Packet Loss Recovery for Streaming Video

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Overview

- Want: High quality streaming video over the Internet

- Problems:
  - Variable bandwidth
  - Variable delay
  - Packet loss

- Packet losses in compressed video are aggravated by propagation of errors.

We present a protocol that enables receiver-driven packet loss recovery using selective retransmission and receiver postprocessing.
Packet loss degrades video quality. Why?
Propagation of Errors
Packet Loss Model
Where does Selective Reliability Fit In?

- **TCP**
  - Favors complete reliability over timely delivery
  - Inordinate buffering
  - Might not want to use TCP’s AIMD congestion control

- **Forward Error Correction**
  - Requires additional (maybe unnecessary) bandwidth
  - FEC packets themselves might be lost!

- **Coding Approaches**
  - Generally must be done offline
Selective Reliability is Beneficial

- Selective reliability can increase perceptual quality of video by up to 3 times.
Challenges

- **Compatibility with existing protocols**
  - *Solution:* SR-RTP, a backwards-compatible extension to RTP that allows for receiver-driven selective retransmission

- **Easy application integration**
  - *Solution:* Expose simple SR-RTP API to applications

- **Recourse in the event of high loss/latency**
  - *Solution:* Retransmissions are receiver-driven, and API exposes mechanism for cancelling spurious retransmissions. Easy integration with postprocessing techniques.
Append SR-RTP extension to 12-byte RTP header.

Extension allows for application level framing:

- Detect lost portions of bitstream
- Optionally request retransmission (depending on priority of surrounding fragments)
SR-RTP API Overview

- Client application is in control:
  - Callback-based mechanism upon frame arrival
  - Can force incomplete frames to be read
  - Can cancel retransmissions on played frames
SR-RTP API Overview

- Application
  - receives callback upon arrival of complete ADUs
  - can force read of incomplete ADU

- What can we do if loss is in an important frame?
Receiver-driven Postprocessing

- Use past motion and texture information to recover from lost packets
  - Texture from preceding P-frame
  - Motion from preceding B-frame
SR-RTP Benefits

Frame Rate vs. Packet Loss Rate

- PSNR > 25
- PSNR > 25 (w/o I-frame retransmissions)
If network latency precludes selective retransmission, postprocessing can help mask errors.
Conclusions

- Streaming compressed video must account for packet loss.

- SR-RTP protocol allows for receiver-driven retransmission of only the most important packets.

- SR-RTP API gives application control over packet loss recovery.

- ALF principle allows easy integration with other techniques, such as receiver postprocessing.
Outline

• Motivation
• Packet Loss Model
• Application-Level Framing with SR-RTP
• SR-RTP Protocol/Library
• Receiver Postprocessing using SR-RTP and ALF
Packet Loss Model

![Graph showing the relationship between Frame Rate and Packet Error Rate with different PSNR thresholds and Model Fit.](#)