EE 368C Project

Multi-stream Audio Transmission with Path Diversity

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The Incentives

- Best-effort services vs. strict QoS requirements of real-time speech communication, e.g. latency, loss, delay variation etc.
- Low data rate of the voice stream
- Why alternative path/multi-path?
  - Exists a superior alt. path in 30-80% cases [Savage, 99’]
  - Path diversity – network behavior averaged; burst loss converted to isolated loss; outage probability decreased [Apostolopoulos, 01’]
  - Multi-path – independent jitter behavior
- Realization: explicitly path selection using relay servers

The bottleneck

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Outline

- Background on media playout scheduling algorithms
- Adaptive playout for multiple streams
- Measurements over the Internet and results
- Ns simulation and results
- Performance analysis of multi-stream transmission
Playout Scheduling Algorithms

- Tradeoff between delay and loss, wish to reduce both
- The adaptive playout scheme for *single* stream – jitter adaptation
Can always take the packet with lower delay

Adaptive playout and speech scaling make seamless switching between streams possible; question: *setting playout schedule*?

Multiple description coding; question: *audio quality*?
Determine the Playout Schedule

- To minimize the Lagrange cost function
  \[ C = \text{delay} + \lambda_1 \times p \text{ (packet from both streams lost)} + \lambda_2 \times p \text{ (one packet lost)} \times (\text{audio quality degrad. when losing one packet}) \]
  \[ = d + \lambda_1 \times \text{lp}_1 \cdot \text{lp}_2 + \lambda_2 \times [\text{lp}_1 (1- \text{lp}_2) + \text{lp}_2 (1- \text{lp}_1)] \times SNR_{\text{degrad}} \]

- Maintaining history for both streams; loss probability determined by delay and past history (order statistics)

- Greater \( \lambda_1 \) results in lower loss rate at the cost of higher delay
- Greater \( \lambda_2 \) results in both lower loss rate and better audio quality, at the cost of higher delay
- Increasing \( \lambda_1 \) without big \( \lambda_2 \) leads to lower loss rate, but not necessarily better sound quality
- Small \( \lambda_1 \) and \( \lambda_2 \) result in low delay
MDC over Multiple Streams

- Multi-stream with MDC
  **Stream 1:**
  - Even samples: quantized in finer resolution (8-bit)
  - Odd samples: quantized in coarser resolution (4-bit)
  **Stream 2:** the other way
  [Jiang, 00’]

- Multiple description coding (MDC): generates multiple descriptions of equal importance for the same source signal

- SNR degradation

<table>
<thead>
<tr>
<th>Stream 1</th>
<th>√</th>
<th>√</th>
<th>×</th>
<th>×</th>
</tr>
</thead>
<tbody>
<tr>
<td>Stream 2</td>
<td>√</td>
<td>×</td>
<td>√</td>
<td>×</td>
</tr>
<tr>
<td>SNR_{degrad} (dB)</td>
<td>0</td>
<td>-21</td>
<td>-21</td>
<td>-27 (after conceal.)</td>
</tr>
</tbody>
</table>
Comparison: Single-stream with FEC

Stream sent

Stream received with packet loss

Stream reconstructed

- **FEC protected single-stream**
- For fair comparison
  - Primary copy: quantized in finer resolution (8-bit)
  - Secondary copy quantized in coarser resolution (4-bit)
- **SNR degradation**
  - $\text{SNR}_{\text{degrad}}$ (dB) -24 -27 (after conceal.)
- **Same data rate as multi-stream MDC**

FEC: adds redundancy by sending multiple copies of the source signal in the following packet(s) [Bolot, 96']
Experiments over the Internet (I)

- Path 1 (direct): Netergy – MIT
- Path 2 (alternative): Netergy – Harvard – MIT
- Direct path: 30ms UDP packets sent from source to dest.; routes determined by routing algorithm
- Alt. path: packets sent from source to relay server, then forwarded to dest.
Delay – loss curve obtained by varying $\lambda_1$ while keeping $\lambda_2$ small and fixed.

