VoIP Simulation Network

Team 3
Benson Lam
Winfield Zhao
Mincong(Logan) Luo

Objective of Project

- To simulate a VoIP network and study the behaviour of VoIP under different scenarios:
  - local VoIP call vs. external VoIP call
  - Overload VoIP network
  - Quality of the Internet (discard ratio and link usage)
  - Different Encoder Scheme
- VoIP quality is mainly impaired by:
  - End to End delay
  - Traffic send/receiver
  - Jitter
  - Packet dropped
  - And more…
- In this project, we will analyze these parameters

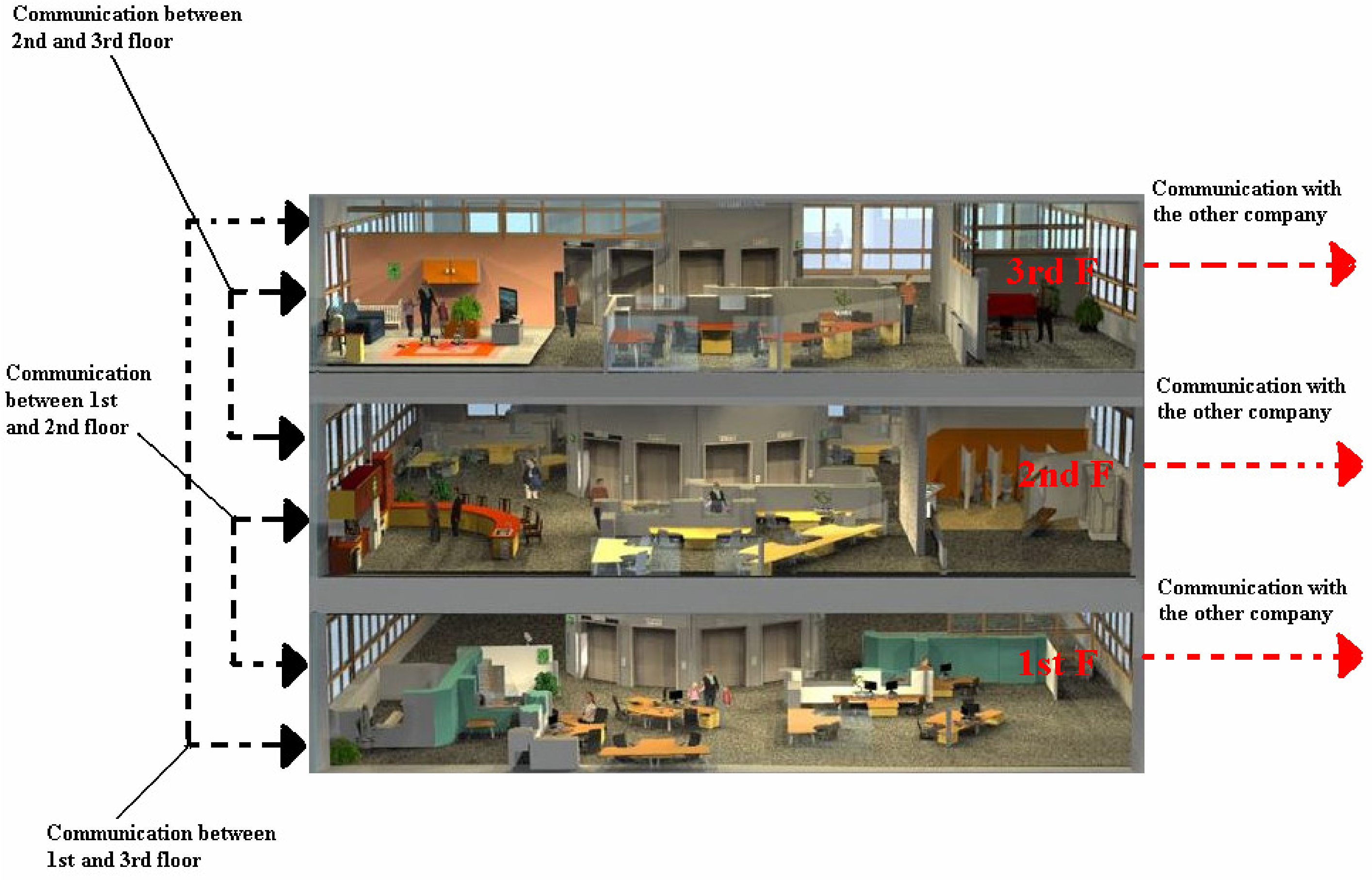
What is VoIP

- Permits communication calls to be made over the internet
- Very similar to the idea of using a microphone to record a voice and saved it in the memory of a computer
- However, audio samples are not stored locally; instead, they are sent over the IP network to another computer

