".

[Note to RFC Editor: please replace MMQ with the IANA provided RTCP XR block type for this block.]

5.2. New RTCP XR SDP Parameter

This document also registers a new parameter "qoe-metrics" in the "RTCP XR SDP Parameters Registry".

5.3. Contact information for registrations

The contact information for the registrations is:

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5.4. New registry of calculation algorithms for single stream segment

This document creates a new registry for single stream per SSRC segment defined in the section 3.2.1 to be called "RTCP XR QoE metric block - multimedia application Calculation Algorithm" as a sub-registry of the "RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry". This registry applies to the multimedia session where each type of media are sent in a separate RTP stream. Specially this registry also applies to the layered video session where each layer video are sent in a separate RTP stream. Policies for this new registry are as follows:

- o The information required to support this assignment is an unambiguous definition of the new metric, covering the base measurements and how they are processed to generate the reported metric. This should include the units of measurement, how values of the metric are reported in the one 16-bit fields "MoS Value".
- o The review process for the registry is "Specification Required" as described in Section 4.1 of [RFC5226].
- o Entries in the registry are integers. The valid range is 0 to 7 corresponding to the 3-bit field "CAlg" in the block. Values are to be recorded in decimal.
- o Initial assignments are as follows:
 1. ITU-T P.564 Compliant Algorithm [P.564] (Voice)
 2. G.107 [G.107] (Voice)
 3. ETSI TS 101 329-5 Annex E [ETSI] (Voice)
 4. ITU-T P.NAMS [P.NAMS] (Multimedia)
 5. ITU-T P.NBAMS [P.NBAMS] (Multimedia)

5.5. New registry of calculation algorithms for multi-channel audio segment

This document creates a new registry for multi-channel audio per SSRC segment defined in the section 3.2.2 to be called "RTCP XR QoE metric block - multi-channel application Calculation Algorithm" as a sub-registry of the "RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry" if multi-channel voice data are carried in the same RTP stream. Policies for this new registry are as follows:

- o The information required to support this assignment is an unambiguous definition of the new metric, covering the base measurements and how they are processed to generate the reported metric. This should include the units of measurement, how values of the metric are reported in the one 16-bit fields "MoS Value".
- o The review process for the registry is "Specification Required" as described in Section 4.1 of [RFC5226].
- o Entries in the registry are integers. The valid range is 0 to 7 corresponding to the 3-bit field "CAlg" in the block. Values are to be recorded in decimal.
- o Initial assignments are as follows:
 1. ITU-T P.564 Compliant Algorithm [P.564] (Voice)
 2. G.107 [G.107] (Voice)
 3. ETSI TS 101 329-5 Annex E [ETSI] (Voice)

6. Security Considerations

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

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This draft merges ideas from two drafts addressing the QoE metric Reporting issue. The authors of these drafts are listed below (in alphabetical order):

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8. Acknowledgements

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Appendix A. Change Log

A.1. draft-ietf-xrblock-rtcp-xr-qoe-02

The following are the major changes compared to previous version:

- o Remove leftmost second bit since it is useless.
- o Change 13bits MoS value field into 14 bits to increase MoS precision.
- o Fix some typo and make some editorial changes.

A.2. draft-ietf-xrblock-rtcp-xr-qoe-01

The following are the major changes compared to previous version:

- o Remove layered support from the QoE metric draft.
- o Allocate 7 bits in the block header for payload type to indicate what type of payload format is in use and add associated definition of payload type.
- o Clarify using Payload Type to determine the appropriate channel mapping in the definition of Channel Identifier.

A.3. draft-ietf-xrblock-rtcp-xr-qoe-00

The following are the major changes compared to previous version:

- o Allocate one more bit in the single stream per SSC segment to get alignment with the other two segment type.

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RTP Control Protocol (RTCP) Extended Report (XR) Blocks for MOS Metric
Reporting
draft-ietf-xrblock-rtcp-xr-qoe-17

Abstract

This document defines an RTP Control Protocol (RTCP) Extended Report (XR) Block including two new segment types and associated SDP parameters that allow the reporting of mean opinion score (MOS) Metrics for use in a range of RTP applications.

Status of this Memo

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1. Introduction

1.1. MOS Metrics Report Block

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

The new block type provides information on media quality using one of several standard metrics (i.e. Mean Opinion Score(MOS)).

The metrics belong to the class of application level metrics defined in [RFC6792].

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. RFC3611 defined an extensible structure for reporting using an RTCP Extended Report (XR). This document defines a new Extended Report block for use with [RFC3550] and [RFC3611].

1.3. Performance Metrics Framework

The Performance Metrics Framework [RFC6390] provides guidance on the definition and specification of performance metrics. The RTP Monitoring Architectures [RFC6792] provides guidelines for reporting block format using RTCP XR. The XR block type described in this document are in accordance with the guidelines in [RFC6390] and [RFC6792].

1.4. Applicability

The MOS Metrics Report Block can be used in any application of RTP for which QoE (Quality of Experience) measurement algorithms are defined.

The factors that affect real-time audio/video application quality can be split into two categories. The first category consists of transport-specific factors such as packet loss, delay and jitter (which also translates into losses in the playback buffer). The factors in the second category consists of content and codec related factors such as codec type and loss recovery technique, coding bit rate, packetization scheme, and content characteristics

Transport-specific factors may be insufficient to infer real time media quality as codec related parameters and the interaction between transport problems and application layer protocols can have a substantial effect on observed media quality. Media quality may be measured using algorithm that directly compare input and output media

streams, or may be estimated using algorithms that model the interaction between media quality, protocol and encoded content. Media quality is commonly expressed in terms of Mean Opinion Score (MOS) however is also represented by a range of indexes and other scores.

The measurement of media quality has a number of applications:

- o Detecting problems with media delivery or encoding that is impacting user perceived quality.
- o Tuning the content encoder algorithm to satisfy real time data quality requirements.
- o Determining which system techniques to use in a given situation and when to switch from one technique to another as system parameters change (for example as discussed in [P.1082]).
- o Pre-qualifying a network to assess its ability to deliver an acceptable end-user perceived quality level.

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].

The terminology used is

Numeric formats X:Y

where X the number of bits prior to the decimal place and Y the number of bits after the decimal place.

