



OPEN

BROADCAST SYSTEMS

Reliable Internet Stream Transport

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Introduction

- Goals
 - Low-latency video between professional devices (not necessarily end-users)
 - Packet recovery on a lossy network
 - Backwards compatible with existing MPEG-TS in UDP streams



Background

- Video Services Forum
 - (Closed) Industry forum of broadcast manufacturers
- Interest in replacing proprietary protocols (building on IETF)
- Published in October 2018
 - Unusually RIST published under Creative Commons Licence



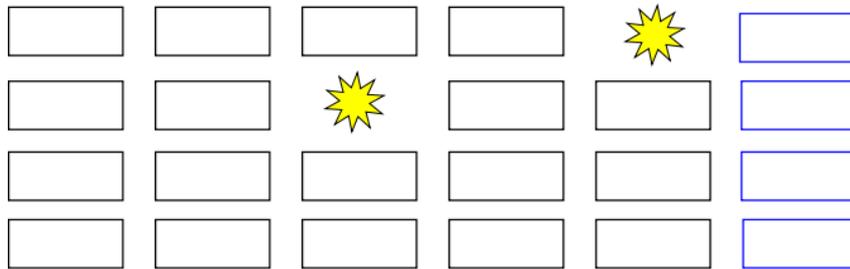
Why not TCP?

- A whole research field in itself!
- Traditional TCP drops transmission rate with packet loss, quite aggressively in many cases
 - Not usually throughput drops in our use-cases
- UDP has native support for multicast
 - Many receivers requesting retransmits
- Backwards compatibility
 - Send RIST stream to existing devices
- Easier to do multipathing
 - Lets application handle the congestion-control decisions

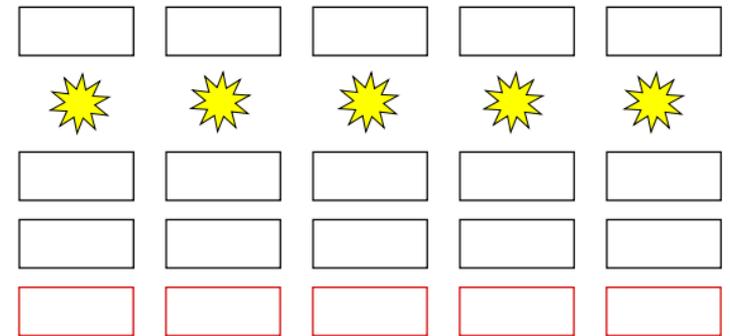
Forward Error Correction (FEC)

- SMPTE 2022-1 FEC

Row FEC



Column FEC



- XOR based, very basic
- Can't handle loss > matrix

Similar protocols/software

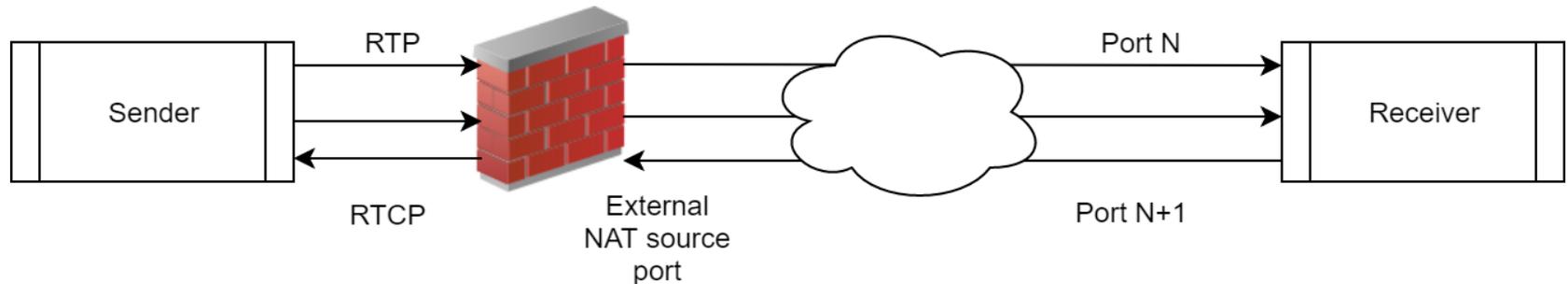
- Aggregatrtp (VideoLAN)
 - Allows aggregation of links with retransmission
 - From 2007
- SRT (Haivision)
 - Monolithic application
 - Very complex codebase around UDT (file transfer!)
 - Supports single-links only
 - Has encryption built-in
 - 89 page document!
- Many other proprietary solutions (mixing FEC and retransmissions)



RTP/RTCP

- Uses port N (RTP – main) and N+1 (RTCP – control)
- Sender periodically sends RTCP packets to keep state alive
 - Receiver sends Receiver Report RTCP packets
 - Allows for NAT “traversal”
- This can be over multiple links

