Packet Loss Recovery for Streaming Video

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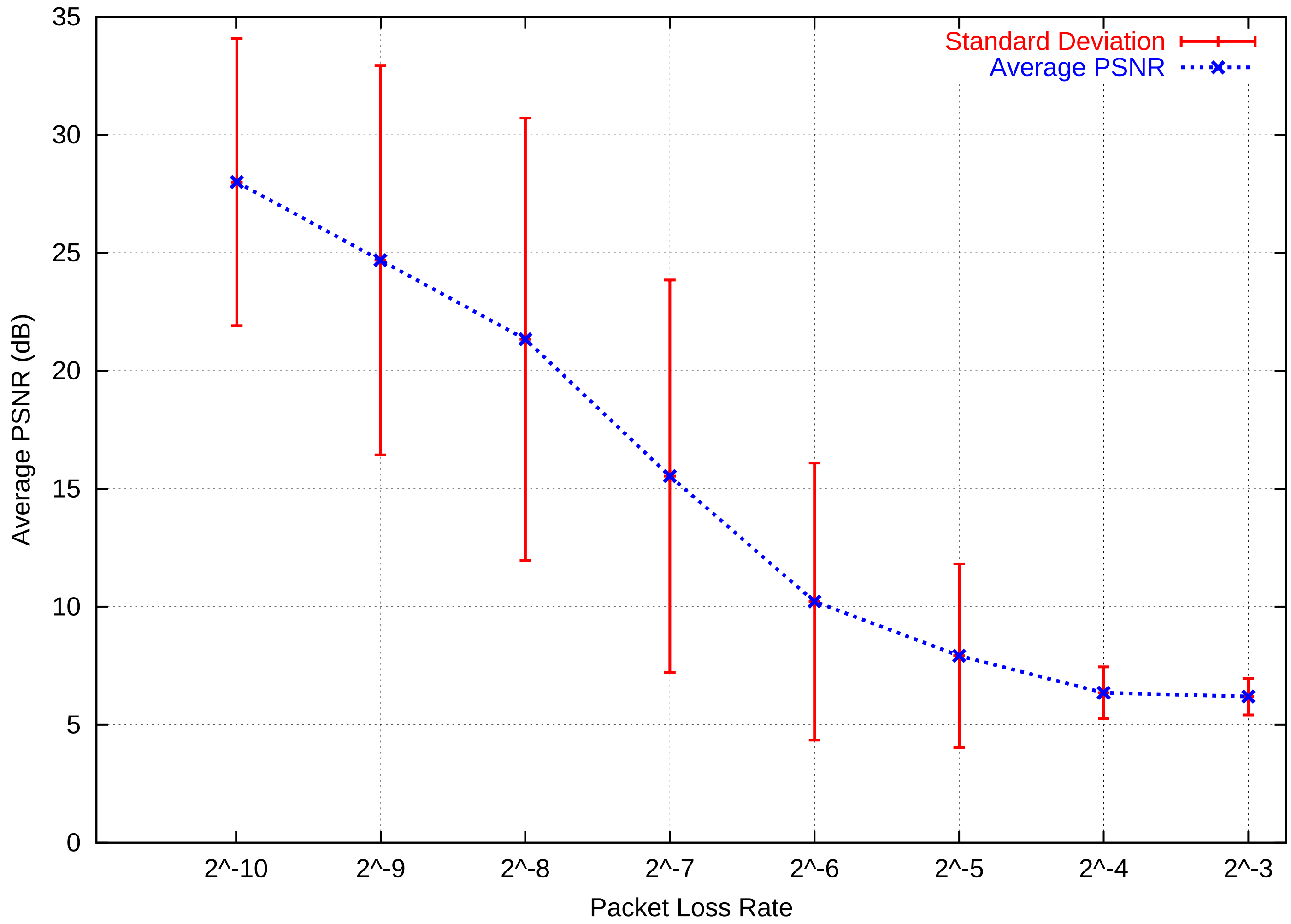
11th International Packet Video Workshop
April 25, 2002

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.

