Simulation of VoIP using NS-2

ENSC 427: Communication In Networking

Group 9:
Behdad Jamshidi
Victor Mateescu

Background

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

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

Why VoIP?

Reason	Percentage
Lower telecommunications costs	66%
Desire to merge voice and data networks	43%
Obtain a platform for one-stop communications in two or more areas	41%
Increase collaboration benefits in two or more areas	36%
Ease of management	31%
Scalability	24%

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

- Jishu Das Gupta, Srecko Howard, and Angela Howard (2006), "Traffic Behaviour of VoIP in a Simulated Access Network," *Proceedings of World Academy of Science, Engineering and Technology (PWASET)*, *18*, pp. 189-194.
    - Studied two VoIP calls made over a bottleneck link with a Droptail queue
    - Used UDP and TCP with CBR for each respective call
    - 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

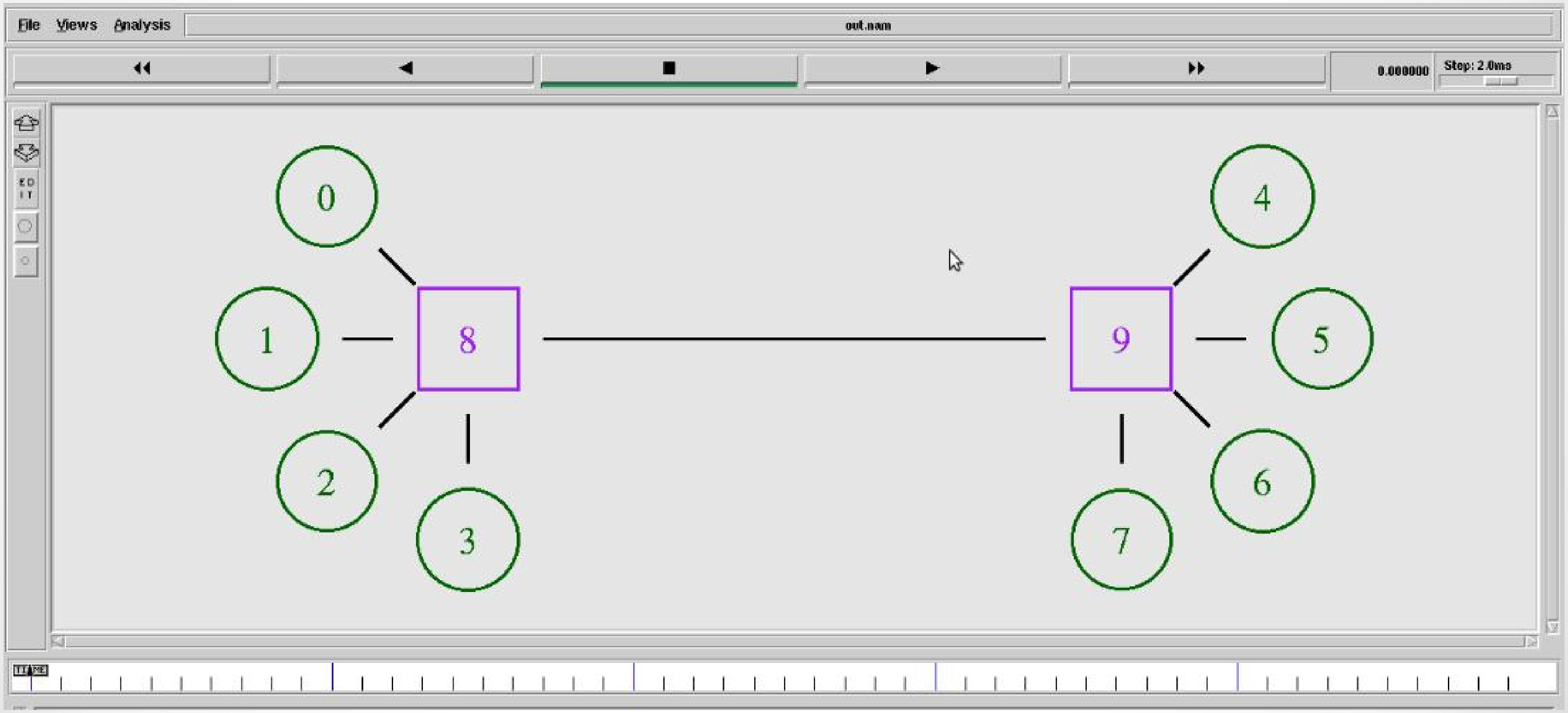
Implementation

- 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.

