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A No-Op Payload Format for RTP
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Status of this Memo

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Abstract

This document defines an no-op payload format for the Real-time Transport Protocol (RTP). This packet is not played out by receivers. It can be useful as a way to keep Network Address Translator (NAT) bindings and Firewall pinholes open. Other uses are discussed in the document.

Table of Contents

1. Introduction 3
2. Conventions Used in this Document 4
3. RTP Payload Format for No-Op 4
3.1. Registration 4
3.2. Use of RTP Header Fields 4
3.3. Payload Format 5
3.4. Sender Operation 5
3.5. Mixer, Translator Operation 5
3.6. Receiver Operation 6
3.7. Indication of No-OP Capability using SDP 6
4. Example SDP Offer/Answer 6
5. Media Type Registration 7
5.1. audio/no-op 7
5.2. video/no-op 8
5.3. text/no-op 8
6. Security Considerations 9
7. IANA Considerations 9
8. Acknowledgments 10
9. References 10
9.1. Normative References 10
9.2. Informational References 10
Authors' Addresses 11
Intellectual Property and Copyright Statements 12

1. Introduction

This memo defines a new RTP payload format called "no-op". This payload behaves like a normal RTP payload, except the RTP packet is not used to play out media.

This new payload format is useful for:

- o facilitating media session reception quality assessment, such as at the beginning of a session;
- o keepalives to keep NAT bindings and/or firewall pinholes open when RTP media traffic is not otherwise being transmitted.
- o measurement-based admission control by probing available bandwidth, and
- o synthetic load generation for performance testing and other minimally-intrusive instrumentation.

When an endpoint has a media stream marked as 'recvonly' or 'inactive' the endpoint is not supposed to send any media (i.e., RTP packets). However, to keep a NAT binding alive, the endpoint will need to periodically send packets over the RTP and RTCP ports. RTP No-Op is ideally suited to this. In comparison, if one participant in an audio multicast conference has a 'recvonly' or 'inactive' media stream yet occasionally sends comfort noise packets in order to keep its NAT binding open, these comfort noise packets are interpreted as audio packets by receivers and mixers which can cause undesirable behavior -- such as selection of the primary speaker or the playout of comfort noise when no audio should be played.

Unlike Comfort noise [RFC3389], which is specific to voice RTP streams, RTP No-Op is applicable to any kind of RTP stream including video, audio, realtime text, or any other media types that would benefit from the capabilities listed above. This gives RTP No-Op an advantage as a NAT keepalive mechanism. Certain functions and RTP payload types can use RTP No-Op without re-inventing their own payload-specific NAT keepalive mechanism -- such as video muting, Clearmode [RFC4040], and text [RFC4103].

Some audio codecs have their own 'silence' packets. However, some codecs only send such silence packets if the noise floor changes; G.729b [G729B] is an example of such a codec. RTP No-Op allows the RTP stack itself, rather than the codec, to send periodic packets as a keepalive mechanism.

Multiplexing RTP and RTCP over the same port

[I-D.ietf-avt-rtp-and-rtcp-mux] provides an separate keepalive mechanism which uses the periodic RTCP transmission to keep middleboxes aware of the flow.

2. 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 [RFC2119].

3. RTP Payload Format for No-Op

3.1. Registration

The RTP payload format is designated as "no-op" and the media types are "audio/no-op", "video/no-op", and "text/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 out-of-band, e.g., through SDP [RFC4566].

3.2. Use of RTP Header Fields

Timestamp: The RTP timestamp reflects the measurement point for the current packet. The receiver uses this timestamp to calculate jitter for RTCP sender and receiver reports per normal RTP procedures. 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.

3.3. 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
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |                            reserved                           |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
   |                       padding (OPTIONAL)                      |
   |                              ....                             |
   +-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

Figure 1: Payload Format

The payload contains at least 4 bytes, the first 32 bits are reserved for future use. These bits SHOULD be set to 0. Receivers MUST ignore the value of these bits.

Additional padding bytes MAY be appended up to the ptime or maxptime value in SDP (see Section 3.7). These bytes MUST be ignored. Padding may be useful to generate RTP packets that are the same size as a normal media payload.

