        Reactions to Signaling from ECN Support for RTP/RTCP
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Abstract

   This document presents various responses to Congestion Experience (CE)
   notifications by real time applications that have negotiated end-to-end
   support of Explicit Congestion Notification (ECN). This document is a
   follow-on effort of [rfc6679], which specifies the signaling used to
   provide ECN support for RTP/RTCP flows.

1. Introduction

   This document presents various responses to Congestion Experience (CE)
   notifications by real time applications that have negotiated end-to-end
   support of Explicit Congestion Notification (ECN). [rfc6679] defines
   the signaling for support of ECN by RTP based sessions, and also covers
   the case where a set of nodes do not respond to CE notifications. A
   more detailed discussion about how back-off algorithms can be achieved
   and supported for specific applications is viewed as out of scope of
   that document and may be addressed by a companion document.

1.1 Background

   ECN is a mechanism used to explicitly signal the presence of congestion
   without relying on packet loss. It was initially designed using a dual
   layer signaling model; negotiation and feedback at the transport layer,
   and downstream notification of congestion at the network layer. For IP,
   a new two bit field was used to both indicate the successful negotiated
   support for ECN signaling, as well as indicate the presence of
   congestion via the CE flag. In the case of TCP [rfc3168], a new TCP
   header flag was defined that provides upstream end-to-end indication of
   congestion occurring somewhere along the downstream path.

   There should be no difference in congestion response if ECN-CE marks or
   packet drops are detected. However it is noted that there MAY be other
   reactions to ECN-CE specified in the future. Such an alternative
   reaction MUST be specified and considered to be safe for deployment
   under any restrictions specified. We specify such an alternative in
   this document.

   With respect to ECN for TCP, [rfc3168] specifies an indication of
   congestion, but it does so once per Round Trip Time (RTT). [rfc6679] is
   an effort that proposes a finer grained notification reflecting a more
   accurate indication of the number of ECN marked packets received within
   one RTT.

1.2 Terminology and Abbreviations

   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 [RFC2119].

2. Issues

The initial discussions and presentation of [rfc6679] produced a
consensus that the specification of signaling was to be done within the
AVTcore working group, and any subsequent discussion on end-to-end
reactions to the signaling would be accomplished in the Transport
Services (TSV) working group. This draft satisfies the latter effort.

Another issue that needs to be recognized is that the reactions to CE in
the context of [rfc6679] are the responsibility of the application.
This is in contrast to ECN support for TCP, where explicit signaled
feedback of, and reaction to, CE is kept transparent to the application.
The issue of placing the feedback responsibility in the application is
that each application needs to add specific support for that reaction.
On the other hand, multiple reactions may be considered by the
application. For this reason, [rfc6679] states the need for a default
congestion control reaction that MUST be supported. Section 3 through 5
expands on this topic.

3. Congestion Control Algorithms

The transport of any data flow across the Internet produces a need for
some form of congestion control to attain a suitable share of the
capacity of the path through a network. Most of the existing work on
realtime congestion control algorithms has been rooted in TCP-friendly
approaches but with smoother adaptation cycles. TCP congestion control
is unsuitable for interactive media for a number of reasons including
the fact that it is loss-based so it maximises the latency on a path, it
changes its transmit rate to quickly for multimedia, and favours
reliability over timeliness. In the case of real time media transport,
one requires:

> Smoother rate variation: (than for bulk data) to accommodate
>   the underlying media flow's characteristics.

> Low latency: Maintaining latencies sufficient to be usable, where
>   150ms one way delay is understood to be a good target
>   [ITU.G114.2003].

> Fairness: The algorithm must be fair to both itself and other flows

3.1 TCP Friendly Rate Control (TFRC)

TFRC has a smoother response to congestion than TCP-like approaches,
thus making it more suitable for real-time interactive multimedia

   applications. It has been cited in a number of other documents within
   the IETF for use with UDP and media flows [rfc3714, bcp145] and is
   seeing full and partial deployment in related solutions such as
   Empathy/Farsight, and GoogleTalk [goog1].