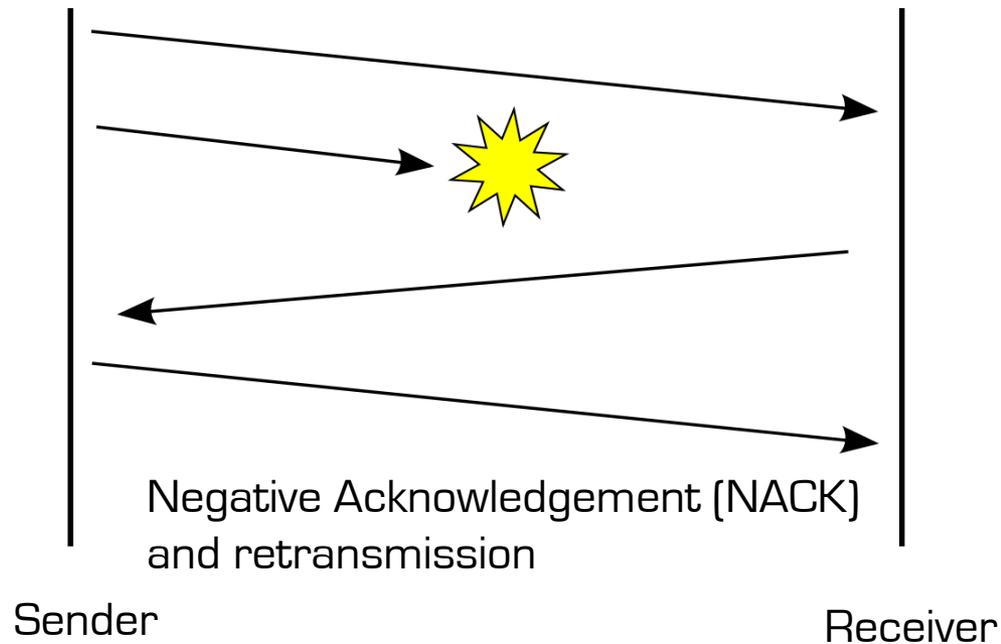
RTP/RTCP



- Receiver sends to external (possibly NATed) source port, stateful NAT passes through
- Exactly like DNS

Acknowledgements

- Every RTP packet has a sequence number (16-bit) for identification

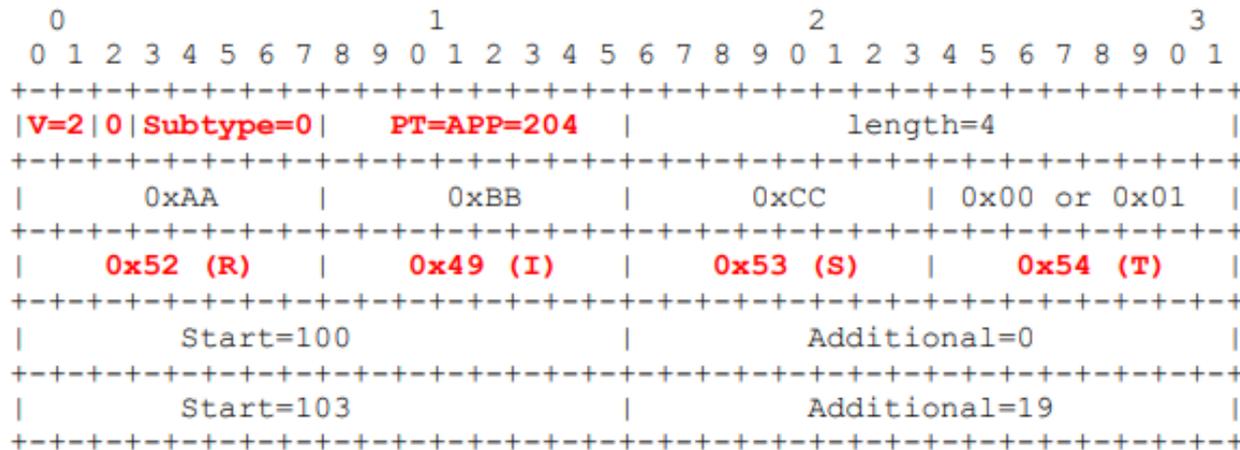


Bitmask NACK

```
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
+-----+-----+-----+-----+
| V=2 | 0 | FMT=1 | PT=205 | length=4 |
+-----+-----+-----+-----+
|          SSRC of packet sender (ignored by RIST sender2)          |
+-----+-----+-----+-----+
|    0xAA    |    0xBB    |    0xCC    | 0x00 or 0x01 |
+-----+-----+-----+-----+
|          PID=100          | 1 1 1 1 1 1 1 1 1 1 1 1 0 0 |
+-----+-----+-----+-----+
|          PID=117          | 0 0 0 0 0 0 0 0 0 0 1 1 1 1 |
+-----+-----+-----+-----+
```

- Uses base RTP sequence number + bitmask
- Useful for "random" packet loss

Range NACK



- Custom RTCP packet (RIST)
- Allows for range of loss

Retransmissions

- Signalling retransmission by setting LSB of SSRC to 1
 - Really strange way of doing it, should have used RFC 4588
- Decoder can identify retransmissions by SSRC
- Rate of retransmit requests implementation defined within latency bounds
 - Also how to handle “thundering herd” of retransmits

Implementations

- Open Source
 - Upipe (in examples)
 - VLC rist://
- Proprietary
 - VideoFlow
 - Zixi
 - Couple of other hardware ones

Future work

- Encryption (probably DTLS)
- Null packet removal
- Encoder bitrate changes
- Pull mode (?), otherwise end user can implement

- Scalable video (hard in practice)
- RIST on uncompressed video (completely crazy!)

Live demo!

Credits

- Thanks to Adi Rozenberg (VideoFlow) and Rafaël Carré for their comments