MMUSIC
Internet-Draft
Intended status: Standards Track
Expires: September 12, 2012

A. Begen
Y. Cai
H. Ou
Cisco

March 11, 2012

Delayed Duplication Attribute in the Session Description Protocol
draft-begen-mmusic-temporal-interleaving-04

Abstract

A straightforward approach to provide protection against packet losses due to network outages with a longest duration of T time units is to simply duplicate the original packets and send each copy separated in time by at least T time units. This approach is commonly referred to as Time-shifted Redundancy, Temporal Redundancy or simply Delayed Duplication. This document defines an attribute to indicate the presence of temporally redundant media streams and the duplication delay in the Session Description Protocol.

Status of this Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at http://datatracker.ietf.org/drafts/current/.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on September 12, 2012.

Table of Contents

1. Introduction

Consider that a media sender transmits an original source packet and transmits its duplicate after a certain delay following the original transmission. If a network outage hits the original transmission, the expectation is that the second transmission arrives at the receiver. Alternatively, the second transmission may be hit by an outage and gets dropped, and the original transmission completes successfully. On the receiver side, both transmissions can also arrive and in that case, the receiver (or the node that does the duplicate suppression) needs to identify the duplicate packets and discard them appropriately, producing a duplicate-free stream.

Delayed duplication can be used in a variety of multimedia applications where there is sufficient bandwidth for the duplicated traffic and the application can tolerate the introduced delay. However, it must be used with care since it might easily result in a new series of denial-of-service attacks. Furthermore, delayed duplication must not be used in cases where the primary cause of packet loss is congestion, rather than a network outage due to a temporary link or network element failure. Duplication can make congestion only worse.

One particular use case for delayed duplication is to improve the reliability of real-time video feeds inside a core IP network [IC2011]. Compared to other popular redundancy approaches such as Forward Error Correction (FEC) [RFC6363] and redundant data encoding (e.g., [RFC2198]), delayed duplication is quite easy to implement since it does not require any special type of encoding or decoding.

For duplicate suppression, the receiver has to be able to identify the identical packets. This is straightforward for media packets that carry one or more unique identifiers such as the sequence number field in RTP header [RFC3550]. In non-RTP applications, the receiver can use unique sequence numbers if available or other alternative approaches to compare the incoming packets and discard the duplicate ones.

In this specification, we are not concerned about how the sender should determine the duplication delay. We are not concerned about how the receiver can suppress the duplicate packets and merge the incoming streams to produce a hopefully loss-free and duplication-free output stream (called stream merging), either. These considerations are out of the scope for this specification. Rather, our goal is simply to introduce a new attribute for the Session Description Protocol (SDP) [RFC4566] that indicates that the media stream is to be duplicated and sent two or more times, and also indicates the relative delay for each additional duplication.

In practice, more than two redundant streams are unlikely to be used since the additional delay and increased overhead are not easily justified. However, we define the new attribute in a general way so that it could be used with more than two redundant streams if needed. While the primary focus in this specification is the RTP-based transport, the new attribute is applicable to both RTP and non-RTP streams. Details on duplicating RTP streams are presented in [I-D.begen-avtcore-rtp-duplication].

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. The 'duplication-delay' Attribute

The following ABNF [RFC5234] syntax formally describes the 'duplication-delay' attribute:

```
delaying-attribute     = "a=duplication-delay:" periods CRLF
periods                = period *( ":" period)
period                 = 1*DIGIT ; in milliseconds
```

Figure 1: ABNF syntax for the 'interleaving-period' attribute

The 'duplication-delay' attribute is defined as both a media-level and session-level attribute. It specifies the relative delay for each duplication in milliseconds (ms). If used as a media-level attribute, it MUST be used with the 'ssrc-group' attribute and "DUP" grouping semantics as defined in [I-D.begen-mmusic-redundancy-grouping]. If used as a session-level attribute, it MUST be used with 'group' attribute and "DUP" grouping semantics as defined in [I-D.begen-mmusic-redundancy-grouping].

4. SDP Examples

In the first example below, the multicast stream is duplicated with a duplication delay of 100 ms. The streams have Synchronization Sources (SSRC) of 1000 and 1010, and they are grouped together using the 'ssrc-group' attribute defined in [RFC5576]. The "DUP" grouping semantics are defined in [I-D.begen-mmusic-redundancy-grouping]. The reason for using explicit grouping is that not all the media streams

in the same "m" line are necessarily duplicates of each other.

```
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
```

Note that in actual use, SSRC values, which are random 32-bit numbers, could be much larger than the ones shown in this example.