Hence 8:8 represents an unsigned number in the range 0.0 to 255.996 with a granularity of 0.0039. 0:16 represents a proper binary fraction with range

0.0 to 1 - 1/65536 = 0.9999847

though note that use of flag values at the top of the numeric range slightly reduces this upper limit. For example, if the 16-bit values 0xffffe and 0xffff are used as flags for "over-range" and "unavailable" conditions, a 0:16 quantity has range 0.0 to 1 - 3/65536 = 0.9999542

3. MoS Metrics Block

Multimedia application MOS Metric is commonly expressed as a MOS ("Mean Opinion Score"), MOS is usually on a scale from 1 to 5, in which 5 represents excellent and 1 represents unacceptable however

can use other ranges (for example 0 to 10) . The term "MOS score" originates from subjective testing, and is used to refer to the Mean of a number of individual Opinion Scores. There is therefore a well understood relationship between MOS and user experience, hence the industry commonly uses MOS as the scale for objective test results. Subjective tests can be used for measuring live network traffic however the use of objective or algorithmic measurement techniques allows much larger scale measurements to be made. Within the scope of this document, MOS scores are obtained using objective or estimation algorithms. ITU-T or ITU-R recommendations (e.g., [BS.1387-1],[G.107],[G.107.1],[P.862],[P.862.1],[P.862.2],[P.863],[P.564],[G.1082],[P.1201.1],[P.1201.2],[P.1202.1],[P.1202.2]) define methodologies for assessment of the performance of audio and video streams. Other international and national standards organizations such as EBU, ETSI, IEC and IEEE also define QoE algorithms and methodologies, and the intent of this document is not to restrict its use to ITU recommendations but to suggest that ITU recommendations be used where they are defined.

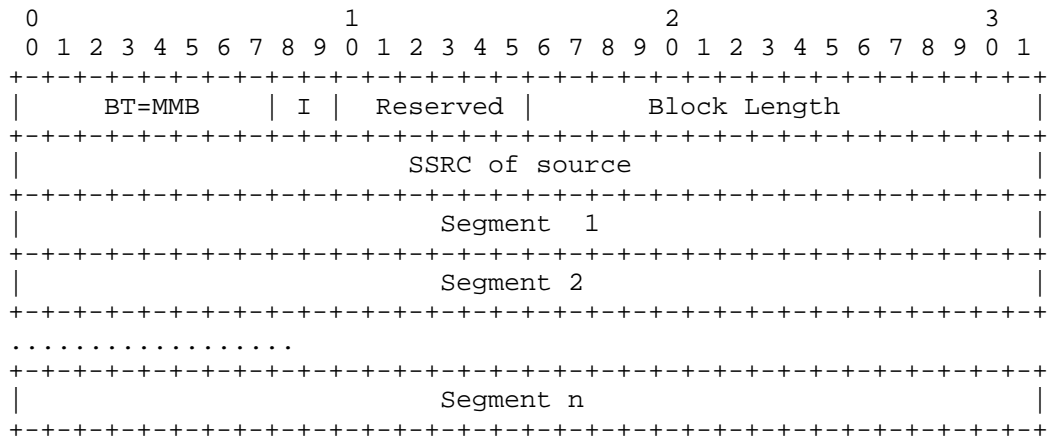
This block reports the media quality in the form of a MOS range (e.g., 1-5, 0-10, or 0-100, as specified by the calculation algorithm) however does not report the MoS score that include parameters outside the scope of the RTP stream, for example signaling performance, mean time to repair (MTTR) or other factors that may affect the overall user experience.

The MOS Metric reported in this block gives a numerical indication of the perceived quality of the received media stream, which is typically measured at the receiving end of the RTP stream. Instances of this Metrics Block refer by Synchronization source (SSRC) to the separate auxiliary Measurement Information block [RFC6776] which describes measurement periods in use (see RFC6776 section 4.2).

This Metrics Block relies on the measurement period in the Measurement Information block indicating the span of the report. Senders MUST send this block in the same compound RTCP packet as the measurement information block. Receivers MUST verify that the measurement period is received in the same compound RTCP packet as this Metrics Block. If not, this Metrics Block MUST be discarded.

3.1. Report Block Structure

The MOS Metrics Block has the following format:



3.2. Definition of Fields in MoS Metrics Block

Block type (BT): 8 bits

The MOS Metrics Block is identified by the constant <MMB>.

Interval Metric flag (I): 2 bits

This field is used to indicate whether the MOS Metrics are Sampled, Interval or Cumulative [RFC6792]:

- I=10: Interval Duration - the reported value applies to the most recent measurement interval duration between successive metrics reports.
- I=11: Cumulative Duration - the reported value applies to the accumulation period characteristic of cumulative measurements.
- I=01: Sampled Value - the reported value is a sampled instantaneous value.
- I=00: Reserved

In this document, MOS Metrics MAY be reported for intervals or for the duration of the media stream (cumulative). The value I=01, indicating a sampled value, MUST NOT be sent, and MUST be discarded when received.

Reserved: 6 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and ignored by the receiver (See RFC6709 section 4.2).

Block Length: 16 bits

The length of this report block in 32-bit words, minus one. For the MOS Metrics Block, the block length is variable length.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

Segment i: 32 bits

There are two segment types defined in this document: single stream Audio/Video per SSRC segment, multi-channel audio per SSRC segment. Multi-channel audio per SSRC segment is used to deal with the case where Multi-channel audios are carried in one RTP stream while single channel Audio/Video per SSRC segment is used to deal with the case where each media stream is identified by SSRC and sent in separate RTP stream. The leftmost bit of the segment determines its type. If the leftmost bit of the segment is zero, then it is single channel segment. If the leftmost bit is one, then it is multi-channel audio segment. Note that two segment types can not be present in the same metric block.

3.2.1. Single Channel audio/video per SSRC Segment

```

+-----+-----+-----+-----+-----+-----+-----+-----+
|S|      CAID      |      PT      |      MOS Value      |
+-----+-----+-----+-----+-----+-----+-----+-----+

```

Segment Type (S): 1 bit

This field is used to identify the segment type used in this report block. A zero identifies this as a single channel Audio/Video per SSRC segment. Single channel means there is only one media stream carried in one RTP stream. The single channel Audio/Video per SSRC segment can be used to report the MOS value associated with the media stream identified by SSRC. If there are multiple media streams and they want to use the single channel Audio/Video per SSRC segment to report the MOS value, they should be carried in the separate RTP streams with each identified by different SSRC. In this case, multiple MOS Metrics Blocks are required to report the MOS value corresponding to each media stream using single channel Audio/Video per SSRC segment in the same RTCP XR packet.

Calculation Algorithm ID (CAID) : 8 bits

The 8-bit CAID is the session specific reference to the calculation algorithm and associated qualifiers indicated in SDP (see Section 4.1) and used to compute the MOS score for this segment.

Payload Type (PT): 7 bits

MOS Metrics reporting depends on the payload format in use. This field identifies the RTP payload type in use during the reporting interval. The binding between RTP payload types and RTP payload formats is configured via a signalling protocol, for example an SDP offer/answer exchange. If the RTP payload type used is changed during an RTP session, separate reports SHOULD be sent for each RTP payload type, with corresponding measurement information blocks indicating the time period to which they relate.

