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RTP Control Protocol (RTCP) Extended Report (XR) Block for Jitter Buffer
Metric Reporting
draft-ietf-xrblock-rtcp-xr-jb-08.txt

Abstract

This document defines an RTP Control Protocol (RTCP) Extended Report (XR) Block that allows the reporting of Jitter Buffer metrics for a range of RTP applications.

Status of this Memo

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1. Introduction

1.1. Jitter Buffer Metrics 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 jitter buffer configuration and performance.

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

Instances of this Metrics Block refer by Synchronization source (SSRC) to the separate auxiliary Measurement Information block [RFC6776] which contains information such as the SSRC of the measured stream, and RTP sequence numbers and time intervals indicating the span of the report.

1.2. RTCP and RTCP XR Reports

The use of RTCP for reporting is defined in [RFC3550]. [RFC3611] defines 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. Metrics described in this draft are in accordance with the guidelines in [RFC6390] and [RFC6792].

1.4. Applicability

Real-time applications employ a jitter buffer to absorb jitter introduced on the path from source to destination. These metrics are used to report how the jitter buffer at the receiving end of RTP stream behaves as a result of jitter in the network and are applicable to a range of RTP applications.

These metrics reflect how terminal-related factors affect real-time application quality and are useful to provide better end-user quality of experience (QoE).

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. Jitter Buffer Operation

A jitter buffer is required to absorb delay variation in network delivery of media packets. A jitter buffer works by holding media data for a period of time after it is received and before it is played out. Packets that arrive early are held in the jitter buffer longer. If packets arrive too early they may be discarded if there is no available jitter buffer space. If packets are delayed excessively by the network they may be discarded if they miss their playout time.

The jitter buffer can be considered as a time window with one side (the early window) aligned with the delay corresponding to the earliest arriving packet and the other side (the late window) representing the maximum permissible delay before a late arriving packet would be discarded. The delay applied to packets that arrive at their expected time is known as the Nominal Delay and this is equivalent to the late window.

The "expected arrival time" is the time that a packet would arrive if there was no delay variation. If all packets arrived at their expected arrival time then every packet would be delayed by exactly the Nominal Delay. Early packets arrive before their expected arrival time and late packets arrive after. The reference for the expected arrival time may, for example, be the first packet in the session or the running average delay.

Jitter Buffer delay is the time spent by a packet in the jitter buffer. The Jitter Buffer Nominal Delay is the delay applied to packets arriving at their expected time. The Jitter Buffer maximum delay is the delay that is applied to an earliest arriving packet that is not discarded and corresponds to the early window of the jitter buffer.

3.1. Fixed Jitter Buffer

A receiver can use either a fixed or adaptive jitter buffer. A fixed jitter buffer is a simple implementation however may not give optimum performance in terms of packet discard rate and delay.

3.2. Adaptive Jitter Buffer

An adaptive jitter buffer allows the nominal delay to be set to a low value initially, to minimize user perceived delay, however can automatically increase the late window if a significant proportion of packets are arriving late (and hence being discarded).

4. Jitter Buffer Metrics Block

This block describes the configuration and operating parameters of the jitter buffer in the receiver of the RTP end system or RTP mixer which sends the report. Instances of this Metrics Block refer by SSRC to the separate auxiliary Measurement Information block [RFC6776] which describes the measurement interval in use. This Metrics Block relies on the measurement interval 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 interval is not received in the same compound RTCP packet as this Metrics Block, this Metrics Block should be discarded.

4.1. Report Block Structure

JB Metrics Block

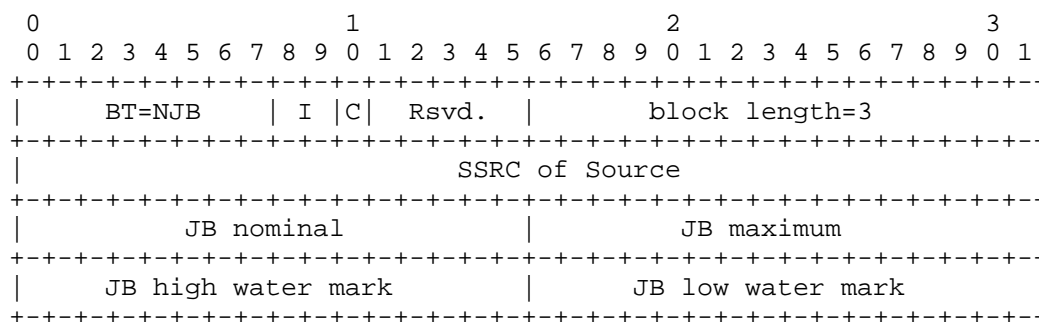


Figure 1: Report Block Structure

4.2. Definition of Fields in Jitter Buffer Metrics Block

Block type (BT): 8 bits

A Jitter Buffer Metrics Report Block is identified by the constant NJB.

