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Duplicating RTP Streams
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Abstract

Packet loss is undesirable for real-time multimedia sessions, but can occur due to congestion, or other unplanned network outages. This is especially true for IP multicast networks, where packet loss patterns can vary greatly between receivers. One technique that can be used to recover from packet loss without incurring unbounded delay for all the receivers is to duplicate the packets and send them in separate redundant streams. This document explains how Real-time Transport Protocol (RTP) streams can be duplicated without breaking RTP media streams, or RTP Control Protocol (RTCP) rules.

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1. Introduction

The Real-time Transport Protocol (RTP) [RFC3550] is widely used today for delivering IPTV traffic, and other real-time multimedia sessions. Many of these applications support very large numbers of receivers, and rely on intra-domain UDP/IP multicast for efficient distribution of traffic within the network.

While this combination has proved successful, there does exist a weakness. As [RFC2354] noted, packet loss is not avoidable, even in a carefully managed network. This loss might be due to congestion, it might also be a result of an unplanned outage caused by a flapping link, link or interface failure, a software bug, or a maintenance person accidentally cutting the wrong fiber. Since UDP/IP flows do not provide any means for detecting loss and retransmitting packets, it leaves up to the RTP layer and the applications to detect, and recover from, packet loss.

One technique to recover from packet loss without incurring unbounded delay for all the receivers is to duplicate the packets and send them in separate redundant streams. Variations on this idea have been implemented and deployed today [IC2011]. However, duplication of RTP streams without breaking the RTP and RTCP functionality has not been documented properly. This document explains how duplication can be achieved for RTP streams.

Stream duplication offers a simple way to protect media flows from packet loss. It has a comparatively high bandwidth overhead, since everything is sent twice, but with a low processor overhead. It is also very predictable in its overheads. Alternative approaches may be suitable in some cases, for example retransmission-based recovery [RFC4588] or forward error correction [RFC5109].

2. Terminology and Requirements Notation

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in [RFC2119].

3. Dual Streaming Use Cases

Dual streaming refers to a technique that involves transmitting two redundant RTP streams of the same content, with each stream capable of supporting the playback when there is no packet loss. Therefore, adding an additional RTP stream provides a protection against packet

loss. The level of protection depends on how the packets are sent and transmitted inside the network.

It is important to note that dual streaming can easily be extended to support cases when more than two streams are desired. However, using three or more streams is rare in practise, due to the high overhead that it incurs.

3.1. Temporal Redundancy

From a routing perspective, two streams are considered identical if the following two IP header fields are the same, since they will be both routed over the same path:

- IP Source Address

- IP Destination Address

Two routing-plane identical RTP streams might carry the same payload, but can use different Synchronization Sources (SSRC) to differentiate the RTP packets belonging to each stream. In the context of dual RTP streaming, we assume that the source duplicates the RTP packets and sends them in separate RTP streams, each with a unique SSRC. All the redundant streams are transmitted in the same RTP session.

For example, one main and one redundant RTP stream can be sent to the same IP destination address and UDP destination port with a certain delay between them [I-D.begen-mmusic-temporal-interleaving]. The streams carry the same payload in their respective RTP packets with identical sequence numbers. This allows receivers (or other nodes responsible for gap filling and duplicate suppression) to identify and suppress the duplicate packets, and subsequently produce a hopefully loss-free and duplication-free output stream. This process is called stream merging.

3.2. Spatial Redundancy

An RTP source might be associated with multiple network interfaces, allowing it to send two redundant streams from two separate source addresses. Such streams can be routed over diverse or identical paths depending on the routing algorithm used inside the network. At the receiving end, the node responsible for duplicate suppression can look into various RTP header fields, for example SSRC and sequence number, to identify and suppress the duplicate packets.

If source-specific multicast (SSM) transport is used to carry such redundant streams, there will be a separate SSM session for each redundant stream since the streams are sourced from different

interfaces (i.e., IP addresses). Thus, the receiving host has to join each SSM session separately.

Alternatively, an RTP source might send the redundant streams to separate IP destination addresses.

