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Congestion Control Using FEC for Conversational Media
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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 the media packets to probe for available network capacity. A straightforward interpretation is, the sending endpoint increases the transmission 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.

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Table of Contents

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 RTP 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 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 transmission 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].

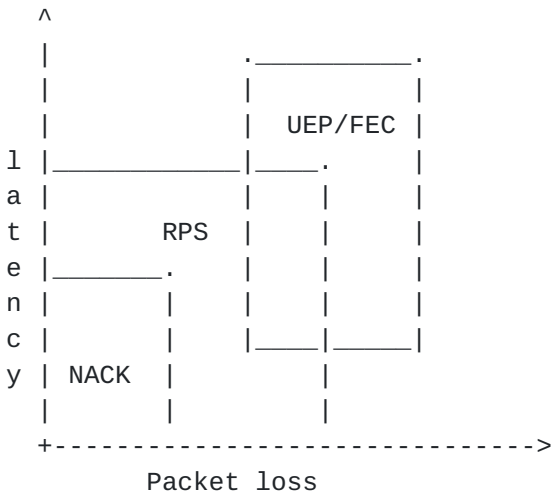

Figure 1: Applicability of Error Resilience Schemes based on the network delay and observed packet loss

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 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, [RFC2119] and indicate requirement levels for compliant implementations.

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], RTP Retransmission Payload Format [RFC4588], Forward Error Correction (FEC) Framework [RFC6363], and Support for Reduced-Size RTCP [RFC5506] apply.

3. Concept: FEC for Congestion Control

FEC is one method for providing error-resilience, it improves reliability by adding redundant data to the primary media flow, which is used by received to recover packets that have been lost due to congestion or bit-errors. The congestion control algorithm on the other hand aims at maximizing the network path utilization, but risks over-estimating the avaiable end-to-end network capacity leading to congestion (and therefore losses).

Figure 2 shows the timeline of enabling and disabling FEC. The main idea behind using FEC for congestion control is as follows: the sending endpoint chooses a high FEC rate to aggressively probe for available capacity and conversely chooses a low FEC rate to conservatively probe for available capacity. During the ramp up, if a packet is lost and the FEC packet arrives in time for decoding, the receiver is be able to recover the lost packet; if no packet is lost, the sender is able to increase the media encoding rate by swapping out a part of the FEC rate. This method can be especially useful when the transmission rate is close to the bottleneck link rate: by choosing an appropriate FEC rate, the endpoint is able to probe for available capacity without changing the target media rate and therefore not affecting the user-experience.

Hence, the congestion control algorithm is always able to probe for available capacity, as improved reliability compensates for possible errors resulting from probing for additional capacity (i.e., increase in observed latency and/or losses).

