Performance Analysis of VoIP over LTE Network

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OVERVIEW

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- LTE Topology
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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

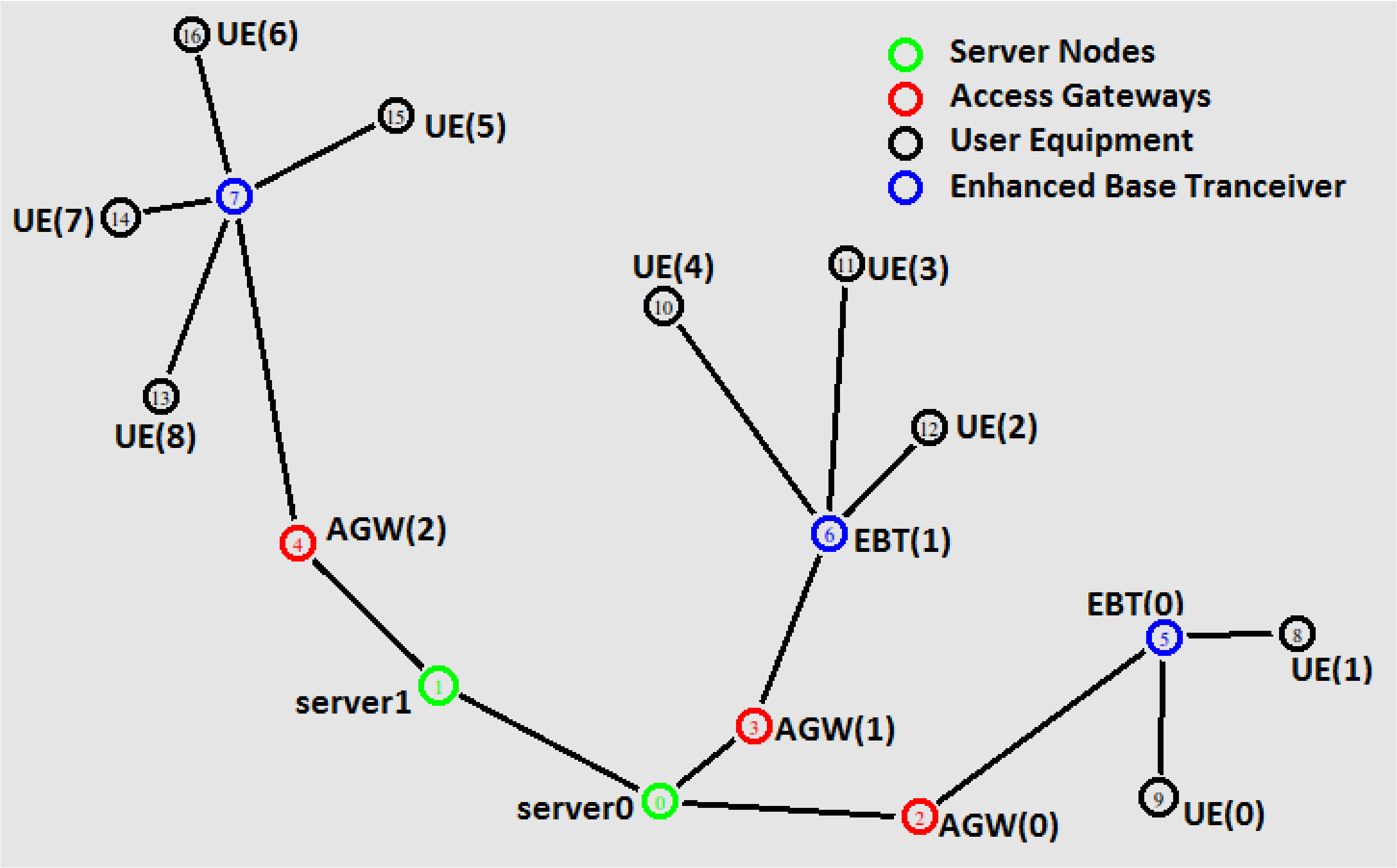