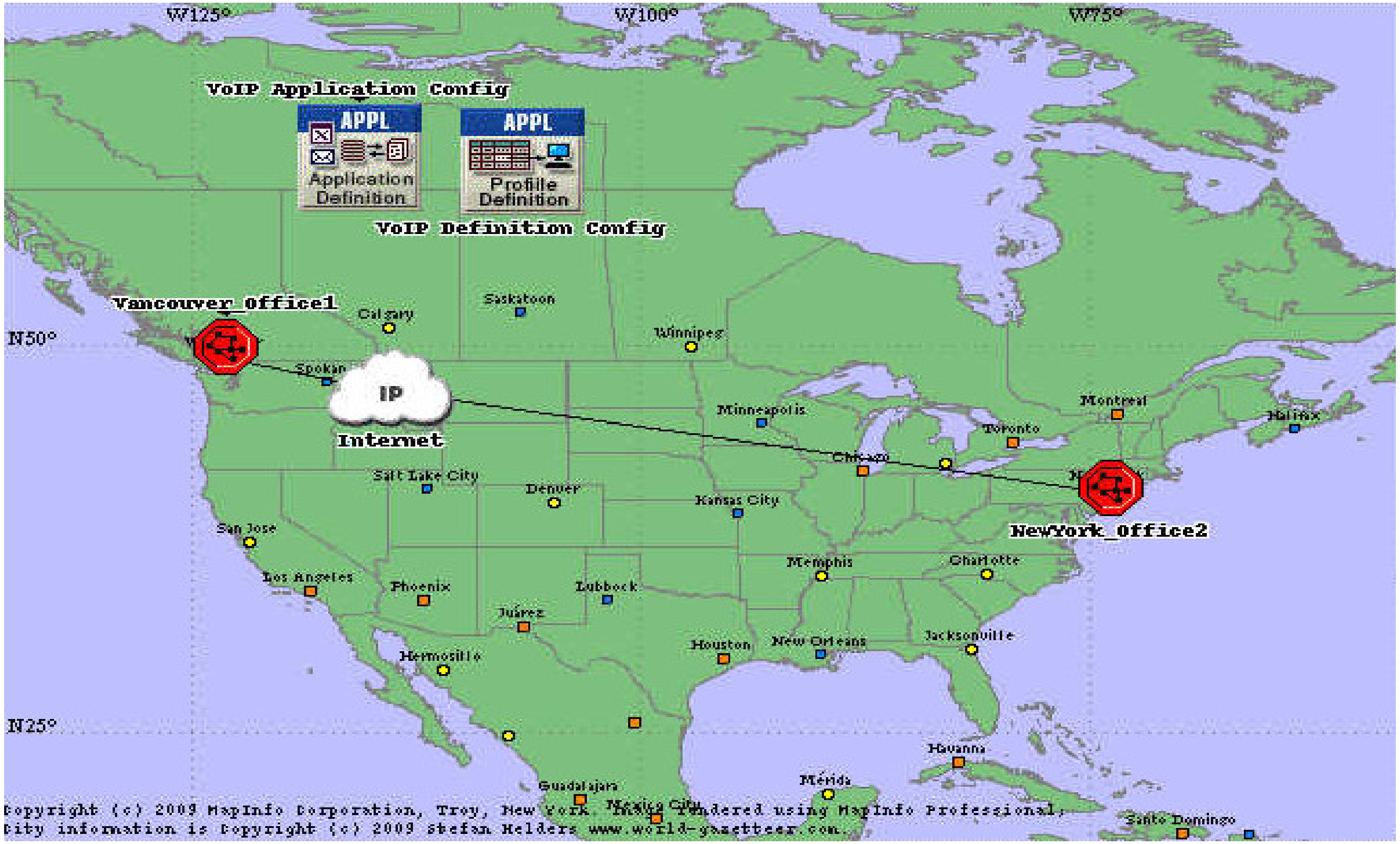
Application

Description of Project

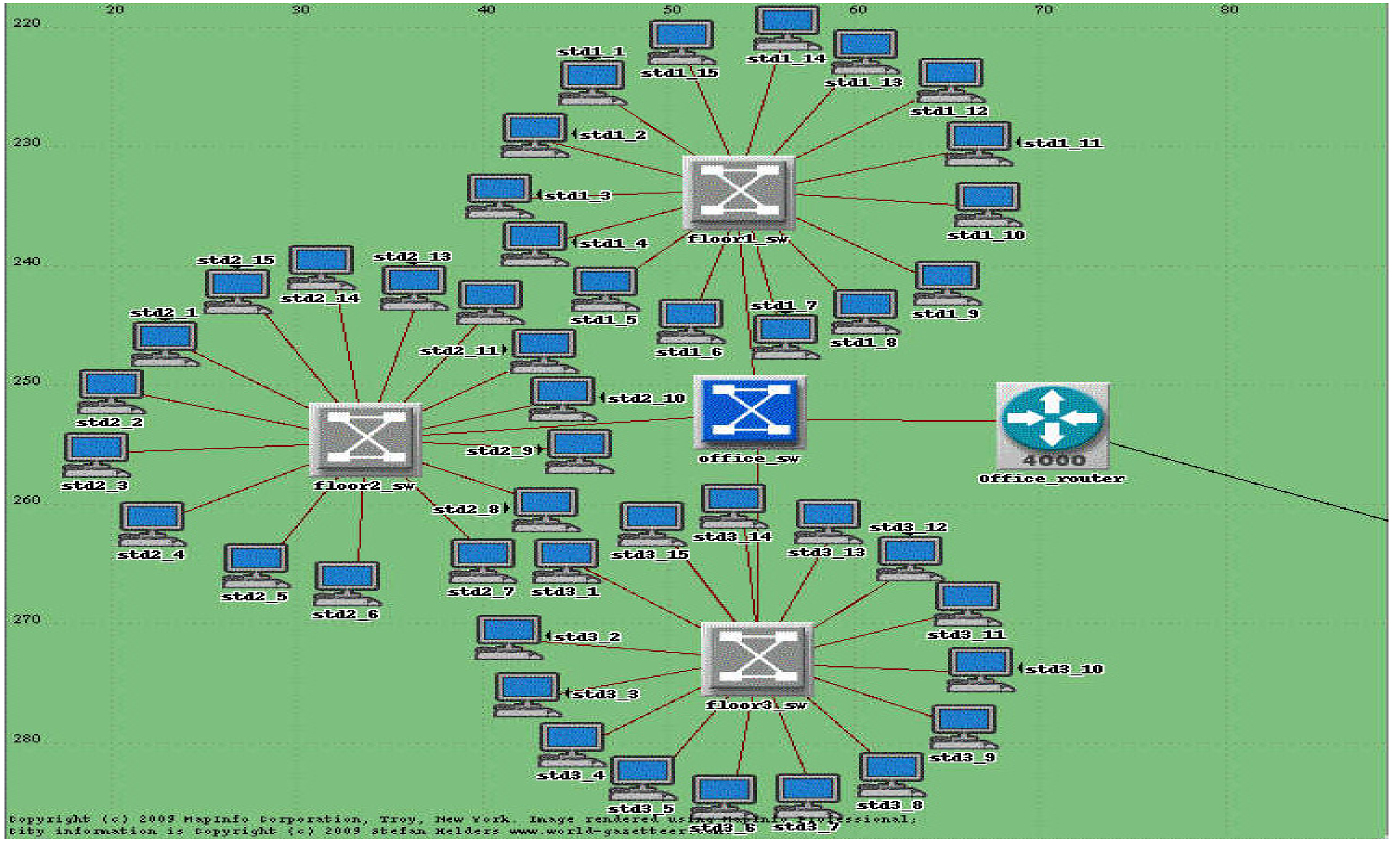
Description of Project



Implementation



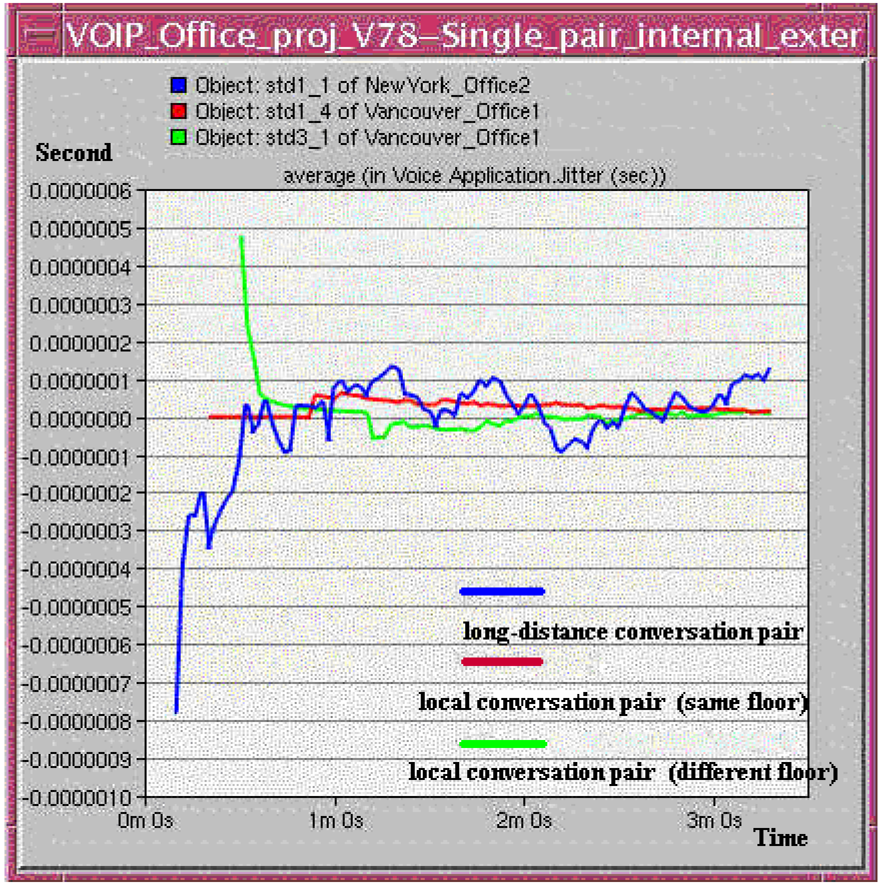
Implementation



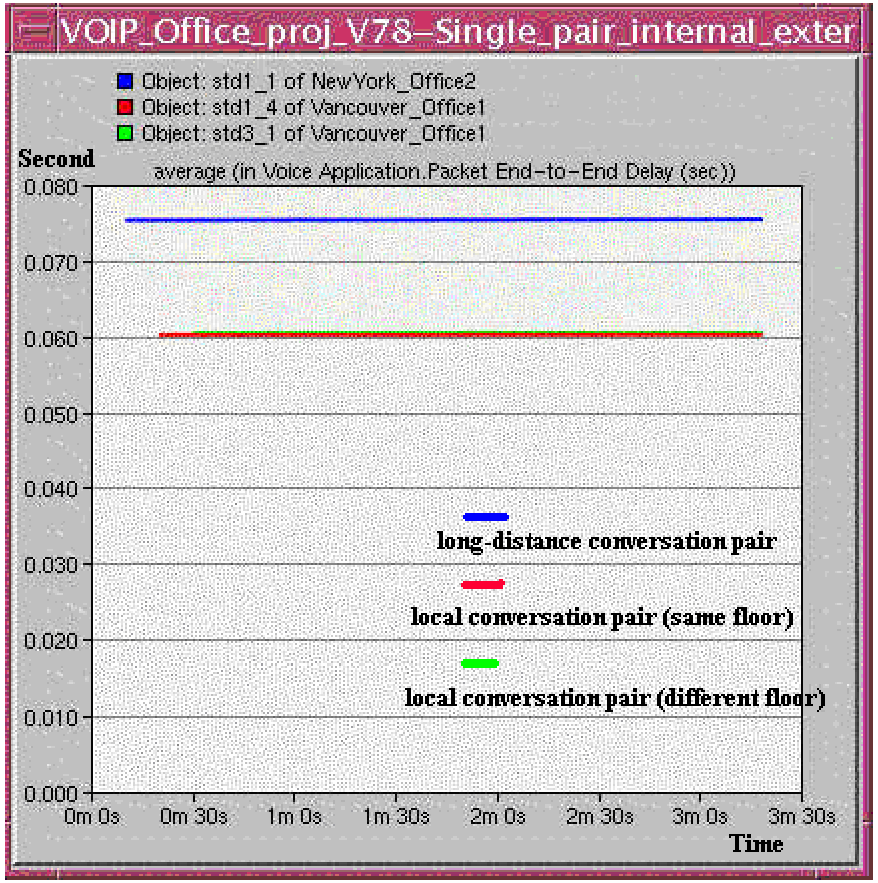
Scenario 1

- Comparison between local and long-distance VoIP calls in term of different parameters
  - Local call in the same floor
    - Vancouver_office.std1_2 -> NewYork_office.std1_4
  - Local call in different floors
    - Vancouver_office.std1_3 -> NewYork_office.std1_4
  - Local call in different offices
    - Vancouver_office.std1_1 -> NewYork_office.std1_1

