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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-bi-xrblock-rtcp-xr-psi-dep-decodability-03

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.

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1. Introduction

1.1. MPEG2 Transport Stream Decodability Metrics

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

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

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

1.2. RTCP and RTCP XR Reports

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

1.3. Performance Metrics Framework

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

1.4. Applicability

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

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

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

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

The other parameters are ignored since they do not apply to all MPEG2 implementations. For further 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	2	3	4	5	6	7	8	9
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                                     |
+-----+

```

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

SSRC of source: 32 bits

As defined in Section 4.1 of RFC 3611.

begin_seq: 16 bits

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: 32 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 it does not occur at least every 0.5s or the table has an ID other than 0x 00, as defined in 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. Note that

PAT_error_count and PAT_error_2_count MUST NOT be reported at the same time in the same metric block. If PAT_error_count is reported, PAT_error_2_count MUST be set to 0xFFFF.

PAT_error_2_count: 32 bits

A count of the number of PAT2 errors that occurred in the above sequence number interval. A PAT2 error occurs when it does not occur at least every 0.5s or the table has an ID other than 0x 00 or there is more than one table ID 0x 00 inside the packet with the PAT PID, as defined in section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported. Note that PAT_error_count and PAT_error_2_count MUST NOT be reported at the same time in the same metric block. If PAT_error_2_count is reported, PAT_error_count MUST be set to 0xFFFF.

PMT_error_count: 32 bits

A count of the number of PMT_errors that occurred in the above sequence number interval. A PMT_error occurs when the program map table (PMT) does not come up at least every 0.5s on the PID that is referred to in the PAT, as defined in the section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported. Note that PMT_error_count and PMT_error_2_count MUST NOT be reported at the same time in the same metric block. If PMT_error_count is reported, PMT_error_2_count MUST be set to 0xFFFF.

PMT_error_2_count: 32 bits

A count of the number of PMT2 errors that occurred in the above sequence number interval. A PMT2_error occurs when the program map table (PMT) does not come up at least every 0.5s on the PID that is referred to in the PAT, as defined in the section 5.2.1 of [ETSI].

The measured value is unsigned value. If the measurement is unavailable, the value 0xFFFF MUST be reported. Note that PMT_error_count and PMT_error_2_count MUST NOT be reported at the same time in the same metric block. If PMT_error_2_count is reported, PMT_error_count MUST be set to 0xFFFF.

PID_error_count: 32 bits

A count of the number of PID_errors that occurred in the above sequence number interval. A PID_error occurs when MPEG TS streams are remultiplexed and any PID doesn't refer to an actual data stream, as defined in the section 5.2.2 of [ETSI].

CRC_error_count: 32 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 -\u002D 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].

CAT_error_count: 32 bits

A count of the number of CAT_errors that occurred in the above sequence number interval. A CAT_error occurs when the table has an ID other than 0x 01, as defined in the section 5.2.2 of [ETSI].

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-attr 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-tpdd-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].

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:

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6. Security Considerations

This proposed RTCP XR report block introduces no new security considerations beyond those described in RFC 3611.

7. References

7.1. Normative References

- [ETSI] ETSI, "Digital Video Broadcasting (DVB); Measurement guidelines for DVB systems", Technical Report TR 101 290, 2001.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- [RFC3550] Schulzrinne, H., "RTP: A Transport Protocol for Real-Time Applications", RFC 3550, July 2003.