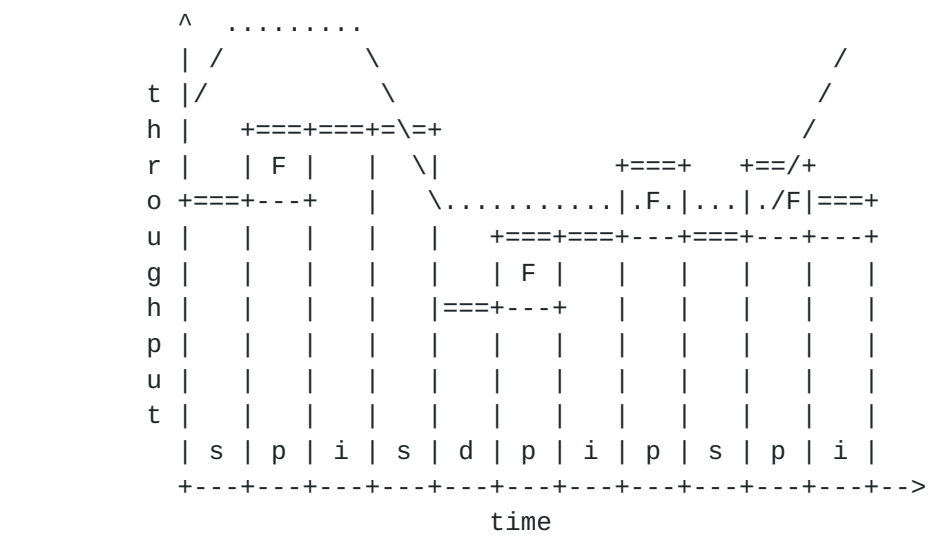

Figure 2: Congestion Control enabling FEC.

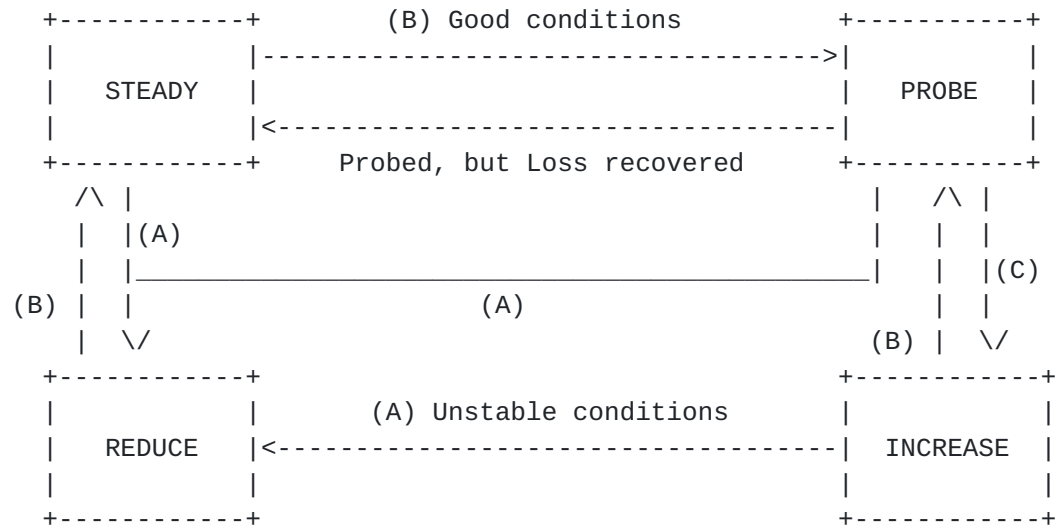

Figure 3: State machine of a Congestion Control enabling FEC.

3.1. States

The Figure 3 illustrates the the state machine of a congestion control algorithm incorporating FEC for probing. The state transitions occur based on the information reported in the feedback packet. In Figure 3 (A) indicates congestion, i.e., the congestion control observes increasing delay and/or packet loss, or any other congestion metric, and in response the congestion control reduces the transmission rate. In Figure 3 (B) occurs when the congestion cues report improvement in congestion metrics, and in response the congestion cue increases the transmission rate. For probing using FEC, the congestion control needs to map to the following 4 states: STEADY, PROBE, INCREASE, and REDUCE.

- STEADY state: The congestion control keeps the same target media rate and no additional FEC packets are generated for probing. This is a transient state, after which the congestion control either attempts to increase the transmission rate, or observes congestion and reduces the transmission rate.

- REDUCE state: The congestion control reduces the transmission rate based on the observed congestion cues, and generated no additional FEC packets than the minimum required for error-resilience. If in subsequent reports the conditions improve, the congestion control can directly transition to the PROBE state.

- PROBE state: The congestion control observes no congestion for two reporting intervals (i.e., the transmission rate should be increased). The endpoint maintains the same target media bit rate, and instead increases the amount of FEC packets, therby increasing the transmission rate.