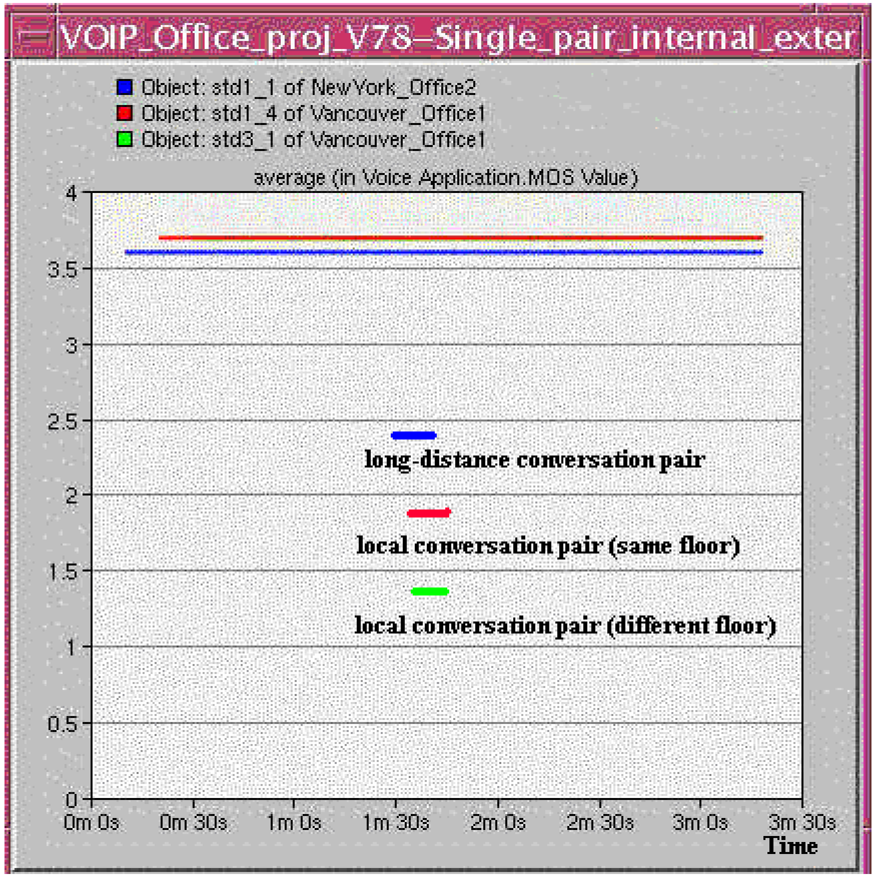
Result: Jitter



Result: End-to-End Delay



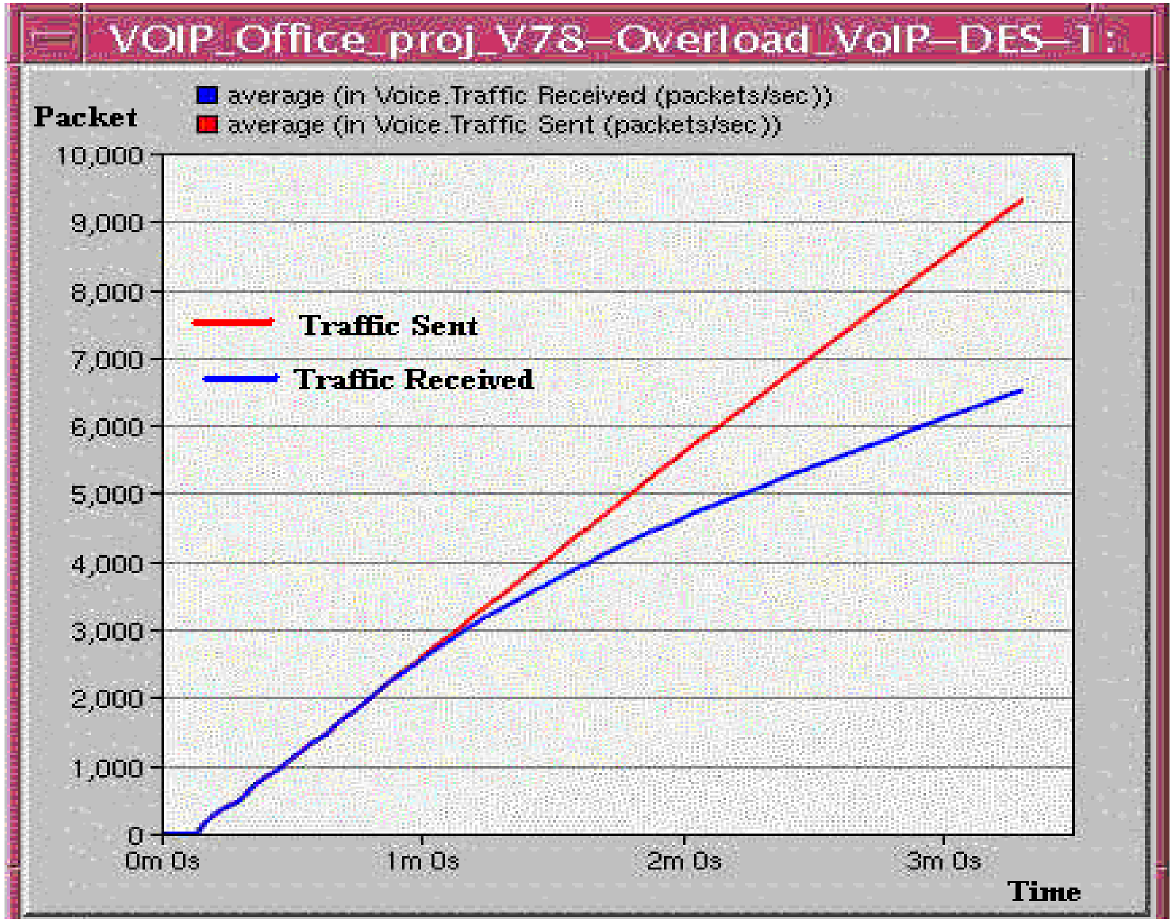
Result: MOS Value



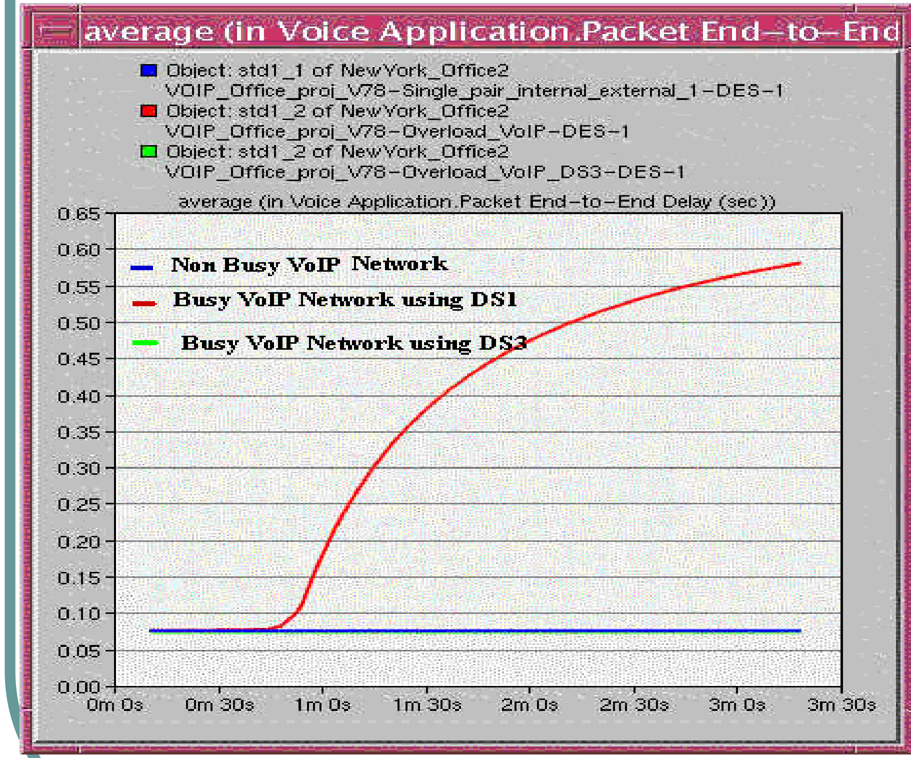
Scenario 2: Busy VoIP Network

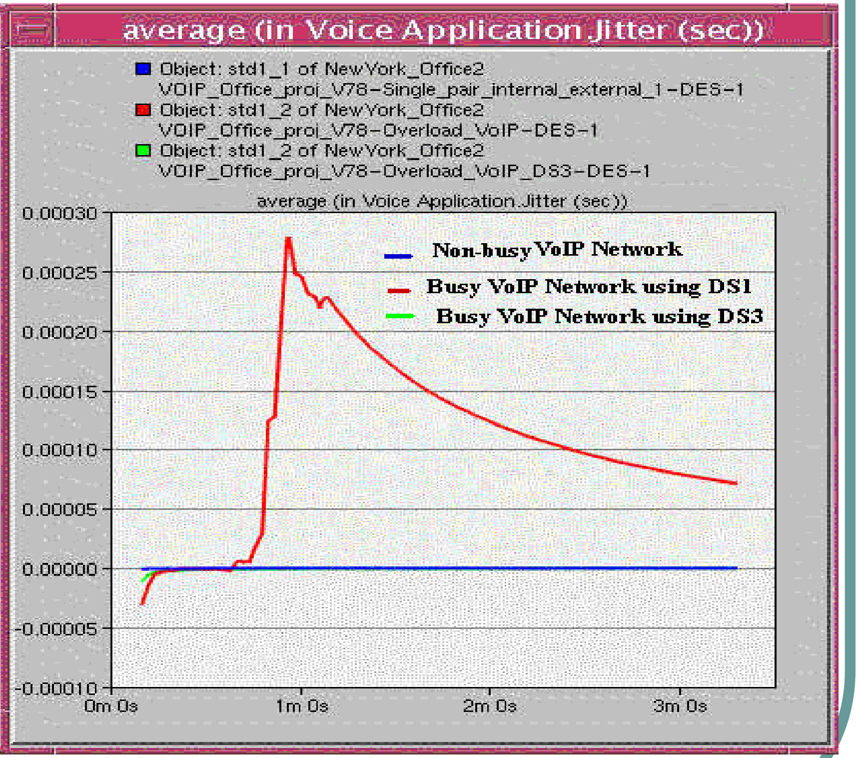
- To compare a busy VoIP network with a Non-busy VoIP network
  - In order to create a busy VoIP network, 15 workstations in each company are set to communicate with 15 workstations in the second company – 15 long-distance conversation pair
- Different link capacity is used in the busy VoIP

