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Duplication Grouping Semantics in the Session Description Protocol
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Abstract

Packet loss is undesirable for real-time multimedia sessions, but it is not avoidable due to congestion or other unplanned network outages. This is especially the case for IP multicast networks. 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. This document defines the semantics for grouping redundant streams in the Session Description Protocol (SDP). The semantics defined in this document are to be used with the SDP Grouping Framework [RFC5888]. SSRC-level (Synchronization Source) grouping semantics are also defined in this document for RTP streams using SSRC multiplexing.

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1. Introduction

RTP [RFC3550] transport is widely used today for delivering real-time multimedia streams. Most of the applications also rely on IP multicast to reach many receivers efficiently.

While the combination proves successful, there does exist a weakness. As [RFC2354] noted, packet loss is not avoidable. This 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 does not provide any means for detecting loss and retransmitting packets, it leaves up to the RTP or the applications to detect and recover from the loss. For retransmission-based recovery, one example is described in [RFC4588].

In this document, we describe a technique that involves transmitting redundant streams to overcome packet loss. Variations of this technique have already been implemented and deployed today [IC2011]. We also describe the semantics needed in the Session Description Protocol (SDP) [RFC4566] to support this technique.

A work-in-progress draft specification [I-D.singh-avtcore-mprtp] proposes changes to the RTP protocol so that a single RTP session can benefit from using multiple paths between two endpoints (to increase the aggregated throughput and improve reliability). While we also discuss spatial diversity in this document, we use diverse paths solely for sending redundant streams. For our purposes, we do not require changes in the RTP protocol.

2. 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

Dual streaming refers to a technique that involves transmitting two redundant (often RTP) streams of the same content, with each stream itself capable of supporting the playback when there is no packet loss. Therefore, adding an additional 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 the technique and specification described by this document can easily be extended to support cases when more than two streams are desired. But triple, quadruple, or more, streaming is rarely used in practice.

3.1. (Routing-Plane) Identical Streams

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

- o IP Source Address

- o IP Destination Address

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

For example, two redundant RTP streams 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 the receiver (or any other node responsible for duplicate suppression) to identify and suppress the duplicate packets, and subsequently produce a hopefully loss-free and duplication-free output stream (called stream merging).

In such a scenario (where the RTP streams are routing-plane identical and share the same UDP destination port), there will be only one "m" line in the SDP description regardless of how many redundant streams are generated. Thus, the SDP Grouping Framework [RFC5888] cannot be used to indicate the grouping for the redundant streams. Instead, the 'ssrc-group' attribute [RFC5576] with new semantics has to be used to describe the redundancy relation (See Section 4.2).

If the two routing-plane identical RTP streams were sent to different UDP destination ports, there would have been two "m" lines in the SDP description and in this case, the 'group' attribute [RFC5888] with new semantics would have to be used to describe the redundancy relation (See Section 4.1).

3.2. Using Separate Source Interfaces

An RTP source might have multiple network interfaces associated with it and it can send two redundant streams from two separate interfaces. Such streams can be routed over diverse or identical paths depending on the routing algorithm inside the network. At the receiving end, the node responsible for duplicate suppression can look into various RTP related fields 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. The receiving host has to join each SSM session separately via Internet Group Management Protocol (IGMP) version 3 [RFC3376] or the Multicast Listener Discovery Protocol (MLD) version 2 [RFC3810]. Note that despite being transmitted in separate SSM sessions, there is still only one RTP session and the redundant streams still have to use unique SSRC identifiers.

3.3. Using Separate Destination Addresses and/or Ports

An RTP source might send the redundant streams to separate IP destination addresses and/or UDP ports. In this case, there will be multiple "m" lines in the SDP description and the 'group' attribute [RFC5888] with new semantics will be used to describe the redundancy relation.

3.4. 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 all three types of dual streaming described in Section 3.1, Section 3.2 and Section 3.3.

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 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. Duplication Grouping

4.1. "DUP" Grouping Semantics

Each "a=group" line is used to indicate an association relationship between the redundant streams. The streams included in one "a=group" line are called a Duplication Group.

Using the framework in [RFC5888], this document defines "DUP" as the grouping semantics for redundant streams.

The "a=group:DUP" semantics MUST be used to group the redundant streams except when the streams are specified in the same media description, i.e., in the same "m" line (See Section 4.2).

The SSRC identifiers for the RTP streams that are carried in the same RTP session MUST be unique per [RFC3550]. Thus, each redundant RTP stream MUST have its own unique SSRC identifier. This way, dual streaming does not break RTCP reporting. When the redundant streams are described in separate "m" lines and the ’group’ attribute is used to describe the redundancy relation, the SSRCs for each redundant stream MUST be announced in the SDP description using the 'ssrc' attribute [RFC5576].

4.2. DUP Grouping for SSRC-Multiplexed RTP Streams

[RFC5576] defines an SDP media-level attribute, called 'ssrc-group', for grouping the RTP streams that are SSRC multiplexed and carried in the same RTP session. The grouping is based on the SSRC identifiers. Since SSRC-multiplexed RTP streams are defined in the same "m" line, the 'group' attribute cannot be used.

