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RTP Profile for TCP Friendly Rate Control

Status of this Memo

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Abstract

This memo specifies a profile called "RTP/AVPCC" for the use of the real-time transport protocol (RTP) and its associated control protocol, RTCP, with the TCP Friendly Rate Control (TFRC). TFRC is

an equation based congestion control scheme for unicast flows operating in a best effort Internet environment. This profile provides RTP flows with the mechanism to use congestion control in best effort IP networks.

1. Introduction

[Note to RFC Editor: All references to RFC XXXX are to be replaced with the RFC number of this memo, when published]

This memo defines a profile called "RTP/AVPCC" for the use of the real-time transport protocol (RTP) [RTP] and its associated control protocol, RTCP, with the TCP Friendly Rate Control (TFRC) [TFRC]. TFRC is an equation based congestion control scheme for unicast flows operating in a best effort Internet environment and competing with TCP traffic.

Due to a number of inherent TFRC characteristics, the RTP/AVPCC profile differs from other RTP profiles [AVP] in the following ways:

- TFRC is a unicast congestion control scheme, therefore by extension the RTP/AVPCC profile can only be used by unicast RTP flows.

- A TFRC sender relies on receiving feedback from the receiver either once per round-trip time (RTT) or per data packet. For certain flows (depending on RTTs and data rates) these TFRC requirements can result in control traffic that exceeds RFC 3550's bandwidth and/or timing recommendations for control traffic. The RTP/AVPCC profile recommends modifications to these recommendations in order to satisfy TFRCs timing needs for control traffic in a safe manner.

This memo primarily addresses the means of supporting TFRC's exchange of congestion control information between senders and receivers via the following modifications to RTP and RTCP: (1) RTP data header additions; (2) extensions to the RTCP Receiver Reports; and (3) modifications to the recommended RTCP timing intervals. For details on TFRC congestion control readers are referred to [TFRC].

The current TFRC standard, RFC3448, only targets applications with fixed packet size. TFRC-PS is a variant of TFRC for applications with varying packet sizes. The RTP/AVPCC profile is applicable to both congestion control schemes.

2. Relation to the Datagram Congestion Control Protocol

The Datagram Congestion Control Protocol (DCCP) is a minimal general purpose transport-layer protocol with unreliable yet congestion-controlled packet delivery semantics and reliable connection setup and teardown. DCCP currently supports both TFRC and TCP-like congestion control. In addition DCCP supports a host of other features, such as: use of Explicit Congestion Notification (ECN) and the ECN Nonce, reliable option negotiation and Path Maximum Transfer Unit (PMTU). Naturally an application using RTP/DCCP as its transport protocol will benefit from the protocol features supported by DCCP.

In contrast the RTP Profile for TFRC only provides RTP applications a standardized means for using the TFRC congestion control scheme, without any of the protocol features of DCCP. However there are a number of benefits to be gained by the development and standardization of a RTP Profile for TFRC:

- o Media applications lacking congestion control can incorporate congestion controlled transport without delay by using the RTP/AVPCC profile. The DCCP protocol is currently under development and widespread deployment is not yet in place.

- o Use of the RTP/AVPCC profile is not contingent on any OS level changes and can be quickly deployed, as the AVPCC profile is implemented at the application layer.

- o AVPCC/RTP/UDP flows face the same restrictions in firewall traversal as do UDP flows and do not require NATs and firewall modifications. DCCP flows, on the other hand, do require NAT and firewall modifications, however once these modifications are in place, they can result in easier NAT and firewall traversal for RTP/DCCP flows in the future.

- o Use of the RTP/AVPCC profile with various media applications will give researchers, implementors and developers a better understanding of the intricate relationship between media quality and equation based congestion control. Hopefully this experience with congestion control and TFRC will ease the migration of media applications to DCCP once DCCP is deployed.

Overall, the RTP/AVPCC profile provides an immediate means for congestion control in media streams, in the time being until DCCP is deployed.

Additionally, there are also a number of differences in the exchange of congestion control information between DCCP with CCID3 and the

RTP/AVPCC profile:

o A RTP/AVPCC sender transmits the round trip time and the send timestamp to the RTP/AVPCC receiver. In addition to congestion control the send timestamp can be used by the receiver for jitter calculations.

In contrast DCCP with CCID3 transmits a quad round trip counter to the receiver.

o A RTP/AVPCC receiver only provides the RTP/AVPCC sender with the loss event rate as computed by the receiver.

