Rapid Synch for RTP Multicast Sessions

draft-versteeg-avt-rapid-synchronization-for-rtp-00

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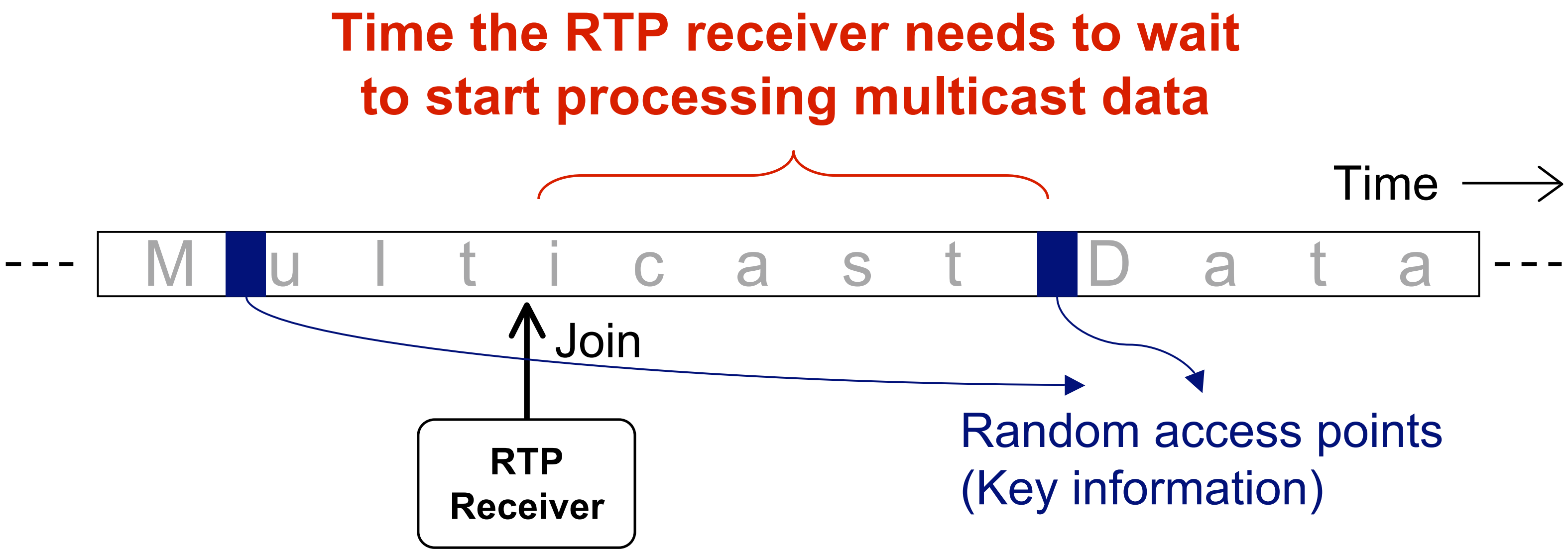
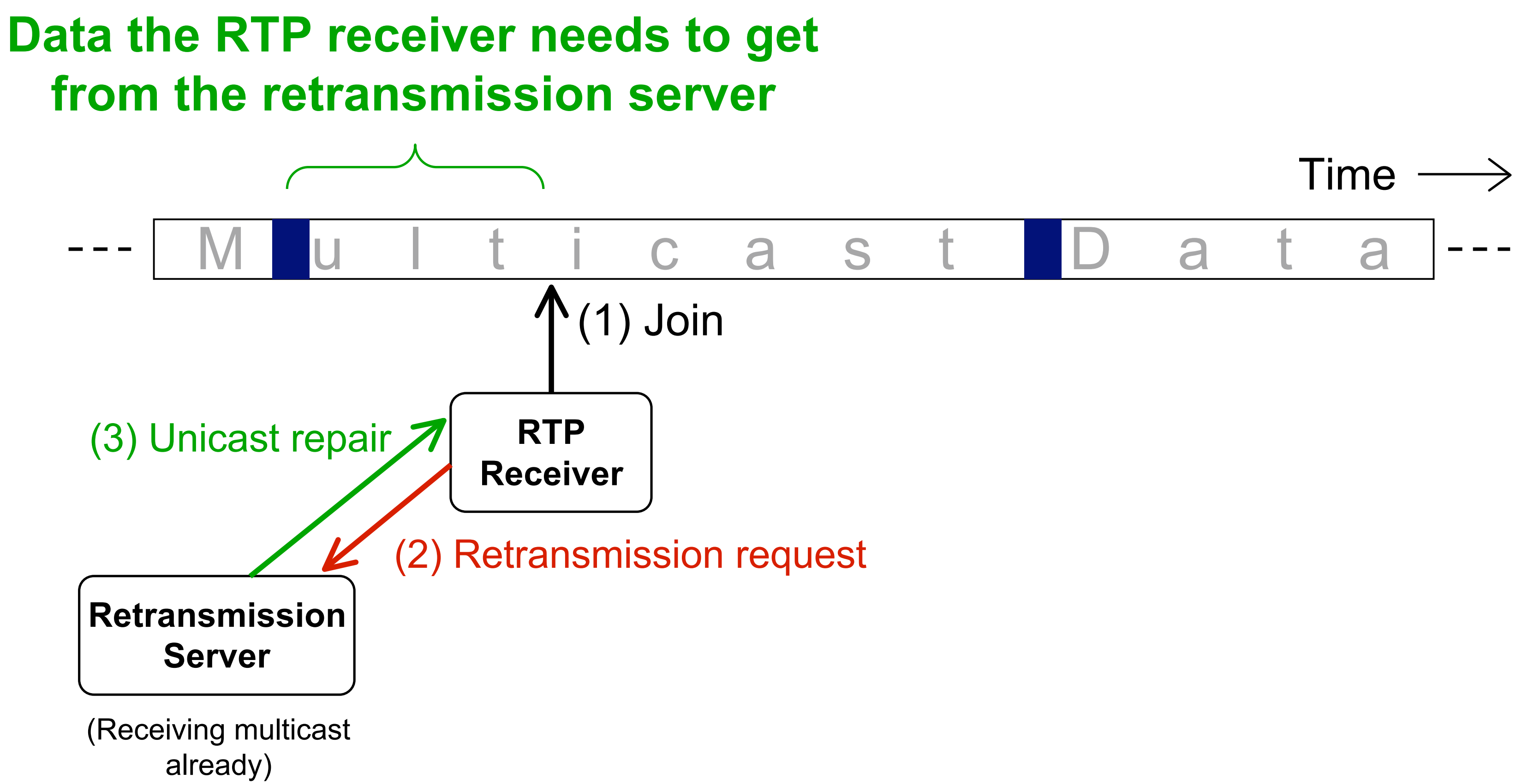
A Typical Multicast Join Scenario