This section specifies how duplication is used with SSRC-multiplexed streams using the 'ssrc-group' attribute [RFC5576].

The semantics of "DUP" for the 'ssrc-group' attribute are the same as the one defined for the 'group' attribute except that the SSRC identifiers are used to designate the duplication grouping associations: a=ssrc-group:DUP *(SP ssrc-id) [RFC5576].

4.3. SDP Offer/Answer Model Considerations

When offering duplication grouping using SDP in an Offer/Answer model [RFC3264], the following considerations apply.

A node that is receiving an offer from a sender may or may not understand line grouping. It is also possible that the node understands line grouping but it does not understand the "DUP" semantics. From the viewpoint of the sender of the offer, these cases are indistinguishable.

When a node is offered a session with the "DUP" grouping semantics but it does not support line grouping or the duplication grouping semantics, as per [RFC5888], the node responds to the offer either (1) with an answer that ignores the grouping attribute or (2) with a refusal to the request (e.g., 488 Not Acceptable Here or 606 Not Acceptable in SIP).

In the first case, the original sender of the offer must send a new offer without any duplication grouping. In the second case, if the sender of the offer still wishes to establish the session, it should retry the request with an offer without the duplication grouping. This behavior is specified in [RFC5888].

5. SDP Examples

5.1. Separate Source Interfaces

In this example, the redundant streams use the same IP destination address (232.252.0.1) but they are sourced from different addresses (198.51.100.1 and 198.51.100.2). Thus, the receiving host needs to join both SSM sessions separately.

```
v=0
o=ali 1122334455 1122334466 IN IP4 dup.example.com
s=DUP Grouping Semantics
t=0 0
m=video 30000 RTP/AVP 100 101
c=IN IP4 232.252.0.1/127
a=source-filter:incl IN IP4 232.252.0.1 198.51.100.1 198.51.100.2
a=rtpmap:100 MP2T/90000
a=rtpmap:101 MP2T/90000
a=ssrc:1000 cname:ch1@example.com
a=ssrc:1010 cname:ch1@example.com
a=ssrc-group:DUP 1000 1010
a=mid:Group1
```

Note that in actual use, SSRC values, which are random 32-bit numbers, can be much larger than the ones shown in this example. Also, note that before receiving an RTP packet for each stream, the receiver cannot know which SSRC identifier is associated with which payload type.

5.2. Separate Destination Addresses

In this 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=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=mid:S1b
```

Editor’s note: What if there are multiple streams per "m" line but grouping has to take place across "m" lines? Could we implicitly use the CNAMEs to infer the redundancy relation (note that ’ssrc-group’ attribute is media-level only)?

5.3. Delayed Duplication

In this example, the redundant streams have the same IP source and destination addresses but different UDP port numbers. Due to the same source and destination addresses, the packets in both streams will be routed over the same path. To provide resiliency against packet loss, the duplicate of an original packet is transmitted 50 ms later as indicated by the ’duplication-delay’ attribute (defined in [I-D.begen-mmusic-temporal-interleaving]).

```
v=0
o=ali 1122334455 1122334466 IN IP4 dup.example.com
s=DUP Grouping Semantics
t=0 0
a=group:DUP S1a S1b
a=duplication-delay:50
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=mid:S1a
m=video 40000 RTP/AVP 101
c=IN IP4 233.252.0.1/127
a=source-filter:incl IN IP4 233.252.0.1 198.51.100.1
a=rtpmap:101 MP2T/90000
a=ssrc:1010 cname:ch1@example.com
a=mid:S1b
```

6. Performance Evaluation and Reporting

Each duplicated stream has a separate (unique) SSRC identifier. Thus, individual RTCP receiver reports can be sent as usual for each of them from the receiving node that suppresses the duplicate packets. This way, the sender can be notified about the delivery performance of the individual streams.

Editor’s note: The receiving node can also produce a new XR report to report on the (loss/delay/jitter/etc.) performance of the output stream after the stream merging process.

7. Security Considerations

There is a weak threat for the receiver that the duplication grouping can be modified to indicate relationships that do not exist. Such attacks might result in failure of the duplication mechanisms, and/or mishandling of the media streams by the receivers.

In order to avoid attacks of this sort, the SDP description needs to be integrity protected and provided with source authentication. This can, for example, be achieved on an end-to-end basis using S/MIME [RFC5652] [RFC5751] when the SDP is used in a signaling packet using MIME types (application/sdp). Alternatively, HTTPS [RFC2818] or the authentication method in the Session Announcement Protocol (SAP) [RFC2974] could be used as well.

8. IANA Considerations

This document registers the following semantics with IANA in Semantics for the 'group' SDP Attribute under SDP Parameters:

Note to the RFC Editor: In the following registrations, please replace "XXXX" with the number of this document prior to publication as an RFC.

Semantics	Token	Reference
Duplication	DUP	[RFCXXXX]

This document also registers the following semantics with IANA in Semantics for the 'ssrc-group' SDP Attribute under SDP Parameters:

Token	Semantics	Reference
DUP	Duplication	[RFCXXXX]

9. Acknowledgments

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10.1. Normative References

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