   However it should be noted that TFRC is only recommended for real-time
   media use with ECN response. TFRC is not recommended for non-ECN paths
   due to its loss based operation which leads to full queues with
   maximised latencies. It is assumed that ECN markings will usually occur
   with lower queue occupancy and thus lower latency. However it is
   understood that ECN marks may not provide for sufficiently low
   latencies in some situations so other congestion control solutions
   would be preferable.

   [rfc4342] specifies the profile for TFRC for use in the Datagram
   Congestion Control Protocol (DCCP) [rfc4340] for a half connection. A
   DCCP half connection is defined as application data sent downstream with
   corresponding acknowledgements sent upstream. These half-connections
   can be realized in the form of one-way pre-recoded media, one-way live
   media, or two-way interactive. A perceived drawback in this profile
   concerns its application to interactive media that use small packets.
   [RFC4828] is an experimental protocol defining a variation of TFRC used
   to address this drawback and achieve the same bandwidth as a TCP flow
   using packets of size 1500 bytes.

   [rfc6679] is an standard that specifies how RTP flows can
   be supported using the RTP/AVPF profile and the general RTP header
   extension mechanism.

3.2 Related Work

3.2.1 3GPP

   Outside of this previous and on-going work with TFRC, it is understood
   that some parties have issues with the behavior of TFRC under certain
   conditions. A notable mention of this is made in the 3GPP's document on
   IP Multimedia Subsystem (IMS) Media handling and interaction [TR26.114],
   where it is mentioned:

      "Note that for IMS networks, which normally have nonzero packet loss and
      fairly long round-trip delay, the amount of bitrate reduction specified
      in RFC 3448 is generally too restrictive for video and may, if used as
      specified, result in very low video bitrates already at (for IMS)
      moderate packet loss rates."

   Though it is unclear exactly what the 3GPP community consider as too
   restrictive and whether some alteration of the response may be suitable.
   It should be noted that the 3GPP document only referred to an older

version of TFRC defined in [RFC3448]. Given that the current version
of TFRC [RFC5348] has made significant changes to the idle and data-
limited responses it is unclear whether their assessment is relevant
to current TFRC implementations.

Furthermore the specification [TR26.114] only outlines a rudimentary
approach to congestion control, providing an example of a 60% back-off
reaction to loss within an RTCP reporting period. The proposed signalling
employs Temporary Maximum Media Stream Bit Rate Request (TMMBR)
[RFC5104] and Codec Mode Request (CMR) [RFC4867] for video and audio
respectively, which would only provide for very basic rate control
if used as specified. We note that [TR26.114] specifies terminal
behavior, while [TS36.300] specifies base station behaviour, though
neither specify any standardised congestion control approach.

It is understood that there are a number of proprietary and patented
approaches that provide more sophisticated response in the case of
3G/LTE, but since these are neither endorsed nor standardized this
document advocates a standardized approach such as TFRC.

We also acknowledge that there are many congestion control algorithms
available for implementers to choose from, with a subset that are
specifically suited to real time media transmission. However, given a
variety of real time applications and their various characteristics
(sender-only broadcast, interactive unicast, etc), we need to expand the
notion of how back-off can be achieved. Hence, the focus needs to be on
an output that would resemble the characteristics of TFRC.

Within the RTCweb Working Group the need for a more media friendly
congestion control mechanism has been made apparent. Currently, TFRC is
perceived as having deficiencies (e.g. its loss-based design, lack of
cross-stream congestion control functionality etc) that make it an
incomplete or insufficient solution for the envisioned RTCWEB media
flows. The RTP Media Congestion Avoidance Techniques (rmcat) working
group has now been formed which aims to lead to the formation of a
working group on these issues. The group aims to develop one or more
congestion control algorithms, associated extensions, and evaluation
criteria. Furthermore it has been proposed that certain practices, such
as 'circuit-breaker' conditions, to provide operational limits on
congestion control algorithms, and feedback messages, may be tackled in
other groups such as AVTCORE and AVTEXT respectively.