Result: Packet Sent Rate and Received Rate

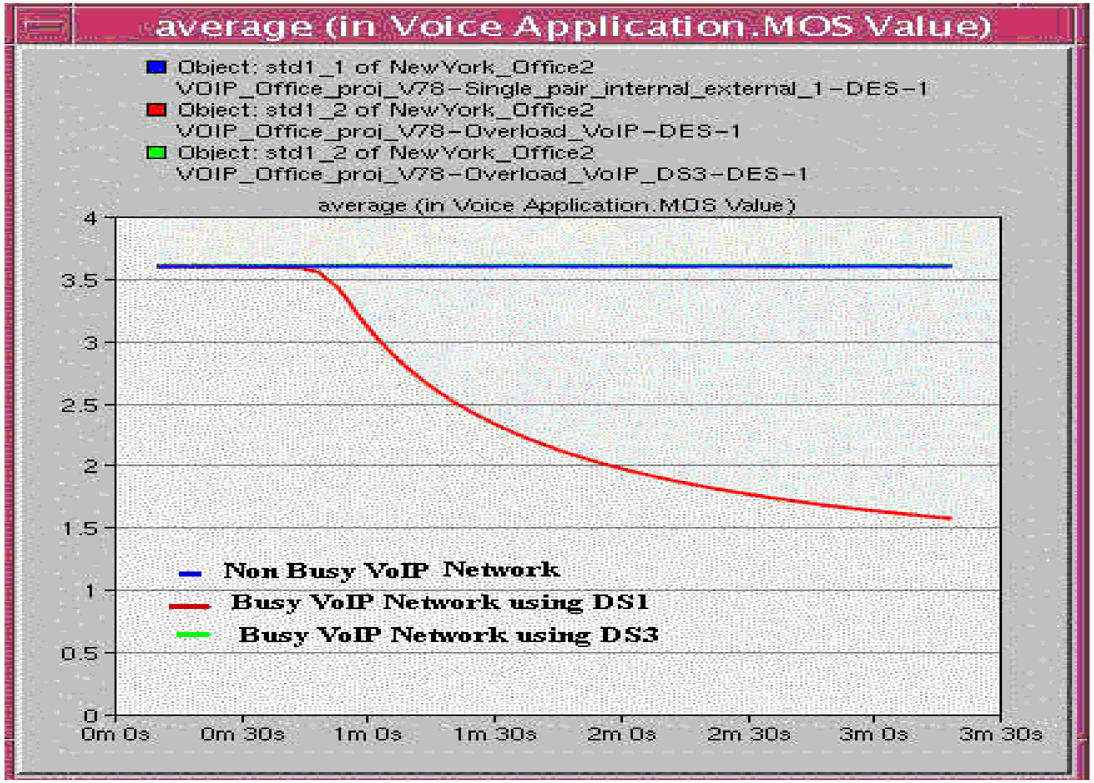


Result: E2E Delay and Jitter





Result: MOS Value



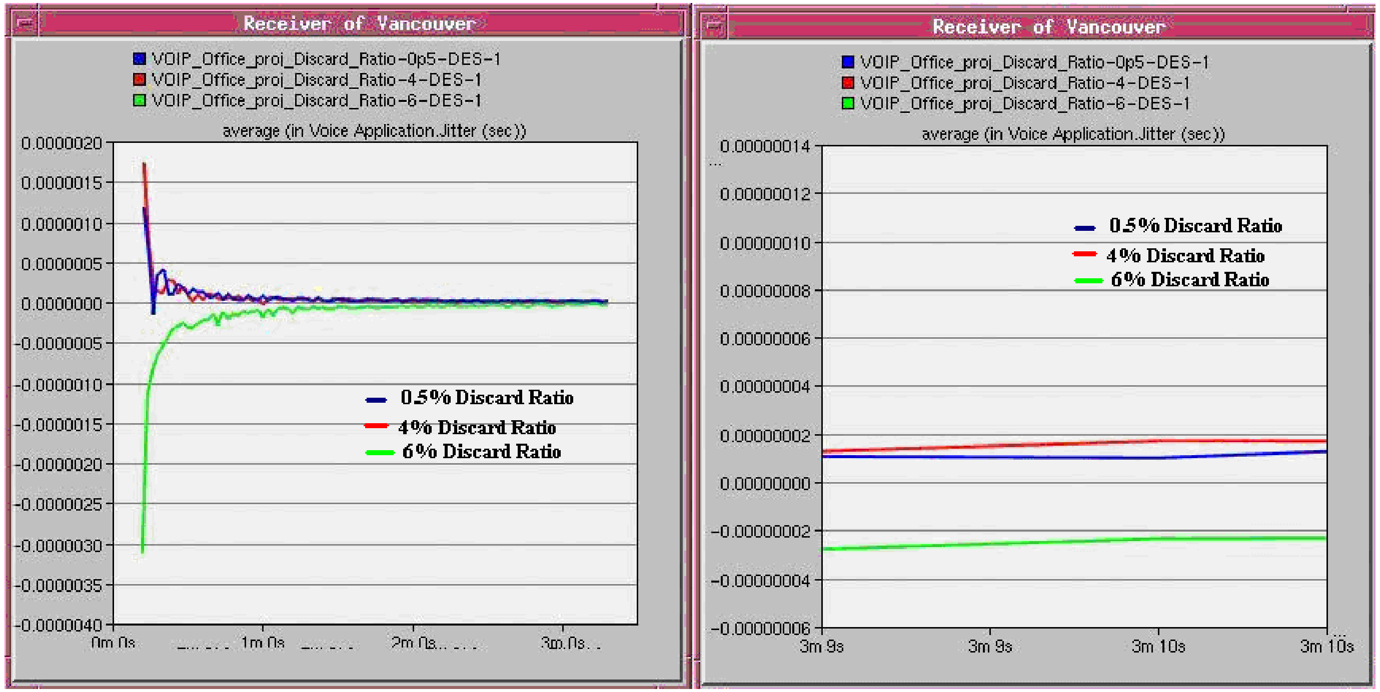
Scenario 3: Discard Ratio

- Adjusting the discard ratio in the IP cloud and observe the VoIP parameters
  - Definition: Specifies the percentage of packets dropped (ratio of packets dropped to the total packets submitted to this cloud multiplied by 100.)