Note that the use of this Report Block with MPEG Transport streams carried over RTP is undefined as each MPEG Transport stream may use distinct audio or video codecs and the indication of the encoding of these is within the MPEG Transport stream and does not use RTP payloads.

MOS Value: 16 bits

The estimated Mean Opinion Score (MOS) for multimedia application performance is defined as including the effects of delay, loss, discard, jitter and other effects that would affect media quality. This is a unsigned fixed-point 7:9 value representing the MOS, allowing the MOS score up to 127 in the integer part. MOS ranges are defined as part of the specification of the MOS estimation algorithm (Calculation Algorithm in this document), and are normally ranges like 1-5, 0-10, or 0-100. Two values are reserved: A value of 0xFFFE indicates out of range and a value of 0xFFFF indicates that the measurement is unavailable. Values outside of the range defined by the Calculation Algorithm, other than the two reserved values, MUST NOT be sent and MUST be ignored by the receiving system.

3.2.2. Multi-Channel audio per SSRC Segment

```

+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+
|S|      CAID      |      PT      |CHID|      MOS Value      |
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+

```


Segment Type (S): 1 bit

This field is used to identify the segment type used in this report block. A one identifies this as a multi-channel audio segment.

Calculation Algorithm ID (CAID) : 8 bits

The 8-bit CAID is the session specific reference to the calculation algorithm and associated qualifiers indicated in SDP (see Section 4.1) and used to compute the MOS score for this segment.

Payload Type (PT): 7 bits

As defined in Section 3.2.1 of this document

Channel Identifier (CHID): 3 bits

If multiple channels of audio are carried in one RTP stream, each channel of audio will be viewed as a independent channel (e.g., left channel audio, right channel audio). This field is used to identify each channel carried in the same media stream. The default Channel mapping follows static ordering rule described in the section 4.1 of [RFC3551]. However there are some payload formats that use different channel mappings, e.g., AC-3 audio over RTP [RFC4184] only follow AC-3 channel order scheme defined in [ATSC]. Enhanced AC-3 Audio over RTP [RFC4598] uses dynamic channel transform mechanism. In order that the appropriate channel mapping can be determined, MOS metrics reports need to be tied to an RTP payload format, i.e., including the payload type of the reported media according to [RFC6792] and using Payload Type to determine the appropriate channel mapping.

MOS Value: 13 bits

The estimated Mean Opinion Score (MOS) for multimedia application performance is defined as including the effects of delay, loss, discard, jitter and other effects that would affect media quality. This is a unsigned fixed-point 7:6 value representing the MOS, allowing the MOS score up to 127 in the integer part. MOS ranges are defined as part of the specification of the MOS estimation algorithm (Calculation Algorithm in this document), and are normally ranges like 1-5, 0-10, or 0-100. Two values are reserved: A value of 0x1FFE indicates out of range and a value of 0x1FFF indicates that the measurement is unavailable. Values outside of the range defined by the Calculation Algorithm, other than the two reserved values, MUST NOT be sent and MUST be ignored

by the receiving system.

4. SDP Signaling

[RFC3611] defines the use of SDP (Session Description Protocol) [RFC4566] for signaling the use of XR blocks. However XR blocks MAY be used without prior signaling (see section 5 of RFC3611).

4.1. SDP rtcp-xr-attr Attribute Extension

This section augments the SDP [RFC4566] attribute "rtcp-xr" defined in [RFC3611] by providing an additional value of "xr-format" to signal the use of the report block defined in this document. Within the "xr-format", the syntax element "calgextmap" is an attribute as defined in [RFC4566] and used to signal the mapping of the local identifier (CAID) in the segment extension defined in section 3.2 to the calculation algorithm. Specific extension attributes are defined by the specification that defines a specific extension name; there might be several.

```

xr-format =/ xr-mos-block
xr-mos-block = "mos-metric" ["=" calgextmap *("," calgextmap)]
calgextmap = mapentry "=" extensionname [SP extentionattributes]
direction = "sendonly" / "recvonly" / "sendrecv" / "inactive"
mapentry = "calg:" 1*3 DIGIT ["/" direction]
           ; Values in the range 1-255 are valid
           ; if needed, 0 can be used to indicate that
           ; an algorithm is rejected
extensionname = "P564";ITU-T P.564 Compliant Algorithm [P.564]
               / "G107";ITU-T G.107 [G.107]
               / "G107_1";ITU-T G.107.1 [G.107.1]
               / "TS101_329";ETSI TS 101 329-5 Annex E [ETSI]
               / "JJ201_1 ";TTC JJ201.1 [TTC]
               / "P1201_1";ITU-T P.1201.2 [P.1201.1]
               / "P1201_2";ITU-T P.1201.2 [P.1201.2]
               / "P1202_1";ITU-T P.1202.1 [P.1202.1]
               / "P1202_2";ITU-T P.1202.2 [P.1202.2]
               / "P.862.2";ITU-T P.862.2 [P.862.2]
               / "P.863"; ITU-T P.863 [P.863]
               / non-ws-string
extensionattributes = mosref
                   /attributes-ext
mosref = "mosref=" ("l"; lower resolution
                  /"m"; middle resolution
                  / "h";higher resolution
                  / non-ws-string)
attributes-ext = non-ws-string
SP = <Define in RFC5234>
non-ws-string = 1*(%x21-FF)

```

Each local identifier (CAID) of calculation algorithm used in the segment defined in the section 3.2 is mapped to a string using an attribute of the form:

```
a=calg:<value> ["/"<direction>] <name> [<extensionattributes>]
```

where <name> is a calculation algorithm name, as above, <value> is the local identifier (CAID) of the calculation algorithm associated with the segment defined in this document and is an integer in the valid range inclusive.

Example:

```
a=rtcp-xr:mos-metric=calg:1=G107,calg:2=P1202_1
```

A usable mapping MUST use IDs in the valid range, and each ID in this range MUST be unique and used only once for each stream or each channel in the stream.

The mapping MUST be provided per media stream (in the media-level section(s) of SDP, i.e., after an "m=" line).

The syntax element "mosref" is referred to the media resolution relative reference and has three values 'l', 'm', 'h'. (e.g., Narrowband (3.4kHz) Speech and Standard Definition (SD) or lower Resolution Video have 'l' resolution, Super Wideband (>14kHz) Speech or higher and High Definition (HD) or higher Resolution Video have 'h' Resolution, Wideband speech(7khz) and Video with resolution between SD and HD has 'm' resolution). The MOS score reported in the MOS metrics block might vary with the MOS reference; For example, MOS values for narrowband, wideband, super wideband codecs occupy the same range but SHOULD be reported in different value. For video application, MOS scores for SD resolution, HD resolution video also occupy the same ranges and SHOULD be reported in different value.

