

RTP Redundancy Update

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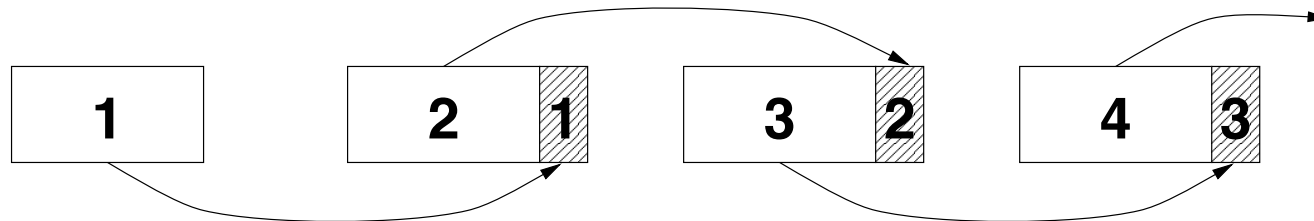
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Status

- RTP redundancy mechanism published as RFC2198 in September 1997.
- Simple packet format, allows bundling of multiple frames of audio into a single packet as a form of media specific FEC.



- Optimised for audio data, but can be used for other media types.

Example packet

```

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|P|X|  CC   |M|       PT   | sequence number of primary |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     timestamp of primary encoding |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     synchronization source (SSRC) identifier |
+-----+-----+-----+-----+-----+-----+-----+-----+
|1| block PT=7 | timestamp offset | block length |
+-----+-----+-----+-----+-----+-----+-----+-----+
|0| block PT=5 |                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     LPC encoded redundant data |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     DVI4 encoded primary data |
+-----+-----+-----+-----+-----+-----+-----+-----+
|                                     |
+-----+-----+-----+-----+-----+-----+-----+-----+

```

Problem: Start of Talkspurt

- The redundant (FEC) data is typically piggy-backed one packet after the primary.
- The first packet in a talkspurt cannot contain FEC data, since there are no preceeding packets.
- This causes two problems:
 1. Changing payload type
 2. Unknown buffering requirement

Issues: Payload Type

- In a standard RTP session, all packets sent by a source will have the same payload type.
- However, senders using redundant audio send the first packet in a talkspurt with no FEC data (ie: payload type of the primary codec) and the following packets with the redundancy payload type.



- This makes implementations needlessly complex, since they have to associate packets with different payload types into a single stream.

Issues: Buffer Space

- The FEC data can be sent any number of packets after the primary. This delay isn't known until a packet containing FEC data is received...
- ...by which time the playout buffer length for this talkspurt has already been calculated.
- Adapting the playout buffer mid-talkspurt will cause an glitch in the audio. Not adapting may make it impossible to use the FEC data (since it arrives too late)

Solution

- Send *all* packet with the redundancy payload type.
- For those at the start of the talkspurt, advertise the FEC offset and set the block length to zero.

[illegible]

Solution

- This solves both problems noticed.
- Requires a change to the *usage* of the protocol, but not to the protocol specification itself.
- Believed backwards compatible with existing implementations...