INTRODUCTION

- Simulation
  - Simulate VoIP with UDP agent and CBR traffic

    ```
    $cbrg($i) set packetSize_ 480
    $cbrg($i) set interval_ 0.03
    ```

  - Performance on a single user scenario and multiple user scenario
  - Analyse4 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

```
$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

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

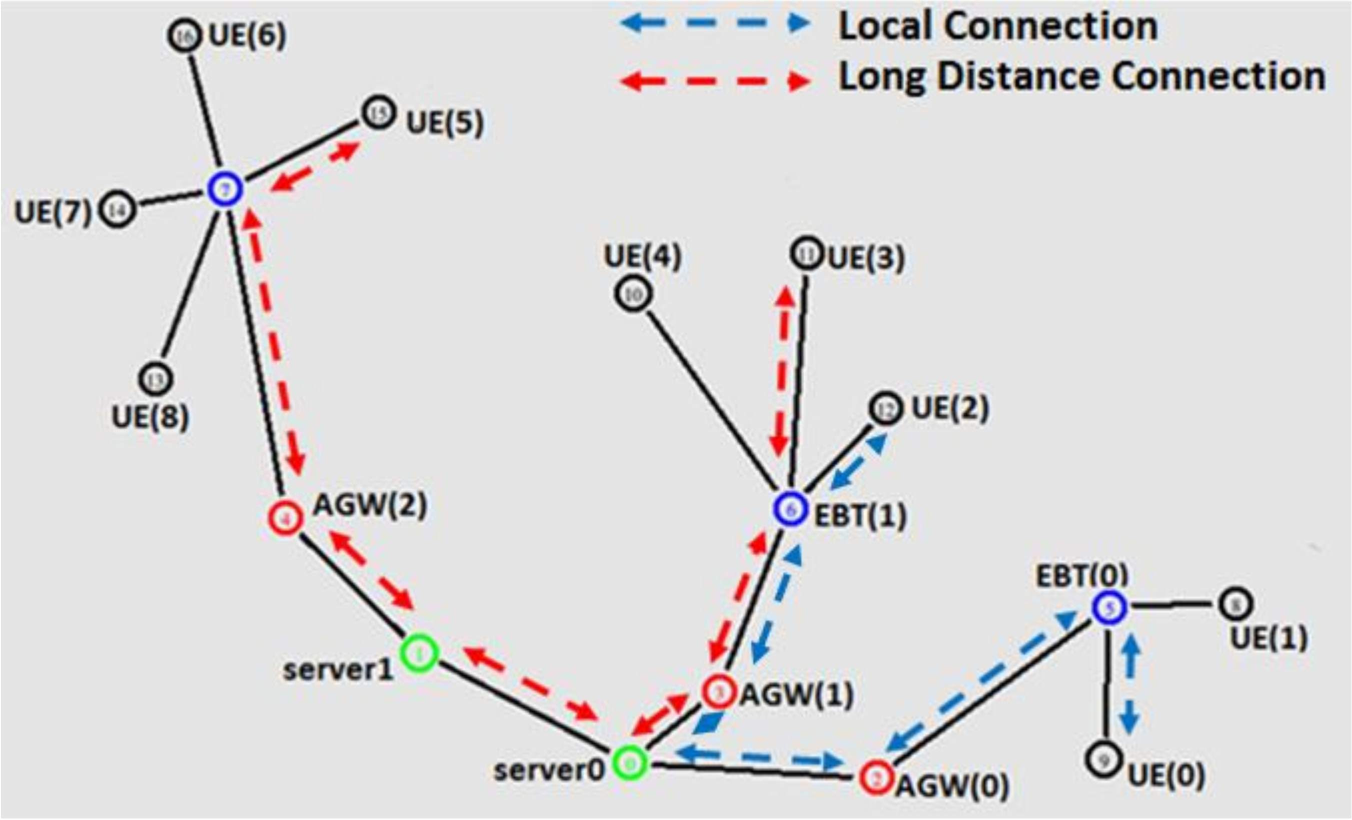
Simulation Scenarios

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

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

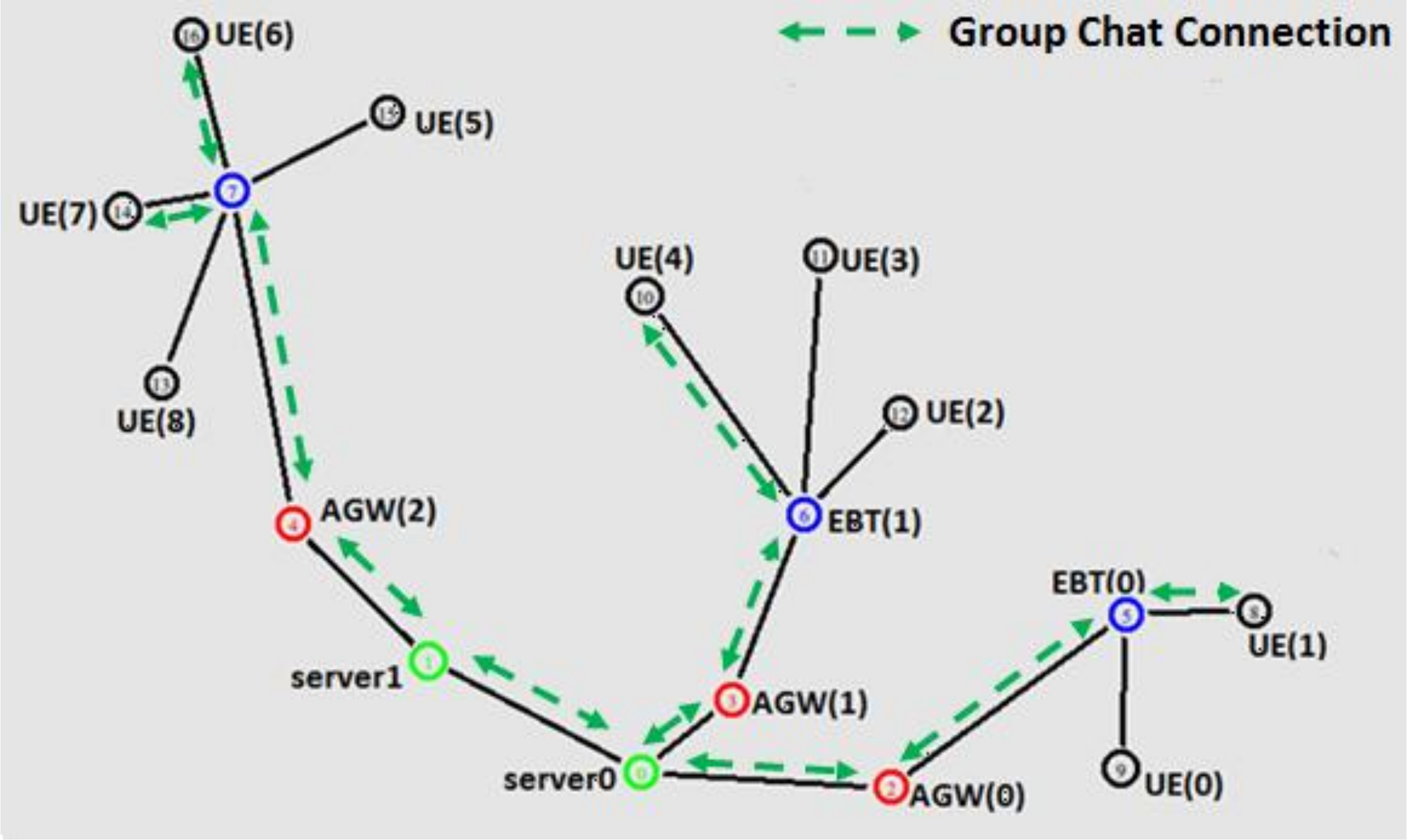