In contrast DCCP with CCID3, provides 2 other options for the transport of loss event rate. A sender may choose to receive loss intervals or an Ack Vector. These two options provide the sender with the necessary information to compute the loss event rate.

o Sequence number: DCCP supports a 48 bit and a 24 bit sequence number. RTP supports a 16 bit sequence number.

3. Conventions Used in this Document

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 RFC 2119 [2119].

4. RTP and RTCP Packet Forms and Protocol Behavior

The section "RTP Profiles and Payload Format Specifications" of RFC 3550 enumerates a number of items that can be specified or modified in a profile. This section addresses each of these items and states which item is modified by the RTP/AVPCC profile:

RTP data header: The standard format of the fixed RTP data header is used (one marker bit).

Payload types: This profile does not define new payload types, and has no payload type restrictions.

RTP data header additions: A 16 bit fixed field is added to the RTP data header for the transport of the quad RTT counter to the TFRC receiver.

RTP data header extensions: No RTP header extensions are

defined, but applications operating under this profile MAY use such extensions. Thus, applications SHOULD NOT assume that the RTP header X bit is always zero and SHOULD be prepared to ignore the header extension. If a header extension is defined in the future, that definition MUST specify the contents of the first 16 bits in such a way that multiple different extensions can be identified.

RTCP packet types: No additional RTCP packet types are defined by this profile specification.

RTCP report interval: This profile is restricted to unicast flows, therefore at all times there is only one active sender and one receiver. Sessions operating under this profile MAY specify a separate parameter for the RTCP traffic bandwidth rather than using the default fraction of the session bandwidth. In particular this may be necessary for data flows were the the RTCP recommended reduced minimum interval is still greater than the RTT.

SR/RR extension: A 16 octet RR extension is defined for the RTCP RR packet.

SDES use: Applications MAY use any of the SDES items described in the RTP specification.

Security: <TBC> See Section 9.

String-to-key mapping: No mapping is specified by this profile.

Congestion: This profile specifies how to use RTP/RTCP with TFRC congestion control.

Underlying protocol: The profile specifies the use of RTP over unicast UDP flows only, multicast MUST NOT be used.

Transport mapping: The standard mapping of RTP and RTCP to transport-level addresses is used.

Encapsulation: This profile is defined for encapsulation over UDP only.

5. The TFRC Feedback Loop

TFRC depends on the exchange of congestion control information between a sender and receiver. In this section we reiterate which items are exchanged between a TFRC sender and receiver as discussed in [TFRC]. We note how the RTP/AVPCC profile accommodates these

exchanges.

5.1. Data Packets

As stated in [TFRC] a TFRC sender transmits the following information in each data packet to the receiver:

o A sequence number, incremented by one for each data packet transmitted.

o A timestamp indicating the packet send time and the sender's current estimate of the round-trip time, RTT. This information is then used by the receiver to compute the TFRC loss intervals.
- or -
A course-grained timestamp incrementing every quarter of a round trip time, which is then used to determine the TFRC loss intervals.

The standard RTP sequence number suffices for TFRCs functionality. For the computation of the loss intervals the RTP/AVPCC profile extends the RTP data header as follows: a 32 bit field to transmit a send timestamp and an additional 32 bit field, present only when the RTT changes, to transmit the RTT. The presence of the RTT is indicated by the R bit in the RTP header (see Section 6).

5.2. Feedback Packets

As stated in [TFRC] a TFRC receiver provides the following feedback to the sender at least once per RTT or per data packet received (which ever time interval is larger):

o The timestamp of the last data packet received, t_i.

o The amount of time elapsed between the receipt of the last data packet at the receiver, and the generation of this feedback report, t_delay. This is used by the sender for RTT computations (see Section 9).

o The rate at which the receiver estimates that data was received since the last feedback report was sent, x_recv

o The receiver's current estimate of the loss event rate, p.

To accommodate the feedback of these values the RTP/AVPCC profile defines a 16 octet extension to the RTCP Receiver Reports (see Section 7).

6. RTP Data Header Additions

```
     0                   1                   2                   3
     0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |V=2|P|X|  CC   |M|R|     PT      |       sequence number         |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |                           timestamp                             |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |           synchronization source (SSRC) identifier              |
    +=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
    |                       send time-stamp                           |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |            contributing source (CSRC) identifiers               |
    |                             ....                                |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 1: RTP header and additions with R=0, no RTT included.