Thus there is some movement to attempt to develop new algorithms better
suited to media transport, but these efforts will clearly take a
considerable time to reach fruition. Whilst TFRC has some perceived
issues it still provides the best existing solution for media transport.

3.3 ECN response

As mentioned above and in accordance to [rfc3168], the actual response
to the reception of an ECN-CE marked packet MUST normally be the same as
that of a lost packet. However there are a number of contexts where one
may also be interested in more varied approaches. We expand on this in
Section 5 below.

4. Application Layer Congestion Response

Whilst the congestion control algorithm may decide to alter the rate at
which the application should operate, in the case of media applications
this process is not as straightforward as the case of bulk data. The
different media engines and codecs in use may only have limited
adaptation ranges, thus, this limitation needs to be a consideration
when adapting the rate. Furthermore the application needs to be aware
of the capability of the specific codecs in terms of their ability to
switch configuration mid-stream (without loss of fidelity), which may
impose further limits on the modes of operation.

One approach for achieving a lower generation of data is through reduced
sampling of the media (e.g., voice or video). In the case of video,
this may also involve slower frame rates. Specific recommendations that
describe how applications should respond to congestion in the context of
supporting the algorithmic characteristics of a congestion control
algorithm are outside the scope of this document.

5. Other Reactions

In addition to the activation of congestion control algorithm, other
reactions can be used or leveraged by an application in response to CE.
We divide these other potential reactions into two categories: signaling
and fault tolerance. We note that these other reactions are considered
symmetric because they require downstream peer support. We also point
out that activation of other reactions represents an example of an
on-demand and as-needed approach in responding to CE.

5.1 Signaling

5.1.1 RSVP

The resource Reservation Protocol (RSVP) can be used to signal a desired
set of path characteristics (e.g., bandwidth, delay) in response to CE
feedback [rfc2205]. Its operation is based on the use of PATH messages
sent downstream hop-by-hop from the source to a destination that specify
requested forwarding characteristics. In return, the destination sends
a hop-by-hop RESV message upstream towards the source confirming the
resources that have been reserved for that flow.

[rfc3181] defines a priority policy element that specifies both an allocation and defending priority. This dual specification supports the use of preemption of existing reservations. [draft-priority-rsvp] is a work-in-progress that defines a new policy element that only conveys priority during reservation establishment. This latter effort also presents several reservation models, including one that describes engineered resources set aside for priority users.

5.1.2 Differentiated Services

Unlike RSVP and its use of a separate signaling mechanism to reserve resources, Differentiated Services (diff-serv) uses code points within the IP header to convey the forwarding behavior of that packet [rfc2474]. This may range from various drop precedence values to a code point that signifies low delay and low loss (i.e., characteristics attributed to real time flows).

As in the case of RSVP, applications could rely on the reception of CE feedback to initiate a subsequent setting of diff-serv code points to provide additional protection or explicit association of forwarding characteristics of a given flow of packets. In addition, the setting of diff-serv code points would be done on an as-needed basis in reaction to CE feedback. Recommendations concerning specific diff-serv values are outside the scope of this document.

5.2 Fault Tolerance

Fault tolerance is another category of reactions that may be used by applications in response to CE feedback. In some cases, these efforts may contribute to an increase in traffic load in order to add protection and resiliency to a flow.

Redundant Transmissions: This approach is based on a source sending duplicate payloads that can be used to compensate for lost packets. Given that ECN marks the packet and forwards it towards the destination (instead of dropping it), this approach can be considered extreme in terms of being network unfriendly. Its positive value may emerge in cases where a path has several downstream congestion points. However, its actions of producing redundant packets still associates a high measure of greedy use of resources.