One-to-One Scenario

- 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

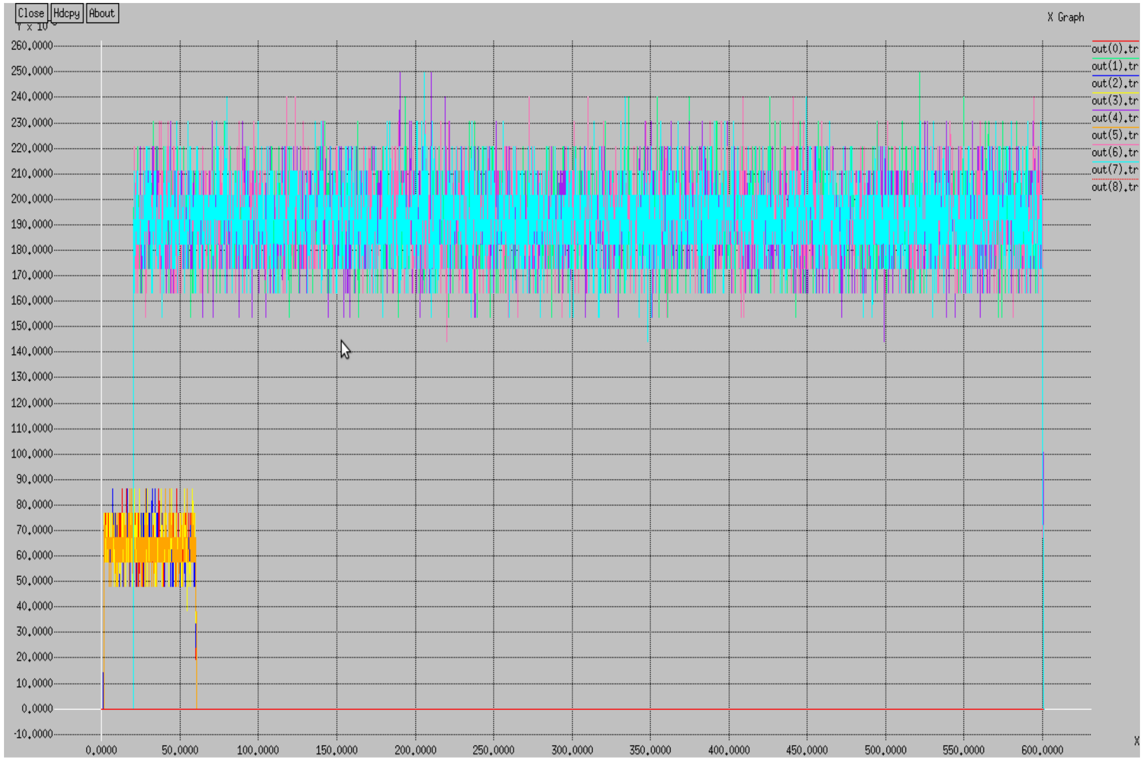
- QoS Standards for VoIP Quality Performance

Network Parameter	Good	Acceptable	Poor
End-to-End Delay (ms)	0-150	150-300	>300
Jitter (ms)	0-20	20-50	>50
Packet Loss	0-0.5%	0.5%-1.5%	>1.5%
Throughput (Mbps)	0-50	50-144	>144

