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Considerations for Selecting RTCP Extended Report (XR) Metrics for the RTCWEB Statistics API

draft-huang-xrblock-rtcweb-rtcp-xr-metrics-04

Abstract

This document describes monitoring features related to RTCWEB. It provides a list of RTCP XR metrics, which are useful and may need to be supported in some RTCWEB implementations. It also describes a list of additional identifiers for WebRTC's statistics API. These identifiers are a set of RTCP XR metrics related to the transport of multimedia flows.

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1. Introduction

Web-based real-time communication (WebRTC) deployments are emerging and applications need to be able to estimate the service quality. If sufficient information (metrics or statistics) are provided to the applications, it can attempt to improve the media quality. [I-D.ietf-rtcweb-use-cases-and-requirements] specifies a requirement for statistics:

F38 The browser must be able to collect statistics, related to the transport of audio and video between peers, needed to estimate quality of experience.

The [I-D.alvestrand-rtcweb-stats-registry] describes a registration procedure for metrics reported by the WebRTC Stats API [W3C.WD-webrtc-20130910]. It currently lists basic metrics reported in the RTCP Sender and Receiver Report (SR/RR) [RFC3550] to fulfill this requirement. However, the basic metrics from RTCP SR/RR are not sufficient for precise quality monitoring, or diagnosing potential issues.

In this document, we provide some guidelines on choosing additional RTP metrics for the WebRTC Stats API [W3C.WD-webrtc-20130910]. Furthermore, we expose additional RTCP XR metrics to complement the identifiers that already exist in the statistics registry [I-D.alvestrand-rtcweb-stats-registry].

2. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

3. RTP Statistics in WebRTC Implementations

Currently, the statistics registry [I-D.alvestrand-rtcweb-stats-registry] exposes the basic RTCP SR and RR metrics for the local and remote media streams. The exposed identifiers are: SentPacketCount, SentOctetCount, packetsLost, Jitter, ReceivedPacketCount, ReceivedOctetCount. However, these metrics provides only partial or limited information, which may not be sufficient for diagnosing problems or quality monitoring. For example, it may be useful to distinguish between packets lost and packets discarded due to late arrival, even though they have the same impact on the multimedia quality, it helps in diagnosing and identifying issues.

RTP Control Protocol Extended Reports [RFC3611] and other extensions discussed in the XRBLOCK working group provide more detailed statistics, which complement the basic metrics reported in the RTCP Sender and Receiver Reports. Section Section 5 discusses the use of XR metrics that may be useful for monitoring the performance of WebRTC applications.

The WebRTC application extracts the statistic from the browser by querying the Stats API [W3C.WD-webrtc-20130910], but the browser currently only reports the local variables i.e., the statistics related to the outgoing RTP media streams and the incoming RTP media streams. Without the support of RTCP XRs or some other signaling mechianism, the WebRTC application cannot expose the remote endpoints' statistics. At the moment [I-D.ietf-rtcweb-rtp-usage] does not mandate the use of any RTCP XRs and since their usage is optional. If the use of RTCP XRs is successfully negotiated between endpoints (via SDP), thereafter the application has access to both local and remote statistics. Alternatively, once the WebRTC application gets the local information, they can report it to an application server or a third-party monitoring system, which provides quality estimations or diagnosis services for application developers. The exchange of statistics between endpoints or between a monitoring server and an endpoint is outside the scope of this document.

4. Considerations for Impact of Measurement Interval

RTCP extensions like RTCP XR usually share the same timing interval with the RTCP SR/RR, i.e., they are sent as compound packets, together with the RTCP SR/RR. Alternatively, if the RTCP XR uses a different measurement interval, all XRs using the same measurement interval are compounded together and the measurement interval is indicated in a specific measurement information block defined in [RFC6776].

When using WebRTC Statistics APIs (see section 7 of [W3C.WD-webrtc-20130910]), the applications can query this information at arbitrary intervals. For the statistics reported by the remote endpoint, e.g., those conveyed in an RTCP SR/RR/XR, these will not change until the next RTCP report is received. Some applications may choose 1 second or a different polling interval, but the statistics from the remote endpoint may not change when using intervals shorter than the average RTCP reporting interval. However, statistics generated by the local endpoint have no such restrictions as long as the endpoint is sending and receiving media.

5. Candidate Metrics

Since following metrics are all defined in RTCP XR which is not mandated in WebRTC, all of them are local. However, if RTCP XR is supported by negotiation between two browsers, following metrics can also be generated remotely and be sent to local by RTCP XR packets.

Following metrics are classified into 3 categories: network impact metrics, application impact metrics and recovery metrics. Network impact metrics are the statistics recording the information only for network transmission. They are useful for network problem diagnosis. Application impact metrics mainly collect the information in the viewpoint of application, e.g., bitrate, frames rate or jitter buffers. Recovery metrics reflect how well the repair mechanisms, e.g. loss concealment, retransmission or FEC, perform. All of the 3 types of metrics are useful for quality estimations of services in WebRTC implementations. WebRTC application can use these metrics to better calculate MoS values or Media Delivery Index (MDI) for their services.