```
     0                   1                   2                   3
     0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |V=2|P|X|  CC   |M|R|     PT      |       sequence number         |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |                           timestamp                             |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |           synchronization source (SSRC) identifier              |
    +=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
    |                       send time-stamp                           |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
    |                              RTT                                |
    +=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
    |            contributing source (CSRC) identifiers               |
    |                            ....                                 |
    +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 2: RTP header and additions with R=1, RTT included.

7. Receiver Report Extensions

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|    RC   |   PT=RR=201   |             length            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                     SSRC of packet sender                     |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|                   SSRC (SSRC of first source)                 |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
| fraction lost |       cumulative number of packets lost       |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           extended highest sequence number received           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      interarrival jitter                      |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                         last SR (LSR)                         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                   delay since last SR (DLSR)                  |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|                             t_i                               |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           t_delay                             |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                  data rate at the receiver (x_recv)           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      loss event rate (p)                      |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
```

Figure 3: RTCP Receiver Report extensions.

8. RTCP Timing Intervals

The RTP/AVPCC profile recommends the use of the TFRC timing feedback requirements for the RTCP timing intervals, only in instances where control traffic bandwidth does not exceed RFC 3550's recommended 5% of data traffic.

A TFRC sender requires feedback from its receiver at least once per RTT or per packet received (based on the larger time interval). These requirements are to ensure timely reaction to congestion.

In some instances TFRC's timing requirements may result in timing intervals for RTCP traffic that are smaller than RFC 3550's recommended scaled reduced minimum timing interval of 360 divided by session bandwidth in kilobits/second or t(s) = 360/X(kbps).

For example, Figure 4 depicts two AVPCC flows and their relationship

with RTCP's reduced minimum interval: t(ms) = 360/X (Mbps). The two flows have data rates of 2 Mbps and 4 Mbps with RTTs of 70 ms and 130 ms, respectively.

The 4 Mbps flow's TFRC feedback requirements of 130 ms falls within RFC 3550's recommended reduced minimum interval for RTCP traffic. However the 2 Mbps flow's TFRC feedback requirement of once per 70 ms is more frequent than the 180 ms recommended by RFC 3550.

However in this case, it is safe to use TFRC's 70 ms interval, as at the rate of roughly one 88 octet RTCP compound packet per 70 ms, the feedback traffic for the 2 Mbps flow amounts to 10 kbps, that is less than 1% of the data flow and well with the 5% recommended by RFC 3550.

```
Bandwidth (Mbps)
      ^
      | \
      |  \
      |   \
      |    \                   360
      |     \         t(ms)= -------
      |      \               X (Mbps)
      |       \
      |        \_
   4  |         \__   x
      |            \___
   2  |       x        \____
      |                     \_________
      +----------------------------------->
              70      130
                    Time (ms)
```

Figure 4: Relationship between RFC 3550 recommended reduced minimum interval and session bandwidth (Mbps).

9. Open Issues

There are a number of open issues on the AVPCC on which we are soliciting input from the community:

o RFC 3550 recommends that the percentage of control traffic

relative to data, be fixed at 5%. For some flows, the feedback traffic for AVPCC may exceed this recommendation. Should AVPCC mandate a strict limit on the percentage of control traffic bandwidth? At what point is feedback too much feedback? (i.e., does it make sense for control traffic be 50% of data traffic?)

What are the implications of this limit, in terms of congestion control, for flows which cannot abide by the limit? This is particularly the case for low bandwidth flows, under 1 Mbps, and RTTs of say less than 10 ms.

- Security: Is it possible for the AVPCC to use the security mechanisms of SRTP as defined in RFC 3711 or is it necessary to define alternative security profile and mechanisms?

- RTT calculations by the sender: As an alternative to including t_i and t_delay in each RTCP packet, could the sender use the LSR and DLSR fields of the Receiver Reports to calculate the RTT? This is mainly a question of: is the frequency of the RTTs computed from Sender Reports sufficient for the sender to react to changes in the RTT and congestion?

- How does this profile relate to other RTP profiles?

10. IANA Considerations

The RTP profile for TCP Friendly Rate Control extends the profile for audio- visual conferences with minimal control and needs to be registered for the Session Description Protocol [SDP] as "RTP/AVPCC".

SDP Protocol ("proto"):

```
Name:               RTP/AVPCC
Long form:          RTP Profile for TCP Friendly Rate Control
Type of name:       proto
Type of attribute:  Media level only
Purpose:            RFC XXXX
Reference:          RFC XXXX
```

11. Security Considerations

See Section 9 (Open Issues).

12. Acknowledgments

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Informative References

14. IPR Notice

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