Packet Loss Recovery for Streaming Video

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Introduction (1)

- Streaming is growing
- Commercial streaming successful (ie RealPlayer and MediaPlayer)
  - but proprietary and inflexible
  - → Use MPEG-4 since open
- Current streaming inflexible
  - Suboptimal
  - Want to adapt to current network
  - → Present system that adapts to loss

Introduction (2)

- MPEG video under loss suffers from propagation of errors (what is this)
  - → Fundamental tradeoff between bandwidth efficiency and error resilience
- Current FEC approaches effective but
  - Reduces benefits of compression
  - Tough to get adaptation right
- Some say cannot use retransmission for streaming but
  - Selective retransmission (I-frames) ok
- Build model + system
  - (Also TCP-Friendly using CM, but not focus)

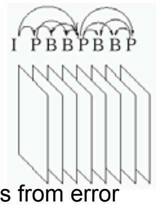
Outline

Introduction	(done)
Model	(next)
System Architecture	
Performance Evaluation	
Related Work	
Conclusions	

Model (Outline)

- Problem Description (next)
  - MPEG-4
  - Error Propagation
- Packet loss model
  - Experiments
  - Analytic Model
- Benefits of Selective Repair

Problem Description



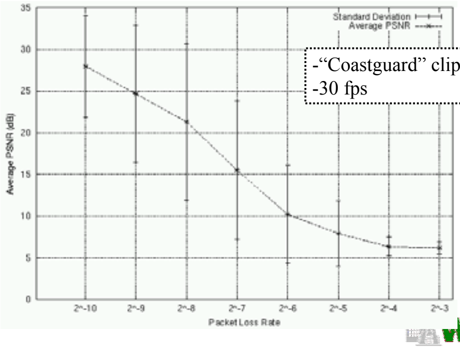
- MPEG-4 has three frame types: I, B, P
  - Note, MPEG-4 calls them “Video Object Planes” but frames is fine
- While high compression, suffers from error propagation

→ Loss of I-frame packets can affect subsequent frames, too

Loss in an I-frame

Propagation to Next P-frame

Average PSNR versus Loss Rate



-“Coastguard” clip
-30 fps

Model (Outline)

- Problem Description (done)
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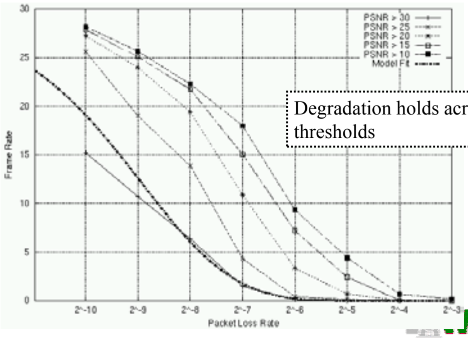
Packet Loss Model

- Assume packet loss degrades quality
  - True, in general, unless FEC
- Assume below PSNR threshold, would discard
  - But PSNR doesn’t model perceived quality
  - This viewability threshold varies with picture
    - So will analyze several thresholds
    - Also, can use other quality metrics
  - Generally, under 20 db is bad
    - Loss of 2^8 (about .4%) trouble and needs correction
    - (See previous figure)

Effects of Loss on Frame Rate (with Thresholds)



Degradation holds across thresholds

Analytic Model (1)

- Observed frame rate $f = f_0(1-\phi)$
  - ϕ is fraction of frames dropped
  - f_0 is original frame rate (ie- 30 fps)

$$\phi = \sum_i P(f_i) \cdot P(\overline{F}|f_i)$$

- Where i runs over types I, P and B
- P(f_i) can be determined by fraction in stream
- F is event that a frame is useless (PSNR below threshold)
- f_i is event that frame is of type i

Analytic Model (2)

$$P(\overline{F}|I) = 1-(1-p)^{S_I}$$

- p is packet loss rate
- S_I is size of I frame (similar for S_B, S_P, too)
- Assume if any packet lost, frame useless

$$P(\overline{F}|P) = \frac{1}{N_P}\sum_{k=1}^{N_P}\left(1-(1-p)^{S_I+kS_P}\right)$$

$$P(\overline{F}|B) \leq \frac{1}{N_P}\sum_{k=1}^{N_P}\left(1-(1-p)^{S_I+(k+1)S_P+S_B}\right)$$