- Mean delays (ms): 72.4/60.3
- Link loss rate: 0.02%/0.85%

- Observed significant reduction in delay and loss rate by using multiple streams.
- Total/burst loss rate greatly reduced since jitter averaged.

Path 1 (direct): Netergy – MIT
Path 2 (alt): Netergy – Harvard - MIT
Results of Experiment I (2)

- Results obtained by varying $\lambda_2$ while keeping $\lambda_1$ fixed
- With higher delay: better chances to play both descriptions
- Observed lower playout rate variation by using multiple streams
- Jitter averaged; lower STD of $\min(d_i, d_j)$
Results of Experiment I (3)

Playout of packets from multiple streams
Experiments over the Internet (II)

- Path 1 (direct): N. J. – Germany
- Path 2 (alternative): N. J. – Harvard – Germany
Results of Experiment II (1)

- **Mean delays (ms):** 61.3/65.0
- **Link loss rate:** 0.6%/1.1%
- Burst loss rate can still be reduced by more than 3%, since jitter averaged.

Path 1 (direct): N. J. – Germany
Path 2 (alternative): N. J. – Harvard – Germany
Results of Experiment II (2)

Path 1: N.J. - Germany; Path 2: N.J. - Harvard - Germany

Path 1: N.J. - Germany; Path 2: N.J. - Harvard - Germany

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More Comments on Our Experiments

Measurements by [Savage, 99]:

- Not all data collected at the same time; conclusions made on averaged large aggregate of data
- Round trip time; ICMP packets; sampling frequency not high enough
- Not able to observe jitter behavior for streaming multimedia study

Measurements by us:

- Transmission and receiving over multiple paths made at the same time
- One-way delay; UDP packets; 20 or 30 ms sampling rate
- Can observe jitter behavior for different paths
- Collected data valuable for further study of streaming multimedia
Simulations Using Network Simulator

- **Ns**: packet by packet event driven simulator
- **Simulation parameters**
  - Link BW: 10Mbps
  - Switch buffer: 100k byte/port
  - Prop. delay on each link: 20ms
  - TCP window size: 16k byte
  - **N**: # of data sources attached to each intermediate hop
  - **Load**: amount of traffic sent by each host
- **Voice traffic model**: CBR 64kbps
- **Data traffic model**: log normal, based on “Workload Characterization of the 1998 World Cup Web Site” [Arlitt, 99]

The simulation topology
Loss Reduction

- Losses of each path increase as network load goes up.
- Multiple paths: loss rate reduced; burst loss isolated.
- Multiple paths: loss/burst loss rate increases more gracefully as traffic load goes up.

![Link loss rate vs. N graph]

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Delay Reduction

- Average delay reduction from stream $i$ defined as:
  $\mathbb{E}[d_i - \min(d_i, d_j)]$
- Most gain from multi-path when prop. delays are close
- Delay reduction increases as delay STD goes up
Conclusions and Future Work

- Multiple streams with path diversity:
  - Reduces loss rate – due to averaged jitter, isolated burst loss and outage period
  - Reduced delay by taking packet with lower delay from multiple streams
  - Smoothed delay variation
- MDC works well with multi-stream adaptive playout, and makes audio quality scalable
- Performance gain affected by prop. delay and delay STD
- Multi-stream transmission can be realized by future peer-to-peer frame

- Applying the scheme onto more loss-sensitive applications, such as streaming video
- Improving topology and traffic model in ns simulation
Coupling of Multiple Paths

- This topology studies using two paths that are not completely independent.
- One link is shared by two streams, over which loss/delay behavior is not independent.

Simulation topology B
Losses over Coupled Paths

Topology A

Conditional loss probability

Number of Hosts Attached to Each Node

Loss rate reduction

Topology B

Conditional loss probability

Number of Hosts Attached to Each Node

Loss rate reduction

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