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Transmission Time Offsets in RTP Streams

Status of This Memo

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Abstract

This document describes a method to inform Real-time Transport Protocol (RTP) clients when RTP packets are transmitted at a time other than their 'nominal' transmission time. It also provides a mechanism to provide improved inter-arrival jitter reports from the clients, that take into account the reported transmission times.

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1. Introduction

In the Real-time Transport Protocol (RTP) specification [RFC3550], network jitter calculations are based on the presumption that packets are transmitted essentially in accordance with their RTP timestamps. This must be true, of course, on average over longer time intervals, as the client is playing the packets out according to those timestamps. However, for individual packets, this may not be true under some circumstances, such as:

- When the data rate of the stream is bursty, such as with video where I-frames may be significantly larger than P or B frames, traffic smoothing may need to be applied to maintain an appropriate data rate.

- In video that has forward-decode dependencies, frames may need to be transmitted in decoding order (the sequence number order) but with, of course, presentation timestamps. Under these circumstances, the transmission time of a frame sent early in sequence does not correspond to its RTP timestamp.

- When retransmissions are sent, the retransmitted packet clearly has a different actual transmission time from the original, even though they share the same timestamp.

Under some circumstances, it can help the receiver, or intermediate network elements, to know the actual transmission time of the packet. This RTP header extension element allows the communication of this information.

The RTP specification does not define a transmission timestamp; nor does this specification. This specification merely provides information on the relationship between the relative transmission times and relative RTP timestamps.

This specification allows the transmitter to indicate to the receiver any known variation between the spacing of transmission times and the spacing of RTP timestamps; any unreported variation introduced at or after the point of measurement of the transmission time will be treated as network jitter by the receiver. The definition of the point where the transmission time is measured or defined is left to the transmitter, though it should, of course, be consistent from packet to packet.

This information can also be of use to report the inter-arrival jitter caused by the network, excluding that introduced by the source. A new RTP Control Protocol (RTCP) packet is defined to enable this reporting.

2. Requirements Notation

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].

3. Transmission Offset

Classically, a pair of RTP packets with timestamps S2 and S1 are transmitted with a time interval between them of (S2 - S1). This specification permits sending an offset value O in each packet, O1 and O2. One characteristic of these offsets is that the original transmission interval can be deduced to be (S2 + O2) - (S1 + O1).

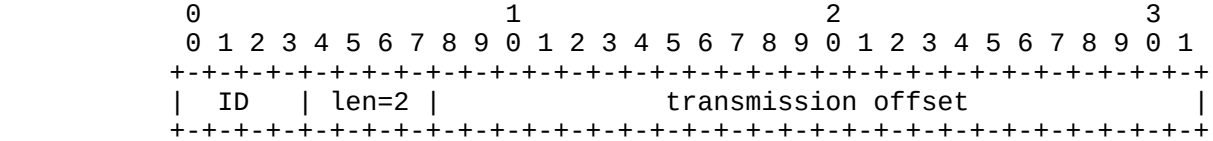
More precisely, the offset is defined as follows (with the function RtoN converting from RTP to Network Time Protocol (NTP) times, and NtoR doing the reverse):

- Take an RTP stream that has a recent RTCP sender report relating RTP timestamp S0 to NTP timestamp N0;
- Consider a packet sent after that with RTP timestamp S1. Nominally, this is sent at N1 = (N0 + RtoN(S1 - S0));
- If it was actually sent at a different time, Na, then the offset value O1 is O1 = NtoR(Na - N1).

The transmission time is signaled to the receiver in-band using the general mechanism for RTP header extensions [RFC5285]. The payload of this extension (the transmitted value) is a 24-bit signed integer. When added to the RTP timestamp of the packet, it represents the "effective" RTP transmission time of the packet, on the RTP timescale. The reported transmission time T1 of a packet with

timestamp S1 and an offset of O1, from the above equations, is T1 = S1+O1 (though of course the transmission time values only have meaning when two or more are compared).

The form of the transmission offset extension block is as follows:



The length field takes the value 2 to indicate that 3 bytes follow.

The sign of the offset value depends greatly on the choice of the initial mapping of RTP to NTP times. In general, without scanning a stream entirely it is not possible to ensure that this mapping would keep all the offsets positive; therefore, this specification allows negative values.

