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RTCP XR Report Block for TS Decodability Statistics Metric reporting  
draft-ietf-xrblock-rtcp-xr-decodability-00

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.

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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"

The contact information for the registrations is:

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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

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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.

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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 0xffff are used as flags for "over-range" and "unavailable" conditions, a 0:16 quantity has range

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

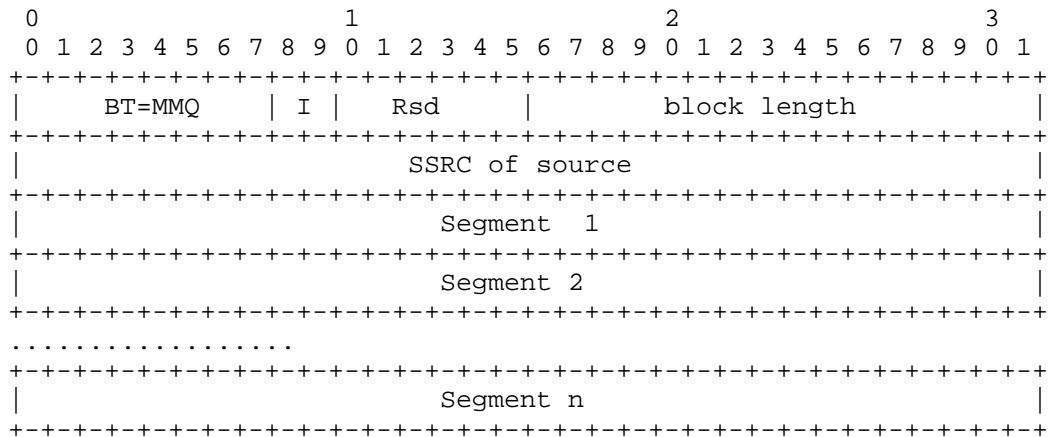
### 3. QoE Metrics Block

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-attrib = "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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## 8. Acknowledgements

The authors would like to thank Alan Clark, Bill Ver Steeg, David R Oran, Ali Begen, Colin Perkins, Roni Even, Youqing Yang, Wenxiao Yu and Yinliang Hu for their valuable comments and suggestions on this document.

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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  
Synchronization Delay and Offset Metrics Reporting  
draft-ietf-xrblock-rtcp-xr-synchronization-01

Abstract

This document defines two RTP Control Protocol (RTCP) Extended Report (XR) 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

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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 SDES packets [RFC3550]. Information is recorded about the synchronization offset time of each RTP stream relative to the reference RTP stream with the same CNAME and General Synchronization Offset of zero.

These metrics belong to the class of 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. 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. 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 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 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.

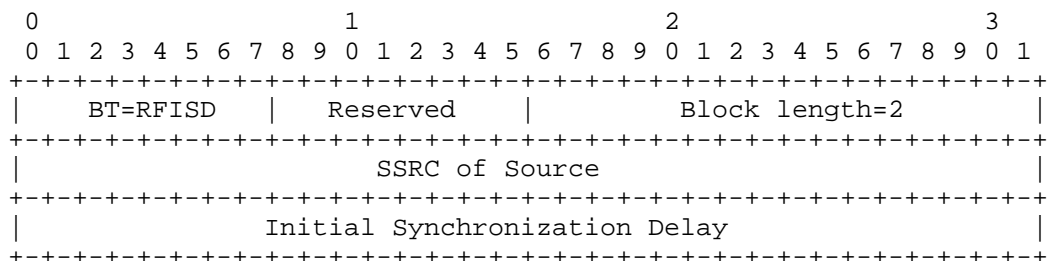
In an RTP multimedia session, there can be an arbitrary number of streams carried in different RTP sessions, with the same RTCP CNAME. These streams may be not synchronized with each other. For example, one audio stream and one video stream belong to the same session and audio stream is transmitted lag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

### 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 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 RTCP packets received on all of the components RTP sessions to the beginning of session [RFC6051]. The value 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 SHOULD be reported.

## 5. 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 time difference between two media streams with the same CNAME. In two media streams, one media stream is the reference stream, which 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:

```

      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=RFSO      |L|  Reserved  |               Block length=4      |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                               SSRC of source                          |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                               SSRC of reference                       |
+-----+-----+-----+-----+-----+-----+-----+-----+
|      Synchronization Offset, most significant word                    |
+-----+-----+-----+-----+-----+-----+-----+-----+
|      Synchronization Offset, least significant word                    |
+-----+-----+-----+-----+-----+-----+-----+-----+

```

## 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>.

Lag and Lead Indication (L): 1 bit

This field is used to indicate whether the reporting stream lead before the reference stream or lag behind the reference stream. The value of this field is set to 1, if the reporting stream lead before the reference stream, 0 otherwise.

Reserved: 7 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 RTP timestamp from the reporting RTP packet  $i$ , and  $R_i$  is the time of arrival in RTP timestamp units for reporting RTP packet  $i$ ,  $S_j$  is the RTP timestamp from the reference RTP packet  $j$ , and  $R_j$  is the time of arrival in RTP timestamp units for reference RTP packet  $j$ , then the value of the synchronization offset  $D$  may be expressed as

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

The value of the synchronization offset 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

[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.

### 6.1. 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]:

```
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"  
RTP-flow-syn-offset = "RTP-flows-syn-offset"
```

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

### 6.2. Offer/Answer Usage

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

## 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"

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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.

## 10. References

### 10.1. Normative References

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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-01

The following are the major changes compared to previous version:

Editorial change based on comments raised on the list.

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