4.2. Offer/Answer Usage

When SDP is used in offer-answer context, the SDP Offer/Answer usage defined in [RFC3611] applies. In the offer answer context, the signaling described above might be used in three ways:

- o asymmetric behavior (segment extensions sent in only one direction),
- o the offer of mutually exclusive alternatives, or
- o the offer of more segments than can be sent in a single session.

A direction attribute MAY be included in a calgextmap; without it, the direction implicitly inherits, of course, from the RTCP stream direction.

Segment extensions, with their directions, MAY be signaled for an "inactive" stream. An extension direction MUST be compatible with the stream direction. If a segment extension in the SDP offer is marked as "sendonly" and the answerer desires to receive it, the extension MUST be marked as "recvonly" in the SDP answer. An answerer that has no desire to receive the extension or does not understand the extension SHOULD NOT include it in the SDP answer.

If a segment extension is marked as "recvonly" in the SDP offer and the answerer desires to send it, the extension MUST be marked as "sendonly" in the SDP answer. An answerer that has no desire to, or is unable to, send the extension SHOULD NOT include it in the SDP answer.

If a segment extension is offered as "sendrecv", explicitly or implicitly, and asymmetric behavior is desired, the SDP MAY be modified to modify or add direction qualifiers for that segment

extension.

A mosref attribute and MOS type attribute MAY be included in an calgextmap; without it, the mosref and most type attribute implicitly inherits, of course, from the name attribute (e.g., P.1201.1 [P.1201.1] indicates lower resolution used while P.1201.2 [P.1201.2] indicates higher resolution used) or payload type carried in the segment extension (e.g., EVRC-WB [RFC5188] indicates using Wideband Codec). However not all payload types or MOS algorithm names indicate resolution to be used and MOS type to be used. If an answerer receives an offer with an mosref attribute value it doesn't support (e.g., the answerer only supports "l" and receives "h" from offerer), the answerer SHOULD reject the mosref attribute value offered by the offerer.

If the answerer wishes to reject a mosref attribute offered by the offerer, it sets identifiers associated with segment extensions in the answer to the value in the range 4096-4351. The rejected answer MUST contain 'mosref ' attribute whose value is the value of the SDP offer.

Local identifiers in the valid range inclusive in an offer or answer must not be used more than once per media section. A session update MAY change the direction qualifiers of segment extensions under use. A session update MAY add or remove segment extension(s). Identifier values in the valid range MUST NOT be altered (remapped).

If a party wishes to offer mutually exclusive alternatives, then multiple segment extensions with the same identifier in the (unusable) range 4096-4351 MAY be offered; the answerer SHOULD select at most one of the offered extensions with the same identifier, and remap it to a free identifier in the valid range, for that extension to be usable. Note that two segment types defined in section 3 are also two exclusive alternatives.

If more segment extensions are offered in the valid range, the answerer SHOULD choose those that are desired, and place the offered identifier value "as is" in the SDP answer.

Similarly, if more segment extensions are offered than can be fit in the valid range, identifiers in the range 4096-4351 MAY be offered; the answerer SHOULD choose those that are desired, and remap them to a free identifier in the valid range.

Note that the range 4096-4351 for these negotiation identifiers is deliberately restricted to allow expansion of the range of valid identifiers in future. Segment extensions with an identifier outside the valid range cannot, of course, be used.

Example (port numbers, RTP profiles, payload IDs and rtpmaps, etc. all omitted for brevity):

The offer:

```
a=rtcp-xr:mos-metric=calg:4906=P1201_1,calg:4906=P1202_1, calg:4907=G107
```

The answerer is interested in transmission P.1202.1 on lower resolution application, but doesn't support P.1201.1 on lower resolution application at all. It is interested in transmission G.107. It therefore adjusts the declarations:

```
a=rtcp-xr:mos-metric=calg:1=P1202_1,calg:2=G107
```

5. IANA Considerations

New block types for RTCP XR are subject to IANA registration. For general guidelines on IANA considerations for RTCP XR, refer to [RFC3611].

5.1. New RTCP XR Block Type value

This document assigns the block type value MMB in the IANA " RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry" to the "MOS Metrics Block".

[Note to RFC Editor: please replace MMB with the IANA provided RTCP XR block type for this block.]

5.2. New RTCP XR SDP Parameter

This document also registers a new parameter "mos-metric" in the " RTP Control Protocol Extended Reports (RTCP XR) Session Description Protocol (SDP) Parameters Registry".

5.3. The SDP calgextmap Attribute

This section contains the information required by [RFC4566] for an SDP attribute.

- o contact name, email address: RAI Area Directors
<rai-ads@tools.ietf.org>
- o attribute name (as it will appear in SDP): calgextmap
- o long-form attribute name in English: calculation algorithm map definition

- o type of attribute (session level, media level, or both): both
- o whether the attribute value is subject to the charset attribute: not subject to the charset attribute
- o a one-paragraph explanation of the purpose of the attribute: This attribute defines the mapping from the local identifier (CAID) in the segment extension defined in section 3.2 into the calculation algorithm name as documented in specifications and appropriately registered.
- o a specification of appropriate attribute values for this attribute: see RFC xxxx.

5.4. New registry of calculation algorithms

This document creates a new registry to be called "RTCP XR MOS Metric block - multimedia application Calculation Algorithm" as a sub-registry of the "RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry". This registry applies to the multimedia session where each type of media are sent in a separate RTP stream and also applies to the session where Multi-channel audios are carried in one RTP stream. Policies for this new registry are as follows:

- o The information required to support this assignment is an unambiguous definition of the new metric, covering the base measurements and how they are processed to generate the reported metric.
- o The review process for the registry is "Specification Required" as described in Section 4.1 of [RFC5226].
- o Entries in the registry are identified by entry name and mapped to the local identifier (CAID) in the segment extension defined in section 3.2.
- o Registration Template

The following information must be provided with each registration:

- * Name: A string uniquely and unambiguously identifying the Calculation algorithm for use in protocols.
- * Name Description: A valid Description of the Calculation algorithm name.
- * Reference: The reference which defines the calculation algorithm corresponding to the Name and Name Description.
- * Type: The media type to which the calculation algorithm is applied

- o Initial assignments are as follows:

Name	Name Description	Reference	Type
=====	=====	=====	=====
P564	ITU-T P.564 Compliant Algorithm	[P.564]	Voice
G107	ITU-T G.107	[G.107]	Voice
TS101_329	ETSI TS 101 329-5 Annex E	[ETSI]	Voice
JJ201_1	TTC JJ201.1	[TTC]	Voice
G107_1	ITU-T G.107.1	[G.107.1]	Voice
P862	ITU-T P.862	[P.862]	Voice
P862_2	ITU-T P.862.2	[P.862.2]	Voice
P863	ITU-T P.863	[P.863]	Voice
P1201_1	ITU-T P.1201.1	[P.1201.1]	Multimedia
P1201_2	ITU-T P.1201.2	[P.1201.2]	Multimedia
P1202_1	ITU-T P.1202.1	[P.1202.1]	Video
P1202_2	ITU-T P.1202.2	[P.1202.2]	Video

6. Security Considerations

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

7. Authors

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Appendix A. Metrics represented using RFC6390 Template

RFC EDITOR NOTE: please change XXXX in [RFCXXXX] by the new RFC number, when assigned.

a. MOS Value Metric

- * Metric Name: MOS in RTP
- * Metric Description: The estimated Mean Opinion Score for multimedia application performance of RTP stream is defined as including the effects of delay, loss, discard, jitter and other effects that would affect audio or video quality.
- * Method of Measurement or Calculation: See section 3.2.1, MOS value definition [RFCXXXX].
- * Units of Measurement: See section 3.2.1, MOS value definition [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 3, 2nd paragraph [RFCXXXX].