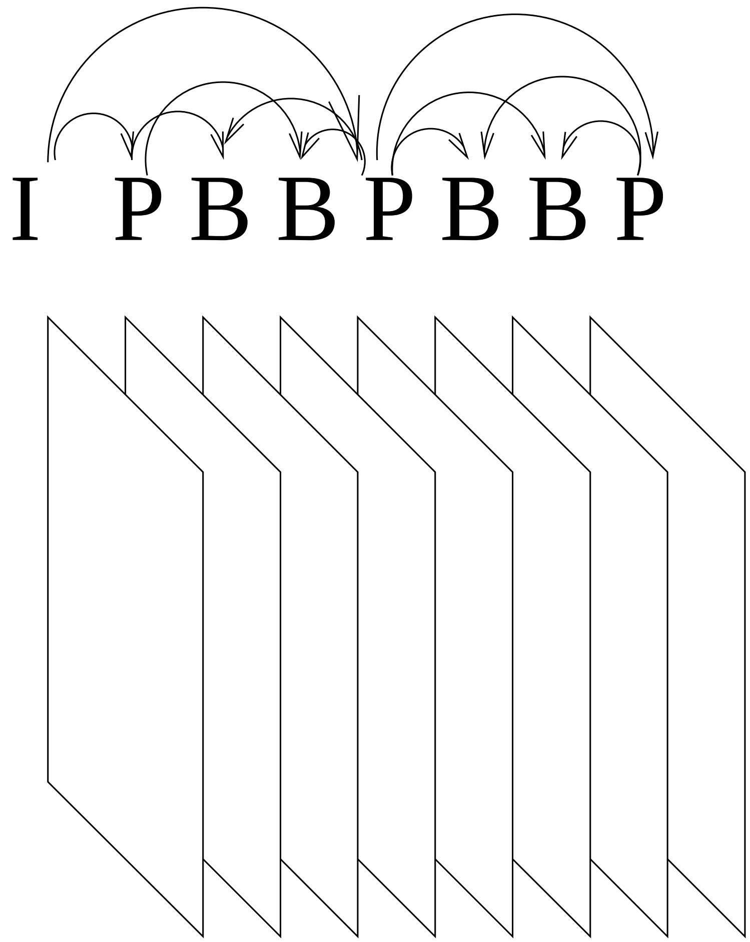
Impact of Packet Loss



Packet loss degrades video quality. Why?