5.1. Network Impact Metrics

5.1.1. Loss and Discard Packet Count Metric

In multimedia transport, packets which are received abnormally are classified into 3 types: lost, discarded and duplicate packets. Packet loss may be caused by network device breakdown, bit-error corruption or network congestion (packets dropped by an intermediate router queue). Duplicate packets may be a result of network delays, which causes the sender to retransmit the original packets. Discarded packets are packets that have been delayed long enough (perhaps they missed the playout time) and are considered useless by the receiver. Lost and discarded packets cause problems for multimedia services, as missing data and long delays can cause degradation in service quality, e.g., missing large blocks of contiguous packets (lost or discarded) may cause choppy audio, and long network transmission delay time may cause audio or video buffering. The RTCP SR/RR defines a metric for counting the total number of RTP data packets that have been lost since the beginning of reception. But this statistic does not distinguish lost packets from discarded and duplicate packets. Packets that arrive late will be discarded and are not reported as lost, and duplicate packets will be regarded as a normally received packet. Hence, the loss metric can be misleading if many duplicate packets are received or packets are discarded, which causes the quality of the media transport to appear okay from the statistic point of view, but meanwhile the users may actually be experiencing bad service quality. So in such cases, it is better to use more accurate metrics in addition to those defined in RTCP SR/RR.

The lost packets and duplicated packets metrics defined in Statistics Summary Report Block of [RFC3611] extend the information of loss carried in standard RTCP SR/RR. They explicitly give an account of lost and duplicated packets. Lost packets counts are useful for network problem diagnosis. It is better to use the loss packets metrics of [RFC3611] to indicate the packet lost count instead of the cumulative number of packets lost metric of [RFC3550]. Duplicated packets are usually rare and have little effect on QoS evaluation. So it may not be suitable for use in WebRTC.

Using loss metrics without considering discard metrics may result in inaccurate quality evaluation, as packet discard due to jitter is often more prevalent than packet loss in modern IP networks. The discarded metric specified in [RFC7002] counts the number of packets discarded due to the jitter. It augments the loss statistics metrics specified in standard RTCP SR/RR. For those RTCWEB services with jitter buffer requiring precise quality evaluation and accurate troubleshooting, this metric is useful as a complement to the metrics of RTCP SR/RR.

5.1.2. Burst/Gap Pattern Metrics for Loss and Discard

RTCP SR/RR defines coarse metrics regarding loss statistics, the metrics are all about per call statistics and are not detailed enough to capture some transitory nature of the impairments like bursty packet loss. Even if the average packet loss rate is low, the lost packets may occur during short dense periods, resulting in short periods of degraded quality. Distributed burst provides a higher subjective quality than a non-burst distribution for low packet loss rates whereas for high packet loss rates the converse is true. So capturing burst gap information is very helpful for quality evaluation and locating impairments. If the WebRTC application needs to evaluate the services quality, burst gap metrics provides more accurate information than RTCP SR/RR.

[RFC3611] introduces burst gap metrics in VoIP report block. These metrics record the density and duration of burst and gap periods, which are helpful in isolating network problems since bursts correspond to periods of time during which the packet loss/discard rate is high enough to produce noticeable degradation in audio or video quality. Burst gap related metrics are also introduced in [RFC7003] and [RFC6958] which define two new report blocks for usage in a range of RTP applications beyond those described in [RFC3611]. These metrics distinguish discarded packets from loss packets that occur in the bursts period and provides more information for diagnosing network problems. Additionally, the block reports the frequency of burst events which is useful information for evaluating the quality of experience. Hence, if WebRTC application need to do quality evaluation and observe when and why quality degrades, these metrics should be considered.

5.1.3. Run Length Encoded Metrics for Loss, Discard

Run-length encoding uses a bit vector to encode information about the packet. Each bit in the vector represents a packet and depending on the signaled metric it defines if the packet was lost, duplicated, discarded, or repaired. An endpoint typically uses the run length encoding to accurately communicate the status of each packet in the interval to the other endpoint. [RFC3611], [RFC7097] define run-length encoding for lost and duplicate packets, and discarded packets, respectively.

The WebRTC application could benefit from the additional information. If losses occur after discards, an endpoint may be able to correlate the two run length vectors to identify congestion-related losses, i.e., a router queue became overloaded causing delays and then overflowed. If the losses are independent, it may indicate bit-error corruption. For the WebRTC StatsAPI, these types of metrics are not recommended for use due to the large amount of data and the computation involved.

5.2. Application Impact Metrics

5.2.1. Discard Octets Metric

The metric reports the cumulative size of the packets discarded in the interval, it is complementary to number of discarded packets. An application measures sent octets and received octets to calculate sending rate and receiving rate, respectively. The application can calculate the actual bitrate in a particular interval by subtracting the discarded octets from the received octets.

For WebRTC, discarded octets supplements the sent and received octets and provides an accurate method for calculating the actual bitrate which is an important parameter to reflect the quality of the media. The discarded bytes metric is defined in [RFC7243].