3.3. Dual Streaming over a Single Path or Multiple Paths

Having described the characteristics of the streams, one can reach the following conclusions:

1. When two routing-plane identical streams are used, the two streams will have identical IP headers. This makes it impractical to forward the packets onto different paths. In order to minimize packet loss, the packets belonging to one stream are often interleaved with packets belonging to the other, and with a delay, so that if there is a packet loss, such a delay would allow the same packet from the other stream to reach the receiver because the chances that the same packet is lost in transit again is often small. This is what is also known as Time-shifted Redundancy, Temporal Redundancy or simply Delayed Duplication [I-D.begen-mmusic-temporal-interleaving] [IC2011]. This approach can be used with both types of dual streaming, described in Section 3.1 and Section 3.2.

2. If the two streams have different IP headers, an additional opportunity arises in that one is able to build a network, with physically diverse paths, to deliver the two streams concurrently to the intended receivers. This reduces the delay when packet loss occurs and needs to be recovered. Additionally, it also further reduces chances for packet loss. An unrecoverable loss happens only when two network failures happen in such a way that the same packet is affected on both paths. This is referred to as Spatial Diversity or Spatial Redundancy [IC2011]. The techniques used to build diverse paths are beyond the scope of this document.

   Note that spatial redundancy often offers less delay in recovering from packet loss provided that the forwarding delay of the network paths are more or less the same. For both temporal and spatial redundancy approaches, packet misordering might still happen and needs to be handled using the sequence numbers of some sort (e.g., RTP sequence numbers).

To summarize, dual streaming allows an application and a network to work together to provide a near zero-loss transport with a bounded or minimum delay. The additional advantage includes a predictable bandwidth overhead that is proportional to the minimum bandwidth

needed for the multimedia session, but independent of the number of receivers experiencing a packet loss and requesting a retransmission. For a survey and comparison of similar approaches, refer to [IC2011].

4. Use of RTP and RTCP with Temporal Redundancy

To achieve temporal redundancy, the main and redundant RTP streams MUST be sent using the same 5-tuple of transport protocol, source and destination IP addresses, and source and destination transport ports. This is perhaps overly restrictive, but with the possible presence of network address and port translation (NAPT) devices, using anything other than an identical 5-tuple can also cause spatial redundancy.

Since main and redundant RTP streams follow an identical path, they are part of the same RTP session. Accordingly, the sender MUST choose a different SSRC for the redundant RTP stream than it chose for the main RTP stream, following the rules in [RFC3550] Section 8.

4.1. RTCP Considerations

If RTCP is being sent for the main RTP stream, then the sender MUST also generate RTCP for the redundant RTP stream. The RTCP for the redundant RTP stream is generated exactly as-if the redundant RTP stream were a regular media stream. The sender MUST NOT duplicate the RTCP packets sent for the main RTP stream when sending the duplicate stream, instead it MUST generate new RTCP reports for the duplicate stream. The sender MUST use the same RTCP CNAME in the RTCP reports it sends for the main and redundant streams, so that the receiver can synchronize them.

Both the main and redundant RTP streams, and their corresponding RTCP reports, will be received. If RTCP is used, receivers MUST generate RTCP reports for both main and redundant streams in the usual way, treating them as entirely separate media streams.

4.2. Signaling Considerations

Signaling is needed to allow the receiver to determine that an RTP stream is a redundant copy of another, rather than a separate stream that needs to be rendered in parallel. There are two parts to this: an SDP extension is needed in the offer/answer exchange to negotiate support for temporal redundancy; and signalling is needed to indicate which stream is the duplicate (the latter can be done in-band using an RTCP extension, or out-of-band by signalling the SSRCs used by the duplicate streams in SDP).

We require out-of-band signalling for both features. The required

SDP attribute to signal duplication in the SDP offer/answer exchange ('duplication-delay') is defined in [I-D.begen-mmusic-temporal-interleaving]. The required SDP grouping semantics are defined in [I-D.begen-mmusic-redundancy-grouping].

In the following SDP example, a video stream is duplicated, and the main and redundant streams are transmitted in two separate SSRCs (1000 and 1010):

```
v=0
o=ali 1122334455 1122334466 IN IP4 dup.example.com
s=Delayed Duplication
t=0 0
m=video 30000 RTP/AVP 100
c=IN IP4 233.252.0.1/127
a=source-filter:incl IN IP4 233.252.0.1 198.51.100.1
a=rtpmap:100 MP2T/90000
a=ssrc:1000 cname:ch1@example.com
a=ssrc:1010 cname:ch1@example.com
a=ssrc-group:DUP 1000 1010
a=duplication-delay:100
a=mid:Group1
```

It is RECOMMENDED that the SSRC listed first in the "a=ssrc-group:" line is sent first, with the other RTP SSRC being the time-delayed duplicate. This is not critical, however, and receivers should size their playout buffers based on the "a=duplication-delay:" attribute, and play the stream that arrives first in preference, with the other stream acting as a repair stream, irrespective of the order in which they are signalled.