[Note to RFC Editor: please replace NJB 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 Jitter Buffer metrics are Sampled, Interval or Cumulative metrics:

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

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.

Jitter Buffer Configuration (C): 1 bit

This field is used to identify the jitter buffer method in use at the receiver, according to the following code:

0 = Fixed jitter buffer

1 = Adaptive jitter buffer

Reserved (Rsvd.): 5 bits

These bits are reserved. They MUST be set to zero by senders 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 3 to match the fixed length of the report block.

jitter buffer nominal delay (JB nominal): 16 bits

This is the current nominal jitter buffer delay in milliseconds, which corresponds to the nominal jitter buffer delay for packets that arrive exactly on time. It is calculated based on the time spend in the jitter buffer for the packet that arrives exactly on time. This parameter MUST be provided for both fixed and adaptive jitter buffer implementations.

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

jitter buffer maximum delay (JB maximum): 16 bits

This is the current maximum jitter buffer delay in milliseconds which corresponds to the earliest arriving packet that would not be discarded. It is calculated based on the time spent in the jitter buffer for the earliest arriving packet. In simple queue implementations this may correspond to the size of the jitter buffer. In adaptive jitter buffer implementations, this value may vary dynamically. This parameter **MUST** be provided for both fixed and adaptive jitter buffer implementations.

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

jitter buffer high water mark (JB high water mark): 16 bits

This is the highest value of the jitter buffer nominal delay in milliseconds which occurred at any time during the reporting interval. This parameter **MUST** be provided for adaptive jitter buffer implementations and its value **MUST** be set to JB maximum for fixed jitter buffer implementations.

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

jitter buffer low water mark (JB low water mark): 16 bits

This is the lowest value of the jitter buffer nominal delay in milliseconds which occurred at any time during the reporting interval. This parameter **MUST** be provided for adaptive jitter buffer implementations and its value **MUST** be set to JB maximum for fixed jitter buffer implementations.

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

5. 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).

5.1. 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-jb-block

xr-jb-block = "jitter-bfr"

5.2. 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].

6. 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].

6.1. New RTCP XR Block Type value

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

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

6.2. New RTCP XR SDP Parameter

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

6.3. Contact information for registrations

The contact information for the registrations is:

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101 Software Avenue, Yuhua District
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China

7. 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.

8. Contributors

Geoff Hunt wrote the initial draft of this document.

9. 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, Hideaki Yamada, Claire Bi, Colin Perkin, Dan Romascanu, Kevin Gross and Glen Zorn.

10. References

10.1. 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] Wu, Q., "Measurement Identity and information Reporting using SDES item and XR Block", RFC 6776, August 2012.

10.2. Informative References

- [RFC6390] Clark, A. and B. Claise, "Framework for Performance Metric Development", RFC 6390, October 2011.
- [RFC6792] Hunt, G., Wu, Q., and P. Arden, "Monitoring Architectures for RTP", RFC 6792, November 2012.

Appendix A. Change Log

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

A.1. draft-ietf-xrblock-rtcp-xr-jb-08

The following are the major changes to previous version :

- o Rewrote descriptive text and definitions for clarification.

A.2. draft-ietf-xrblock-rtcp-xr-jb-07

The following are the major changes to previous version :

- o Add one new section to discuss jitter buffer operation.

A.3. draft-ietf-xrblock-rtcp-xr-jb-05

The following are the major changes to previous version :

- o Some editorial change changes based on the discussion with Glen and Kevin on the list.

A.4. draft-ietf-xrblock-rtcp-xr-jb-03

The following are the major changes to previous version :

- o Reduce the "jb cfg" to 1-bit based on discussion in the WGLC.
- o Other editorial change changes aligning with PDV,Delay draft.