5.2.2. Frame Impairment Summary Metrics

RTP has different framing mechanisms for different payload types. For audio streams, a single RTP packet may contain one or multiple audio frames, each of which has a fixed length. On the other hand, in video streams, a single video frame may be transmitted in multiple

RTP packets. The size of each packet is limited by the Maximum Transmission Unit (MTU) of the underlying network. However, statistics from standard SR/RR only collect information from transport layer, which may not fully reflect the quality observed by the application. Video is typically encoded using two frame types i.e., key frames and derived frames. Key frames are normally just spatially compressed, i.e., without prediction from other pictures. The derived frames are temporally compressed, i.e., depend on the key frame for decoding. Hence, Key frames are much larger in size than derived frames. The loss of these key frames results in a substantial reduction in video quality. Thus it is reasonable to consider this application layer information in WebRTC implementations, which influence sender strategies to mitigate the problem or require the accurate assessment of users' quality of experience.

The following metrics can also be considered for WebRTC's Statistics API: number of discarded key frames, number of lost key frames, number of discarded derived frames, number of lost derived frames. These metrics can be used to calculate Media Loss Rate (MLR) of MDI. Details of the definition of these metrics are described in [RFC7003]. Additionally, the metric provides the rendered frame rate, an important parameter for quality estimation.

5.2.3. Jitter Buffer Metrics

The size of the jitter buffer affects the end-to-end delay on the network and also the packet discard rate. When the buffer size is too small, slower packets are not played out and dropped, while when the buffer size is too large, packets are held longer than necessary and consequently reduce conversational quality. Measurement of jitter buffer should not be ignored in the evaluation of end user perception of conversational quality. Jitter buffer related metrics, such as maximum and nominal jitter buffer, could be used to show how the jitter buffer behaves at the receiving endpoint. They are useful for providing better end-user quality of experience (QoE) when jitter buffer factors are used as inputs to calculate MoS values. Thus for those cases, jitter buffer metrics should be considered. The definition of these metrics is provided in [RFC7005].

5.3. Recovery metrics

[Editor's Note: Concealment Metrics are currently not considered.]

5.3.1. Post-repair Packet Count Metrics

Error-resilience mechanisms, like RTP retransmission or FEC, are optional in RTCWEB because the overhead of the repair bits adding to the original streams. But they do help to greatly reduce the impact of packet loss and enhance the quality of transmission. Web applications could support certain repair mechanism after negotiation between both sides of browsers when needed. For these web applications using repair mechanisms, providing some statistic information for the performance of their repair mechanisms could help to have a more accurate quality evaluation.

The un-repaired packets count and repaired loss count defined in [I-D.ietf-xrblock-rtcp-xr-post-repair-loss-count] provide the recovery information of the error-resilience mechanisms to the monitoring application or the sending endpoint. The endpoint can use these metrics to ascertain the ratio of repaired packets to lost packets. Including this kind of metrics helps the application evaluate the effectiveness of the applied repair mechanisms.

5.3.2. Run Length Encoded Metric for Post-repair

[RFC5725] defines run-length encoding for post-repair packets. When using error-resilience mechanisms, the endpoint can correlate the loss run length with this metric to ascertain where the losses and repairs occurred in the interval. This provides more accurate information for recovery mechanisms evaluation than those in Section 5.3.1. However, it is not suggested to use due to their enormous amount of data when RTCP XR are supported.

For WebRTC, the application may benefit from the additional information. If losses occur after discards, an endpoint may be able to correlate the two run length vectors to identify congestion-related losses, i.e., a router queue became overloaded causing delays and then overflowed. If the losses are independent, it may indicate bit-error corruption. Lastly, when using error-resilience mechanisms, the endpoint can correlate the loss and post-repair run lengths to ascertain where the losses and repairs occurred in the interval. For example, consecutive losses are likely not to be repaired by a simple FEC scheme.

6. Candidate XR Block Metrics for WebRTC Statistics API

This document describes a list of additional identifiers to complement the identifiers in Section 4.1 of [I-D.alvestrand-rtcweb-stats-registry] and these group of identifiers are defined on a ReportGroup corresponding to an SSRC. In practice the application MUST be able to query the statistic identifiers on both an incoming (remote) and outgoing (local) media stream. Depending on the support of the corresponding XR report the endpoint MAY be able to query the reception statistics for its outgoing (local) media stream.

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

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6.1. Variables from XR Blocks

6.1.1. Packets and Octets Discarded

Name: PacketsDiscarded

Definition: Cumulative Number of RTP packets discarded due to late or early-arrival, Appendix A (a) of [RFC7002].

Name: OctetsDiscarded

Definition: Cumulative Number of octets discarded due to late or early-arrival, Appendix A of [RFC7243]

6.1.2. Cumulative Number of Packets Repaired

Name: PacketsRepaired

Definition: The cumulative number of lost RTP packets repaired after applying a error-resilience mechanism, Appendix A (b) of [I-D.ietf-xrblock-rtcp-xr-post-repair-loss-count]. To clarify, the value is upper bound to the cumulative number of lost packets.

6.1.3. Burst Packet Loss or Discarded

Name: BurstPacketDiscarded

Definition: The total number of RTP packets discarded during discard bursts, Appendix A (b) of [RFC7003].

Name: BurstPacketLost

Definition: The total number of RTP packets lost during loss bursts, Appendix A (c) of [RFC6958].