Propagation of Errors

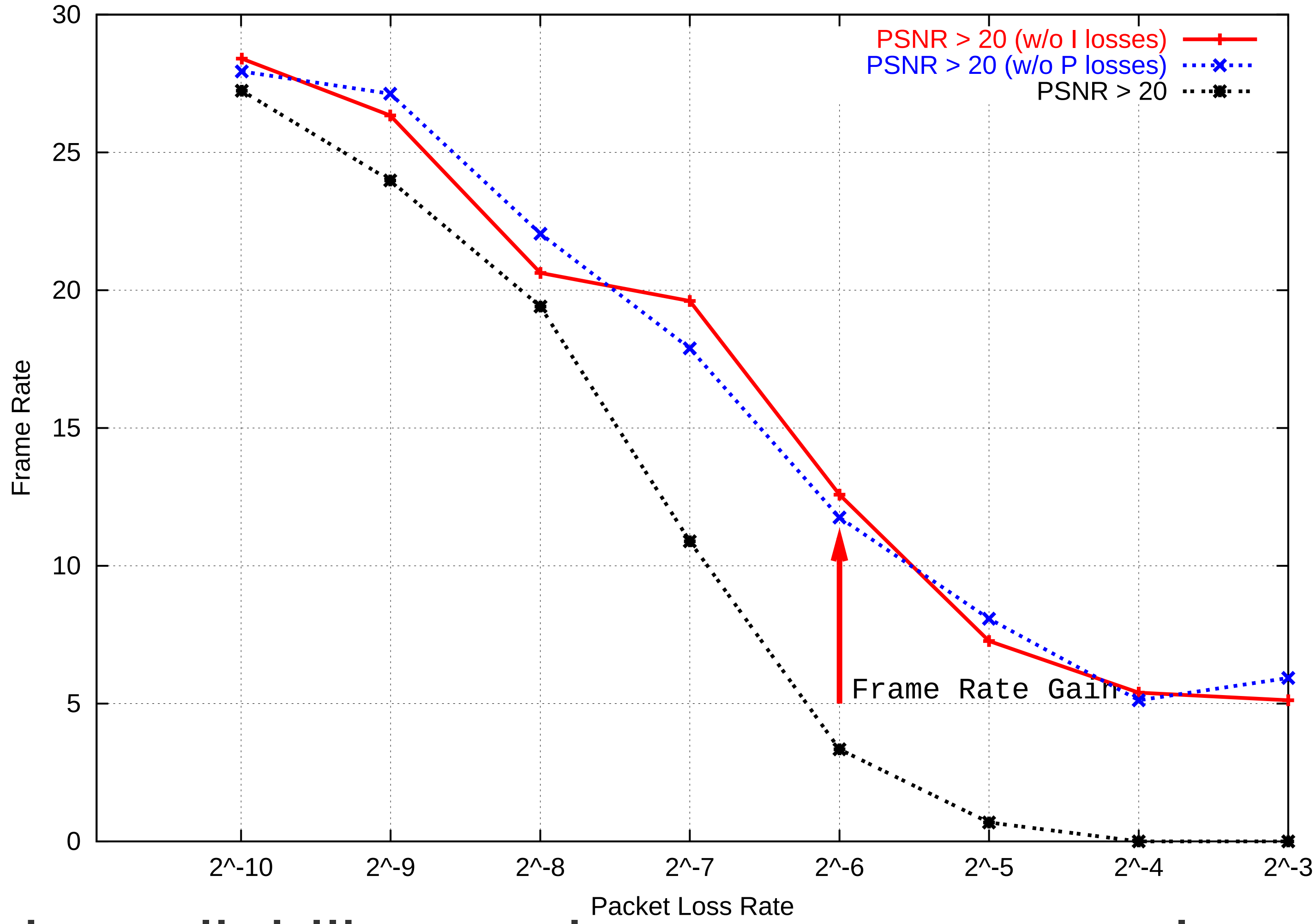
Packet Loss Model



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

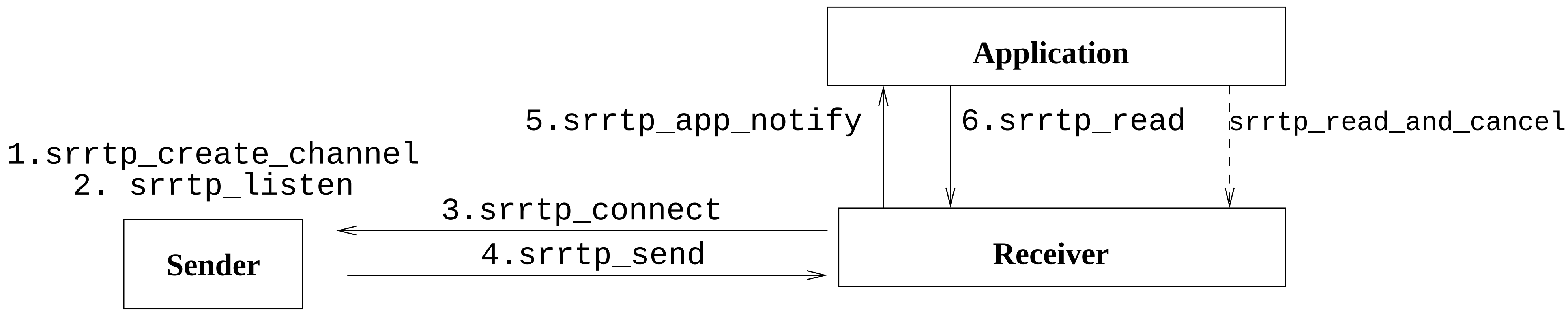
- Compatibility with existing protocols
  - *Solution:* SR-RTP, a backwards-compatible extension to RTP that allows for receiver-driven selective retransmission
- Easy application integration
  - *Solution:* Expose simple SR-RTP API to applications
- Recourse in the event of high loss/latency
  - *Solution:* Retransmissions are receiver-driven, and API exposes mechanism for cancelling spurious retransmissions. Easy integration with postprocessing techniques.

SR-RTP Protocol Overview

<table>
<tr><td colspan="10">0</td><td colspan="10">1</td><td colspan="10">2</td><td colspan="2">3</td></tr>
<tr><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>1</td><td>2</td><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td><td>0</td><td>1</td></tr>
<tr><td colspan="2">V</td><td>P</td><td>X</td><td colspan="4">CC</td><td>M</td><td colspan="7">Payload Type</td><td colspan="16">Sequence Number</td></tr>
<tr><td colspan="32">Timestamp</td></tr>
<tr><td colspan="32">Synchronization Source (SSRC) Identifier</td></tr>
<tr><td colspan="16">Zeros</td><td colspan="16">length of extension (words)</td></tr>
<tr><td colspan="32">ADU sequence number</td></tr>
<tr><td colspan="32">ADU length (bytes)</td></tr>
<tr><td colspan="32">ADU offset (bytes)</td></tr>
<tr><td colspan="16">Priority</td><td colspan="16">Layer Number</td></tr>
</table>