Application Layer Forward Error Correction (FEC): This approach also adds additional overhead to the flow in order to compensate for potential packet loss. And as the case of redundant transmissions, the value of this approach is probably better realized when there exists multiple downstream congestion points. However, the impact of the overhead is minimized by having one (or a few) additional packet(s) used to compensate for the loss of a set of packets.

   Codec Swapping: This approach involves changing codecs to either reduce
   load or achieve an improvement in compensating for lost packets.

5.3 **Alternative Reaction for Emergency Communications**

   As mentioned in [rtp-ecn], the default reaction on the reception of
   these ECN-CE marked packets MUST be to provide the congestion control
   algorithm with a congestion notification that triggers the algorithm to
   react as if packet loss had occurred. There MAY be an alternative
   reaction if it is considered safe for deployment. An example of the
   need for an alternative reaction would be the case of Emergency
   Telecommunications Service (ETS) [rfc3689, rfc4190], where an
   improvement in QoS or a higher probability of session establishment and
   forwarding of traffic is of high interest.

   It is proposed that certain authorized ETS flows may be permitted to
   employ either a substantially less aggressive back-off algorithm than
   the default algorithm, or some level of exemption from reacting to ECN
   marked packets. This alternative reaction will benefit these flows as
   the marks would normally be considered as equivalent to lost packets,
   which would effectively increase the loss level, which in turn will
   generally result in the reduction of flow rate. This applies to all
   flows that utilize some form of the rate control that is inversely
   proportional to the loss rate, which includes TCP-like algorithms or
   equation-based approaches.

   Simulations of the use of ECN exemption with TFRC and have found that it
   has limited effect on the normal flows with low numbers of exempt
   flows. A half-dumbbell network was used with a RED router queue
   configured using the settings recommended by Sally Floyd. The candidate
   flows are 1Mbit/s each with a backhaul 100Mbit/s link. In the standard
   case where 1% of flows would be exempt the remaining flows achieve
   99.99% of the bandwidth that they would achieve without the presence of
   the exempt flows. This is what would be expected from the simple
   calculation of the allocation, given that the exempt flows achieve their
   full rate (1Mbit/s); With 100 normal plus 1 exempt flow, assuming that
   the except flow uses 1Mbit/s, the remaining capacity is 99Mbit/s which
   is divided between the 100 normal flows. Whilst when 101 normal flows
   are run over the 100Mbit/s link they would have to share it evenly, so
   it works out thus: ((99/100)/(100/101))*100=99.99%. In the case of 5%
   exempt flows then the proportion is very slightly lower at
   ((95/100)/(100/105))*100=99.75%. Both these calculations are borne out
   in the simulation runs.

   The level of exemption employed can be altered in a number of ways. Two
   simple approaches would be to either set a threshold number of ECN
   marked packets that could be considered as a loss, and another approach
   would be to set a percentage threshold of ECN marked packet that would

be considered as a loss.

It should be noted that in the simulations the end-to-end delay of the packets within the flows was monitored and the relative delay of the exempt flows apparently rises somewhat when exemption is enacted. However what is actually occurring is that the 'normal' flows are reducing their throughput and are thus reducing their latency somewhat. There is normally some limited latency when using loss-based techniques such as TFRC because it fills the queues to ascertain the link capacity and maintains that level of delay throughout a session. However the level of latency is clearly limited by the queue sizes in the network and on media specific links these queue sizes are typically quite small, so the resulting latency is limited.

Furthermore in the case where media flows employing TFRC, or any other congestion control algorithm (e.g. delay-based), are sharing a bottleneck link with TCP flows then the queues will be filled by the TCP flows and the latency will be kept near or at a their maximum despite any other flows.

6. IANA Considerations

This document requires no actions from IANA.

7. Security Considerations

The reliance on accurate and un-modified RTCP information means that SRTP needs to be used, or any other mechanism that helps prevent modification of RTCP feedback packets.

8. Acknowledgements

TBD

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