- [RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 2003.
- [RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", RFC 4566, July 2006.

7.2. Informative References

- [ISO-IEC.13818-1.2007]
International Organization for Standardization,
"Information technology - Generic coding of moving
pictures and associated audio information: Systems", ISO
International Standard 13818-1, October 2007.
- [PIDCB] Wu, Q., "RTP Control Protocol (RTCP) Extended Report (XR)
Block for MPEG2 Transport Stream (TS) Program Specific
Information (PSI) Independent Decodability Statistics
Metrics reporting", ID ietf-xrblock-rtcp-xr-
decodability-12, May 2013.
- [RFC6390] Clark, A. and B. Claise, "Guidelines for Considering New
Performance Metric Development", BCP 170, RFC 6390,
October 2011.
- [RFC6709] Carpenter, B., Aboba, B., and S. Cheshire, "Design
Considerations for Protocol Extensions", RFC 6709,
September 2012.
- [RFC6792] Wu, Q., Hunt, G., and P. Arden, "Guidelines for Use of the
RTP Monitoring Framework", RFC 6792, November 2012.

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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 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 mechanism, 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.

5.1.1.2. 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-usagel], [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.

10. References

10.1. Normative References

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- [I-D.alvestrand-rtcweb-stats-registry]
Alvestrand, H., "A Registry for WebRTC statistics identifiers", draft-alvestrand-rtcweb-stats-registry-00 (work in progress), September 2012.
- [RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.
- [RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 2003.
- [RFC4588] Rey, J., Leon, D., Miyazaki, A., Varsa, V., and R. Hakenberg, "RTP Retransmission Payload Format", RFC 4588, July 2006.
- [RFC5725] Begen, A., Hsu, D., and M. Lague, "Post-Repair Loss RLE Report Block Type for RTP Control Protocol (RTCP) Extended Reports (XRs)", RFC 5725, February 2010.
- [RFC6776] Clark, A. and Q. Wu, "Measurement Identity and Information Reporting Using a Source Description (SDS) Item and an RTCP Extended Report (XR) Block", RFC 6776, October 2012.
- [RFC6958] Clark, A., Zhang, S., Zhao, J., and Q. Wu, "RTP Control Protocol (RTCP) Extended Report (XR) Block for Burst/Gap Loss Metric Reporting", RFC 6958, May 2013.
- [RFC7002] Clark, A., Zorn, G., and Q. Wu, "RTP Control Protocol (RTCP) Extended Report (XR) Block for Discard Count Metric Reporting", RFC 7002, September 2013.
- [RFC7003] Clark, A., Huang, R., and Q. Wu, "RTP Control Protocol (RTCP) Extended Report (XR) Block for Burst/Gap Discard Metric Reporting", RFC 7003, September 2013.
- [RFC7004] Zorn, G., Schott, R., Wu, Q., and R. Huang, "RTP Control Protocol (RTCP) Extended Report (XR) Blocks for Summary Statistics Metrics Reporting", RFC 7004, September 2013.

- [RFC7005] Clark, A., Singh, V., and Q. Wu, "RTP Control Protocol (RTCP) Extended Report (XR) Block for De-Jitter Buffer Metric Reporting", RFC 7005, September 2013.
- [RFC7097] Ott, J., Singh, V., and I. Curcio, "RTP Control Protocol (RTCP) Extended Report (XR) for RLE of Discarded Packets", RFC 7097, January 2014.
- [RFC7243] Singh, V., Ott, J., and I. Curcio, "RTP Control Protocol (RTCP) Extended Report (XR) Block for the Bytes Discarded Metric", RFC 7243, May 2014.
- [I-D.ietf-xrblock-rtcp-xr-post-repair-loss-count]
Huang, R. and V. Singh, "RTP Control Protocol (RTCP) Extended Report (XR) for Post-Repair Loss Count Metrics", draft-ietf-xrblock-rtcp-xr-post-repair-loss-count-05 (work in progress), June 2014.

10.2. Informative References

- [I-D.ietf-rtcweb-use-cases-and-requirements]
Holmberg, C., Hakansson, S., and G. Eriksson, "Web Real-Time Communication Use-cases and Requirements", draft-ietf-rtcweb-use-cases-and-requirements-14 (work in progress), February 2014.
- [W3C.WD-webrtc-20130910]
Bergkvist, A., Burnett, D., Jennings, C., and A. Narayanan, "WebRTC 1.0: Real-time Communication Between Browsers", World Wide Web Consortium WD WD-webrtc-20130910, September 2013, <<http://www.w3.org/TR/2013/WD-webrtc-20130910>>.
- [I-D.ietf-rtcweb-rtp-usage]
Perkins, C., Westerlund, M., and J. Ott, "Web Real-Time Communication (WebRTC): Media Transport and Use of RTP", draft-ietf-rtcweb-rtp-usage-15 (work in progress), May 2014.
- [I-D.ietf-rtcweb-security]
Rescorla, E., "Security Considerations for WebRTC", draft-ietf-rtcweb-security-06 (work in progress), January 2014.

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 Run Length Encoding
