Simulation of VoIP using NS-2

ENSC 427: Communication In Networking

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Background

• VoIP is being used more and more every year (Rogers and Vonage)

• Capitalizes on the versatility of IP networks:
  o Lower operating costs (common computer equipment)
  o Integrate many web services with VoIP
  o Potentially more bandwidth-efficient due to availability of different codecs
Why VoIP?

<table>
<thead>
<tr>
<th>Reason</th>
<th>Percentage</th>
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<tbody>
<tr>
<td>Lower telecommunications costs</td>
<td>66%</td>
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<tr>
<td>Desire to merge voice and data networks</td>
<td>43%</td>
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<tr>
<td>Obtain a platform for one-stop communications in two or more areas</td>
<td>41%</td>
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<tr>
<td>Increase collaboration benefits in two or more areas</td>
<td>36%</td>
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<tr>
<td>Ease of management</td>
<td>31%</td>
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<tr>
<td>Scalability</td>
<td>24%</td>
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Objectives

• Implement VoIP phone call between two users

• Create background traffic to simulate real life situation

• Background traffic increases as time elapses

• Test UDP, TCP, and RTP
Overview of Related Work

  o Studied two VoIP calls made over a bottleneck link with a Droptail queue
  o Used UDP and TCP with CBR for each respective call
  o Mainly looked at packet loss

• Marc Greis’ Tutorial on NS-2
Quality of Service

- ITU-T Recommendation G.114
- 150ms end-to-end delay or less is recommended
- 400ms maximum acceptable delay for international calls
- Keep packet delay variation (jitter) as low as possible
- Packet losses of about 5% are tolerable (based on distribution)
- In general, large delay is more undesirable than loss of quality
Implementation

• Simulation done in NS-2 v2.35

• NS-2 trace file filtered with AWK to remove background traffic

• Resulting trace file parsed with MATLAB

• MATLAB script used to calculate and plot throughput, end-to-end delay, packet loss, and jitter.
Technical Specifications

• OC-1 Link (51.84Mbps)

• G.711 Audio Codec (64kbps)

• Nation-wide call (Vancouver to Toronto)

• Background traffic increase as time elapses
  o 25.89 Mb/s both ways
  o 25.91 Mb/s both ways
  o 25.92 Mb/s both ways
Initial layout
During simulation:
Throughput - UDP
Throughput - TCP
Delay - UDP
Delay - TCP
Packet loss - UDP
Packet loss - TCP

TCP: Node 0 -> Node 4 Instantaneous Packet Loss

TCP: Node 0 -> Node 4 Cumulative Packet Loss
Jitter - UDP

UDP: Node 0 --> Node 4 Jitter

UDP: Node 4 --> Node 0 Jitter
Jitter - TCP

TCP: Node 0 -> Node 4 Jitter

TCP: Node 4 -> Node 0 Jitter
Results

- Much more packets are lost for UDP/RTP
- Very low end-to-end delay and jitter for UDP/RTP
- The large end-to-end delay and jitter of TCP makes it unacceptable for VoIP
- Throughput/packet loss of UDP/RTP acceptable for network under minimal load
Future work

• Finish the rest of the work for the project and reports

• Future future future work (aka not now)
  o Adding SIP
What did we end up with?

- A pretty awesome project
- A better knowledge of how the three protocols work
- Better understanding of NS2 and its capabilities
- A presentation :D
References

Any Questions?