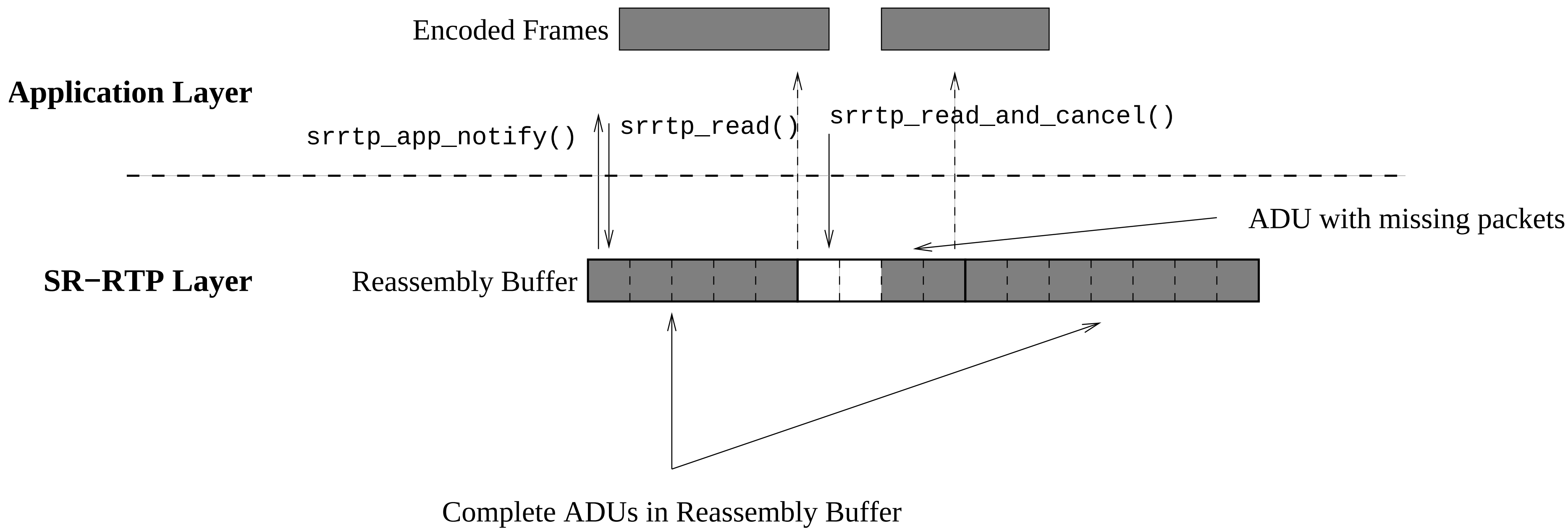
- Append SR-RTP extension to 12-byte RTP header.
- Extension allows for application level framing:
  - Detect lost portions of bitstream
  - Optionally request retransmission (depending on priority of surrounding fragments)

SR-RTP API Overview



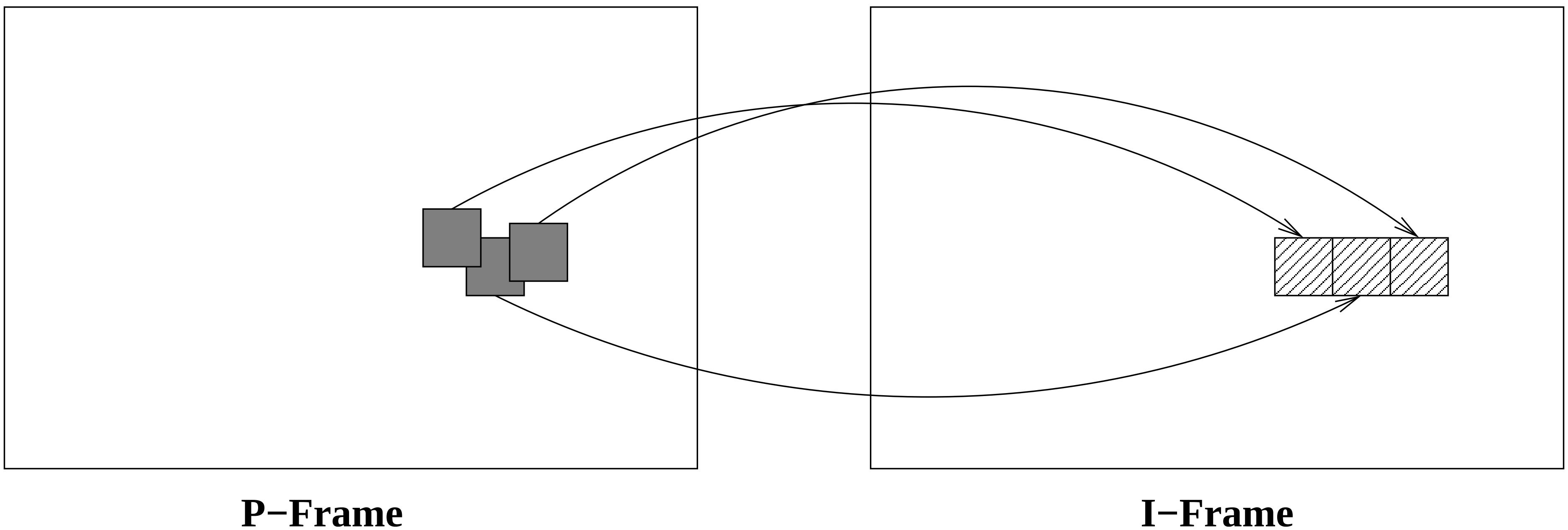
- Client application is in control:
  - Callback-based mechanism upon frame arrival
  - Can force incomplete frames to be read
  - Can cancel retransmissions on played frames