A.5. draft-ietf-xrblock-rtcp-xr-jb-02

The following are the major changes to previous version :

- o Add some explanation text in the SDP offer/answer section.
- o Add some text in applicability section to explain the use to report jitter buffer metrics.
- o Other editorial change changes aligning with PDV,Delay draft.

A.6. draft-ietf-xrblock-rtcp-xr-jb-01

The following are the major changes to previous version :

- o Outdated reference update
- o Add one Editor notes to ask clarification on the use of reporting jitter buffer metrics.
- o Other Editorial changes.

A.7. draft-ietf-xrblock-rtcp-xr-jb-00

The following are the major changes to previous version :

- o Boilerplate updates.
- o references updates
- o allocate 32 bit field in report block for SSRC
- o Other editorial changes to get alignment with MONARCH draft.

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RTCP XR Report Block for Concealment metrics Reporting on Audio
Applications
draft-ietf-xrblock-rtcp-xr-loss-conceal-04.txt

Abstract

This document defines two RTCP XR Report Blocks that allows the reporting of loss concealment metrics for audio applications of RTP.

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. Loss Concealment and Concealment Seconds Metrics Reporting Block

At any instant, the audio output at a receiver may be classified as either 'normal' or 'concealed'. 'Normal' refers to playout of audio payload received from the remote end, and also includes locally generated signals such as announcements, tones and comfort noise. Concealment refers to playout of locally-generated signals used to mask the impact of network impairments or to reduce the audibility of jitter buffer adaptations.

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

The first block type provides metrics for actions taken by the receiver to mitigate the effect of packet loss and packet discard. Specifically, the first metric (On-Time Playout Duration) reports the duration of normal playout of data which the receiver obtained from the sender's stream. A second metric (Loss Concealment Duration) reports the total time during which the receiver played out media data which was manufactured locally, because the sender's data for these periods was not available due to packet loss or discard. A similar metric (Buffer Adjustment Concealment Duration) reports the duration of playout of locally-manufactured data replacing data which is unavailable due to adaptation of an adaptive de-jitter buffer. Further metrics (Playout Interrupt Count and Mean Playout Interrupt Size) report the number of times normal playout was interrupted, and the mean duration of these interruptions.

Loss Concealment Duration and Buffer Adjustment Concealment Duration are reported separately because buffer adjustment is typically arranged to occur in silence periods so may have very little impact on user experience, whilst loss concealment may occur at any time.

The second block type provides metrics for concealment. Specifically, the first metric (Unimpaired Seconds) reports the number of whole seconds occupied only with normal playout of data which the receiver obtained from the sender's stream. The second metric (Concealed Seconds) reports the number of whole seconds during which the receiver played out any locally-generated media data. A third metric (Severely Concealed Seconds) reports the number of whole seconds during which the receiver played out locally-generated data for more than SCS Threshold (ms).

These metrics belongs to the class of transport-related terminal 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 draft defines a new Extended Report block that MUST be used as defined in [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 those guidelines.

1.4. Applicability

These metrics are only applicable to audio applications of 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 RFC2119 [RFC2119].

3. Loss Concealment Block

3.1. Report Block Structure

Loss Concealment metrics block

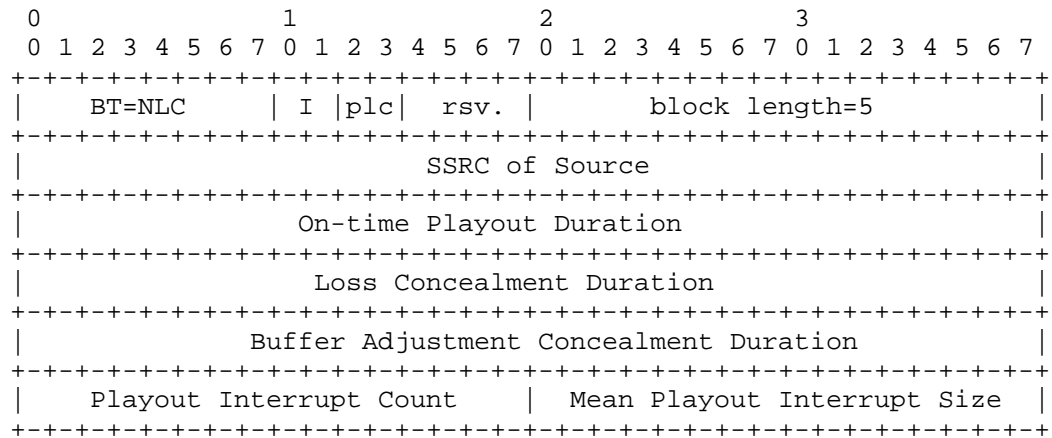


Figure 1: Report Block Structure

3.2. Definition of Fields in Loss Concealment Report Block

Block type (BT): 8 bits

A Loss Concealment Metrics Report Block is identified by the constant NLC.

