VoIP Simulation Network

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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

- Skype
- Gizmo
- Ego
- Yahoo! Messenger
Description of Project

Company A Located in Vancouver

Company B located in New York
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

![Graph showing Packet Sent Rate and Received Rate over time. The graph includes two lines: one for Traffic Sent and one for Traffic Received, both showing an increase over time.]
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)


Question?