Name: BurstCount

Definition: The cumulative number of bursts of lost RTP packets, Appendix A (e) of [RFC6958].

[RFC3611] recommends a Gmin value of 16.

6.1.4. Frame Impairment Metrics

Name: FullFramesLostCount

Definition: Number of full frames lost, Appendix A (i) of [RFC7004]

Name: PartialFramesLostCount

Definition: Number of frames partially lost, Appendix A (j) of [RFC7004]

Name: FramesDiscardedCount

Definition: Number of full frames discarded, Appendix A (g) of [RFC7004]

7. IANA Considerations

This document requests IANA to update the registry described in [I-D.alvestrand-rtcweb-stats-registry] with the identifiers defined in Section 6.

8. Security Considerations

The monitoring activities are implemented between two browsers or between a browser and a server. Therefore encryption procedures, such as the ones suggested for a Secure RTCP (SRTCP), need to be used. Currently, the monitoring in RTCWEB introduces no new security considerations beyond those described in [I-D.ietf-rtcweb-rtp-usage], [I-D.ietf-rtcweb-security], and [I-D.alvestrand-rtcweb-stats-registry].

9. Acknowledgements

The authors would like to thank Bernard Aboba , Al Morton , Colin Perkins , and Shida Schubert , for their valuable comments and suggestions on earlier version of this document.

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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. changes in draft-huang-xrblock-rtcweb-rtcp-xr-metrics-04

- o Addressed comments from the London IETF meeting:

- o Removed ECN metrics.

- o Merged draft-singh-xrblock-webrtc-additional-stats-01

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RTP Control Protocol (RTCP) Extended Report (XR) for Post-Repair Loss Count Metrics

draft-ietf-xrblock-rtcp-xr-post-repair-loss-count-11

Abstract

This document defines an RTP Control Protocol (RTCP) Extended Report (XR) Block that allows reporting of post-repair loss count metric for a range of RTP applications. In addition, another metric, repaired loss count, is also introduced in this report block for calculating the pre-repair loss count when needed so that the RTP sender or a third-party entity is able to evaluate the effectiveness of the repair methods used by the system.

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1 Introduction

RTCP Sender Reports (SR)/Receiver Reports (RR) [RFC3550] contain some rough statistics about the data received from the particular source indicated in that block. One of them is the cumulative number of packets lost, which is called pre-repair loss metric in this document. This metric conveys information regarding the total number of RTP data packets that have been lost since the beginning of the RTP session.

However, this metric is measured on media stream before any loss repair mechanism, e.g., retransmission [RFC4588] or Forward Error Correction (FEC) [RFC5109], is applied. Using a repair mechanism usually results in recovering some or all of the lost packets. Hence, the sending endpoint cannot assess the performance of the repair mechanism by observing the change in fraction loss and the cumulative loss statistics from RTCP SR/RR [RFC3550].

Consequently, [RFC5725] specifies a post-repair loss Run-length Encoding (RLE) XR report block to address this issue. The sending endpoint is able to infer which packets were repaired from the RLE report block, but the reporting overhead for the packet-by-packet report block is higher compared to other report blocks.

When applications use multiple XR blocks, the endpoints may require more concise reporting to save bandwidth. This document defines a new XR block type to augment those defined in [RFC3611] and complement the report block defined in [RFC5725] for use in a range of RTP applications. This new block type reports the post-repair loss count metric which records the number of primary source RTP packets that are still lost after applying one or more loss repair mechanisms. In addition, another metric, repaired loss count, is also introduced in this report block for calculating the pre-repair loss count during this range, so that the RTP sender or a third-party entity is able to evaluate the effectiveness of the repair methods used by the system. The metrics defined in this document are packet level rather than slice/picture level, which means the partial recovery of a packet will not be regarded as a repaired packet.

The metrics defined in this document belong to the class of transport-related metrics defined in [RFC6792] and are specified in accordance with the guidelines in [RFC6390] and [RFC6792]. These metrics are applicable to any RTP application, especially those that use loss repair mechanisms.

2 Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT",

"SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [KEYWORDS].

primary source RTP packet: The original RTP packet sent from the RTP sender for the first time. A lost primary source RTP packet may be repaired by some other RTP packets used in repair mechanisms like FEC or retransmission.

3 Post-Repair Loss Count Metrics Report Block

This block reports the number of packets lost after applying repair mechanisms (e.g., FEC). It complements the RTCP XR metrics defined in [RFC5725]. As noted in [RFC5725], ambiguity may occur when comparing this metric with pre-repair loss metric reported in an RTCP SR/RR, i.e., some packets were not repaired in the current RTCP interval, but they may be repaired later. Therefore, this block uses a begin sequence number and an end sequence number to explicitly indicate the actual sequence number range reported by this RTCP XR. Accordingly, only packets that have no further chance of being repaired and that have been repaired are included in this report block.