3.4. Sender Operation

As discussed in the introduction, endpoints must occasionally send a packet to their RTP and RTCP peer to keep NAT and firewall bindings active, even if the media stream is marked 'recvonly' or 'inactive'. No matter if the media stream is marked 'recvonly', 'sendrecv', 'sendonly', or 'inactive', if approximately 20 seconds elapse with no packets transmitted from the RTP port (either RTP packets or non-RTP packets (e.g., STUN [I-D.ietf-behave-rfc3489bis] packets), then an RTP No-Op packet SHOULD be sent.

3.5. Mixer, Translator Operation

An RTP mixer or unicast-to-unicast RTP translator SHOULD forward RTP No-Op payload packets normally; if the input stream is made up of RTP No-Op packets only, a corresponding RTP No-Op packet SHOULD be generated. If the input stream consists of other packets than No-Op, then the No-Op packets SHOULD simply be discarded. A unicast-to-multicast RTP translator SHOULD replicate RTP No-Op payload packets normally.

3.6. Receiver Operation

Upon receipt of an RTP packet with the No-Op payload format 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 media.

3.7. 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 [RFC4566].

The default packetization interval for this payload type is 20ms but alternate values can be advertised in SDP using the ptime or maxptime attributes [RFC4566].

4. Example SDP Offer/Answer

Offer:

```
v=0
o=alice 2890844526 2890844526 IN IP4 host.atlanta.example.com
s=-
c=IN IP4 host.atlanta.example.com
t=0 0
m=audio 49170 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 no-op/8000
m=video 41372 RTP/AVP 31 96
a=rtpmap:31 H261/90000
a=rtpmap:96 no-op/90000
```

Answer:

```
v=0
o=bob 2808844564 2808844564 IN IP4 host.biloxi.example.com
s=-
c=IN IP4 host.biloxi.example.com
t=0 0
m=audio 59174 RTP/AVP 0 96
a=rtpmap:0 PCMU/8000
a=rtpmap:96 no-op/8000
m=video 59170 RTP/AVP 32 96
a=rtpmap:31 H261/90000
a=rtpmap:96 no-op/90000
```

5. Media Type Registration

This section registers media types for audio/no-op, video/no-op, and text/no-op, per [RFC4855].

5.1. audio/no-op

Media type name: audio

Subtype name: no-op

Required parameters: none

Optional parameters: none

Encoding considerations: This media type is framed and binary; see Section 4.8 in [RFC4288].

Security considerations: See Section 6, "Security Considerations", in this document.

Interoperability considerations: none

Published specification: This document.

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

Additional information: none.

Person and email address to contact for further information: Dan Wing <dwing@cisco.com>.

Intended usage: COMMON

Restrictions on usage: This media type depends on RTP framing and is only defined for transfer via RTP [RFC3550]. Transfer within other framing protocols is not defined at this time.

Author: Flemming Andreasen, David Oran, and Dan Wing

Change controller: IETF Audio/Video Transport working group delegated from the IESG.

5.2. video/no-op

Media type name: video

Subtype name: no-op

Required parameters: none

Optional parameters: none

Encoding considerations: This media type is framed and binary; see Section 4.8 in [RFC4288].

Security considerations: See Section 6, "Security Considerations", in this document.

Interoperability considerations: none

Published specification: This document.

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

Additional information: none.

Person and email address to contact for further information: Dan Wing <dwing@cisco.com>.

Intended usage: COMMON

Restrictions on usage: This media type depends on RTP framing and is only defined for transfer via RTP [RFC3550]. Transfer within other framing protocols is not defined at this time.

Author: Flemming Andreasen, David Oran, and Dan Wing

Change controller: IETF Audio/Video Transport working group delegated from the IESG.

5.3. text/no-op

Media type name: audio

Subtype name: no-op

Required parameters: none

Optional parameters: none

Encoding considerations: This media type is framed; see Section 4.8 in [RFC4288].

Security considerations: See Section 6, "Security Considerations", in this document.

Interoperability considerations: none

Published specification: This document.

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

Additional information: none.

Person and email address to contact for further information: Dan Wing <dwing@cisco.com>.

Intended usage: COMMON

Restrictions on usage: This media type depends on RTP framing and is only defined for transfer via RTP [RFC3550]. Transfer within other framing protocols is not defined at this time.

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Change controller: IETF Audio/Video Transport working group delegated from the IESG.

6. Security Considerations

There are no additional security considerations for this new RTP payload format; the RTP security considerations from RTP [RFC3550] apply.

7. IANA Considerations

IANA is requested to make media type registrations as specified above in Section 5

8. Acknowledgments

The authors thank Bob Biskner and Rajesh Kumar for their contributions to this specification.

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