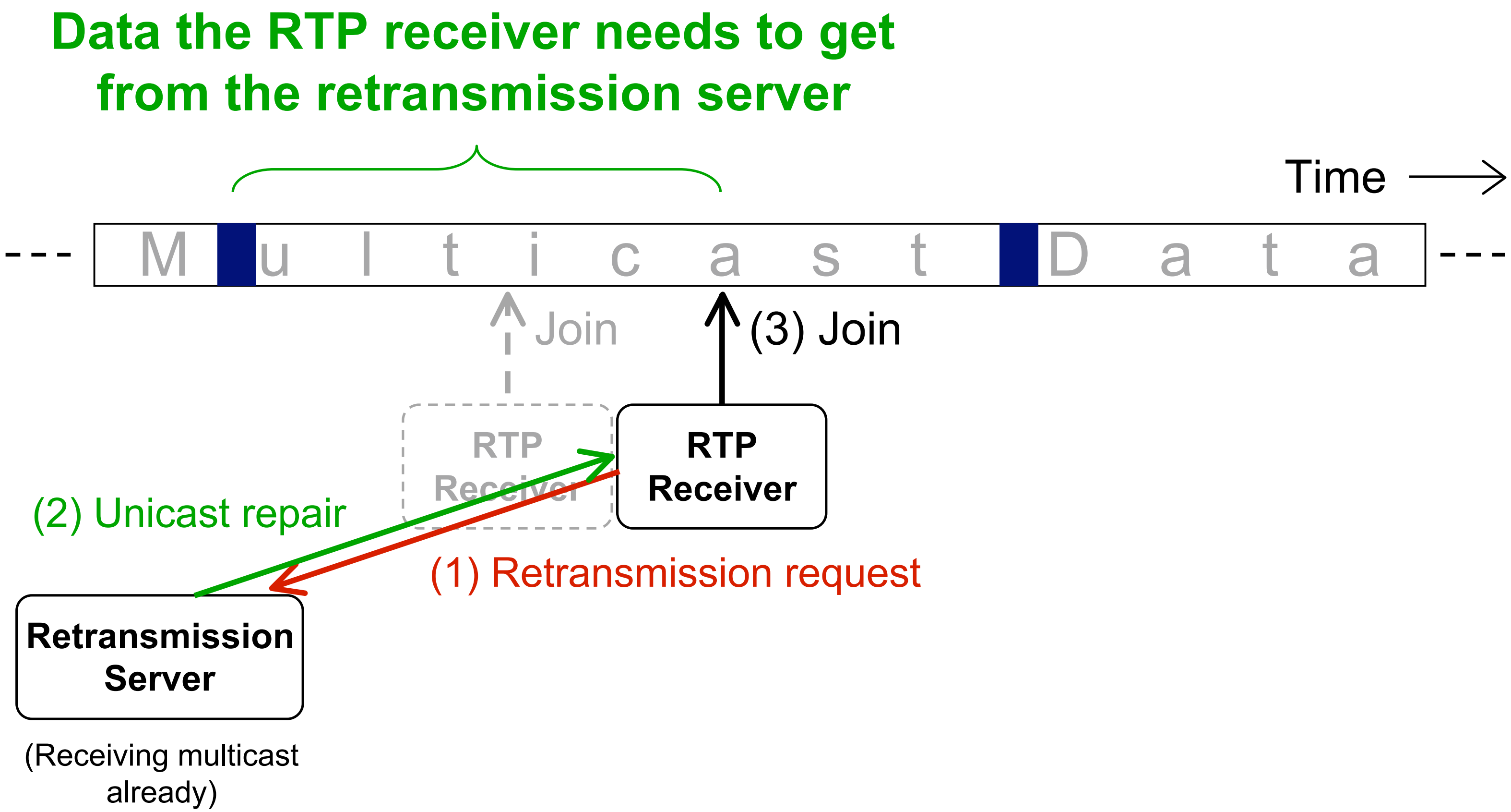
- Note that
  - RAPs might be far away from each other
  - Key information might be large in size and non-contiguous

