A Selective Retransmission Protocol for Multimedia on the Internet

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Applications:
Text-Based vs. Multimedia
• Text
  – Strict loss constraints
  – Minimal timing constraints
• Multimedia
  – Forgiving to loss
  – Requires timing constraints

Protocols:
TCP vs. UDP
• TCP
  – No loss
  – Retransmits all lost messages
  – Potentially large latency
• UDP
  – Potentially unbounded loss
  – Does no retransmission
  – Minimal latency
• Neither is what you want!

Our Solution:
A Selective Retransmission Protocol
• Balances the extremes of TCP and UDP
• Tradeoff between loss and latency
• Retransmits a percentage of lost packets
  – If end-to-end delay is large, may accept loss
  – If end-to-end delay is small, may always request retransmission
  – If loss rate is very high, may request retransmission
  – How to decide?

Groupwork
• Measure of loss
• Measure of latency
• Packet is lost
• … Do you request retransmission?
• Consider:
  – Quiet WAN, interactive audio
  – LAN, broadcast video
  – Lossy MAN, interactive audio

Decision Algorithms

Increasing Loss
(Kleinrock, 1992)
Decision Algorithms

- Increasing Loss
- Increasing Latency

Policies
- Acceptable Quality
- Equal Loss Latency (ELL)
- Optimum Quality (OQ)

Approach

- Implement SRP and “application”
- Setup “WAN” test-bed
- Run “application” over
  - TCP - No loss - Low latency
  - UDP - Medium loss - Medium latency
  - SRP - High loss - High latency
- Measure “Quality”
- Analyze Results

Implementation of SRP

- Application layer client/server protocol
  - No “kernel hacking” (yet)
  - Built on top of UDP
- Measure loss and latency
  - Use to decide when to request retransmission
- Decision algorithm modular
  - Equal Loss Latency (ELL)
  - Optimum Quality (OQ)

Sample SRP Session

- Data Block
- Time
- (Sequence Numbers)

Experiments

- UDP traffic generator
- Token bucket router to control loss and latency
- Audio session 8000 bytes/sec
  - Sample rate 160ms, packet size 1280
Sample Data

Low Loss, Low Latency

High Loss, High Latency

Conclusions

• TCP and UDP provide extremes
  – Not what Multimedia wants
• SRP can provide a balance
• Tuning of SRP depends upon
  – Application
  – Measure of “quality”
  – Measurement of network (loss, RTT)

Future Work

• Repair (FEC)
• Congestion control
• Loss detection (timeout)
• Additional decision algorithms
• Multicast

Future Work?