PERFORMANCE ANALYSIS OF VOIP OVER LTE NETWORK

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OVERVIEW

- Background
- Introduction
- LTE Topology
- LTE Module
- Simulation Scenarios
- Result Analysis
- Discussion
- Conclusion
- References
BACKGROUND

- **LTE (Long-Term Evolution)**
  - Deliver voice over dedicated fixed bandwidth channels to user equipment
  - Provides higher capacity, data rates
  - Reduces latency

- **VoIP (Voice over Internet Protocol)**
  - Provide voice communication that access to the internet
  - Compress digital voice into packet

- **NS-2 (Network Simulator)**
  - Series of discrete event network simulators
  - Create an open simulation environment for networking research
INTRODUCTION

• Simulation
  • Simulate VoIP with UDP agent and CBR traffic
    \[ \text{$cbreg($i) set packetSize_480}$]  
    \[ \text{$cbreg($i) set interval_0.03}$]

• Performance on a single user scenario and multiple user scenario

• Analyse 4 aspects
  ○ Delay
  ○ Jitter
  ○ Throughput
  ○ Packet Loss
LTE Topology
LTE Module

- **Simplex-link**
  - Connection between UE and EBT, AGW and EBT
    - Uplink & Downlink have different frequency bandwidth

```bash
$ns simplex-link $UE($i) $EBT(0) 200Mb 2ms LTEQueue/ULAIRQueue
$ns simplex-link $EBT(0) $UE($i) 500Mb 2ms LTEQueue/DLAIRQueue
$ns simplex-link $EBT($i) $AGW($i) 5Gb 10ms LTEQueue/ULS1Queue
$ns simplex-link $AGW($i) $EBT($i) 5Gb 10ms LTEQueue/DLS1Queue
```

- **Duplex-link**
  - Connection between AGW and Servers

```bash
$ns duplex-link $AGW(0) $server0 10Gb 50ms DropTail
```
SIMULATION SCENARIOS

- One-to-One
  - Local (UE0 & UE2)
  - Long Distance (UE3 & UE5)

- Group Chat
  - Mix of Local and Long Distance Users
    - UE1, UE4, UE6, UE7
One-to-One Scenario

- Local call between UE(0) and UE(2)
- Long distance call between UE(3) and UE(5)
GROUP-CHAT SCENARIO

- Group chat between UE(1), UE(4), UE(6) & UE(7)
  - Mix between long distance and local users
OVERLAP

- Simulation time overlaps where one-to-one user ends and group chat begins
  - One-to-one: 0s-60s
  - Group: 20s-600s

Performance
- Will an overlap show any performance issues?
## Parameters

### QoS Standards for VoIP Quality Performance

<table>
<thead>
<tr>
<th>Network Parameter</th>
<th>Good</th>
<th>Acceptable</th>
<th>Poor</th>
</tr>
</thead>
<tbody>
<tr>
<td>End-to-End Delay (ms)</td>
<td>0-150</td>
<td>150-300</td>
<td>&gt;300</td>
</tr>
<tr>
<td>Jitter (ms)</td>
<td>0-20</td>
<td>20-50</td>
<td>&gt;50</td>
</tr>
<tr>
<td>Packet Loss</td>
<td>0-0.5%</td>
<td>0.5%-1.5%</td>
<td>&gt;1.5%</td>
</tr>
<tr>
<td>Throughput (Mbps)</td>
<td>0-50</td>
<td>50-144</td>
<td>&gt;144</td>
</tr>
</tbody>
</table>

*VoIP Quality Parameter Measures [9]*
**THROUGHPUT**

- One-to-One call: 60Mbs average
- Group chat: 190mbs average
- Traffic increases 3 times (expected due to increase of traffic lines)
- No performance change in overlap
**Packet Loss**

- One-to-One call: very low packet loss, almost 0
- Group chat: increases with time. Expected since increase of traffic
**Delay**

- Delay is stable for both scenarios, almost 0
- Spikes due to contention window adjustment at the beginning of each scenario
DELAY (CLOSE UP)
**Jitter**

- Similar behavior as delay since jitter is based on delay \( Jitter = (T4 - T3) - (T2 - T1) \)
- Stable for both one-to-one and group chat
- Spikes due to contention window adjustment
DISCUSSION

- Difficulties
  - Riverbed Modeler
  - Ns2 LTE module set up

- Future Work
  - Mobile nodes
  - Multicast
  - STCP and DCCP instead of UDP
CONCLUSION

- Jitter and Delay falls in good range of QoS standards, except for spikes from the beginning of each scenario due to contention windows.
- Packet loss fell into the poor range of QoS standards.

Further modification need to be done to make VoIP a reliable voice solution for LTE (VoLTE, an evolved VoIP had been published from 3GPP)
Questions?
REFERENCE