- * Measurement Timing: See section 3, 3rd paragraph [RFCXXXX] for measurement timing and section 3.1 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

b. Segment Type Metric

- * Metric Name: Segment Type in RTP
- * Metric Description: It is used to identify the segment type of RTP stream used in this report block. For more details, see section 3.2.1, Segment type definition.
- * Method of Measurement or Calculation: See section 3.2.1, Segment Type definition [RFCXXXX].
- * Units of Measurement: See section 3.2.1, Segment Type definition [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 3, 2nd paragraph [RFCXXXX].
- * Measurement Timing: See section 3, 3rd paragraph [RFCXXXX] for measurement timing and section 3.1 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

c. Calculation Algorithm Identifier Metric

- * Metric Name: RTP Stream Calculation Algorithm Identifier
- * Metric Description: It is the local identifier of RTP Stream calculation Algorithm associated with this segment in the range 1-255 inclusive.
- * Method of Measurement or Calculation: See section 3.2.1, Calculation Algorithm ID definition [RFCXXXX].
- * Units of Measurement: See section 3.2.1, Calg Algorithm ID definition [RFCXXXX].

- * Measurement Point(s) with Potential Measurement Domain: See section 3, 2nd paragraph [RFCXXXX].
- * Measurement Timing: See section 3, 3rd paragraph [RFCXXXX] for measurement timing and section 3.1 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

d. Payload Type Metric

- * Metric Name: RTP Payload Type
- * Metric Description: It is used to identify the format of the RTP payload. For more details, see section 3.2.1, payload type definition.
- * Method of Measurement or Calculation: See section 3.2.1, Payload type definition [RFCXXXX].
- * Units of Measurement: See section 3.2.1, payload type definition [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 3, 2nd paragraph [RFCXXXX].
- * Measurement Timing: See section 3, 3rd paragraph [RFCXXXX] for measurement timing and section 3.1 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

e. Channel Identifier Metric

- * Metric Name: Audio Channel Identifier in RTP
- * Metric Description: It is used to identify each audio channel carried in the same RTP stream. For more details, see section 3.2.2, channel identifier definition.
- * Method of Measurement or Calculation: See section 3.2.2, Channel Identifier definition [RFCXXXX].

- * Units of Measurement: See section 3.2.2, channel identifier definition[RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 3, 2nd paragraph [RFCXXXX].
- * Measurement Timing: See section 3, 3rd paragraph [RFCXXXX] for measurement timing and section 3.1 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

Appendix B. Change Log

B.1. draft-ietf-xrblock-rtcp-xr-qoe-15

The following are the major changes compared to previous version:

- o Some Editorial Changes.

B.2. draft-ietf-xrblock-rtcp-xr-qoe-14

The following are the major changes compared to previous version:

- o Add some texts to address IESG review comments.

B.3. draft-ietf-xrblock-rtcp-xr-qoe-10

The following are the major changes compared to previous version:

- o Replace QoE metrics with MoS metrics.

B.4. draft-ietf-xrblock-rtcp-xr-qoe-09

The following are the major changes compared to previous version:

- o Address comments recieved from WGLC, PM-DIR Review and SDP review.
- o Change an existing SDP attribute 'extmap' to new SDP attribute 'calgextmap' and add new SDP attribute registry.
- o Add Reference to G.107.1, P.862.1, P.862.2 and P.863 for new calculation algorithms.
- o Add MoS type attribute to distinguish different MoS type.
- o Other Editorial changes.

B.5. draft-ietf-xrblock-rtcp-xr-qoe-08

The following are the major changes compared to previous version:

- o Remove mostype attribute from SDP extension since it can be inferred from payload type.
- o Clarify mosref attribute usage in the O/A.

B.6. draft-ietf-xrblock-rtcp-xr-qoe-07

The following are the major changes compared to previous version:

- o Some editorial changes to get in line with burst gap related draft.
- o Add an appendix to apply RFC6390 template.

B.7. draft-ietf-xrblock-rtcp-xr-qoe-06

The following are the major changes compared to previous two versions:

- o A few Contact information update.
- o A few Acknowledgement section update.

B.8. draft-ietf-xrblock-rtcp-xr-qoe-04

The following are the major changes compared to previous version:

- o Split two references P.NAMS and P.NBAMS into four references.
- o SDP signaling update.
- o Add one example to explain User QoE evaluation for video stream

B.9. draft-ietf-xrblock-rtcp-xr-qoe-03

The following are the major changes compared to previous version:

- o Add one new reference to support TTC JJ201.01.
- o Update two references P.NAMS and P.NBAMS.
- o Other Editorial changes based on comments applied to PDV and Delay drafts.

B.10. draft-ietf-xrblock-rtcp-xr-qoe-02

The following are the major changes compared to previous version:

- o Remove leftmost second bit since it is useless.
- o Change 13bits MoS value field into 14 bits to increase MoS precision.
- o Fix some typo and make some editorial changes.

B.11. draft-ietf-xrblock-rtcp-xr-qoe-01

The following are the major changes compared to previous version:

- o Remove layered support from the QoE Metric draft.
- o Allocate 7 bits in the block header for payload type to indicate what type of payload format is in use and add associated definition of payload type.

- o Clarify using Payload Type to determine the appropriate channel mapping in the definition of Channel Identifier.

B.12. draft-ietf-xrblock-rtcp-xr-qoe-00

The following are the major changes compared to previous version:

- o Allocate one more bit in the single channel per SSC segment to get alignment with the other two segment type.

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RTCP XR Blocks for Synchronization Delay and Offset Metrics Reporting
draft-ietf-xrblock-rtcp-xr-synchronization-00

Abstract

This document defines two RTCP XR Report Blocks and associated with 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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1. Introduction

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

The first 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].

The second new block type 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 SDPS 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:

Initial Synchronization Delay:

A multimedia session comprises a set of concurrent RTP sessions among a common group of participants, using one RTP session for each media type. Initial synchronization Delay is the average time for receiver to synchronize the components of a multimedia session [RFC6051].