VoIP Quality Parameter Measures [9]

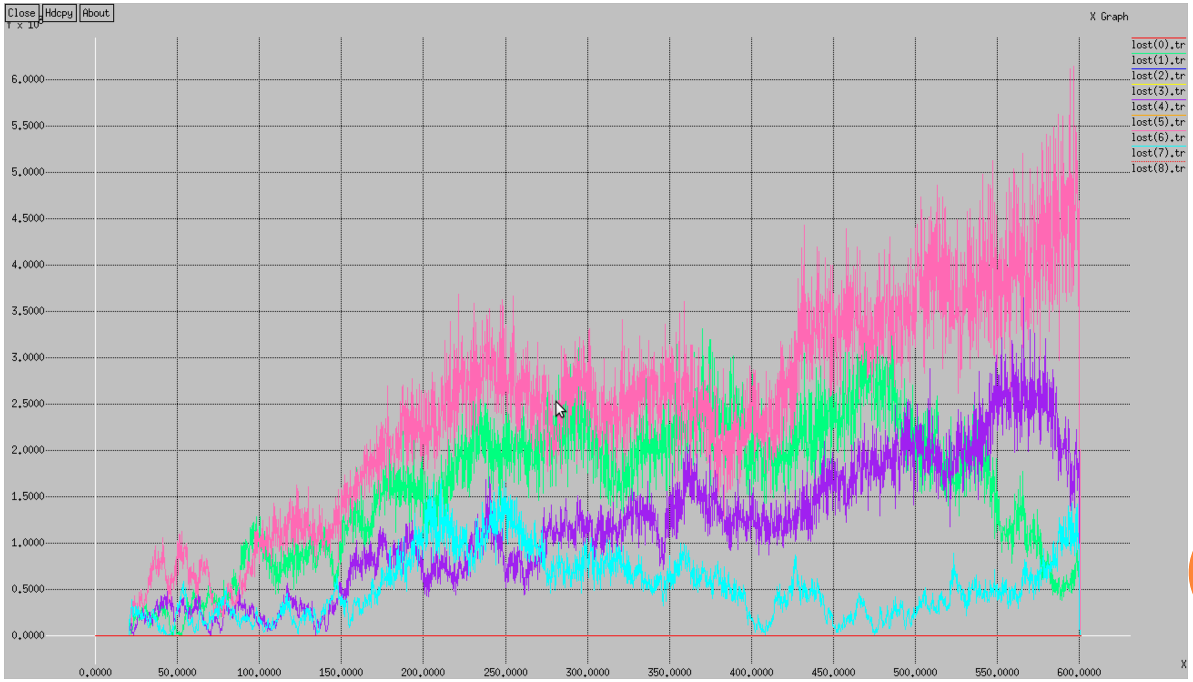
THROUGHPUT

- 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



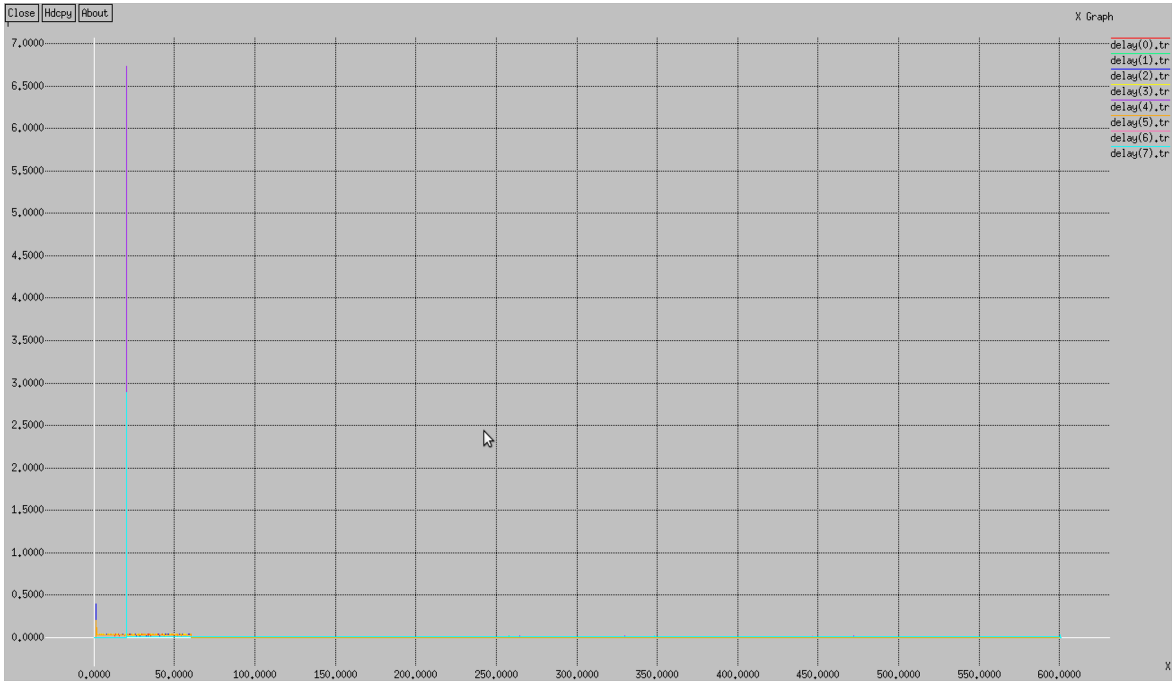
Packet loss

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

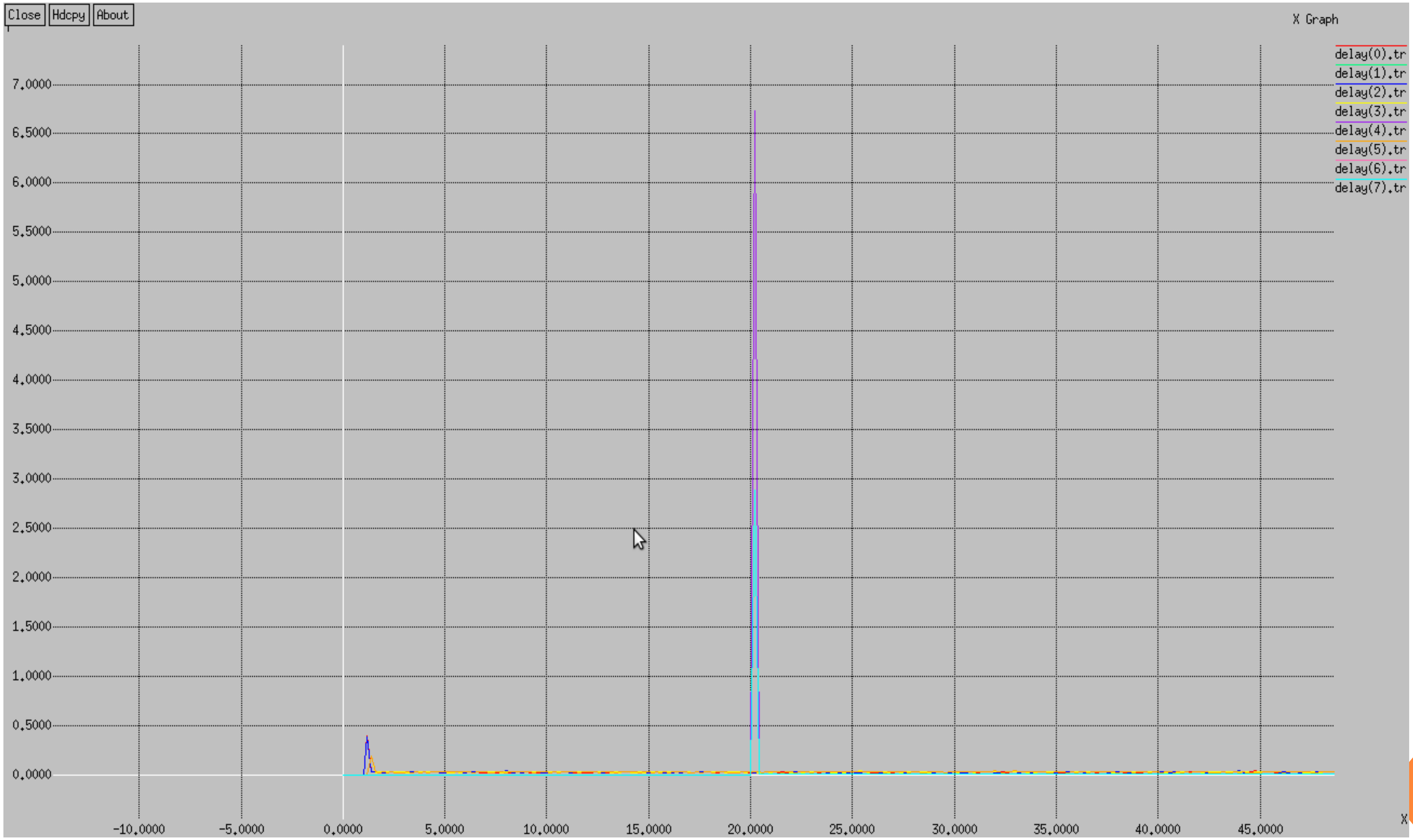


DELAY