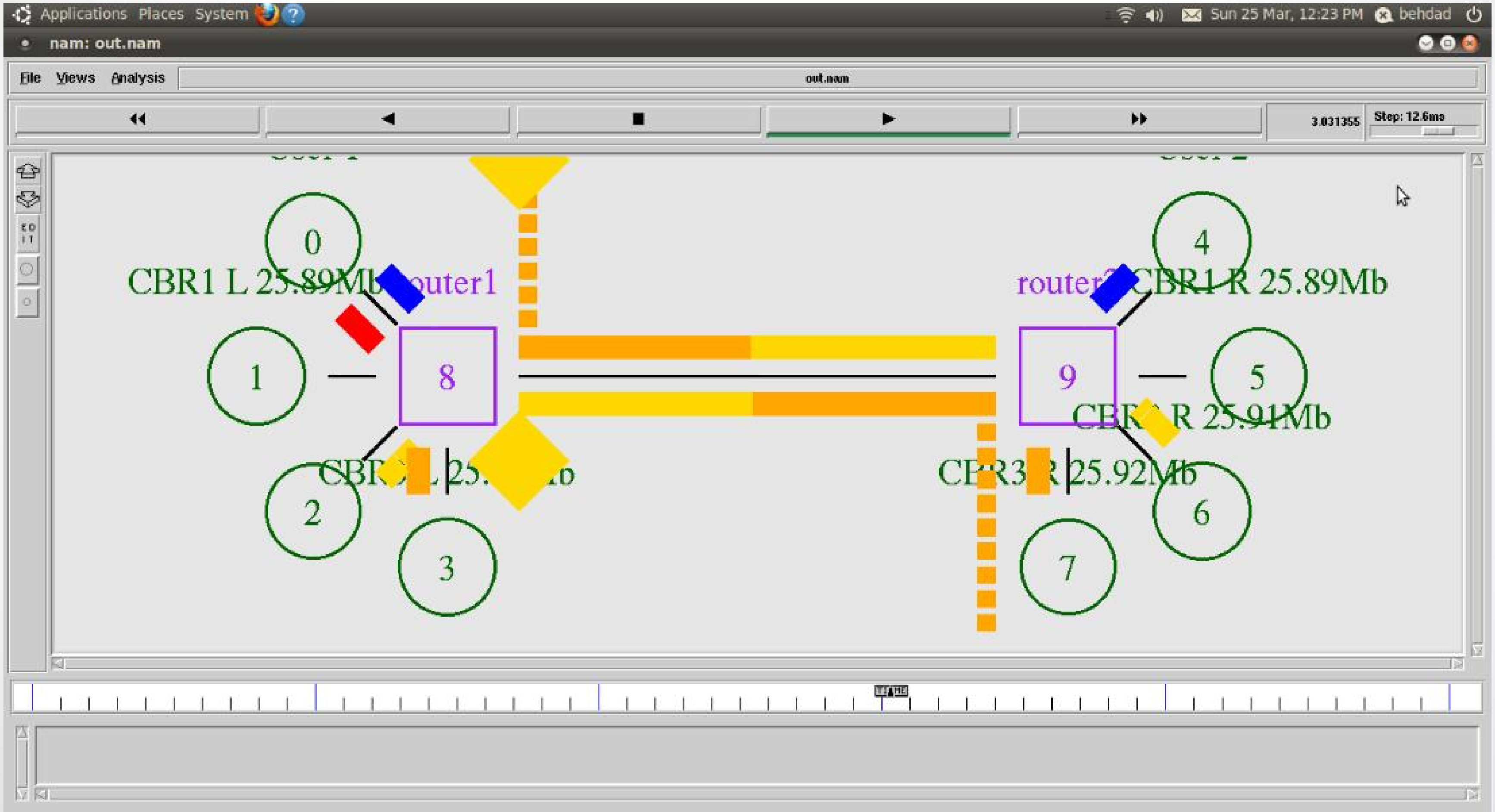
Technical Specifications

- OC-1 Link (51.84Mbps)
- G.711 Audio Codec (64kbps)
- Nation-wide call (Vancouver to Toronto)
- Background traffic increase as time elapses
  - 25.89 Mb/s both ways
  - 25.91 Mb/s both ways
  - 25.92 Mb/s both ways