  → These increase the synchronization delay for the RTP receiver

If There is Sufficient Bandwidth…



A Better (and Faster) Solution



Introduction

- RTP receiver says to the retransmission server:

  "I have no synch with the stream. Send me a repair burst that will get me back on the track with the multicast session"

- Differences compared to conventional retransmission:
  - Receiver does not know exactly what it is missing
  - Retransmission server
    - May need to parse data from earlier in the stream than it is needed for retransmission (Key information may be dispersed)
    - May need to burst faster than real time
    - Needs to coordinate the time for multicast join and ending the burst
- We define a method that enables a multicast receiver to acquire and process a multicast flow quickly
  - The method is applicable to any RTP-encapsulated multicast flow
  - The only requirement is to have RTP transport

Rapid Synch through an Intermediary

```
                      +------------------+
                 +--->|   Intermediary   |
                 | ...| Network Element  |
                 | :  |(Feedback Target) |
                 | :  +------------------+
                 | v
+---------+     +-------+         +---------+
|   RTP   |---->|Router|........>|Joining  |
| Sender  |     |       |-------->|   RTP   |
+---------+     +-------+         |Receiver |
                    |             +---------+
                    |
                    |             +---------+
                    +------------>|Existing |
                                  |   RTP   |
                                  |Receiver |
                                  +---------+

   ---> RTP Multicast Flow
   ...> RTP Unicast Flow
```