Synchronization Offset:

The absolute delay variance of the measured RTP stream relative to the reference RTP stream in the multimedia session.

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 session, a receiver may not synchronize playout across the multimedia session or layered video session until RTCP SR packets have been received on all of the component RTP sessions. The component RTP session are referred to as each RTP session for each media type in multimedia session or separate RTP session for each layer in the layered video session. 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. 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 be 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.

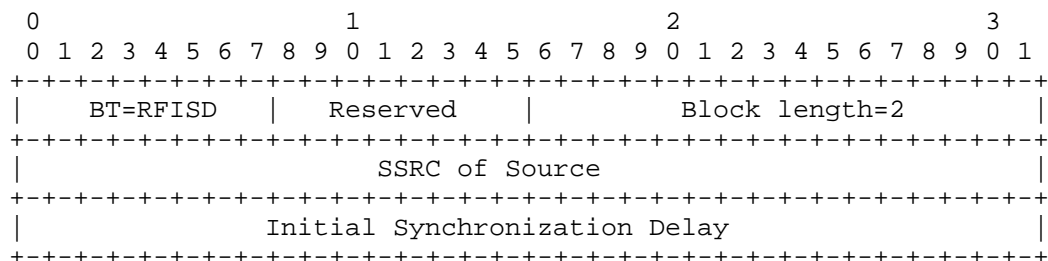
In an RTP multimedia session, there can be an arbitrary number of streams carried in different RTP sessions, with the same RTCP CNAME. These streams may be not synchronized with each other. For example, one audio stream and one video stream belong to the same session and audio stream are transmitted lag behind video stream for multiple tens of milliseconds. The RTP Flows 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 an arbitrary RTP stream belonging to the same multimedia session.

Initial Synchronization Delay: 32 bits

The average delay, expressed in units of 1/65536 seconds, from the RTCP packets received on all of the components RTP sessions to 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 synchronization 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.

If the measurement is unavailable, the value of this field with all bits set to 1 SHOULD be reported.

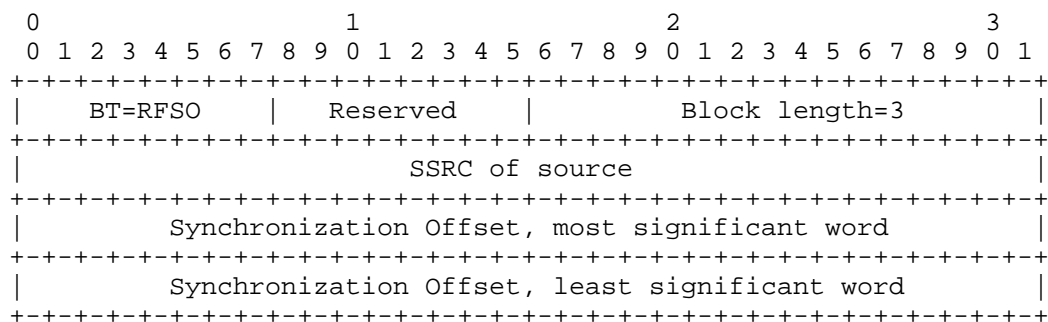
5. RTP Flows Synchronization Offset Metrics Block

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 synchronized by means of RTCP CNAME contained in the RTCP Source Description (SDS) packets [RFC3550].

This block is sent by RTP receivers and reports synchronization offset of the arbitrary two RTP streams that needs to be synchronized in the RTP multimedia session. Information is recorded about the actual delay variance of the measured RTP stream relative to the reference RTP stream with the same CNAME. The reference RTP stream can be chosen as the arbitrary stream with minimum delay according to the common criterion 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 <RFSO>.

Block length: 16 bits

The constant 3, 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.

Synchronization Offset: 64 bits

The synchronization offset of one RTP stream relative to the reference RTP stream with the same CNAME. The Synchronization Offset of the reference stream should be zero. 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. The value of this field is represented using a 64-bit NTP-format timestamp as defined in [RFC5905], which is 64-bit unsigned fixed-point number with the integer part in the first 32 bits and the fractional part in the last 32 bits.

If the measurement is unavailable, the value of this field with all bits set to 1 SHOULD be reported.

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-attr = "a=rtcp-xr:"  
               [xr-format *(SP xr-format)] CRLF  
xr-format = RTP-flows-init-syn-delay  
           / RTP-flows-syn-offset  
RTP-flows-init-syn-delay = "RTP-flows-init-syn-delay"  
                           ["=" max-size]  
RTP-flow-syn-offset = "RTP-flows-syn-offset"  
                     ["=" 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:	RFSO
Long Name:	RTP Flows Synchronization Offset Metrics Block
Value	<RFSO>
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-delay"
- * "RTP-flows-syn-offset"

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

The authors would like to thank Bill Ver Steeg, David R Oran, Ali Begen, Colin Perkins, Roni Even, Kevin Gross, Jing Zhao, Fernando Boronat Segui, Youqing Yang, Wenxiao Yu and Yinliang Hu for their valuable comments and suggestions on this document.

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- [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

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- [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-synchronization-07

Editorial changes are made from the previous version 06.

A.2. draft-asaeda-xrblock-rtcp-xr-synchronization-06

The following are the major changes compared to previous version 05:

- o Define synchronization offset as 64 bit NTP-format timestamp to meet synchronization resolution requirements for some RTP applications.
- o Add the definition of Initial Synchronization Delay in section 2.
- o Other editorial changes.

A.3. draft-asaeda-xrblock-rtcp-xr-synchronization-05

The following are the major changes compared to previous version 04:

- o Remove per packet reporting and only report a single value of general synchronization offset.

A.4. draft-asaeda-xrblock-rtcp-xr-synchronization-04

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

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

A.5. draft-asaeda-xrblock-rtcp-xr-synchronization-03

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

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

A.6. draft-asaeda-xrblock-rtcp-xr-synchronization-02

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

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

A.7. draft-asaeda-xrblock-rtcp-xr-synchronization-01

The following are the major changes compared to previous version:

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

A.8. draft-asaeda-xrblock-rtcp-xr-synchronization-00

The following are the major changes compared to previous version:

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.

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RTP Control Protocol (RTCP) Extended Report (XR) Blocks for
Synchronization Delay and Offset Metrics Reporting
draft-ietf-xrblock-rtcp-xr-synchronization-09

Abstract

This document defines two RTP Control Protocol (RTCP) Extended Report (XR) Blocks 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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1. Introduction

1.1. Synchronization Delay and Offset Metrics Reporting Blocks

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

The first 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].

The second new block type 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 Source description items (SDES) packets [RFC3550].

These metrics belong to the class of transport level metrics defined in [RFC6792].