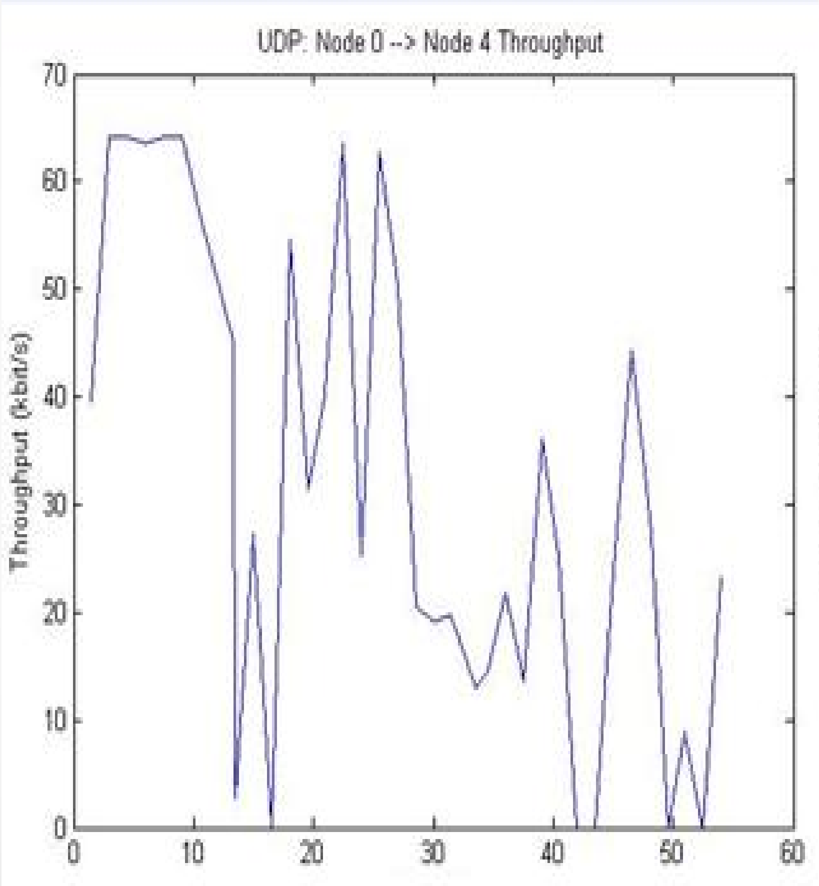
Initial layout

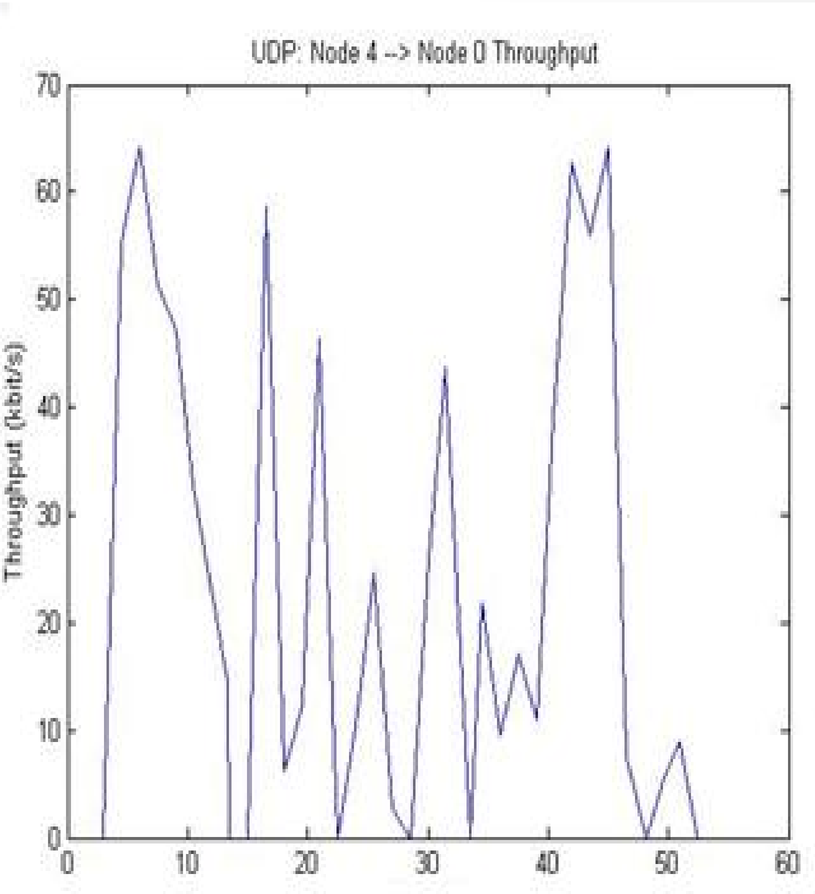


During simulation:

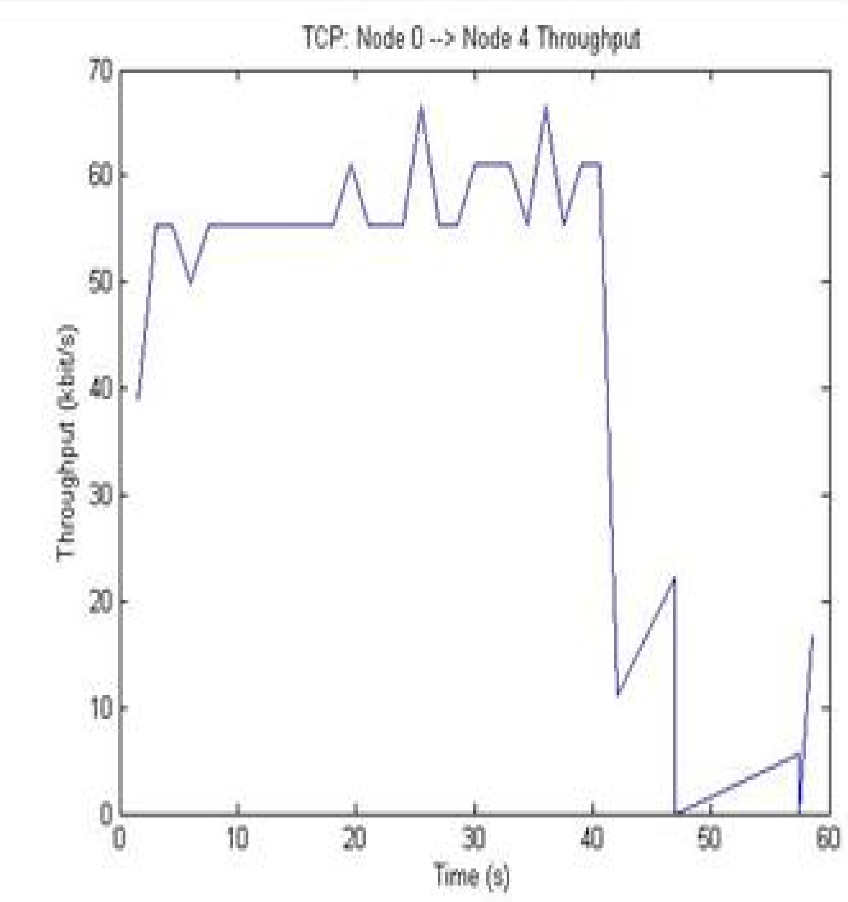


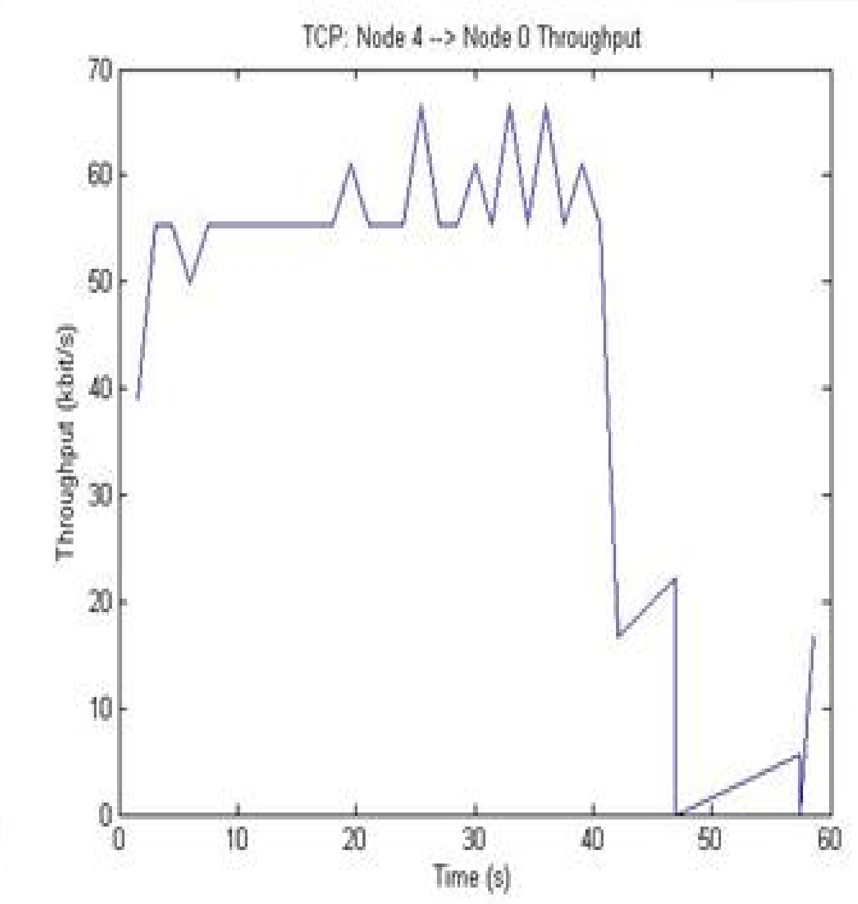
Throughput - UDP



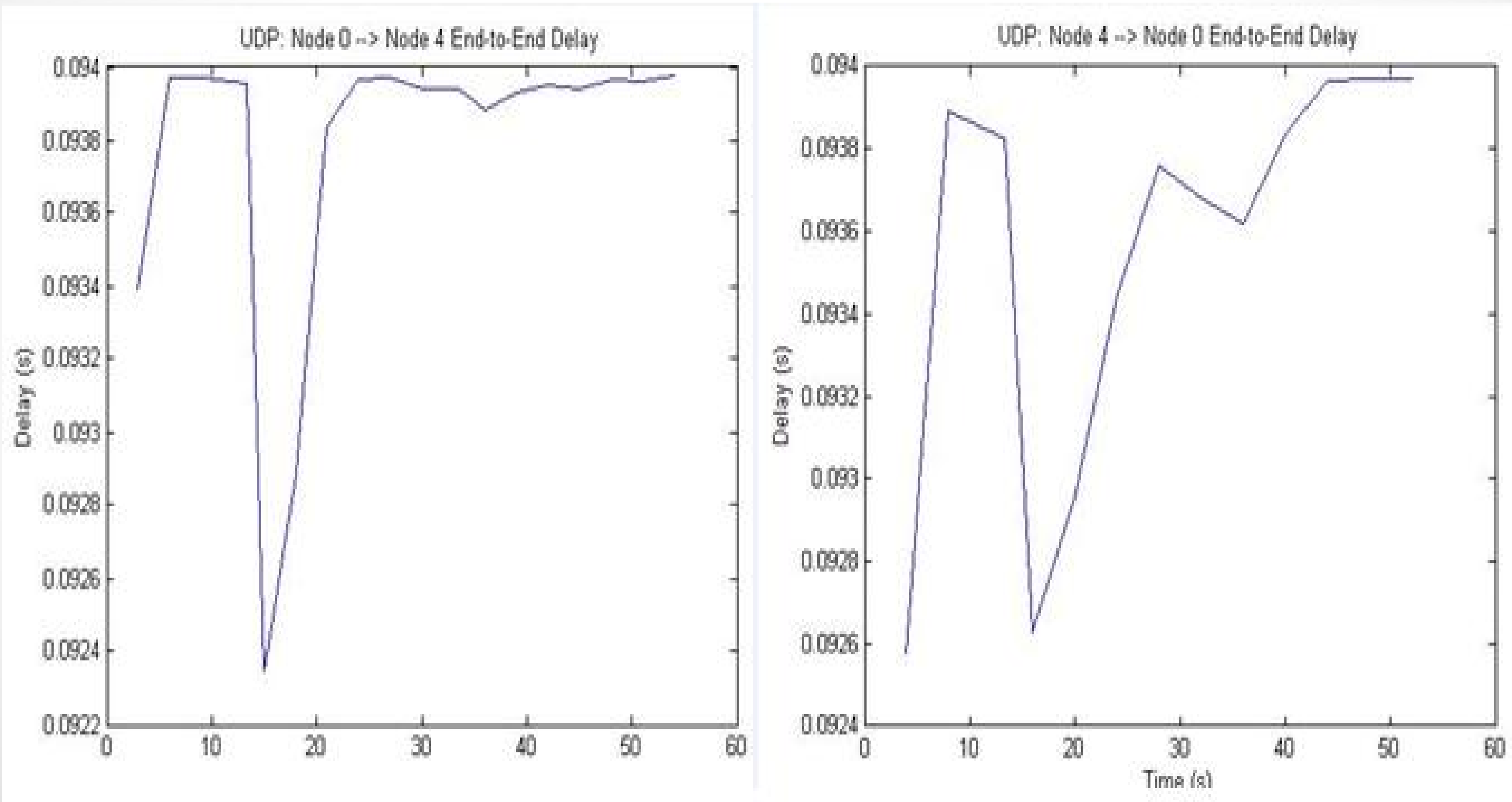


Throughput - TCP



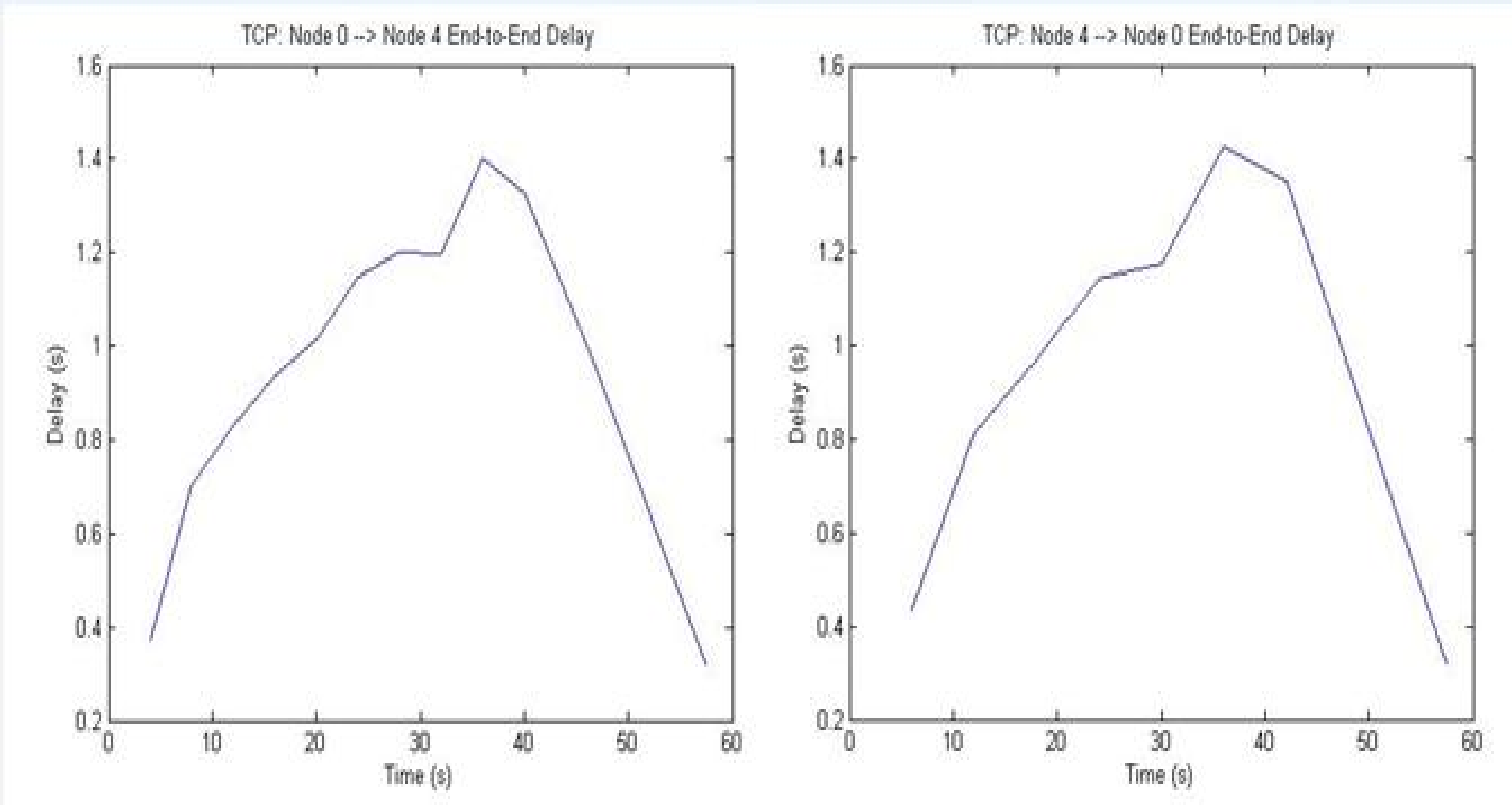


Delay - UDP

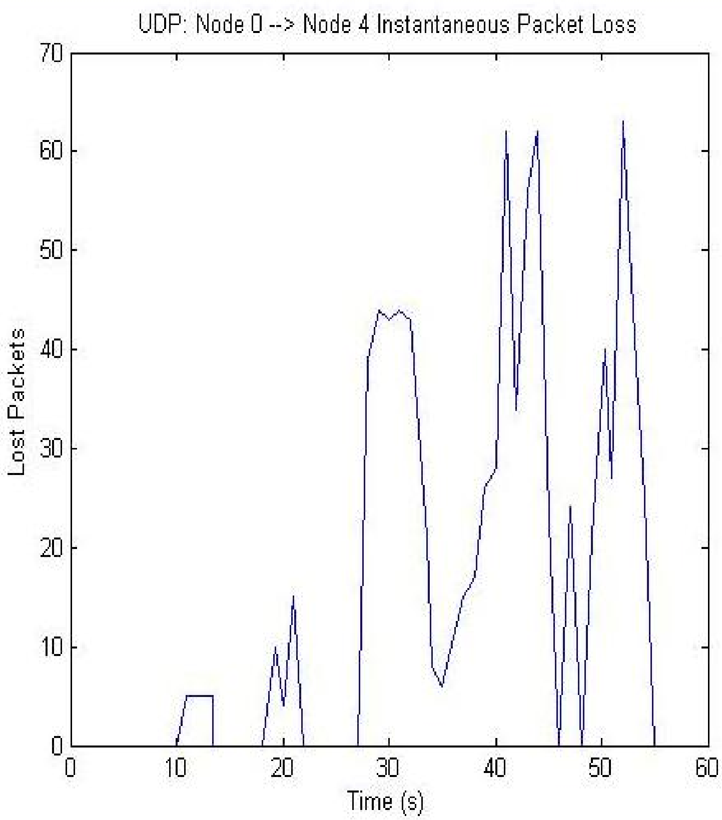