SR-RTP API Overview



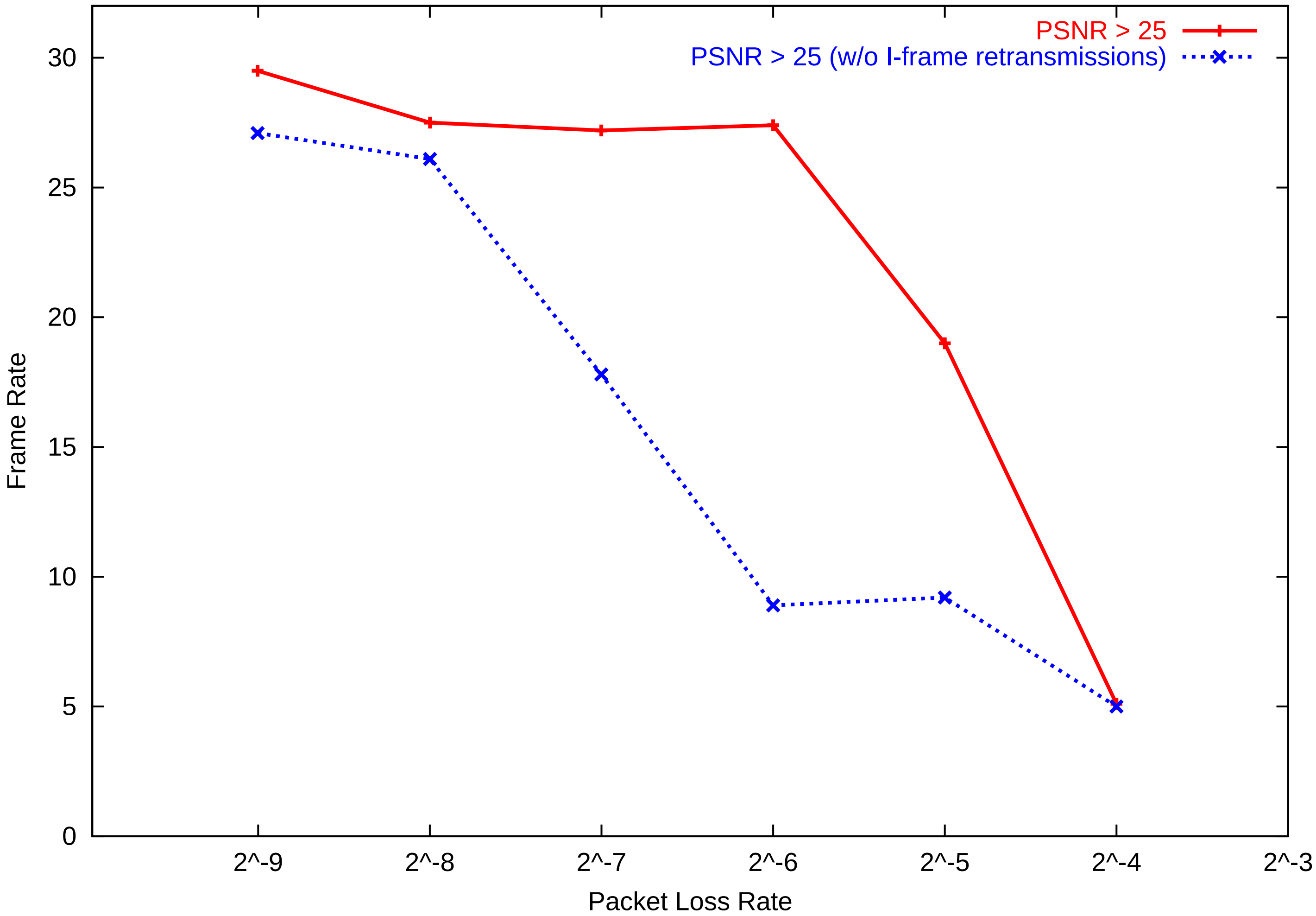
- Application
  - receives callback upon arrival of complete ADUs
  - can force read of incomplete ADU
- What can we do if loss is in an important frame?