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. [RFC3611] defined an extensible structure for reporting using an RTCP Extended Report (XR). This document defines a new Extended Report block for use with [RFC3550] and [RFC3611].

1.3. Performance Metrics Framework

The RTP Monitoring Architectures [RFC6792] provides guideline for reporting block format using RTCP XR. The new report block described in this memo is in compliance with the monitoring architecture specified in [RFC6792].

1.4. Applicability

When joining each session in layered video sessions [RFC6190] or the multimedia session, a receiver may not synchronize playout across the multimedia session or layered video session until RTCP Sender Report (SR) packets have been received on all components of RTP sessions. The component RTP session are referred to as each RTP session for each media type in multimedia session or separate RTP session for each layer in the layered video session. For multicast session, the initial synchronization delay metric varies with the session bandwidth, the number of members, and the number of senders in the session. The RTP flow Initial synchronization delay block defined in this document can be used to report such metric, i.e., the initial

synchronization delay to receive all the RTP streams belonging to the same multimedia session or layered video session. 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 should rely 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.

Receivers of the RTP flow initial synchronization delay block could use this metric to compare with targets (i.e., Service Level Agreement or thresholds of the system) to help ensure the quality of real-time application performance.

In an RTP multimedia session, there can be an arbitrary number of streams carried in different RTP sessions, 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 the audio stream is transmitted lagging behind video stream for multiple tens of milliseconds [TR-126]. The RTP Flows Synchronization Offset block can be used to report such synchronization offset between video stream and audio stream. This block is also applied to the case where an RTP session can contain media streams with media from multiple media types. The metrics defined in the RTP flows synchronization Offset block can be used by the network manager for trouble shooting and dealing with user experience issues.

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:

Initial Synchronization Delay:

A multimedia session comprises a set of concurrent RTP sessions among a common group of participants, using one RTP session for each media type. The initial synchronization Delay is the average time for receiver to synchronize all components of a multimedia session [RFC6051].

Synchronization Offset:

Synchronization between two media streams must be maintained to ensure satisfactory Quality of Experience (QoE). Two media streams can be of the same or different media type belonging to one RTP session or in different media types belonging to one multimedia session. The Synchronization Offset is the relative time difference of the two media streams that need to be synchronized.

3. 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 multimedia session and the time when the RTP receiver acquires all components of RTP sessions [RFC6051] measured at the receiving end of RTP stream.

This block needs only be exchanged occasionally, for example sent once at the start of RTP session.

3.1. Metric Block Structure

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

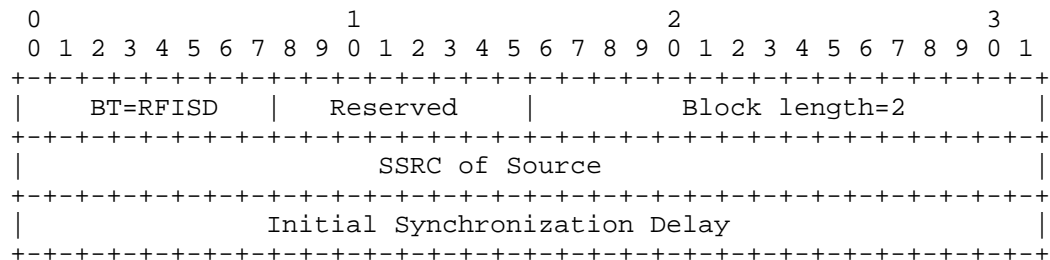


Figure 1: Report Block Structure

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

Block type (BT): 8 bits

The RTP Flows Initial Synchronization Delay Report Block is identified by the constant <RFISD>.

[Note to RFC Editor: please replace RFISD with the IANA provided RTCP XR block type for this block.]

Reserved: 8 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and ignored by the receiver.

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 any arbitrary component of RTP sessions belonging to the same multimedia session.

Initial Synchronization Delay: 32 bits

The average delay, expressed in units of 1/65536 seconds, from the beginning of multimedia session [RFC6051] to the time when RTCP packets are received on all of the components RTP sessions. It is recommended that the beginning of multimedia session is chosen as the time when the receiver has joined the first RTP session of the multimedia session. The value of the initial synchronization delay is calculated based on received RTCP SR packets or the RTP header extension containing in-band mapping of RTP and NTP-format timestamps [RFC6051]. If there is no packet loss, the initial synchronization 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 or the average time taken to receive the first RTP header extension containing in-band mapping of RTP and NTP-format timestamps.

If the measurement is unavailable, the value of this field with all bits set to 1 MUST be reported.

4. RTP Flows Synchronization Offset Metrics Block

In the RTP multimedia sessions or one RTP session, there can be an arbitrary number of Media streams and each media stream (e.g., audio stream or video stream) is sent in a separate RTP stream. In case of one RTP session, each media stream or each medium uses different SSRC. The receiver associates RTP streams to be synchronized by means of RTCP CNAME contained in the RTCP Source Description (SDES) packets [RFC3550].

This block is sent by RTP receivers and reports synchronization offset of two arbitrary RTP streams that needs to be synchronized in the RTP multimedia session. Information is recorded about the relative average time difference between two arbitrary RTP streams (one is reporting stream, the other is reference stream) with the same CNAME and measured at the receiving end of RTP stream. In order to tell what the offset of reporting stream is relative to, the block for reference stream with synchronization offset of zero should be reported.

Instances of this Block refer by Synchronization source (SSRC) to the separate auxiliary Measurement Information block [RFC6776] which describes measurement periods in use (see [RFC6776] section 4.2). This metrics block relies on the measurement period in the Measurement Information block indicating the span of the report and SHOULD be sent in the same compound RTCP packet as the measurement information block. If the measurement period is not received in the same compound RTCP packet as this Block, this Block MUST be discarded.

4.1. Metric Block Structure

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

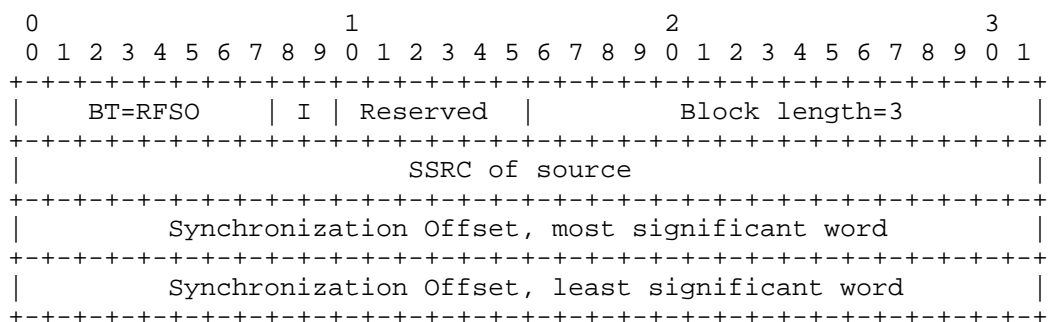


Figure 2: Report Block Structure

4.2. 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 <RFSO>.

