RTP Redundancy Update

Colin Perkins <c.perkins@cs.ucl.ac.uk>
Department of Computer Science
University College London
Gower Street
London WC1E 6BT
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
+-----------------------------
|V=2|P|X| OC |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 preceding 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.

  ![Diagram](image)

  - Talkspurt 1
  - Talkspurt 2

- 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.
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...