Receiver-driven Postprocessing

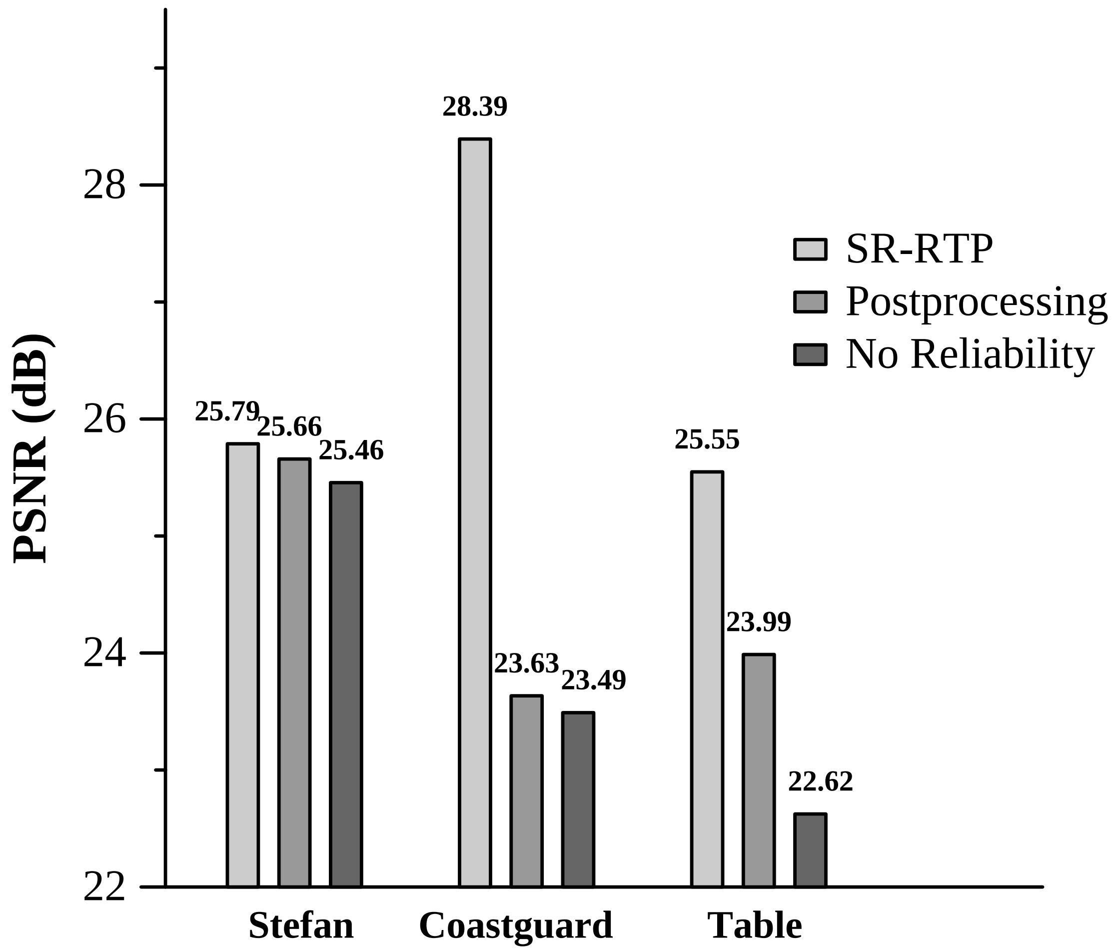


- Use past motion and texture information to recover from lost packets
  - Texture from preceding P-frame
  - Motion from preceding B-frame

SR-RTP Benefits



Recovering I-frames



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