  - Packet loss should never exceed 1%
  - 1% packet loss rate translates into one voice clip or skip every three minutes
  - 0.25% will translate into one error every 53 minutes

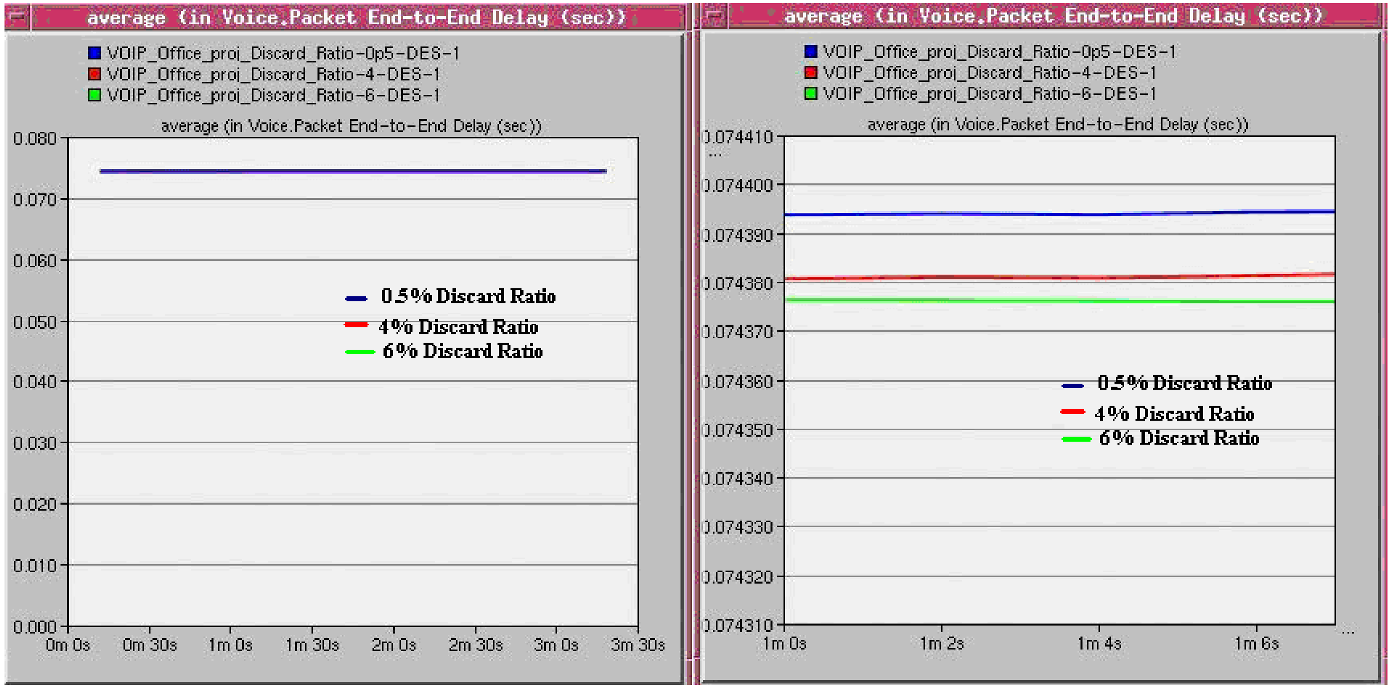
  - Set three packet discard ratios: 0.5% , 4% and 6%

Result: Jitter



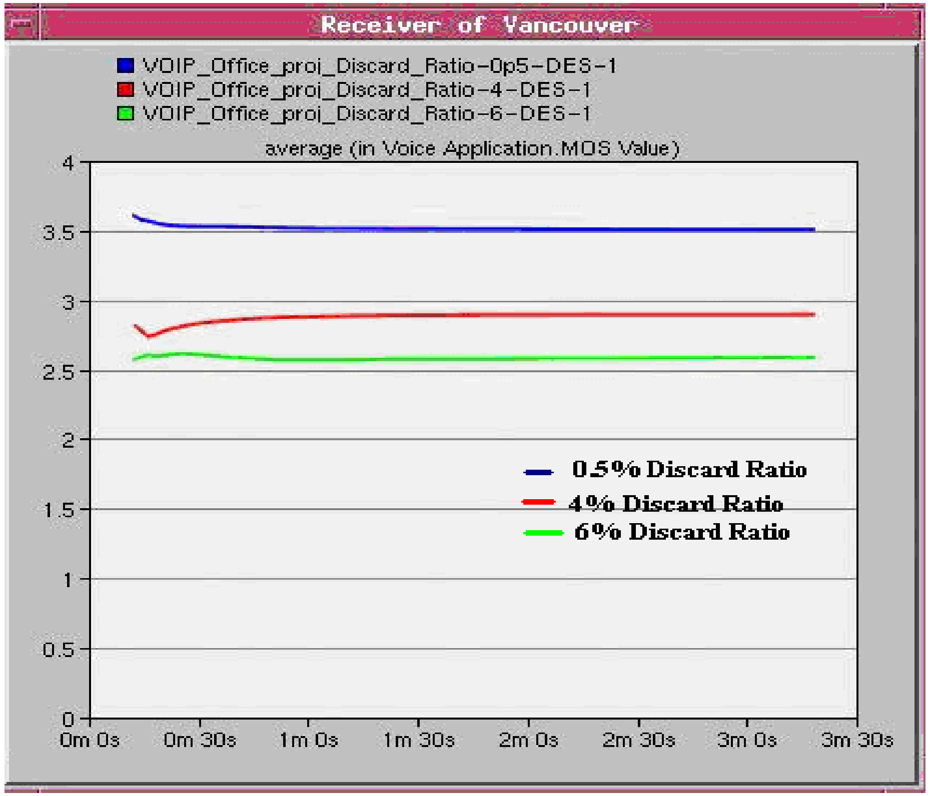
Discard Ratio Comparison--Voice Application Jitter (sec). Left: Original; Right: Zoom in

