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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

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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.

6. References

6.1. Normative References

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[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

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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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