5. Use of RTP and RTCP with Spatial Redundancy

When using spatial redundancy, the redundant RTP stream is sent on using a different source and/or destination address/port pair. This will be a separate RTP session to the session conveying the main RTP stream.

The SSRCs used for the main and redundant streams MUST be chosen randomly, following the rules in Section 8 of [RFC3550]. Accordingly, they will almost certainly not match each other. The sender MUST, however, use the same RTCP CNAME for both the main and redundant streams, and MUST include an "a=ssrc:... srcname:..." attribute to correlate the flows. An "a=group:DUP" attribute is used to indicate duplication.

5.1. RTCP Considerations

If RTCP is being sent for the main RTP stream, then the sender MUST also generate RTCP for the redundant RTP stream. The RTCP for the redundant RTP stream is generated exactly as-if the redundant RTP stream were a regular media stream; the sender MUST NOT duplicate the RTCP packets sent for the main RTP stream. The sender MUST use the same RTCP CNAME in the RTCP reports it sends for the main and redundant streams, so that the receiver can synchronize them.

The main and redundant streams are conceptually synchronised using the standard RTCP SR-based mechanism, deriving a mapping between their timelines. The RTP timestamps and sequence numbers SHOULD be identical in the main and redundant streams, however, making the mapping trivial in most cases.

Both main and redundant streams, and their corresponding RTCP, will be received. If RTCP is used, receivers MUST generate RTCP reports for both main and redundant streams in the usual way, treating them as entirely separate media streams.

5.2. Signaling Considerations

The required SDP grouping semantics have been defined in [I-D.begen-mmusic-redundancy-grouping]. In the following example, the redundant streams have different IP destination addresses. The example shows the same UDP port number and IP source addresses, but either or both could have been different for the two streams.

```
v=0
o=ali 1122334455 1122334466 IN IP4 dup.example.com
s=DUP Grouping Semantics
t=0 0
a=group:DUP S1a S1b
m=video 30000 RTP/AVP 100
c=IN IP4 233.252.0.1/127
a=source-filter:incl IN IP4 233.252.0.1 198.51.100.1
a=rtpmap:100 MP2T/90000
a=ssrc:1000 cname:ch1@example.com
a=ssrc:1000 srcname:45:a8:f4:19:b4:c3
a=mid:S1a
m=video 30000 RTP/AVP 101
c=IN IP4 233.252.0.2/127
a=source-filter:incl IN IP4 233.252.0.2 198.51.100.1
a=rtpmap:101 MP2T/90000
a=ssrc:1010 cname:ch1@example.com
a=ssrc:1010 srcname:45:a8:f4:19:b4:c3
a=mid:S1b
```

6. Use of RTP and RTCP with Temporal and Spatial Redundancy

This uses the same RTP/RTCP mechanisms, plus a combination of both sets of signaling.

7. Security Considerations

The security considerations of [RFC3550], [I-D.begen-mmusic-temporal-interleaving], and [I-D.begen-mmusic-redundancy-grouping] apply.

If stream de-duplication is done by an in-network middlebox, rather than by an end system, that middlebox can work if Secure RTP (SRTP) encryption is used [RFC3711], since the RTP headers are in the clear. Doing so would break the authentication when the SSRC is rewritten, unless the de-duplication middlebox were trusted to re-authenticate the packets. This would require additional signalling which is not specified here, since de-duplication in the receiver end system is expected to be the more common use case.

8. IANA Considerations

No IANA actions are required.

9. Acknowledgments

Thanks to Magnus Westerlund for his suggestions.

10. References

10.1. Normative References

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10.2. Informative References

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[RFC4588] Rey, J., Leon, D., Miyazaki, A., Varsa, V., and R. Hakenberg, "RTP Retransmission Payload Format", RFC 4588, July 2006.

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