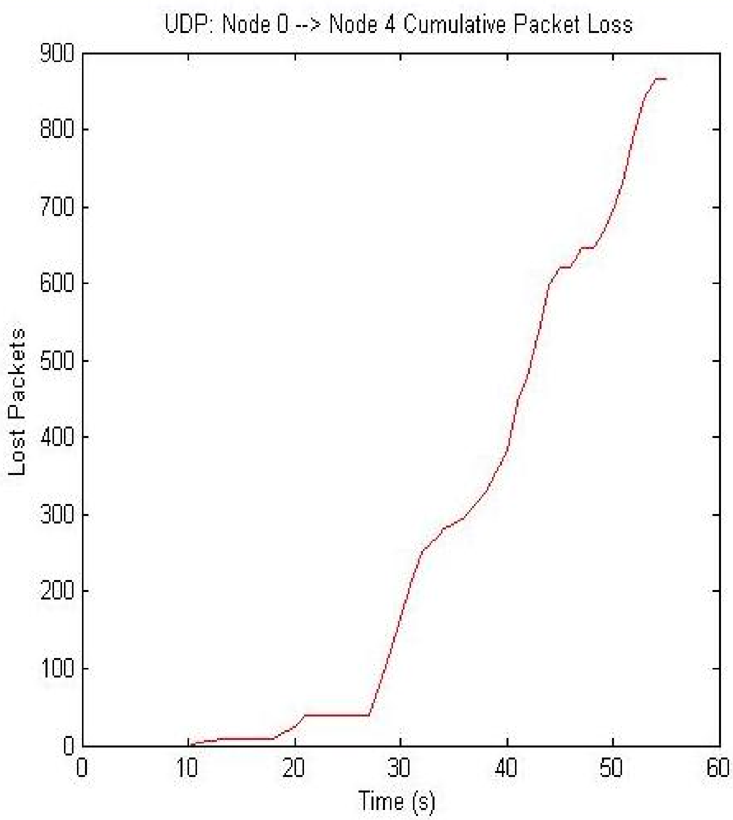
UDP: Node 0 --> Node 4 End-to-End Delay
Delay (s)
0.094
0.0938
0.0936
0.0934
0.0932
0.093
0.0928
0.0926
0.0924
0.0922
0
10
20
30
40
50
60
UDP: Node 4 --> Node 0 End-to-End Delay
Delay (s)
0.094
0.0938