3.1 Report Block Structure

The post-repair loss count metrics report block has the following format:

```
 0               1               2               3               4
 0 1 2 3 4 5 6 7 0 1 2 3 4 5 6 7 0 1 2 3 4 5 6 7 0 1 2 3 4 5 6 7 0
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |    BT=PRLR    |   Reserved    |       block length = 4        |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |                        SSRC of Source                         |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |       begin_seq               |           end_seq             |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
 |  post-repair loss count       |     repaired loss count       |
 +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Block Type (BT): 8 bits

A Post-Repair Loss Count Metrics Report Block is identified by the constant PRLR.

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

Reserved: 8 bits

These bits are reserved for future use. They MUST be set to zero by senders and ignored by receivers (see [RFC6709], Section 4.2).

block length: 16 bits

This field is in accordance with the definition in [RFC3611]. In this report block, it MUST be set to 4. The block MUST be discarded if the block length is set to a different value.

SSRC of source: 32 bits

As defined in Section 4.1 of [RFC3611].

begin_seq: 16 bits

The first sequence number that this block reports on. It can remain fixed when calculating metrics over several RTCP reporting intervals.

end_seq: 16 bits

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

post-repair loss count: 16 bits

Total number of packets finally lost after applying one or more loss-repair methods, e.g., FEC and/or retransmission, during the actual sequence number range indicated by begin_seq and end_seq. This metric MUST NOT count the lost packets for which repair might still be possible. Note that this metric MUST measure only primary source RTP packets.

repaired loss count: 16 bits

Total number of packets fully repaired after applying one or more loss-repair methods, e.g., FEC and/or retransmission, during the actual sequence number range indicated by begin_seq and end_seq. Note that this metric MUST measure only primary source RTP packets.

3.2 Example Usage

The metrics defined in this report block are all measured at the RTP receiver. However, the receiving endpoint can report the metrics in two different ways:

1. Cumulative report

In this case, implementations may set begin_seq to the first packet in the RTP session and it will remain fixed across all reports. Hence, the "post-repair loss count" and "repaired loss count", respectively will correspond to "cumulative post-repair loss count" and "cumulative repaired loss count" in this case. These cumulative metrics when combined with the cumulative loss metrics reported in an RTCP RR (pre-repair) assists in calculating the still to be repaired lost packets:

still to be repaired lost packet = cumulative number of packets lost - cumulative post-repair loss count - cumulative repaired loss count

2. Interval report

Some implementations may align the begin_seq and end_seq number with the highest sequence numbers of consecutive RTCP RRs (RTCP interval). This is NOT RECOMMENDED as packets that are not yet repaired in this current RTCP interval and may be repaired in the subsequent intervals will not be reported. It is illustrated in the following example:

Interval A: The extended highest sequence number received in RTCP RR is 20. Begin_seq is 10 and end_seq is 20.

Interval B: The extended highest sequence number received in RTCP RR is 30. Begin_seq is 20 and end_seq is 30.

If packets 17 and 19 are lost and not yet repaired in interval A and subsequently repaired in interval B, they will not then be reported because their sequence numbers do not belong in the interval B. Therefore, if implementations want these packets to be reported as repaired, they MUST NOT align the begin_seq and end_seq to the RTCP intervals.

Alternatively, implementations may choose the begin_seq and end_seq numbers that cover several RTCP intervals. Additionally, the reported range of sequence numbers may overlap with the previous report blocks, so that the packets that were not yet repaired in one interval but were subsequently repaired or deemed unrepairable were reported in subsequent intervals.

In this case, the "cumulative number of packets lost" cannot be easily compared with the post-repair metrics. However, the sending endpoint can calculate the efficiency of the error resilience algorithm using the post-repair and repaired lost count, respectively.

4 SDP Signaling

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

4.1 SDP rtcp-xr-attrib Attribute Extension

This session augments the SDP attribute "rtcp-xr" defined in Section 5.1 of [RFC3611] by providing an additional value of "xr-format" to signal the use of the report block defined in this document.