In the second example below, the multicast stream is duplicated twice. 50 ms after the original transmission, the first duplicate is transmitted and 100 ms after that, the second duplicate is transmitted. In other words, the same packet is transmitted three times over a period of 150 ms.

```
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:1020 cname:ch1@example.com
a=ssrc-group:DUP 1000 1010 1020
a=duplication-delay:50:100
a=mid:Group1
```

In the third example below, the multicast UDP stream is duplicated with a duplication delay of 50 ms. Both streams are sent in the same source-specific multicast (SSM) session but they are sent to different ports. The "DUP" grouping semantics

[I-D.begen-mmusic-redundancy-grouping] are used to describe the redundany relation.

```
v=0
o=ali 1122334455 1122334466 IN IP4 dup.example.com
s=Delayed Duplication
t=0 0
a=group:DUP S1a S1b
a=duplication-delay:50
m=audio 30000 udp mp4
c=IN IP4 233.252.0.1/127
a=source-filter:incl IN IP4 233.252.0.1 198.51.100.1
a=mid:S1a
m=audio 40000 udp mp4
c=IN IP4 233.252.0.2/127
a=source-filter:incl IN IP4 233.252.0.1 198.51.100.1
a=mid:S1b
```

5. Security Considerations

The 'duplication-delay' attribute is not believed to introduce any significant security risk to multimedia applications. A malevolent third party could use this attribute to misguide the receiver(s) about the duplication delays and/or the number of redundant streams. For example, if the malevolent third party increases the value of the duplication delay, the receiver(s) will unnecessarily incur a longer delay since they will have to wait for the entire period. Or, if the duplication delay is reduced by the malevolent third party, the receiver(s) might not wait long enough for the duplicated transmission and incur unnecessary packet losses. However, these require intercepting and rewriting the packets carrying the SDP description; and if an interceptor can do that, many more attacks are also possible.

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 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.

Another security risk is due to possible software misconfiguration or a software bug where a large number of duplicates could be unwillingly signaled in the 'duplication-delay' attribute. In applications where this attribute is to be used, it is a good

practice to put a hard limit both on the number of duplicate streams and the total delay introduced due to duplication regardless of what the SDP description specifies.

6. IANA Considerations

The following contact information shall be used for all registrations in this document:

Ali Begen
abegen@cisco.com

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

6.1. Registration of SDP Attributes

This document registers a new attribute name in SDP.

SDP Attribute ("att-field"):	
Attribute name:	duplication-delay
Long form:	Duplication delay for temporally redundant streams
Type of name:	att-field
Type of attribute:	Media or session level
Subject to charset:	No
Purpose:	Specifies the relative duplication delay(s) for redundant stream(s)
Reference:	[RFCXXXX]
Values:	See [RFCXXXX]

7. Acknowledgements

Authors would like to thank Colin Perkins for his suggestions and review.

8. References

8.1. Normative References

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

[RFC4566] Handley, M., Jacobson, V., and C. Perkins, "SDP: Session Description Protocol", RFC 4566, July 2006.

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

[RFC5234] Crocker, D. and P. Overell, "Augmented BNF for Syntax Specifications: ABNF", STD 68, RFC 5234, January 2008.

[RFC5576] Lennox, J., Ott, J., and T. Schierl, "Source-Specific Media Attributes in the Session Description Protocol (SDP)", RFC 5576, June 2009.

[I-D.begen-mmusic-redundancy-grouping] Begen, A., Cai, Y., and H. Ou, "Duplication Grouping Semantics in the Session Description Protocol", draft-begen-mmusic-redundancy-grouping-02 (work in progress), October 2011.

8.2. Informative References

[RFC6363] Watson, M., Begen, A., and V. Roca, "Forward Error Correction (FEC) Framework", RFC 6363, October 2011.

[RFC2198] Perkins, C., Kouvelas, I., Hodson, O., Hardman, V., Handley, M., Bolot, J., Vega-Garcia, A., and S. Fosse-Parisis, "RTP Payload for Redundant Audio Data", RFC 2198, September 1997.

[I-D.begen-avtcore-rtp-duplication] Begen, A. and C. Perkins, "Duplicating RTP Streams", draft-begen-avtcore-rtp-duplication-00 (work in progress), October 2011.

[IC2011] Evans, J., Begen, A., Greengrass, J., and C. Filsfils, "Toward Lossless Video Transport (to appear in IEEE Internet Computing)", November 2011.

[RFC5652] Housley, R., "Cryptographic Message Syntax (CMS)", STD 70, RFC 5652, September 2009.

[RFC5751] Ramsdell, B. and S. Turner, "Secure/Multipurpose Internet Mail Extensions (S/MIME) Version 3.2 Message Specification", RFC 5751, January 2010.

[RFC2818] Rescorla, E., "HTTP Over TLS", RFC 2818, May 2000.

[RFC2974] Handley, M., Perkins, C., and E. Whelan, "Session Announcement Protocol", RFC 2974, October 2000.

Authors' Addresses

Ali Begen
Cisco
181 Bay Street
Toronto, ON M5J 2T3
Canada

Email: abegen@cisco.com

Yiqun Cai
Cisco
170 W. Tasman Dr.
San Jose, CA 95134
USA

Email: ycai@cisco.com

Heidi Ou
Cisco
170 W. Tasman Dr.
San Jose, CA 95134
USA

Email: hou@cisco.com