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

Interval Metric flag (I): 2 bit

This field is used to indicate whether the Loss Concealment 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.

Packet Loss Concealment Method (plc): 2 bits

This field is used to identify the packet loss concealment method in use at the receiver, according to the following code:

bits 014-015

0 = silence insertion

1 = simple replay, no attenuation

2 = simple replay, with attenuation

3 = enhanced

Other values reserved

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. For the Loss Concealment block, the block length is equal to 5.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

On-time Playout Duration (ms): 32 bits

'On-time' playout is the uninterrupted, in-sequence playout of valid decoded audio information originating from the remote endpoint. This includes comfort noise during periods of remote talker silence, if VAD is used, and locally generated or regenerated tones and announcements.

An equivalent definition is that on-time playout is playout of any

signal other than those used for concealment.

On-time playout duration MUST include both speech and silence intervals, whether VAD is used or not. This duration is reported in millisecond units.

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

Loss Concealment Duration (ms): 32 bits

The duration, in milliseconds, of audio playout corresponding to Loss-type concealment.

Loss-type concealment is reactive insertion or deletion of samples in the audio playout stream due to effective frame loss at the audio decoder. "Effective frame loss" is the event in which a frame of coded audio is simply not present at the audio decoder when required. In this case, substitute audio samples are generally formed, at the decoder or elsewhere, to reduce audible impairment.

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

Buffer Adjustment Concealment Duration (ms): 32 bits

The duration, in milliseconds, of audio playout corresponding to Buffer Adjustment-type concealment, if known.

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

Buffer Adjustment-type concealment is proactive or controlled insertion or deletion of samples in the audio playout stream due to jitter buffer adaptation, re-sizing or re-centering decisions within the endpoint.

Because this insertion is controlled, rather than occurring randomly in response to losses, it is typically less audible than loss-type concealment. For example, jitter buffer adaptation events may be constrained to occur during periods of talker silence, in which case only silence duration is affected, or

sophisticated time-stretching methods for insertion/deletion during favorable periods in active speech may be employed.

Concealment events which cannot be classified as Buffer Adjustment- type MUST be classified as Loss-type.

Playout Interrupt Count: 16 bits

The number of interruptions to normal playout which occurred during the reporting period.

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

Mean Playout Interrupt Size (ms): 16 bits

The mean duration, in ms, of interruptions to normal playout which occurred during the reporting period.

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

4. Concealment Seconds Block

This sub-block provides a description of potentially audible impairments due to lost and discarded packets at the endpoint, expressed on a time basis analogous to a traditional PSTN T1/E1 errored seconds metric.

The following metrics are based on successive one second intervals as declared by a local clock. This local clock does NOT need to be synchronized to any external time reference. The starting time of this clock is unspecified. Note that this implies that the same loss pattern could result in slightly different count values, depending on where the losses occur relative to the particular one-second demarcation points. For example, two loss events occurring 50ms apart could result in either one concealed second or two, depending on the particular 1000 ms boundaries used.

The seconds in this sub-block are not necessarily calendar seconds. At the tail end of a session, periods of time of less than 1000ms shall be incorporated into these counts if they exceed 500ms and shall be disregarded if they are less than 500ms.