- INCREASE state: The endpoint is sending FEC packets and the congestion control observes no congestion (as reported in RTCP feedback), the media transmission rate is increased while maintaining minimal amount of FEC for error protection. In this case, the combined transmission rate (FEC+media) may remain the same as in the PROBE state. If the feedback reports packet loss, but some of these lost packets are recovered by the FEC packets, the congestion control can keep the same media bit rate and adjust the amount of FEC (compared to the previous PROBE state). If congestion is observed (the target rate calculated by the congestion control is much lower than the current media rate), the congestion control can transition to the REDUCE state and decrease the transmission rate.

3.2. Framework

The Figure 4 shows the interaction between the rate control module, the RTP and the FEC module.

At the sender, the rate control module calculates the new bit rate. If the new bit rate is higher than the previous than the previous bit rate indicates to the FEC module that the congestion control intends to probe. The FEC module depending on its internal state machine decides to add FEC for probing or not. Thereafter it indicates to the rate control module the bit rate remaining for the RTP media stream, which may be less than equal to the calculated bit rate.

At the reciver, the FEC module reconstructs lost packets in the primary stream from the packets received in the repair stream. If packets are repair it generates the post-repair loss report (discussed in Section 3.3) for the corresponding RTP packets.

At the sender, The FEC module also receives the RTCP Feedback related to the primary stream and any post-repair loss report. It uses the information from these RTCP reports to calculate the effectiveness of FEC for congestion control and is also the basis for changing its internal state.

