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

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

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

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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.

RTCWEB-STATS [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 better complement the identifiers already in the statistics registry. In depth discussion about the XR metrics candidates is carried out in [I-D.huang-xrblock-rtcweb-rtcp-xr-metrics].

The Javascript application on the receiving endpoint extracts the statistics for the remote (incoming) media stream from the browser's RTP using the Stats API [W3C.WD-webrtc-20130910]. There are multiple ways to exchange these observed identifiers with the remote party, one of which is to use corresponding RTCP Extension Reports (XRs) [RFC3611] and query the statistic identifiers for the local (outgoing) media stream. However, the exchange of statistics of the local stream are currently outside the scope of WebRTC.

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 the group of identifiers are well defined on a ReportGroup corresponding to an SSRC. In practice the application MUST be able to query the statistic identifiers on an incoming media stream (remote stream). Depending on the support of the corresponding XR report the endpoint MAY be able to query the statistic identifier for the outgoing media stream (local 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, Appendix A of [RFC7002]

Name: OctetsDiscarded

Definition: Cumulative Number of octets discarded, Appendix A of [I-D.ietf-xrblock-rtcp-xr-bytes-discarded-metric]

2.1.2. Cumulative Number of Retransmitted Packets Received

Name: PacketsRetx

Definition: See Appendix A of this document, [RFCXXXX].

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

2.1.3. Cumulative Number of Packets Repaired

Name: PacketsRepaired

Definition: Appendix A (b) of [I-D.huang-xrblock-post-repair-loss-count]

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.

6. References

6.1. 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.

[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.

[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.

[I-D.huang-xrblock-post-repair-loss-count] Huang, R. and V. Singh, "RTP Control Protocol (RTCP) Extended Report (XR) for Post-Repair Non-Run Length Encoding (RLE) Loss Count Metrics", draft-huang-xrblock-post-repair-loss-count-00 (work in progress), 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.

6.2. 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-10 (work in progress), December 2012.

[I-D.huang-xrblock-rtcweb-rtcp-xr-metrics]
Huang, R., Even, R., and V. Singh, "Consideration for Selecting RTCP Extended Report (XR) Metrics for RTCWEB Statistics API", draft-huang-xrblock-rtcweb-rtcp-xr-metrics-01 (work in progress), July 2013.

[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>.

Appendix A. Metrics represented using RFC6390 Template

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

a. Number of Packets Retransmitted Metric

* Metric Name: Cumulative number of RTP Packets retransmitted

* Metric Description: Total number of packets retransmitted from the beginning of the session.

* Method of Measurement or Calculation: Cumulative number of retransmitted packets received from the beginning of the session. The measured value is an unsigned value. If the measured value exceeds 0xFFFFFFFD, the value 0xFFFFFFFE MUST be reported to indicate an over-range measurement. If the measurement is unavailable, the value 0xFFFFFFFF MUST be reported.

* Units of Measurement: The counter is increased by one for every retransmitted RTP packet that is received.

* Measurement Point(s) with Potential Measurement Domain: This metric reports the number of retransmitted RTP packets received by the receiver. The measurement of these metrics are made at the receiving end of the retransmission stream and the association of the retransmission and original streams should refer to section 5.3 of [RFC4588].

* Measurement Timing: This metric is applicable to cumulative measurements, which may be the duration of the ongoing RTP session.

* Use and applications: See section 1 of [RFCXXXX].

* Reporting model: Queried periodically by the WebRTC Statistics API.

Appendix B. Change Log

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

B.1. changes in draft-singh-xrblock-webrtc-additional-stats-00

o Clarified measurement points for remote (incoming) media stream and local (outgoing) media stream.

o Added this section.

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