4.1. Report Block Structure

Concealed Seconds metrics block

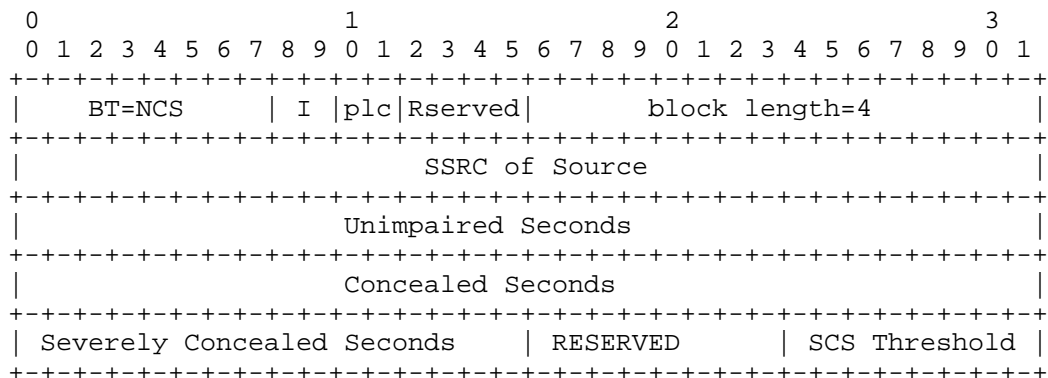


Figure 1: Report Block Structure

4.2. Definition of Fields in Concealed Seconds Metrics Block

Block type (BT): 8 bits

A Concealed Seconds Metrics Report Block is identified by the constant NCS.

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

Interval Metric flag (I): 2 bit

This field is used to indicate whether the Concealment Seconds 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.

Packet Loss Concealment Method (plc): 2 bits

This field is used to identify the packet loss concealment method in use at the receiver, according to the following code:

bits 014-015

0 = silence insertion

1 = simple replay, no attenuation

2 = simple replay, with attenuation

3 = enhanced

Other values reserved

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. For the Concealment Seconds block, the block length is equal to 4.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

Unimpaired Seconds: 32 bits

A count of the number of unimpaired Seconds that have occurred.

An unimpaired Second is defined as a continuous period of 1000ms during which no frame loss or discard due to late arrival has occurred. Every second in a session must be classified as either OK or Concealed.

Normal playout of comfort noise or other silence concealment signal during periods of talker silence, if VAD [VAD] is used, shall be counted as unimpaired seconds.

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

Concealed Seconds: 32 bits

A count of the number of Concealed Seconds that have occurred.

A Concealed Second is defined as a continuous period of 1000ms during which any frame loss or discard due to late arrival has occurred.

Equivalently, a concealed second is one in which some Loss-type concealment has occurred. Buffer adjustment-type concealment SHALL not cause Concealed Seconds to be incremented, with the following exception. An implementation MAY cause Concealed Seconds to be incremented for 'emergency' buffer adjustments made during talkspurts.

Loss-type concealment is reactive insertion or deletion of samples in the audio playout stream due to effective frame loss at the audio decoder. "Effective frame loss" is the event in which a frame of coded audio is simply not present at the audio decoder

when required. In this case, substitute audio samples are generally formed, at the decoder or elsewhere, to reduce audible impairment.

Buffer Adjustment-type concealment is proactive or controlled insertion or deletion of samples in the audio playout stream due to jitter buffer adaptation, re-sizing or re-centering decisions within the endpoint.

Because this insertion is controlled, rather than occurring randomly in response to losses, it is typically less audible than loss-type concealment. For example, jitter buffer adaptation events may be constrained to occur during periods of talker silence, in which case only silence duration is affected, or sophisticated time-stretching methods for insertion/deletion during favorable periods in active speech may be employed. For these reasons, buffer adjustment-type concealment MAY be exempted from inclusion in calculations of Concealed Seconds and Severely Concealed Seconds.

However, an implementation SHOULD include buffer-type concealment in counts of Concealed Seconds and Severely Concealed Seconds if the event occurs at an 'inopportune' moment, with an emergency or large, immediate adaptation during active speech, or for unsophisticated adaptation during speech without regard for the underlying signal, in which cases the assumption of low-audibility cannot hold. In other words, jitter buffer adaptation events which may be presumed to be audible SHOULD be included in Concealed Seconds and Severely Concealed Seconds counts.

Concealment events which cannot be classified as Buffer Adjustment-type MUST be classified as Loss-type.

For clarification, the count of Concealed Seconds MUST include the count of Severely Concealed Seconds.

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

Severely Concealed Seconds: 16 bits

A count of the number of Severely Concealed Seconds.

A Severely Concealed Second is defined as a non-overlapping period of 1000 ms during which the cumulative amount of time that has been subject to frame loss or discard due to late arrival, exceeds the SCS Threshold.

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

Reserved: 8 bits

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

SCS Threshold: 8 bits

The SCS Threshold defines the amount of time corresponding to lost or discarded frames that must occur within a one second period in order for the second to be classified as a Severely Concealed Second. This is expressed in milliseconds and hence can represent a range of 0.1 to 25.5 percent loss or discard.