(RLE) of Discarded Packets
draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-09

Abstract

The RTP Control Protocol (RTCP) is used in conjunction with the Real-time Transport Protocol (RTP) in to provide a variety of short-term and long-term reception statistics. The available reporting may include aggregate information across longer periods of time as well as individual packet reporting. This document specifies a per-packet report metric capturing individual packets discarded from the de-jitter buffer after successful reception.

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1. Introduction

RTP [RFC3550] provides a transport for real-time media flows such as audio and video together with the RTP control protocol (RTCP) which provides periodic feedback about the media streams received in a specific duration. In addition, RTCP can be used for timely feedback about individual events to report (e.g., packet loss) [RFC4585]. Both long-term and short-term feedback enable a media sender to adapt its media transmission and/or encoding dynamically to the observed path characteristics.

RFC3611 [RFC3611] defines RTCP Extended Reports as a detailed reporting framework to provide more than just the coarse Receiver Report (RR) statistics. The detailed reporting may enable a media sender to react more appropriately to the observed networking conditions as these can be characterized better, although at the expense of extra overhead.

Among many other report blocks, RFC3611 specifies the Loss Run Length Encoding (RLE) block which reports runs of packets received and lost with the granularity of individual packets. This can help both error recovery and path loss characterization. In addition to lost packets, RFC3611 defines the notion of "discarded" packets: packets that were received but dropped from the de-jitter buffer because they were either too early (for buffering) or too late (for playout). The "discard rate" metric is part of the VoIP metrics report block even though it is not just applicable to audio: it is specified as the fraction of discarded packets since the beginning of the session. See section 4.7.1 of RFC3611 [RFC3611]. The discard metric is believed to be applicable to a large class of RTP applications which use a de-jitter buffer RFC5481 [RFC5481].

Recently proposed extensions to the Extended Reports (XR) reporting suggest enhancing this discard metric:

- o Reporting the number of discarded packets in a measurement interval, i.e., during either the last reporting interval or since the beginning of the session, as indicated by a flag in the suggested XR report [RFC7002]. If an endpoint needs to report packet discard due to other reasons than early- and late-arrival (for example, discard due to duplication, redundancy, etc.) then it should consider using the Discarded Packets Report Block [RFC7002].
- o Reporting gaps and bursts of discarded packets during a measurement interval, i.e., the last reporting interval or the duration of the session [RFC7003].

- o Reporting the sum of payload bytes discarded during a measurement interval, i.e., the last reporting interval or the duration of the session [I-D.ietf-xrblock-rtcp-xr-bytes-discarded-metric].

However, none of these metrics allow a receiver to report precisely which packets were discarded. While this information could in theory be derived from high-frequency reporting on the number of discarded packets [RFC7002] or from the gap/burst report [RFC7003], these two mechanisms do not appear feasible: The former would require an unduly high amount of reporting which still might not be sufficient due to the non-deterministic scheduling of RTCP packets. The latter incur significant complexity and reporting overhead and might still not deliver the desired accuracy.

This document defines a discard report block following the idea of the run-length encoding applied for lost and received packets in [RFC3611].

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 BCP 14, RFC 2119 [RFC2119].

The terminology defined in RTP [RFC3550] and in the extensions for XR reporting [RFC3611] applies.

3. XR Discard RLE Report Block

The XR Discard RLE report block uses the same format as specified for the loss and duplicate report blocks in [RFC3611]. Figure 1 describes the packet format. The fields "BT", "T", "block length", "SSRC of source", "begin_seq", and "end_seq" have the same semantics and representation as defined in [RFC3611], with the addition of the "E" flag to indicate the reason for discard. The "chunks" encoding the run length have the same representation as in RFC3611, but encode discarded packets. A definition of a discarded packet is given in [RFC7002].