```
xr-format =/ xr-prlr-block

xr-prlr-block = "post-repair-loss-count"
```

4.2 Offer/Answer Usage

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

5 Security Considerations

This proposed RTCP XR 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.

An attacker may put incorrect information in the Post-Repair Loss Count reports, which will be affect the performance of loss repair mechanisms. Implementers should consider the guidance in [RFC7202] for using appropriate security mechanisms, i.e., where security is a concern, the implementation should apply encryption and authentication to the report block. For example, this can be achieved by using the AVPF profile together with the Secure RTP profile as defined in [RFC3711]; an appropriate combination of the two profiles (an "SAVPF") is specified in [RFC5124]. However, other mechanisms also exist (documented in [RFC7201]) and might be more suitable.

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 PRLR in the IANA "RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry" to the "Post-Repair Loss Count Metrics Report Block".

[Note to RFC Editor: please replace PRLR 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 "post-repair-loss-count" in the "RTP Control Protocol Extended Reports (RTCP XR) Session Description Protocol (SDP) Parameters Registry".

6.3 Contact Information for registrations

The contact information for the registration is :

RAI Area Directors <rai-ads@tools.ietf.org>

7 Acknowledgments

The author would like to thank Roni Even, Colin Perkins, and Qin Wu for giving valuable comments and suggestions.

8 References

8.1 Normative References

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Appendix A. Metrics Represented Using the Template from RFC 6390

a. Post-Repair RTP Packet Loss Count Metric

* Metric Name: Post-Repair RTP Packet Loss Count Metric.

* Metric Description: Total number of RTP packets still lost after loss repair methods are applied

* Method of Measurement or Calculation: See section 3.1, Post-Repair RTP Packet Loss Count Metric definition. It is directly measured and must be measured for the primary source RTP packets with no further chance of repair.

* Units of Measurement: This metric is expressed as a 16-bit unsigned integer value giving the number of RTP packets.

* Measurement Point(s) with Potential Measurement Domain: It is measured at the receiving end of the RTP stream.

* Measurement Timing: This metric relies on the sequence number interval to determine measurement timing. See Section 3, 3rd paragraph, for details.

* Use and Applications: These metrics are applicable to any RTP application, especially those that use loss repair mechanisms. See Section 1 for details.

* Reporting Model: See RFC3611.

b. Repaired RTP Packet Loss Count Metric

* Metric Name: Repaired RTP Packet Count Metric.

* Metric Description: The number of RTP packets lost but repaired after applying loss repair methods

* Method of Measurement or Calculation: See section 3.1, Repaired RTP Packet Loss Count Metric definition. It is directly measured and must be measured for the primary source RTP packets with no further chance of repair.

* Units of Measurement: This metric is expressed as a 16-bit unsigned integer value giving the number of RTP packets.

* Measurement Point(s) with Potential Measurement Domain: It is measured at the receiving end of the RTP stream.

* Measurement Timing: This metric relies on the sequence number interval to determine measurement timing. See Section 3, 3rd paragraph, for details.

* Use and Applications: These metrics are applicable to any RTP application, especially those that use loss repair mechanisms. See Section 1 for details.

* Reporting Model: See RFC3611.

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RTP Control Protocol (RTCP) Extended Report (XR) Block for MPEG2 Transport Stream (TS) Program Specific Information (PSI) Decodability Statistics Metrics reporting

draft-ietf-xrblock-rtcp-xr-psi-decodability-07

Abstract

An MPEG2 Transport Stream (TS) is a standard container format used in the transmission and storage of multimedia data. Unicast/Multicast MPEG2 TS over RTP is widely deployed in IPTV systems. This document defines an RTP Control Protocol (RTCP) Extended Report (XR) Block that allows the reporting of MPEG2 TS decodability statistics metrics related to transmissions of MPEG2 TS over RTP. The metrics specified in the RTCP XR Block are related to Program Specific Information carried in MPEG TS.

Status of This Memo

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Table of Contents

1. Introduction

1.1. MPEG2 Transport Stream Decodability Metrics

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

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

This memo defines a new block type for use with MPEG2 Transport Stream (TS) [ISO-IEC.13818-1.2007], to augment those defined in [RFC3611]. The new block type supports reporting of the number of occurrences of each Program Specific Information (PSI) indicator in the first and second priorities listed by [ETSI] sections 5.2.1 and 5.2.2 respectively. Third priority indicators are not supported. The metrics defined here supplement information from the PSI-independent Decodability Statistics Metrics Block [RFC6990].

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 guidelines for RTCP XR reporting block formats. The new report block described in this memo is in compliance with the monitoring architecture specified in [RFC6792] and the Performance Metrics Framework [RFC6390].

1.4. Applicability

These metrics are applicable to any type of RTP application that uses the MPEG2 TS standard format for multimedia data, for example, MPEG4 over MPEG2 TS over RTP. 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 problems 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 tool.

2. Terminology

2.1. Standards Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119].

3. MPEG2 TS PSI Decodability Statistics Metrics Block

ETSI TR 101290 [ETSI] generally defines indicators related to error events, while the XR block defined in this document contains counts of occurrences of the [ETSI] indicators. The block defined in this document reports MPEG2 TS PSI decodability statistics metrics beyond the information carried in the standard RTCP packet format and PSI-independent Decodability Metrics Block [RFC6990], which are measured at the receiving end of the RTP stream. It contains counts of seven metrics defined in ETSI TR 101290 [ETSI]. Information is reported about basic monitoring parameters necessary to ensure that the TS can be decoded including:

- Program Association Table (PAT) errors
- PAT 2 errors
- Program Map Table (PMT) errors
- PMT 2 errors
- Packet Identifier (PID) errors

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

- Cyclic Redundancy Check (CRC) errors
- Conditional Access Table (CAT) errors

In these parameters, PAT 2 errors and PMT 2 errors are actually replacements for and improvements on PAT errors and PMT errors respectively and are therefore preferred in future implementations. In addition, measurement results for some of these parameters (e.g., PAT errors or PMT errors) may be different based on whether scrambling is employed. The other parameters defined in [ETSI] Section 5 are ignored since they do not apply to all MPEG2 implementations. For further detailed information on these parameters, see [ETSI].

The MPEG2 TS PSI 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=MTPD    |   Reserved    |         block length          |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      SSRC of source                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|          begin_seq            |             end_seq           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|        PAT_error_count        |        PAT_error_2_count      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|        PMT_error_count        |        PMT_error_2_count      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|       PID_error_count         |       CRC_error_count         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|        CAT_error_count        |         Reserved              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

block type (BT): 8 bits

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

Reserved: 8 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 constant 6, in accordance with the definition of this field in Section 3 of RFC 3611. The block MUST be discarded if the block length is set to a different value.

SSRC of source: 32 bits

As defined in Section 4.1 of RFC 3611.

begin_seq: 16 bits

As defined in Section 4.1 of RFC 3611.

end_seq: 16 bits

As defined in Section 4.1 of RFC 3611.

PAT_error_count: 16 bits

A count of the number of PAT errors that occurred in the above sequence number interval. The program association table (PAT) is the only packet with packet ID (PID) 0x 0000. A PAT error occurs when: (1) a packet with PID 0x0000 does not occur at least every 0.5 seconds, or (2) a packet with PID 0x0000 does not contain a table_id 0x00 (i.e., a PAT), or (3)Scrambling_control_field in the TS packet header is not 00 for a packet with PID 0x0000. See section 5.2.1 of [ETSI]. Every program within the MPEG TS stream is listed in the PAT; if it is missing, then no programs can be decoded.

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported. As indicated in the NOTE 1 of the table in the section 5.2.1 of TR101.290, TR 101.290 recommends using PAT_error_2_count. Upon reception, If PAT_error_2_count is available (that is, other than 0xFFFF), then receivers MUST ignore PAT_error_count.

PAT_error_2_count: 16 bits

A count of the number of PAT2 errors that occurred in the above sequence number interval. A PAT2 error occurs when: (1) a packet with PID 0x0000 containing table_id 0x00 does not occur at least every 0.5 seconds,or (2) a packet with PID 0x0000 contains a table with table_id other than 0x00, or (3) Scrambling_control_field in the TS packet header is not 00 for a packet with PID 0x0000. See section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported.

PMT_error_count: 16 bits

A count of the number of PMT_errors that occurred in the above sequence number interval. A PMT_error occurs when: (1) a packet containing a table with table_id 0x02(i.e.,a PMT) does not occur at least every 0.5s on the PID that is referred to in the PAT, or(2) Scrambling_control_field in the TS packet header is not 00 for all packets with PID containing a table with table_id 0x02 (i.e. a PMT). See the section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported. As indicated in the NOTE 2 of table in the section 5.2.1 of TR101.290, TR 101.290 recommends using PMT_error_2_count. Upon reception, If PMT_error_2_count is available (that is, other than 0xFFFF), then receivers MUST ignore PMT_error_count.

PMT_error_2_count: 16 bits

A count of the number of PMT2 errors that occurred in the above sequence number interval. A PMT2_error occurs when: (1) a packet containing table_id 0x02(i.e.,a PMT) does not occur at least every 0.5s on each program_map_PID which is referred to in the PAT, or (2) Scrambling_control_field in the TS packet header is not 00 for all packets containing a table with table_id 0x02 (i.e. a PMT) on each program_map_PID which is referred to in the PAT. See section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported.

PID_error_count: 16 bits

A count of the number of PID_errors that occurred in the above sequence number interval. A PID error occurs when no data stream is present corresponding to a given PID. This may be caused by multiplexing or demultiplexing, then remultiplexing. See section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported.

CRC_error_count: 16 bits

A count of the number of CRC_errors that occurred in the above sequence number interval. A CRC_error occurs if data corruption occurred in any of the following tables -- CAT, PAT, PMT, Network Information Table (NIT), Event Information Table (EIT), Bouquet Association Table (BAT), Service Description Table (SDT) or Time Offset Table (TOT), as defined in the section 5.2.2 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported.

CAT_error_count: 16 bits

A count of the number of CAT_errors that occurred in the above sequence number interval. A CAT_error occurs when: (1) a packet with PID 0x0001 contains a table with table_id other than 0x01(i.e.,not a CAT), or (2) A packet does not contain a table with table_id = 0x01 (i.e. a CAT) when scrambling is employed ((i.e., scrambling_control field is set as a value other than 00)). See the section 5.2.2 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported.

Reserved: 16 bits

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

4. SDP Signaling

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

4.1. SDP rtcp-xr-attrib Attribute Extension

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