A default threshold of 50ms (5% effective frame loss per second) is suggested.

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

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

```
xr-format =/ xr-conceal-block  
xr-format =/ xr-conc-sec-block
```

```
xr-conceal-block = "loss-conceal"  
xr-conc-sec-block = "conc-sec" ["=" thresh]
```

```
thresh          = 1*DIGIT           ; threshold for SCS (ms)  
DIGIT           = %x30-39
```

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

6.1. New RTCP XR Block Type values

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

Name: NLC
Long Name: Loss Concealment Block
Value <NLC>
Reference: Section 3.1

Name: NCS
Long Name: Concealment Seconds Block
Value <NCS>
Reference: Section 4.1

[Note to RFC Editor: please replace <NLC> and <NCS> with the RTCP XR block type assigned by IANA for this block.]

6.2. New RTCP XR SDP Parameters

This document also registers two new parameters in the "RTCP XR SDP Parameters Registry":

- o "loss-conceal"
- o "conc-sec"

6.3. Contact information for registrations

The contact information for the registrations is:

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China

7. 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.

8. Contributors

Geoff Hunt wrote the initial draft of this document.

9. Acknowledgements

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Appendix A. Change Log

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

A.1. draft-ietf-xrblock-rtcp-xr-loss-conceal-04

The following are the major changes to previous version :

- o Merge Concealment Seconds draft into this draft (i.e., Loss Concealment draft).
- o Updated references.

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RTP Control Protocol (RTCP) Extended Report (XR) Blocks for QoE Metric
Reporting
draft-ietf-xrblock-rtcp-xr-qoe-06

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 QoE metrics for use in a range of RTP applications.

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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 [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 draft 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 XR Block described in this document are in accordance with the guidelines in [RFC6390] and [RFC6792].

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

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

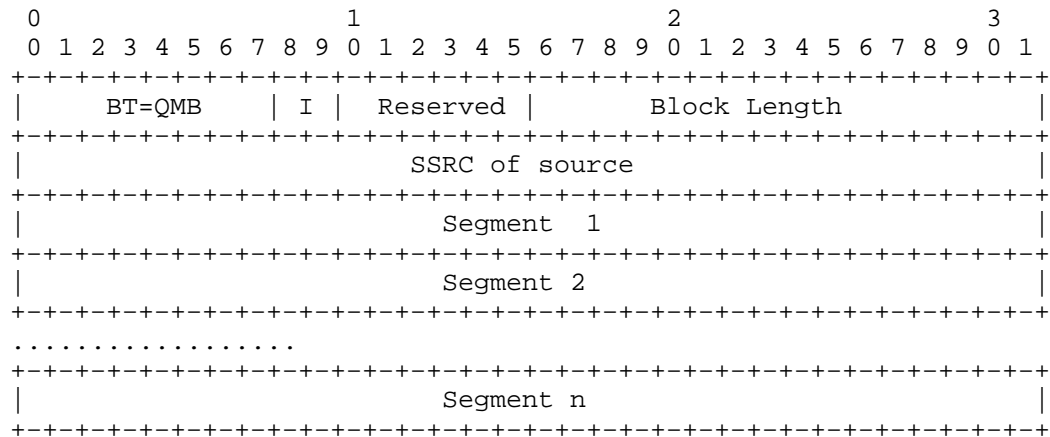
3. QoE Metrics Block

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.1201.1][P.1201.2][P.1202.1][P.NBAMS-HR] 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 <QMB>.

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).

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 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 leftmost bit of the segment determines 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

```

+-----+
|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 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 the 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 each identified by 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 in the same RTCP XR packet.

Calc Algorithm ID (CAID) : 8bits

The 8-bit CAID is the local identifier of calculation algorithm associated with this segment in the range 1-255 inclusive.

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.

MOS Value: 16 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 8:8 with the value in the range 0.0 to 255.996. 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 0xFFFE MUST be reported to indicate an over-range measurement. If the measurement is unavailable, the value 0xFFFF MUST be reported. Values other than 0xFFFE, 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

```

+-----+-----+-----+-----+-----+-----+-----+-----+-----+
|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.

CAID Algorithm ID (CAID) : 8bits

The 8-bit ID is the local identifier of this segment in the range 1-255 inclusive.

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 [RFC6792] and using Payload Type to determine the appropriate channel mapping.