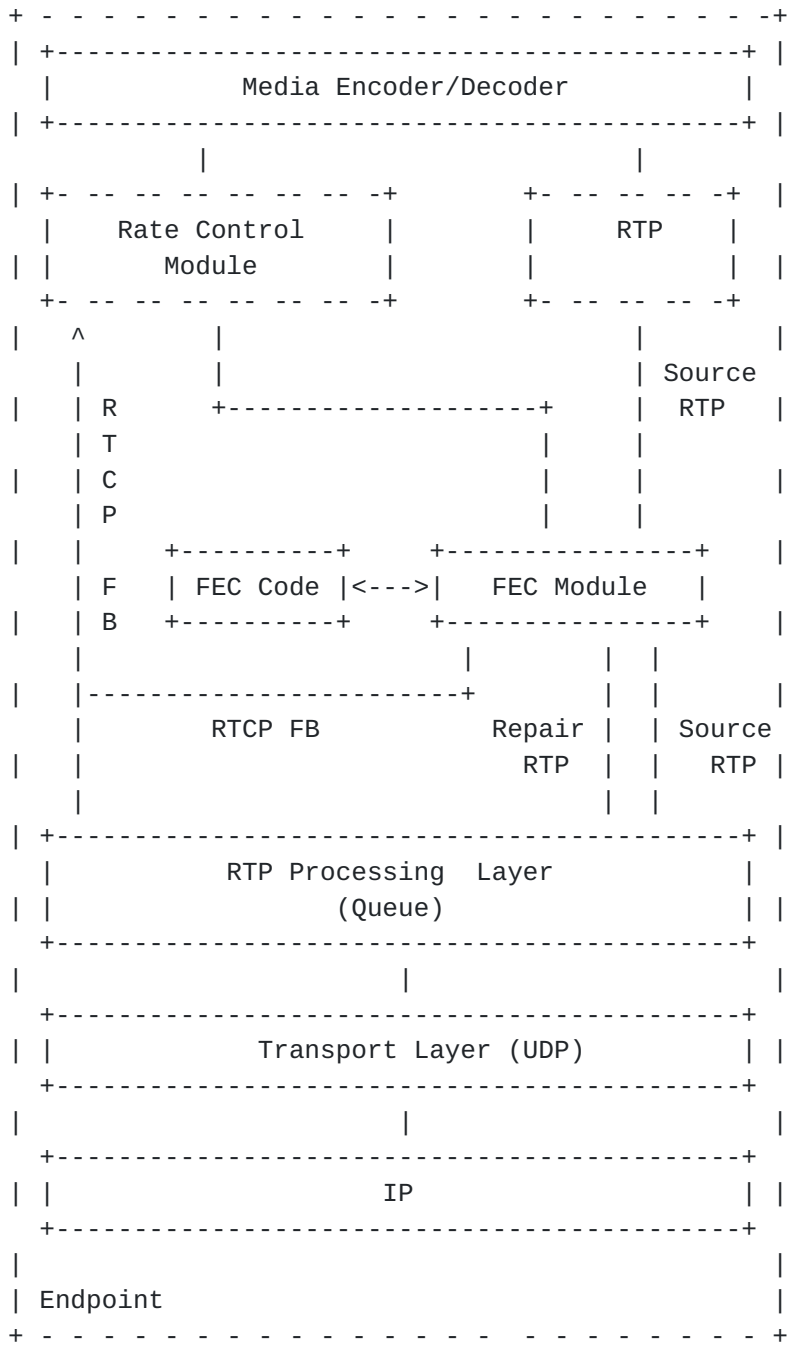

Figure 4: Interaction of Congestion Control and FEC Module.

3.3. FEC Scheme

[RFC6363] describes a framework for using Forward Error Correction (FEC) codes with RTP and allows any FEC code to be used with the framework. For this proposal, the FEC packets are created by XORing RTP media packets, the resulting redundant RTP packets are encoded

using the scheme defined in [I-D.singh-payload-rtp-1d2d-parity-scheme].

The endpoint MAY use a single-frame FEC (1-dimensional) or a multi-frame FEC (2-dimensional) for protecting the primary RTP stream. A single-frame FEC protects against a single packet loss and fails when burst loss occurs. Using multi-frame FEC helps mitigate these issues at the cost of higher overhead and latency in recovering lost packets. [Holmer13] shows examples of using a single- and multi-frame FEC.

The receiving endpoint may report the post-repair loss (or residual loss) using either the report block defined in [RFC5725] (Run-length encoding of packets repaired) or in [I-D.ietf-xrblock-rtcp-xr-post-repair-loss-count] (packet count of repaired packets).

Additionally, the receiving may report the occurance of losses and discards via a run-length encoding (RLE) of lost [RFC3611] (Section 4.1), which enables the sender to detect the burst loss length and apply appropriate FEC scheme.

Packet that arrive too late to be played out by the receiver are discarded by the de-jitter buffer. Typically, the de-jitter buffer adjust the playout delay based on the observed frame inter-arrival delay, so that packets are played out smoothly. Reporting RLE of discarded packets [RFC7097] may further enable a sender to detect losses that occur after packet discards.

3.4. Applicability to other RMCAT Schemes

[Open issue: The current implementation is delay based and is documented in [Nagy14]. However, we would like to generalize the concept and apply it to different RMCAT algorithms for e.g., Google's Congestion Control algorithm [I-D.alvestrand-rmcat-congestion], SCReaM [I-D.johansson-rmcat-scream-cc], etc.]

4. Security Considerations

The security considerations of [RFC3550], RTP/AVPF profile for rapid RTCP feedback [RFC4585], circuit breaker [I-D.ietf-avtcore-rtp-circuit-breakers], and Generic Forward Error Correction [RFC5109] apply.

If non-authenticated RTCP reports are used, an on-path attacker can send forged RTCP feedback packets that can disrupt the operation of the underlying congestion control. Additionally, the forged packets can either indicate no packet loss causing the congestion control to

ramp-up quickly, or indicate high packet loss or RTT causing the circuit breaker to trigger.

5. IANA Considerations

There are no IANA impacts in this memo.

6. Acknowledgements

This document is based on the results published in [Nagy14].

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Appendix A. Simulations

This document is based on the results published in [Nagy14]. See the paper for ns-2 and testbed results; more results based on the scenarios listed in [I-D.sarker-rmcat-eval-test] will be published shorty.

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