Imagine a stream with the following timestamps and sizes (in KB):

```
200    2 KB
300    4 KB
400    2 KB
500   12 KB
600   ...effective end of stream
```

This has 20 KB spread over 400 time units, i.e., on average, 1 KB per 20 time units. We traffic-smooth this, and establish that given a transmission time of x for the first packet, we would transmit the following packets at the given intervals later:

```
x + 000   2 KB
x + 040   4 KB
x + 120   2 KB
x + 160  12 KB
x + 400 ...effective end of stream
```

The choice of x is essentially arbitrary: only relative values of timestamps matter. Now, let’s say I claim on the first packet that it went out *at* its RTP timestamp, i.e., with an offset of 0, meaning that x is 200. Then the offset values are:

```
   0
 -60
 -80
-140
```

This is because in this case, I traffic-smooth by conceptually sending the small packets 'early'. But since only the relative values are significant, it is just as valid to say x is 400, whereupon the offset values are:

200
140
120
60

In a stream where this extension is not in effect (i.e., not declared or negotiated), the actual transmission offset is therefore unknown. However, when the extension is in effect for the stream, it MAY be omitted in those packets for which the offset is 0 (zero); that is, packets sent at their nominal time do not need this to be tagged with this extension. Therefore, the implied transmission time of an un-tagged RTP packet depends on whether the extension is in effect for the stream (and therefore the transmission offset is 0) or not (whereupon the transmission offset is unknown).

The jitter calculations performed by an RTP client MUST NOT use these transmission offsets. In general, the sender (or intermediate network elements doing RTP analysis) cannot always know whether the offsets have been taken into account or not. Therefore, for consistency, the jitter calculation should continue to operate on the 'raw' reception times. However, see Section 4 on extended jitter reports, below.

There are no extensionattributes defined for this extension.

It is structurally possible to have more than one extension of the same type in a packet. However, this extension is only defined for the source to report. Intermediate network nodes that are not the source of the RTP session MUST NOT add this extension (whether or not it was previously present) and MUST NOT alter the existing transmission offset value in a packet, if the extension is already present.

(Of course, it is clear that network elements that terminate an RTP flow, and are the source for a new RTP flow, can add a transmission offset extension header to the RTP packets of the new flow, if desired.)

4. Extended Jitter Reports

The inter-arrival jitter computed as defined in Section 6.4.1 of RFC 3550 provides inter-arrival jitter reports that include any source-introduced jitter (transmission time offsets). If it is desired to

indicate the actual network jitter, excluding the source-introduced jitter, the new RTCP packet type defined here may be used.

It has the following form:

If present, this RTCP packet must be placed after a receiver report (inside a compound RTCP packet), and MUST have the same value for RC (reception report count) as the receiver report. The content is exactly that number of inter-arrival jitter calculations, calculated using the same formula as for sender and receiver reports, but taking into account the transmission offsets for the streams (if any). That is, the formula uses the values T1=S1+O1, T2, etc., as defined above, instead of S1, S2, etc. (If no transmission offset information is given for a stream, then the value of inter-arrival jitter in this packet and in the receiver report will be identical).

Precisely, the replacement equation for the equation in the RTP specification is as follows, where Rj is the most recent arrival time:

```
D(i,j) = (Rj - Ri) - ((Sj + Oj) - (Si + Oi))
       = (Rj - (Sj + Oj)) - (Ri - (Si + Oi))
```

5. Signaling (Setup) Information

The URI for declaring this header extension in an extmap attribute is "urn:ietf:params:rtp-hdrext:toffset". There is no additional setup information needed for this extension (no extensionattributes).

6. Security Considerations

The given transmission offsets are only informative, and it is hard to see security considerations from associating them with media streams.

The underlying security considerations of [RFC3550] should be taken into account.

It is possible that malicious senders (or systems tampering with packets in transit) could send offsets that are implausible, could confuse the receiver, or result in calculated jitter values that might mislead the sender. Both the sender and receiver of the transmission offsets and jitter values should take care that such behavior does not result in denial of service or other problems.

7. IANA Considerations

The RTCP packet type used for the adjusted inter-arrival jitter has been registered, in accordance with Section 15 of [RFC3550]. IANA has added a new value to the RTCP Control Packet types subregistry of the Real-Time Transport Protocol (RTP) Parameters registry, according to the following data:

abbrev.	name	value	Reference
IJ	Extended inter-arrival jitter report	195	RFC 5450

Additionally, IANA has registered a new extension URI to the RTP Compact Header Extensions subregistry of the Real-Time Transport Protocol (RTP) Parameters registry, according to the following data:

```
Extension URI: urn:ietf:params:rtp-hdrext:toffset
Description:   Transmission Time offsets
Contact:       singer@apple.com
Reference:     RFC 5450
```

8. Acknowledgments

Ron Frederick, Colin Perkins, and Steve Casner all contributed substantially to this document, and their help and contributions helped turn an idea into a specification.

9. Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

[RFC3550] Schulzrinne, H., Casner, S., Frederick, R., and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications", STD 64, RFC 3550, July 2003.

[RFC5285] Singer, D. and H. Desineni, "A General Mechanism for RTP Header Extensions", RFC 5285, July 2008.

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