MOS Value: 13 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 6:7 with the value in the range 0.0 to 63.992. 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 0x1FFE MUST be reported to indicate an over-range measurement. If the measurement is unavailable, the value 0x1FFF MUST be reported. Values other than 0x1FFE, 0x1FFF and the valid range defined above 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-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. Within the "xr-format", the syntax element "extmap" 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 may be several.

```

xr-format =/ xr-qoe-block
xr-qoe-block = "qoe-metrics" ["=" extmap *("," extmap)]
extmap = mapentry "=" extensionname [SP extentionattributes]
direction = "sendonly" / "recvonly" / "sendrecv" / "inactive"
mapentry = "calg:" 1*5 DIGIT ["/" direction]
extensionname = "P564";ITU-T P.564 Compliant Algorithm [P.564]
               / "G107";ITU-T G.107 [G.107]
               / "TS101_329";ETSI TS 101 329-5 Annex E [ETSI]
               / "JJ201_01 ";TTC JJ201.01 [TTC]
               / "P1201_01";ITU-T P.1201.2 [P.1201.1]
               / "P1201_02";ITU-T P.1201.2 [P.1201.2]
               / "P1202_01";ITU-T P.1202.1 [P.1202.1]
               / "P1202_02";ITU-T P. NBAMS-HR [P.NBAMS-HR]
               / non-ws-string
extentionattributes = mediatype
                    /mosreference
                    /attributes-ext
mediatype = "a" ;voice
           / "v" ;video
           / "m" ;multimedia
mosreference = "mosref=" ("0"; lower resolution
                    / "1";higher resolution
                    / 1*2DIGIT ) ;Value 2~15 are valid and
                    ;reserved for future use

attributes-ext = non-ws-string
SP = <Define in RFC5234>
DIGIT = <as defined in Section 3.4 of [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=extmap:<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 = 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).

Note that the syntax element "mosreference" is referred to the media resolution(e.g., Narrowband (3.4kHz) Speech and Standard Definition (SD) Resolution Video with lower resolution, Wideband (7kHz) Speech and High Definition (HD) Resolution Video with higher resolution). MOS scores reported in the QoE block may vary with the Mos reference; For example MOS values for narrowband, 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 may 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 an extmap; without it, the direction implicitly inherits, of course, from the stream direction.

Segment extension, with their directions, may be signaled for an "inactive" stream. It is an error to use an extension direction incompatible with the stream direction (e.g., a "sendonly" attribute for a "recvonly" stream).

If an segment extension map 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.

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). Identifiers 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:qoe-  
metrics=calg:4906=P1201.1m,calg:4906=P1202.1v,calg:4907=G107a
```

The answerer is interested in transmission P.1202.1 only on video, but doesn't understand P.1202.1 at all. It is interested in transmission G.107 on audio. It therefore adjusts the declarations:

```
a=rtcp-xr:qoe-metrics=calg:1=P1202.1v, calg:2=G107a
```

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

This document creates a new registry 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 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. This should include the units of measurement, how values of the metric are reported in the one 16-bit fields or 13-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 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_01	TTC JJ201.01	[TTC]	Voice
P1201_01	ITU-T P.1201.01	[P.1201.1]	Multimedia
P1201_02	ITU-T P.1201.02	[P.1201.2]	Multimedia
P1202_01	ITU-T P.1202.01	[P.1202.01]	Video
P1202_02	ITU-T P. NBAMS-HR	[P. NBAMS-HR]	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. Evaluation Example of User Quality of Experience for video stream

To evaluate user quality of experience levels using objective test data. MoS Scores provide a familiar, easily understood numeric representation of video, audio, and overall audiovisual quality. Unlike audio, video is even more sensitive to transport impairments than voice, and even low rates of packet loss can cause severe degradation in perceived quality. However, all occurrences of packet loss do not have an equal impact on perceptual quality, in part

because of the way video frames are structured during the encoding process - such as frame properties including frame type and quantization parameter (QP), frame structure, and in part due to subjective factors - such as the degree to which perception is affected by the levels of motion and detail in the video sequence, demux/decoder statistics characteristic parameters including packet loss concealment metrics, jitter buffer metrics and/or Frame loss rate parameter. Note that Frame loss rate can be derived When a video stream is sent from the media source to RTP receiving end and get monitored, in order to provide accurate evaluation of video quality, One possible evaluation method of QoE is the network nodes that implement network management tools may get frame properties, perception degree as MoS calculation input parameters from media source, and demux/decoder statistics characteristic parameters and transport impairment as other MoS calculation input parameters from the RTP receiving end and use appropriate MoS calculation algorithm to calculate MoS scores. Such MoS Scores value can be useful for troubleshooting or comparing video quality across different service types.

