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Congestion Control Using FEC for Conversational Media

draft-singh-rmcat-adaptive-fec-00

Abstract

This document describes a new mechanism for conversational multimedia flows. The proposed mechanism uses Forward Error Correction (FEC) encoded RTP packets (redundant packets) along side media packets to probe for available network capacity. A straightforward interpretation is, the sending endpoint increases the sending rate by keeping the media rate constant but increases the amount of FEC. If no losses and discards occur, the endpoint can then increase the media rate. If losses occur, the redundant FEC packets help in recovering the lost packets. Consequently, the endpoint can vary the FEC bit rate to conservatively (by a small amount) or aggressively (by a large amount) probe for available network capacity.

Status of This Memo

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1. Introduction

The Real-time Transport Protocol (RTP) [RFC3550] is widely used in voice telephony and video conferencing systems. Many of these systems run over best-effort UDP/IP networks, and are required to implement congestion to adapt the transmission rate of the multimedia streams to match the available network capacity, while maintaing the user-experience [I-D.ietf-rmcat-cc-requirements]. The circuit breakers [I-D.ietf-avtcore-rtp-circuit-breakers] describe a minimal set of conditions when an RTP media stream is causing severe congestion and should cease transmission. Consequently, the congestion control algorithm are expected to avoid triggering these conditions.

Conversational multimedia systems use Negative Acknowlegment (NACK), Forward Error Correction (FEC), and Reference Picture Selection (RPS) to protect against packet loss. These are used in addition to the codec-dependent resilience methods (for e.g., full intra-refresh and error-concealment). In this way, the multimedia system is anyway trading off part of the sending rate for redundancy or retransmissions to reduce the effects of packet loss. An endpoint often prefers using FEC in high latency networks where retransmissions may arrive later than the playout time of the packet (due to the size of the dejitter buffer) [Holmer13]. Therefore, the endpoint needs to adapt the transmission rate to best fit the changing network capacity and the amount of redundancy based on the observed/expected loss rate and network latency. Figure 1 shows the applicatbility of different error-resilience schemes based on the end-to-end latency and the observed packet loss [Devadoss08].

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                        ^             __________
                        |            |          |
                        |            |    UEP   |
                      l |____________|____      |
                      a |            |    |     |
                      t |       RPS  |    |     |
                      e |_______     |    |     |
                      n |       |    |    |     |
                      c |       |    |____|_____|
                      y | NACK  |         |
                        |       |         |
                        +---------------------------->
                               Packet loss
```

Figure 1: Applicability of Error Resilience Schemes based on the network delay and observed packet loss

In this document, we describe the use of FEC packets not only for error-resilience but also as a probing mechanism for congestion control (ramping up the transmission rate).

2. Terminology

The terminology defined in RTP [RFC3550], RTP Profile for Audio and Video Conferences with Minimal Control [RFC3551], RTCP Extended Report (XR) [RFC3611], Extended RTP Profile for RTCP-based Feedback (RTP/AVPF) [RFC4585] and Support for Reduced-Size RTCP [RFC5506] apply.

3. Concept: FEC for Congestion Control

4. Design Features

4.1. FEC Scheme

Block and Parity FEC

4.2. FEC and RTP

RFC4588, RFC5109, RFC6363

5. Applicability to RMCAT

6. Security Considerations

Security issues have not been discussed in this memo.

7. IANA Considerations

There are no IANA impacts in this memo.

8. Acknowledgements

9. References

9.1. Normative References

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Perkins, C. and V. Singh, "Multimedia Congestion Control: Circuit Breakers for Unicast RTP Sessions", draft-ietf-avtcore-rtp-circuit-breakers-05 (work in progress), February 2014.

9.2. Informative References

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[Holmer13]
Holmer, S., Shemer, M., and M. Paniconi, "Handling Packet Loss in WebRTC", Proc. of IEEE International Conference on Image Processing (ICIP 2013) , 9 2013.

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-rmcat-adaptive-fec-00

- Updated references.

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