```
xr-format =/  xr-tpd-block

xr-tpd-block = "ts-psi-decodability"
```

4.2. Offer/Answer Usage

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

4.3. Usage Outside of Offer/Answer

For usage outside of Offer/Answer,refer to section 5.3 of [RFC3611].

5. IANA Considerations

New report block types for RTCP XR are subject to IANA registration. For general guidelines on IANA allocations for RTCP XR, refer to Section 6.2 of RFC 3611.

5.1. New RTCP XR Block Type value

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

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

5.2. New RTCP XR SDP Parameter

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

5.3. Contact information for registrations

The contact information for the registrations is:

RAI Area Directors <rai-ads@tools.ietf.org>

6. Security Considerations

This proposed RTCP XR report block introduces no new security considerations beyond those described in [RFC3611] [RFC6990].

7. References

7.1. Normative References

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7.2. Informative References

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[RFC6990] Wu, Q., "RTP Control Protocol (RTCP) Extended Report (XR) Block for MPEG2 Transport Stream (TS) Program Specific Information (PSI) Independent Decodability Statistics Metrics reporting", RFC 6990, May 2013.

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Additional RTP Control Protocol (RTCP) Extended Report (XR) Metrics for WebRTC Statistics API

draft-singh-xrblock-webrtc-additional-stats-02

Abstract

This document describes a list of additional identifiers used in WebRTC's Javascript statistics API. These identifiers are a set of RTCP XR metrics related to the transport of multimedia flows.

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

Web-based real-time communication (WebRTC) deployments are emerging and applications need to be able to estimate the service quality. If sufficient information (metrics or statistics) are provided to the applications, it can attempt to improve the media quality. [I-D.ietf-rtcweb-use-cases-and-requirements] specifies a requirement for statistics:

F38 The browser must be able to collect statistics, related to the transport of audio and video between peers, needed to estimate quality of experience.

The [I-D.alvestrand-rtcweb-stats-registry] describes a registration procedure for metrics reported by the Javascript API. It currently lists basic metrics reported in the RTCP Sender and Receiver Report (SR/RR) to fulfill this requirement. However, the basic metrics from

RTCP SR/RR are not sufficient for precise quality monitoring or troubleshooting. This document proposes to expose the RTCP XR metrics to complement the identifiers already in the statistics registry [I-D.alvestrand-rtcweb-stats-registry]. In depth discussion about the XR metrics candidates is carried out in [I-D.huang-xrblock-rtcweb-rtcp-xr-metrics].

The WebRTC application has two options to extract the statistics: 1) the browser monitors the local (outgoing) and remote (incoming) RTP stream and exposes metrics to the application via the Stats API [W3C.WD-webrtc-20130910], or 2) the browser measures the remote (incoming) RTP stream and exposes the metrics to the other participant by sending the appropriate RTCP XR. At the moment [I-D.ietf-rtcweb-rtp-usage] does not specify the use of any RTCP XRs and since their usage is optional, the exchange of statistics between participants or a monitoring server is outside the scope of this document.

2. Candidate XR Block Metrics for WebRTC Statistics API

This document describes a list of additional identifiers to complement the identifiers in Section 4.1 of [I-D.alvestrand-rtcweb-stats-registry] and these group of identifiers are defined on a ReportGroup corresponding to an SSRC. In practice the application MUST be able to query the statistic identifiers on both an incoming (remote) and outgoing (local) media stream. Depending on the support of the corresponding XR report the endpoint MAY be able to query the reception statistics for its outgoing (local) media stream.

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

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2.1. Variables from XR Blocks

2.1.1. Packets and Octets Discarded

Name: PacketsDiscarded

Definition: Cumulative Number of RTP packets discarded due to late or early-arrival, Appendix A (a) of [RFC7002].

Name: OctetsDiscarded

Definition: Cumulative Number of octets discarded due to late or early-arrival, Appendix A of [I-D.ietf-xrblock-rtcp-xr-bytes-discarded-metric]

2.1.2. Cumulative Number of Packets Repaired

Name: PacketsRepaired

Definition: The cumulative number of lost RTP packets repaired after applying a error-resilience mechanism, Appendix A (b) of [I-D.huang-xrblock-post-repair-loss-count]. To clarify, the value is upper bound to the cumulative number of lost packets.

2.1.3. Burst Packet Loss or Discarded

Name: BurstPacketDiscarded

Definition: The total number of RTP packets discarded during discard bursts, Appendix A (b) of [RFC7003].

Name: BurstPacketLost

Definition: The total number of RTP packets lost during loss bursts, Appendix A (c) of [RFC6958].

Name: BurstCount

Definition: The cumulative number of bursts of lost RTP packets, Appendix A (e) of [RFC6958].

[RFC3611] recommends a Gmin value of 16.

2.1.4. Frame Impairment Metrics

Name: FullFramesLostCount

Definition: Number of full frames lost, Appendix A (i) of [RFC7004]

Name: PartialFramesLostCount

Definition: Number of frames partially lost, Appendix A (j) of [RFC7004]

Name: FramesDiscardedCount

Definition: Number of full frames discarded, Appendix A (g) of [RFC7004]

3. IANA Considerations

This document requests IANA to update the registry described in [I-D.alvestrand-rtcweb-stats-registry] with the identifiers defined in Section 2.

4. Security Considerations

The security considerations of [I-D.alvestrand-rtcweb-stats-registry], apply.

5. Acknowledgements

This document is a product of discussion in XRBLOCK WG, initial motivation for this documented is discussed in [I-D.huang-xrblock-rtcweb-rtcp-xr-metrics]

The authors would like to thank Al Morton, Colin Perkins, Dan Romascanu, and Shida Schubert, for their valuable comments and suggestions on earlier version of this document.

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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. changes in draft-singh-xrblock-webrtc-additional-stats-01

- Addressed comments from the Vancouver IETF meeting.
- Added Burst discard/loss metric.
- Removed retransmission metric.

A.2. changes in draft-singh-xrblock-webrtc-additional-stats-00

- Clarified measurement points for remote (incoming) media stream and local (outgoing) media stream.
- Added this section.

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