Appendix B. Change Log

B.1. 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.2. 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.3. 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.4. 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.5. 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.6. 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
Synchronization Delay and Offset Metrics Reporting
draft-ietf-xrblock-rtcp-xr-synchronization-02

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 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 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 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. The metrics defined in the RTP flows synchronization Offset block can be used by 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:

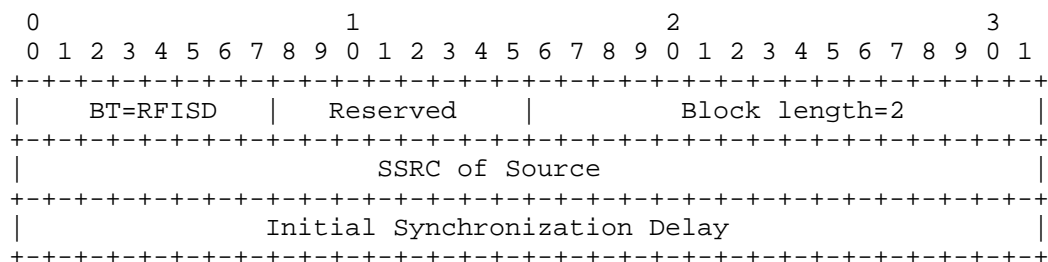
Synchronization between two media streams must be maintained to ensure satisfactory QoE. Two media streams can be of the same media type belonging to one RTP session or in different media types belonging to one multimedia session. The Synchronization Offset is the relative time difference of the two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

3.1. Metric Block Structure

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



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>.

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 MUST be 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

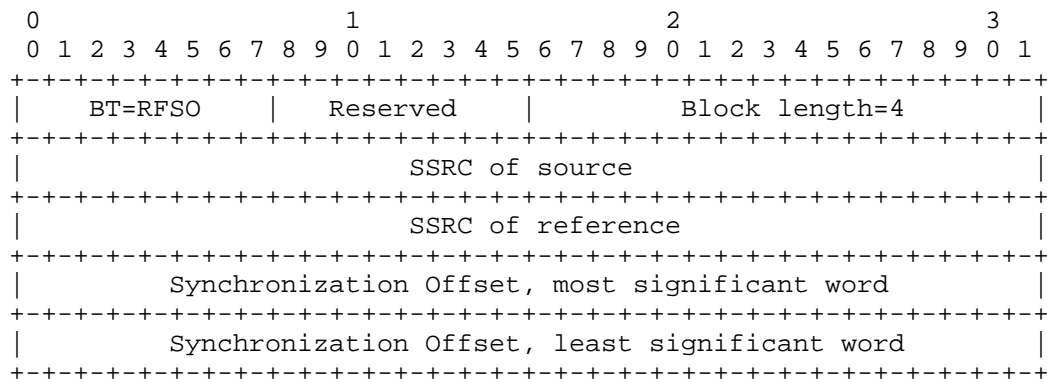
In the RTP multimedia sessions, there can be an arbitrary number of Media streams and each media stream (e.g., audio stream or video stream) is sent in a separate RTP stream. The receiver associates RTP streams to be synchronized by means of RTCP CNAME contained in the RTCP Source Description (SDS) packets [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 relative average time difference between the reporting stream and the reference stream with the same CNAME. For multimedia session with multiple media types (e.g., audio and video), it is recommended to choose the stream with the lower bandwidth as the reference stream. For layered video sessions, it is recommended to use the base layer

stream as the reference stream.

4.1. Metric Block Structure

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



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>.

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 MUST be ignored by the receiver.

Block length: 16 bits

The constant 4, 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.

SSRC of Reference: 32 bits

The SSRC of the reference stream 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 the reporting RTP stream relative to the reference RTP 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 value means that the reporting stream lags behind 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 = " init-syn-delay"  
xr-rfso-block = " 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:

```
* "xr-rfisd "  
* "xr-rfso"
```

The contact information for the registrations is:

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7. Security Considerations

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

8. Acknowledgements

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Appendix A. Change Log

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

A.1. 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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