P needs all previous P (and I) frames
N_P is number of P frames in GOP

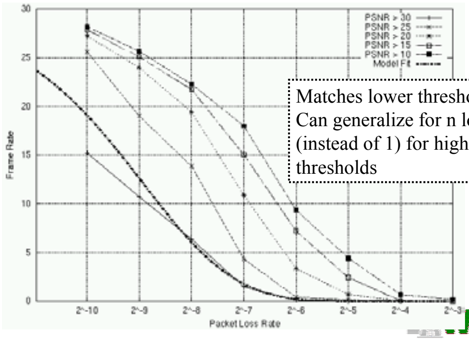
Analytic Model (3)

$$P(\overline{F}|P) = 1-\frac{(1-p)^{S_I}}{N_P\left(1-(1-p)^{S_P}\right)}\left(1-(1-p)^{S_P N_P}\right)$$

$$P(\overline{F}|B) \leq 1-\frac{(1-p)^{S_I+S_P+S_B}}{N_P\left(1-(1-p)^{S_P}\right)}\left(1-(1-p)^{S_P N_P}\right)$$

- Simplify to closed form above
- Now, using equations and given
  - N_P S_I S_B S_P f_0
- Can determine
  - $f = f_0(1-\phi)$
- "Model". Compare to measured

Model vs. Measured



Matches lower thresholds
Can generalize for n losses (instead of 1) for higher thresholds

Model (Outline)

- Problem Description (done)
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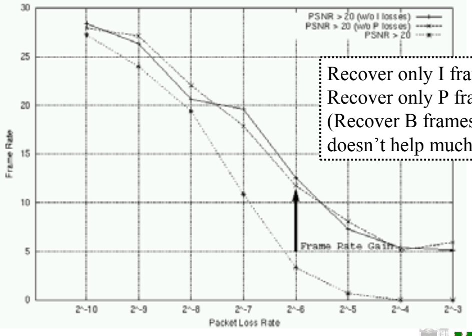
- Benefits of Selective Repair (next)

Benefits of Selective Retransmission



Recover only I frames
Recover only P frames
(Recover B frames doesn't help much)

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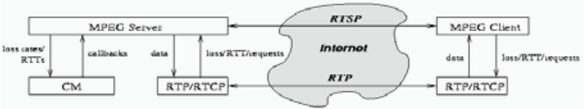
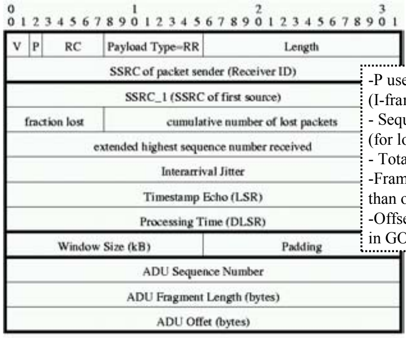
Overview of System



- CM provides TCP-Friendly data rate
  - Calls back when can send data
- Data sent over RTP (using UDP)
- Control over RTSP (over TCP)
- Frames put into Application Data Units (ADU)
  - 1 per ADU

Packet Header



V	P	RC	Payload Type=RR	Length
SSRC of packet sender (Receiver ID)				
SSRC_1 (SSRC of first source)				
fraction lost			cumulative number of lost packets	
extended highest sequence number received				
Interarrival Jitter				
Timestamp Echo (LSR)				
Processing Time (DLSR)				
Window Size (kB)				Padding
ADU Sequence Number				
ADU Fragment Length (bytes)				
ADU Offet (bytes)				

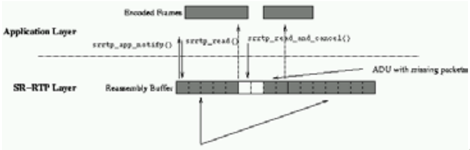
-P used for priority (I-frames)
- Sequence numbers (for loss)
- Total length
-Frames can be more than one packet
-Offset for location in GOP

Loss Detection and Recovery

- Mid-frame
  - Gaps in ADU using offset plus fragment length
- Start-of frame
  - First offset non-zero
- End-of-frame
  - ADU less than reported length
- Complete loss
  - Detected by gap in ADU sequence numbers
- Can use priorities to decide upon retransmissions
  - (Me: which ones *is* the hard part!)