Result: End-to-End Delay



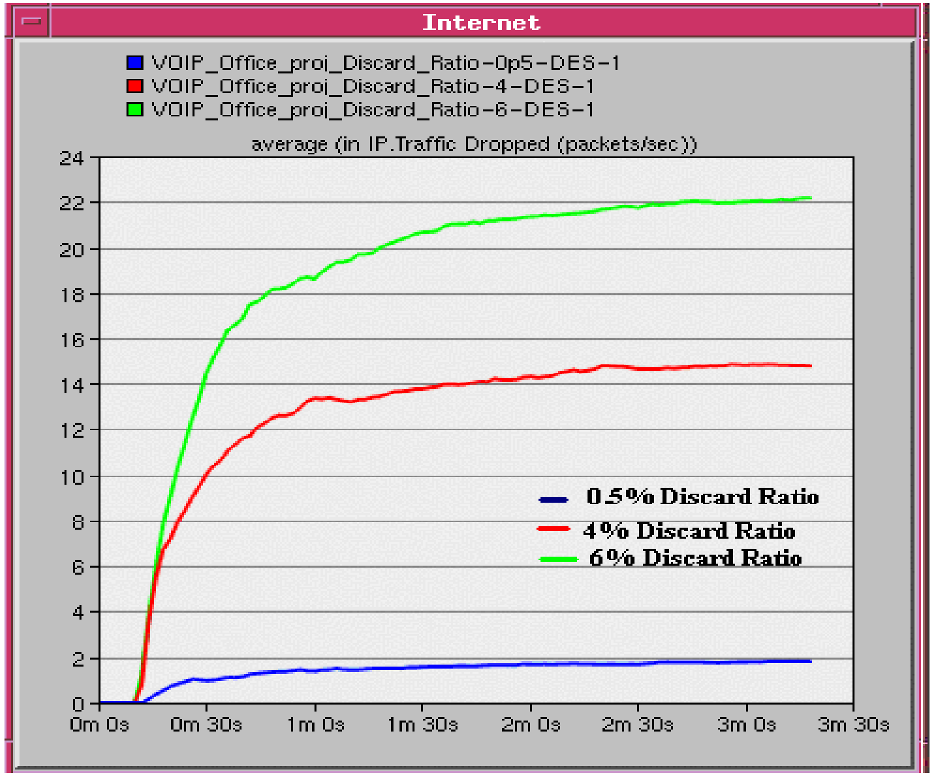
Discard Ratio Comparison--Voice Packet End-to-End Delay (sec). Left: Original; Right: Zoom in

Result: MOS



Discard Ratio Comparison--Voice Application MOS Value

Result: Traffic Dropped



Discard Ratio Comparison--IP Traffic Dropped (packets per sec)

Scenario 4: Encoder Scheme

- Uses different encoder schemes:

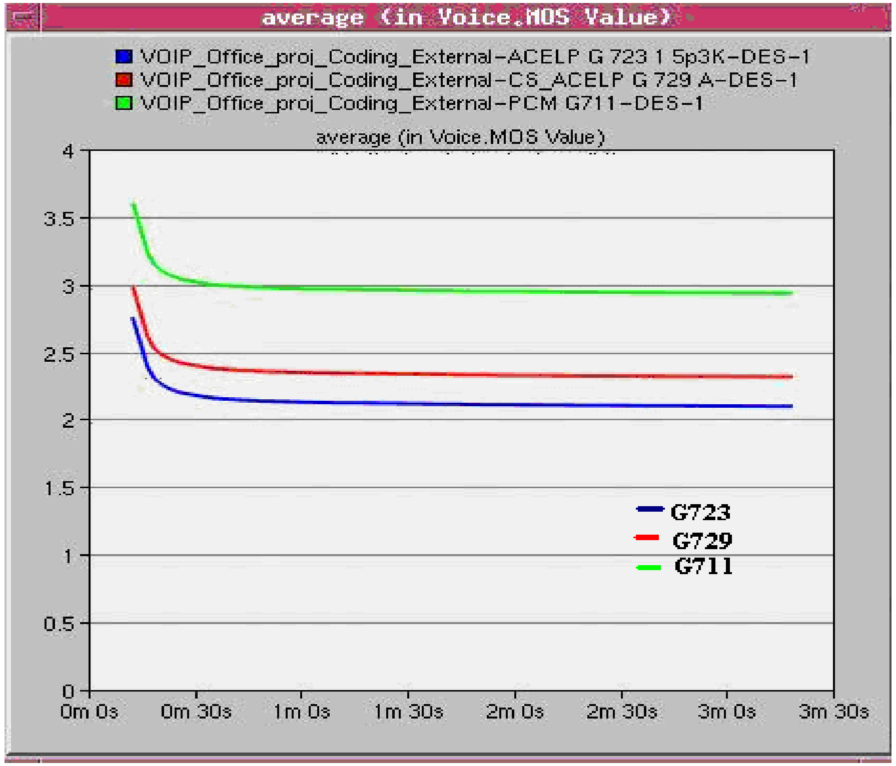
  ACELP – G723

  CS-ACELP – G729 A

  PCM(Pulse Code Modulation) – G711

- To compare VoIP application performance by measuring different parameters

Result: MOS Value



Encoder scheme comparison—average (in Voice.MOS Value)

Reference

- [1] Packetizer Inc., “How Does VoIP Work?” Available: http://www.packetizer.com/ipmc/papers/understanding_VoIP/how_VoIP_works.html, Jan. 2009 [Mar. 1, 2009]
- [2] TSeyva Pte Ltd., “Advantage Disadvantage of VoIP.” Available: http://support.tseyva.com/support/index.php?_m=knowledgebase&_a=viewarticle&kbarticleid=1,Aug.02, 2007 [Feb. 25, 2009]
- [3] P. Curry, J. Hagedorn, J. Hermanowicz and M. Sparks, Synchronized Voice Broadcast Over Congested IP Networks. Available: http://slappy.cs.uiuc.edu/fall07/Rockwell-Collins-VoIP/docs/requirements.pdf
- [4] 2004 Conference Proceedings. Available: http://www.csun.edu/cod/conf/2004/proceedings/265.htm
- [5] VoIP news. Available: http://www.voip-news.com/feature/top-residential-voip-tools-022508/
- [6] OPNET. Available: http://www.opnet.com/

Question?