RTP No-Op Payload Format

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Abstract

This document defines an no-op payload format for the Real-time Transport Protocol (RTP), and a mechanism to request an immediate RTCP report. This can be used to verify RTP connectivity and to keep Network Address Translator (NAT) bindings and Firewall pinholes open.
Requirements Language
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 [1].

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1. Introduction

This memo defines a new RTP payload format called "no-op". This payload behaves like a normal RTP payload, except that it isn't played by the receiver.

This new payload format is useful for:

- bearer continuity testing, such as at the beginning of a call;
- keepalives to keep NAT bindings open when RTP media traffic is not otherwise being transmitted;
- performing an RTP traceroute;

For testing the RTP path, an RTP sender may transmit several No-Op payload packets with the Request Immediate RTCP bit set to 0, followed by one No-Op payload packet with the Request Immediate RTCP bit set to 1. This would cause the RTP receiver to send an RTCP report indicating the quality of the RTP path. The RTP sender could then decide to continue with call setup, abort the session, or perform some other action.

2. RTP Payload Format for No-Op

The no-op payload format is carried as part of the RTP stream, and MUST use the same sequence number space, SSRC, and timestamp base as the regular media.

2.1 Registration

The RTP payload format is designated as "no-op" and the MIME type as "audio/no-op". The default clock rate is 8000 Hz, but other rates MAY be used. In accordance with current practice, this payload format does not have a static payload type number, but uses a RTP payload type number established dynamically and out-of-band.

2.2 Use of RTP Header Fields

Timestamp: The RTP timestamp reflects the measurement point for the current packet. The receiver calculates jitter for RTCP receiver reports based on all packets with a given timestamp. Note: The jitter value should primarily be used as a means for comparing the reception quality between two users or two time-periods, not as an absolute measure.

Marker bit: The RTP marker bit has no special significance for this payload type.
2.3 Payload Format

The payload format is shown below.

```
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
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|R|                         reserved                            |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                      padding (OPTIONAL)                       |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

The payload contains at least 4 bytes. The first 32 bits are defined as follows:

bit 0:  "R", "Request immediate RTCP", is used to request transmission of an immediate RTCP report (see Section 2.6).
bits 1-31: Reserved; contents are ignored.

Additional padding bytes MAY be appended up to the negotiated ptime value in SDP (see Section 2.6). These bytes MUST contain all 0 bits. Padding may be useful to generate RTP packets that are the same size as a normal media payload.

2.4 Sender Operation

A source MAY send normal RTP audio and the no-op payload format for the same time instants (but with different sequence numbers of course). This might be done in conjunction with this payload format’s "Request Immediate RTCP" opcode.

2.5 Mixer, Translator Operation

An RTP mixer or unicast-to-unicast RTP translator SHOULD forward RTP No-Op payload packets normally. A unicast-to-multicast RTP translator SHOULD replicate RTP No-Op payload packets normally.

A multicast-to-unicast RTP translator SHOULD NOT replicate an RTP No-Op packet with the Request Immediate RTCP bit set, because the receivers won’t be able to prevent flooding of the multicast RTP sender.

2.6 Receiver Operation
Upon receipt of an RTP packet with the No-Op payload format and the Send Immediate RTCP Report bit set to 0, the receiver performs normal
RTP receive operations on it -- incrementing the RTP receive counter, calculating jitter, and so on. The receiver then discards the packet -- it is not used to play out data.

Upon receipt of an RTP packet with the No-Op payload format and the Send Immediate RTCP Report bit set to 1, the receiver performs the steps above and:

- if listening on a multicast IP address, the receiver MUST not send an immediate RTCP report, and the receiver MUST follow the normal RTCP transmission rules RFC3550, sections 6.2 and 6.3 [2], and
- if listening on a unicast IP address and sending RTP traffic, the receiver prepares to send an RTCP sender report, and
- if listening on a unicast IP address and receiving RTP traffic, the receiver prepares to send an RTCP receiver report.

In all cases, before actually sending its RTCP report, the RTCP bandwidth limits and randomization interval MUST be observed RFC3550, sections 6.2 and 6.3 [2], most especially when multiple SSRCs are in the same session.

2.7 Indication of No-OP Capability using SDP

Senders and receivers may indicate support for the No-Op payload format, for example, by using the Session Description Protocol SDP [3].

If successful completion of RTP No-Op is required before completing call establishment -- such as to verify the existence or quality of the bearer path -- connectivity preconditions [4] can be used.

The default packetization interval for this payload type is 20ms (ptime:20) but alternate values can be advertised in SDP using the ptime attribute value [3].

3. MIME Registration

3.1 audio/no-op

MIME media type name: audio

MIME subtype name: no-op

Required parameters: none
Optional parameters: none

Encoding considerations: This type is only defined for transfer via
RTP [2].


Interoperability considerations: none

Published specification: This document.

Applications which use this media: The "no-op" audio subtype is used to maintain network state or verify network connectivity, when a more traditional RTP payload type cannot be used.

Additional information:

1. Magic number(s): N/A
2. File extension(s): N/A
3. Macintosh file type code: N/A

4. Security Considerations

Without security of the RTP stream (via SRTP [6], IPsec [5], or other means), it is possible for an attacker to spoof RTP packets, including this new packet type. As this new RTP payload type includes a method to request immediate transmission of RTCP, this could be used to cause endpoints to flood the network with RTCP reports. Thus, the RTCP transmissions MUST NOT exceed the bandwidth recommendations described in section 6.3 of RFC3550 [2].

5. Open Issues

1. Issues brought up during IETF59:
   A. Roni Even asked why only the audio/noop MIME type is listed? The answer is that there is no reason for not having it for all media types in use. The draft is simply missing registration of all the media types that are appropriate.
   B. Dan Romascanu noted that it is not very clear from the draft how you can determine the actual media path characteristics through the use of RTCP, and how to separate the reception quality reports for the NOOP traffic from other media traffic. Flemming noted that the RTP SSRC can be used to separate traffic, and promised to work on the text, clarifying this.
   C. Magnus Westerlund and Colin Perkins noted that the draft is unclear on the interactions between its immediate feedback request and the usual RTCP timers (in both the standard A/V
profile and in the RTCP feedback profile). Clarification was requested.
D. Jonathan Rosenberg was doubtful of the need to request immediate RTCP reports. He supported the functionality of an RTP payload format that is discarded, to keep NAT bindings alive when media is on hold, but asked the group to consider the need for requesting immediate RTCP reports and to clarify the problem and requirements before jumping into the solution space.

E. Jonathan also noted, as he did in the MMUSIC session, that ICE and STUN would still be needed for NAT traversal, and so can’t be replaced with this mechanism.

F. Roni Even also commented that the use case for checking the MTU might need feedback, however the QoS diagnostic does look like a different issue.

G. Colin Perkins concluded that the draft must be updated and clarified on use cases, and what problems it solves.

2. Discuss how this relates to AVPF.

3. Need to expand on the RTCP transmission interval considerations and explain in more detail when you can and cannot transmit (i.e., does not modify/violate normal RTCP considerations)

4. Should "request immediate RTCP" be a generic mechanism?

5. Clarify usefulness of Noop for diagnostics.

6. Clarify how the packet counting, etc. works. Should explain a bit more in document (e.g. how everything is on a per SSRC basis, packet counting works and is revealed in the RTCP reports, etc.). The document isn’t clear on this.

6. Acknowledgments

Thanks to Henning Schulzrinne for suggesting using RTCP as a feedback mechanism.

7. References

7.1 Normative References


7.2 Informational References


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