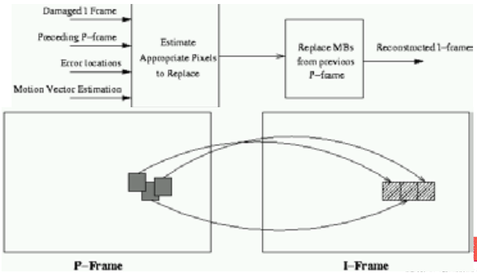
Implementation

- Used OpenDivX for MPEG-4
- Used CM previously built for Linux
- Used RTSP client-server library
- Also, extended `mplayer` for Linux
  - Call-backs give complete ADU to player
  - Retransmit all unless canceled by app



Postprocessing (Receiver-Based)

- May still have some missing frames
- Simple replacement bad if motion
  - Estimate motion and compensate



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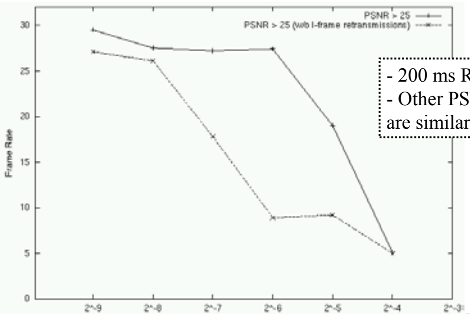
Setup

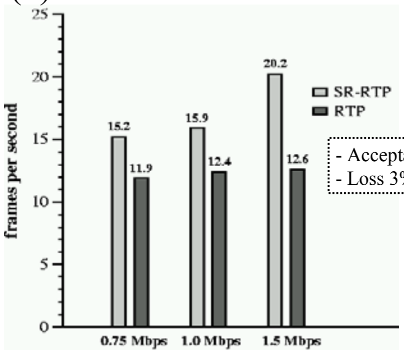
- Server on P4, 1.5 GHz, Linux 2.2.18
- Client on P2, 233 MHz, Linux 2.2.9
- 1.5 Mbps, 50 ms latency, 3% loss
    - using Dummynet, a WAN emulator
- 20 Kbps video at 30 fps
    - Used only 300 frames
- For "Internet", used 200 ms RTT and used Web cross traffic with SURG (traffic emulator)
- Added buffering to combat jitter
    - (Me: how and how much is not specified)

Benefits of Selective Reliability (1)



- 200 ms RTT
- Other PSNRs are similar

Benefits of Selective Reliability (2)

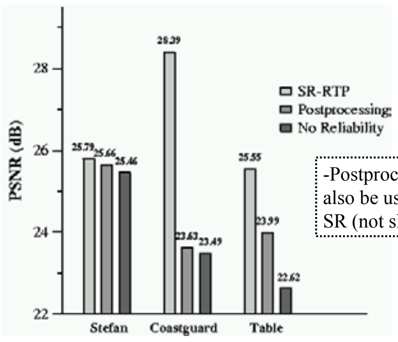


- Acceptable PSNR 25
- Loss 3%

Buffering Requirements

- Buffer for jitter depends upon variance
- Buffer for retrans depends upon RTT
- Buffer for quality adaptation (congestion responsiveness) depends upon data rate (R)
    - O(R) for SQRT (TFRC)
    - O(R^2) for AIMD (TCP)
- Dominant factor depends upon RTT to RTT variance and rate
- (Me: no more analysis than above)

Benefits Receiver Postprocessing



-Postprocessing can also be used with SR (not shown)

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Related Work (1)

- Media Transport
  - RTSP for control [49]
  - RTP is application level protocol with real-time data properties (timing info + sequence) [48]
  - RTCP protocol provides reports to sender [40]
  - TFRC [52], CM, RAP[44]
    - all TCP-Friendly protocols for streaming media

Related Work (2)

- Error and Loss Recovery
  - Survey of techniques [54]
  - Receiver post-processing [22]
  - Avoid propagation but don't delay [54]
  - Effects of MPEG-4's built in repair for bit errors [23]
  - FEC schemes [9,10,33]
  - Priority-based packets [39,50,1]

Conclusion

- Streaming video must account for loss
- This paper models loss to explain effects
- Can use retransmission of important packets for significant gain
- Describe system to do so based on extensions of common tools

Future Work?

Future Work (Me)

- Wider-range of videos
  - Motion content
  - GOPs
  - Resolution
- Alternate measures of quality
- Evaluation of buffering