Toolkit:
- **RFC 3550**
  RTP/RTCP
- **RFC 4585**
  RTP/AVPF
- **RFC 4588**
  RTP Retransmissions
- **RTCP SSM**
  RTCP Extensions for SSM

Elements of Delay for MPEG AV Flows

- Multicast switching delay
  - IGMP joins, leaves → Generally well-bounded
  - Route establishment → Generally well-bounded
- Key information latency
  - PSI (PAT/CAT/PMT) acquisition delay
  - RAP acquisition delay
- Buffering delays
  - Loss repair, de-jittering, application buffering
  - MPEG decoder buffering

→ Key information latency and buffering delays are more critical in MPEG-based AV applications

Rapid Synchronization

```
           +---------+
           |   RTP   |
           |  Sender |
           +---------+
               |
               v
        +--------------+
        |    Router    |<----- (1) IGMP Leave -------+
        |              |<----- (5) IGMP Join ------+ |
        +--------------+                           | |
          |       |                                | |
  +-------+       +--- (6) New Multicast Flow ---+ | |
  |                                              | | |
  v                                              v | |
+--------------+                              +----------+
|              | <.... (2) RTCP NACK PLI .... |          |
|Retransmission|     & Buffer Requirements    |   RTP    |
|    Server    | .. (3) Burst Description ..> | Receiver |
|     (RS)     | .... (4) Unicast Burst ....> |   (RR)   |
|              | <.. (7) Burst Termination .. |          |
+--------------+                              +----------+

---> Multicast Flows and IGMP Messages
...> Unicast Flows
```

How to Carry Messages?

- RTCP NACK PLI and buffer requirements
  - NACK PLI → RFC 4585
  - Max/min buffer requirements of RR
    - An RTCP APP packet?
    - A new transport-layer feedback message?
- Burst description: Seqnum of the first RTP packet to be sent, earliest IGMP join time, burst end time, etc.
  - RS accepts the request
    - An RTCP APP packet?
    - A new transport-layer feedback message?
  - RS rejects the request → RR joins multicast session immediately
    - A null RTCP APP packet?
    - A new transport-layer feedback message?

How to Carry Messages?

- Unicast burst
  - Key information (TSRAP) data
    - An RTCP APP packet?
    - A new transport-layer feedback message?
  - RTP data
- Burst termination
  - If RS hadn’t specified burst duration, RR needs to tell RS to stop
  - RR may switch to another session, thus may want cancel rapid synch
    - RTCP BYE packet?
    - A new transport-layer feedback message?

Open Issue: Which messages/packets shall we use?

Open Issue: Do we need redundancy in these messages?

SDP Example

```
m=video 41000 RTP/AVPF 98
i=Primary Source Stream
c=IN IP4 224.1.1.2/255
a=source-filter: incl IN IP4 224.1.1.2 8.166.1.1
a=recvonly
a=rtpmap:98 MP2T/90000
a=rtcp:41001 IN IP4 9.30.30.1       → Address for the feedback target
a=rtcp-fb:98 nack                   → Retransmission support (RFC 4585/4588)
a=rtcp-fb:98 nack pli               → Rapid synch. support (RFC 4585/4588)
a=ssrc:123321 cname:iptv-ch32@rtp.rocks.com
a=mid:1
m=video 41002 RTP/AVPF 99
i=Unicast Retransmission Stream
c=IN IP4 9.30.30.1
a=recvonly
a=rtpmap:99 rtx/90000
a=rtcp:41003
a=fmtp:99 apt=98
a=fmtp:99 rtx-time=5000
a=mid:2
```

Other Open Issues

- Burst shaping and failures cases
  - These sections are TBC
- NAT considerations
  - This section is TBC
- Measuring and reporting the performance of rapid synch
  - Shall we define a new RTCP XR report block type?
- Defining a new NACK parameter for non-video apps
  - Is PLI acceptable for non-video apps?

Open Source Implementation

Web Access:

http://www.cisco.com/en/US/docs/video/cds/cda/vqe/2_0/user/guide/ch1_over.html

FTP Access:

ftp://ftpeng.cisco.com/ftp/vqec/

Next Steps

- Shall we make this draft a WG item?