0.0936
0.0934
0.0932
0.093
0.0928
0.0926
0.0924
0
10
20
30
40
50
60
Time (s)

Delay - TCP

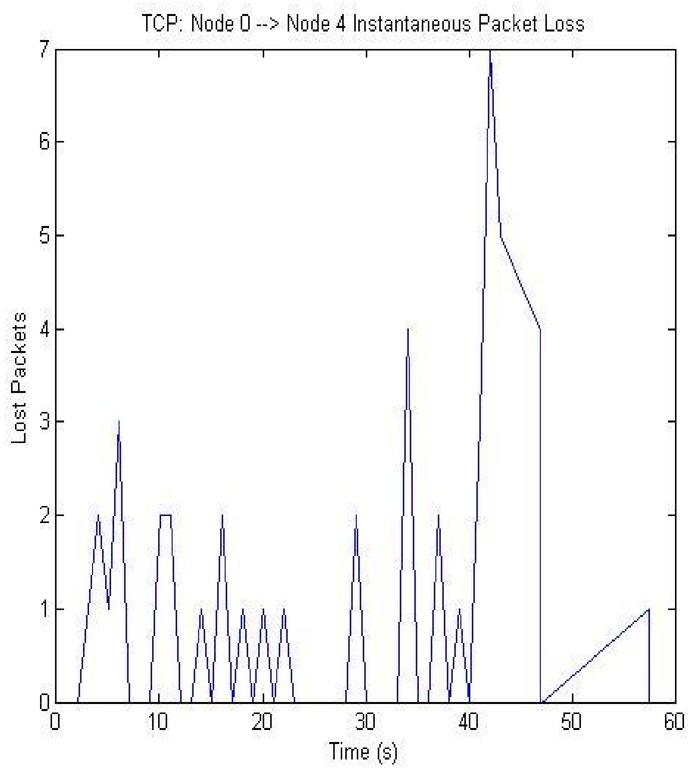

TCP: Node 0 --> Node 4 End-to-End Delay
Delay (s)
Time (s)
TCP: Node 4 --> Node 0 End-to-End Delay
Delay (s)
Time (s)