```

      0               1               2               3
0 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
+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+---+

```

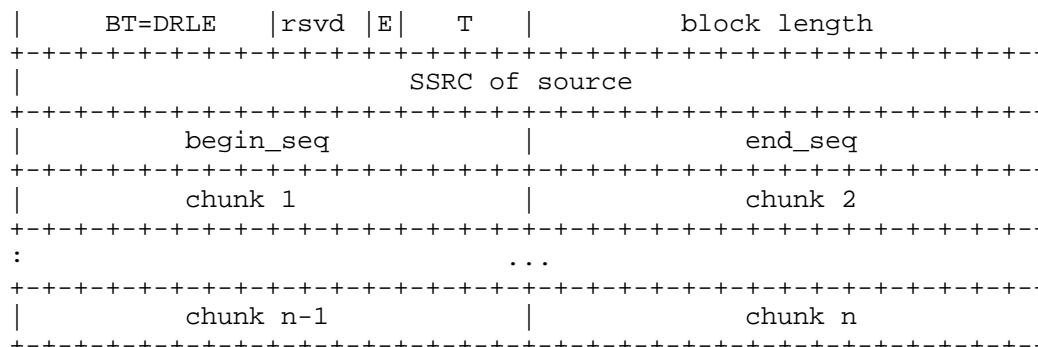


Figure 1: XR Discard RLE Report Block

Block Type (BT, 8 bits): A Run-length encoded Discarded Packets Report Block is identified by the constant DRLE.

[Note to RFC Editor: please replace DRLE with the IANA provided RTCP XR block type for this block. Please remove this note prior to publication as an RFC.]

rsvd (3 bits): This field is reserved for future definition. In the absence of such definition, the bits in this field MUST be set to zero and MUST be ignored by the receiver.

The 'E' bit is introduced to distinguish between packets discarded due to early arrival and those discarded due to late arrival. The 'E' bit is set to '1' if the chunks represent packets discarded due to too early arrival and is set to '0' otherwise.

In case both early and late discarded packets shall be reported, two Discard RLE report blocks MUST be included; their sequence number range MAY overlap, but individual packets MUST only be reported as either early or late and not appear marked in both. If packets appear in both report blocks, the conflicting packets are ignored. Packets reported in neither are considered to be properly received and not discarded.

Discard RLE Report Blocks SHOULD be sent in conjunction with an RTCP RR as a compound RTCP packet.

4. Protocol Operation

This section describes the behavior of the reporting node (= media receiver) and the media sender.

4.1. Reporting Node (Receiver)

Transmission of RTCP XR Discard RLE Reports is up to the discretion of the media receiver, as is the reporting granularity. However, it is RECOMMENDED that the media receiver signals all discarded packets using the method defined in this document. If all packets over a reporting period were discarded, the media receiver MAY use the Discard Report Block [RFC7002] instead. In case of limited available reporting bandwidth, it is up to the receiver whether or not to include RTCP XR Discard RLE reports.

The media receiver MAY send the Discard RLE Reports as part of the regularly scheduled RTCP packets as per RFC3550. It MAY also include Discard RLE Reports in immediate or early feedback packets as per RFC4585.

4.2. Media Sender

The media sender MUST be prepared to operate without receiving any Discard RLE reports. If Discard RLE reports are generated by the media receiver, the media sender cannot rely on all these reports being received, nor can the media sender rely on a regular generation pattern from the media receiver.

However, if the media sender receives any RTCP reports but no Discard RLE report blocks and is aware that the media receiver supports Discard RLE report blocks, it MAY assume that no packets were discarded at the media receiver.

5. SDP signaling

A participant of a media session MAY use SDP to signal its support for the report block specified in this document or use them without any prior signaling (see section 5 of [RFC3611]).

For signaling in SDP, the RTCP XR attribute as defined in [RFC3611] MUST be used. The SDP [RFC4566] attribute 'xr-format' defined in RFC3611 is augmented as described in the following to indicate the the discard RLE metric.

```
rtcp-xr-attrib = "a=" "rtcp-xr" ":" [xr-format *(SP xr-format)]
                  CRLF      ; defined in [RFC3611]