[Note to RFC Editor: please replace RFSO with the IANA provided RTCP XR block type for this block.]

Interval Metric Flag (I): 2 bits

This field is used to indicate whether the Burst/Gap Discard Summary Statistics metrics are Sampled, Interval or Cumulative metrics:

I=10: Interval Duration - the reported value applies to the most recent measurement interval duration between successive metrics reports.

I=11: Cumulative Duration - the reported value applies to the accumulation period characteristic of cumulative measurements.

I=01: Sampled Value - the reported value is a sampled instantaneous value.

In this document, the value I=00 is the reserved value and MUST NOT be used. If the value I=00 is received, then the XR block MUST be ignored by the receiver.

Reserved: 6 bits

This field is reserved for future definition. In the absence of such a definition, the bits in this field MUST be set to zero and MUST be ignored by the receiver.

Block length: 16 bits

The constant 3, 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 reporting RTP stream to which the XR relates.

Synchronization Offset: 64 bits

The synchronization offset of the reporting RTP stream relative to the reference stream with the same CNAME. The calculation of Synchronization Offset is similar to Difference D calculation in the RFC3550. That is to say, if S_i is the NTP timestamp from the reporting RTP packet i , and R_i is the time of arrival in NTP timestamp units for reporting RTP packet i , S_j is the NTP timestamp from the reference RTP packet j , and R_j is the time of arrival in NTP timestamp units for reference RTP packet j , then the value of the synchronization offset D may be expressed as

$$D(i,j) = (R_j - R_i) - (S_j - S_i) = (R_j - S_j) - (R_i - S_i)$$

If in-band delivery of NTP-format timestamps is supported [RFC6051], S_i and S_j should be obtained directly from the RTP packets where NTP timestamps are available. If not, S_i and S_j should be calculated from their corresponding RTP timestamps. The value of the synchronization offset is represented using a 64-bit signed NTP-format timestamp as defined in [RFC5905], which is 64-bit signed fixed-point number with the integer part in the first 32 bits and the fractional part in the last 32 bits. A positive value of the synchronization offset means that the reporting stream leads before the reference stream, while a negative one means the reporting stream lags behind the reference stream. The synchronization offset of zero means the stream is the reference stream.

If the measurement is unavailable, the value of this field with all bits set to 1 MUST be reported.

5. SDP Signaling

[RFC3611] defines the use of SDP (Session Description Protocol) [RFC4566] for signaling the use of XR blocks. XR blocks MAY be used without prior signaling.

5.1. SDP rtcp-xr-attr Attribute Extension

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]:

```
xr-format = xr-rfisd-block  
           / xr-rfso-block
```

```
xr-rfisd-block = "rtp-flow-init-syn-delay"  
xr-rfso-block = "rtp-flow-syn-offset"
```

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

5.2. Offer/Answer Usage

When SDP is used in offer-answer context, the SDP Offer/Answer usage defined in [RFC3611] applies.

6. 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 3
Name:	RFSO
Long Name:	RTP Flows Synchronization Offset Metrics Block
Value	<RFSO>
Reference:	Section 4

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

```
* "rtp-flow-init-syn-delay "  
* "rtp-flow-syn-offset"
```

The contact information for the registrations is:

RAI Area Directors

<rai-ads@tools.ietf.org>

7. Security Considerations

When using Secure RTP [RFC3711], or other media layer security, reporting accurate synchronisation offset information can expose some details about the timing of the cryptographic operations that are used to protect the media. There is a possibility that this timing information might enable a side-channel attack on the encryption. For environments where this attack is a concern, implementations need to take care to ensure cryptographic processing and media compression take the same amount of time irrespective of the media content, to avoid the potential attack.

Besides this, it is believed that this RTCP XR block introduces no new security considerations beyond those described in [RFC3611].

8. Acknowledgements

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Appendix A. Metrics represented using RFC6390 Template

RFC EDITOR NOTE: please change XXXX in [RFCXXXX] by the new RFC number, when assigned.

a. Initial Synchronization Delay Metric

- * Metric Name: RTP Initial Synchronization Delay
- * Metric Description: See Section 2.1, Initial Synchronization Delay term [RFCXXXX].

- * Method of Measurement or Calculation: See section 3.2, Initial Synchronization Delay definition [RFCXXXX].
- * Units of Measurement: See section 3.2, Initial Synchronization Delay definition [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 3, 1st paragraph [RFCXXXX].
- * Measurement Timing: See section 3, 2nd paragraph [RFCXXXX] for measurement timing.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

b. Synchronization Offset Metric

- * Metric Name: RTP Synchronization Offset Delay
- * Metric Description: See Section 2.1, Synchronization Offset term [RFCXXXX].
- * Method of Measurement or Calculation: See section 4.2, Initial Synchronization Delay definition [RFCXXXX].
- * Units of Measurement: See section 4.2, Initial Synchronization Delay definition [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: See section 4, 2nd paragraph [RFCXXXX].
- * Measurement Timing: See section 4, 3rd paragraph [RFCXXXX] for measurement timing and section 4.2 [RFCXXXX] for Interval Metric flag.
- * Use and applications: See section 1.4 [RFCXXXX].
- * Reporting model: See RFC3611.

Appendix B. Change Log

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

B.1. draft-ietf-xrblock-rtcp-xr-synchronization-09

The following are the major changes compared to previous version:

Some Editorial changes based on IESG Review comments.

B.2. draft-ietf-xrblock-rtcp-xr-synchronization-08

The following are the major changes compared to previous version:

Some Editorial changes based on Gen-Art Reviewer comments.

B.3. draft-ietf-xrblock-rtcp-xr-synchronization-07

The following are the major changes compared to previous version:

Minor Editorial changes.

B.4. draft-ietf-xrblock-rtcp-xr-synchronization-06

The following are the major changes compared to previous version:

Some Editorial changes.

B.5. draft-ietf-xrblock-rtcp-xr-synchronization-05

The following are the major changes compared to previous version:

Editorial changes and typo fixed.

B.6. draft-ietf-xrblock-rtcp-xr-synchronization-04

The following are the major changes compared to previous version:

Additional text to clarify on how to distinguish report stream from reference stream.

Other Editorial changes.

B.7. draft-ietf-xrblock-rtcp-xr-synchronization-03

The following are the major changes compared to previous version:

Remove the need to signal the reference source in the synchronisation offset metrics RTCP XR report.

Apply RFC6390 template to metrics in the appendix.

Other editorial changes to get inline with other XRBLOCK drafts.

B.8. draft-ietf-xrblock-rtcp-xr-synchronization-02

The following are the major changes compared to previous version:

Editorial change based on comments raised on the list and in the IETF85 meeting

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