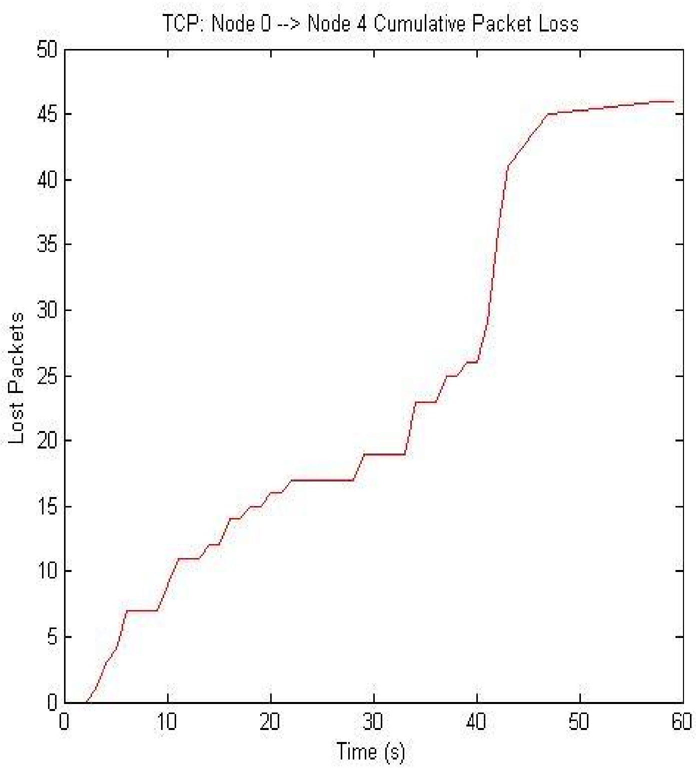
Packet loss - UDP



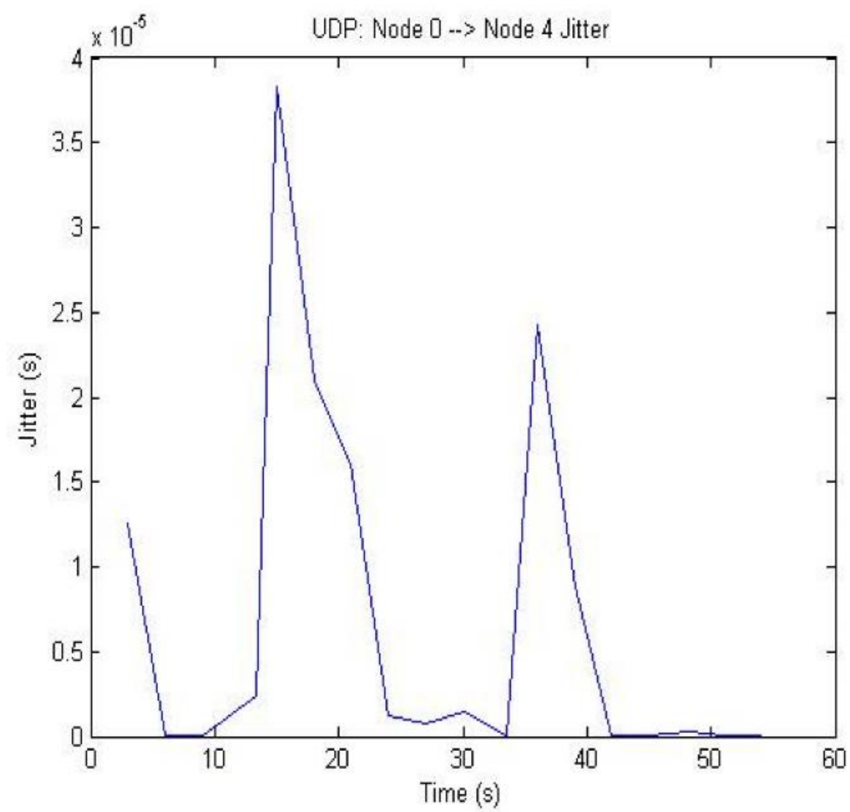


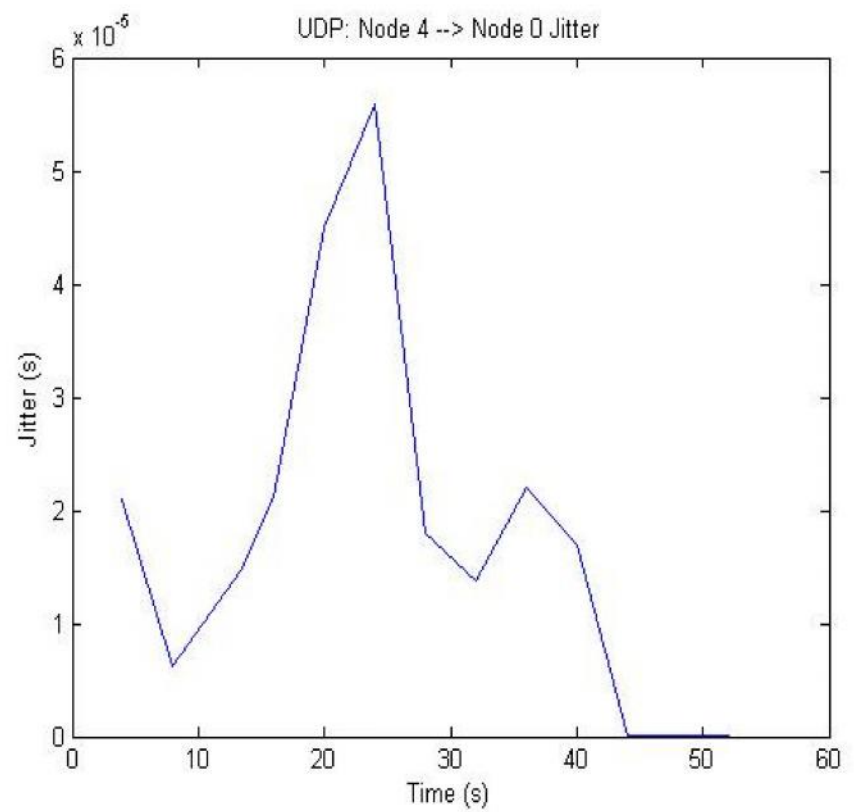
Packet loss - TCP



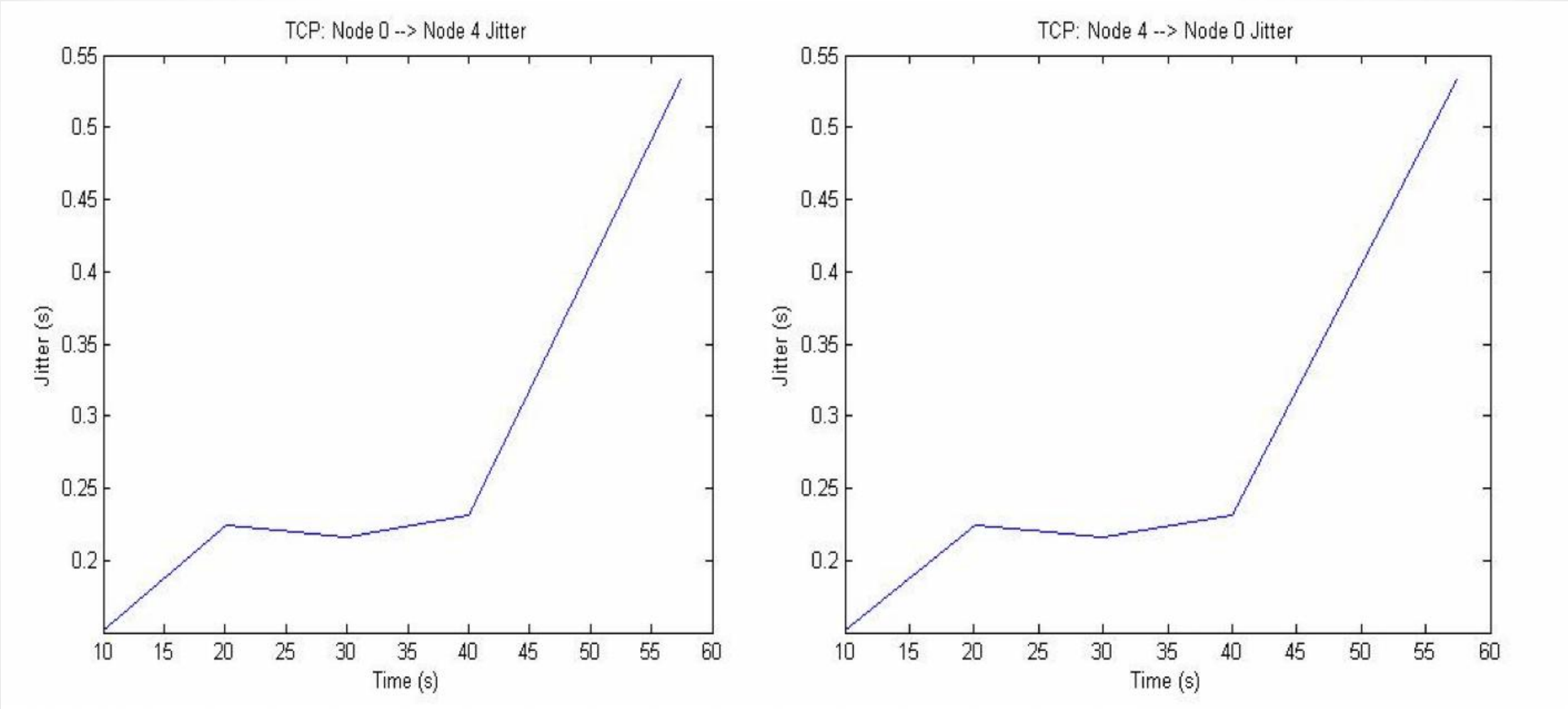


Jitter - UDP


UDP: Node 0 --> Node 4 Jitter
x 10^-5
Jitter (s)
Time (s)
0 0.5 1 1.5 2 2.5 3 3.5 4
0 10 20 30 40 50 60


UDP: Node 4 --> Node 0 Jitter
x 10^-5
Jitter (s)
Time (s)
0 1 2 3 4 5 6
0 10 20 30 40 50 60

Jitter - TCP


TCP: Node 0 --> Node 4 Jitter
Jitter (s)
Time (s)
0.55
0.5
0.45
0.4
0.35
0.3
0.25
0.2
10
15
20
25
30
35
40
45
50
55
60
TCP: Node 4 --> Node 0 Jitter
Jitter (s)
Time (s)
0.55
0.5
0.45
0.4
0.35
0.3
0.25
0.2
10
15
20
25
30
35
40
45
50
55
60

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)
  - 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

- [1] D. Collins, "Carrier Grade Voice Over IP," McGraw-Hill, 2002.
- [2] O. Hersent, "IP Telephony: Deploying VoIP Protocols and IMS Infrastructure," West Sussex, UK: Wiley, 2011.
- [3] C. Demichelis and P. Chimento, "IP Packet Delay Variation Metric for IP Performance Metrics (IPPM)," RFC 3393, November 2002: http://www.ietf.org/rfc/rfc3393.txt (accessed in March 2012).
- [4] A. Leon-Garcia and I. Widjaja, "Communication Networks: Fundamental Concepts and Key Architectures," 2nd edition, McGraw -Hill, 2004.
- [5] Jishu Das Gupta, Srecko Howard, and Angela Howard (2006), "Traffic Behaviour of VoIP in a Simulated Access Network," *Proceedings of World Academy of Science, Engineering and Technology (PWASET)*, *18*, pp. 189-194,. Available: http://www.waset.org/journals/waset/v24/v24-15.pdf (accessed in March 2012)

Any Questions?