xr-format      =/ xr-discard-rle

xr-discard-rle = "discard-rle"
```

The parameter 'discard-rle' is used to indicate support for the Discard RLE Report Block defined in Section 3.

When SDP is used in Offer/Answer context, the mechanism defined in [RFC3611] for unilateral "rtcp-xr" attribute parameters applies (see section 5.2 of [RFC3611]).

6. Security Considerations

The Discard RLE block provides per-packet statistics so the risk to confidentiality documented in Section 7, paragraph 3 of [RFC3611] applies. In some situations, returning very detailed error information (e.g., over-range measurement or measurement unavailable) using this report block can provide an attacker with insight into the security processing. Implementers should consider the guidance in [I-D.ietf-avt-srtp-not-mandatory] 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 [I-D.ietf-avtcore-rtp-security-options]) and might be more suitable.

Additionally, The security considerations of [RFC3550], [RFC3611], and [RFC4585] apply.

7. 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].

7.1. XR Report Block Registration

This document extends the IANA "RTP Control Protocol Extended Reports (RTCP XR) Block Type Registry" by a new value: DRLE (Discard RLE Report).

[Note to RFC Editor: please replace DRLE with the IANA provided RTCP XR block type for this block here and in the diagrams above. Please remove this note prior to publication as an RFC.]

7.2. SDP Parameter Registration

This document registers a new parameters for the Session Description Protocol (SDP), "discard-rle" in the "RTP Control Protocol Extended Reports (RTCP XR) Session Description Protocol (SDP) Parameters Registry".

7.3. Contact information for IANA registrations

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8. Acknowledgments

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9. References

9.1. Normative References

- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.
- [RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.
- [RFC3611] Friedman, T., Caceres, R., and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)", RFC 3611, November 2003.
- [RFC4585] Ott, J., Wenger, S., Sato, N., Burmeister, C., and J. Rey, "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)", RFC 4585, July 2006.
- [RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", RFC 4566, July 2006.
- [RFC7002] Clark, A., Zorn, G., and Q. Wu, "RTP Control Protocol (RTCP) Extended Report (XR) Block for Discard Count Metric Reporting", RFC 7002, September 2013.

9.2. Informative References

- [RFC7003] Clark, A., Huang, R., and Q. Wu, "RTP Control Protocol (RTCP) Extended Report (XR) Block for Burst/Gap Discard Metric Reporting", RFC 7003, September 2013.
- [RFC5481] Morton, A. and B. Claise, "Packet Delay Variation Applicability Statement", RFC 5481, March 2009.
- [RFC3711] Baugher, M., McGrew, D., Naslund, M., Carrara, E., and K. Norrman, "The Secure Real-time Transport Protocol (SRTP)", RFC 3711, March 2004.
- [RFC5124] Ott, J. and E. Carrara, "Extended Secure RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/SAVPF)", RFC 5124, February 2008.
- [I-D.ietf-avt-srtp-not-mandatory]
Perkins, C. and M. Westerlund, "Securing the RTP Protocol Framework: Why RTP Does Not Mandate a Single Media Security Solution", draft-ietf-avt-srtp-not-mandatory-13 (work in progress), May 2013.
- [I-D.ietf-avtcore-rtp-security-options]
Westerlund, M. and C. Perkins, "Options for Securing RTP Sessions", draft-ietf-avtcore-rtp-security-options-04 (work in progress), July 2013.
- [I-D.ietf-xrblock-rtcp-xr-bytes-discarded-metric]
Singh, V., Ott, J., and I. Curcio, "RTP Control Protocol (RTCP) Extended Report (XR) for Bytes Discarded Metric", draft-ietf-xrblock-rtcp-xr-bytes-discarded-metric-00 (work in progress), October 2013.

Appendix A. Metrics represented using RFC6390 Template

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

a. Run-length encoding of Discarded RTP Packets Metric

- * Metric Name: Run-length encoding of Discarded RTP Packets Metric.
- * Metric Description: Instances of RTP packets discarded over the period covered by this report.
- * Method of Measurement or Calculation: See section 3, for the definition of Discard run-length encoding [RFCXXXX] and section 4.1 of RFC3611 for Run-length encoding.

- * Units of Measurement: Every RTP packet in the interval is reported as discarded or not. See section 3 for the definition of [RFCXXXX].
- * Measurement Point(s) with Potential Measurement Domain: The measurement of these metrics is made at the receiving end of the RTP stream.
- * Measurement Timing: Each RTP packet between a beginning sequence number (begin_seq) and ending sequence number (end_seq) are reported as discarded or not. See section 3 for the definition of Discard run-length encoding [RFCXXXX].
- * Use and applications: See section 1, paragraph 1 of [RFCXXXX].
- * Reporting model: See RFC3611.

Appendix B. Change Log

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

B.1. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-00

- o Changed the interval flag from 1 to 2 bits in the discarded bytes report. Also added the measurement identification tag to the block.
- o Added this section.

B.2. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-01

- o Removed the measurement identification tag in the bytes discarded block.

B.3. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-02

- o Removed the extra Tag bits from the Discarded bytes XR block.
- o Clarified use of measurement identity block in Section 4 and 5.2

B.4. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-03

- o Added explanation for block length in bytes discarded block.
- o Added an acknowledgement section.

- B.5. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-04
 - o Added Block Type definition to each XRBlock.
 - o Made changes requested in WGLC.
- B.6. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-05
 - o Made changes requested by SDP directorate.
- B.7. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-06
 - o Editorial fixes based on review from Gen-art and IESG review.
- B.8. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-07
 - o Editorial fixes based on review from IESG.
 - o Editorial fixes based on Security and PM directorate.
 - o Split bytes discarded from this draft to another.
 - o Updated Security Considerations Section.
 - o This draft now normatively cites the definition of discards in 'packets discarded' draft.
- B.9. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-08
 - o Editorial fixes: Updated references from drafts to RFCs.
 - o Updated RFC6390 template with RTP keyword.
- B.10. changes in draft-ietf-xrblock-rtcp-xr-discard-rle-metrics-09
 - o Removed (RLE) from RFC6390 template.

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