- Delay is stable for both scenarios, almost 0
- Spikes due to contention window adjustment at the beginning of each scenario

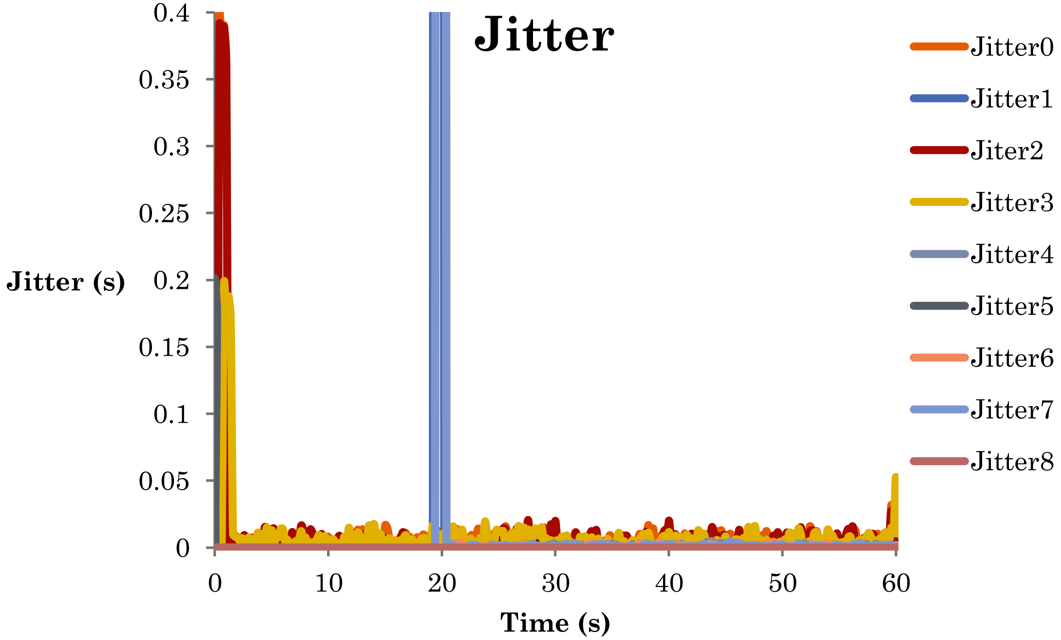


DELAY (CLOSE UP)



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.

	Good	Acceptable	Poor	LEGEND
Packet Loss			☑☑	☑: One-to-One
Delay	☑☑			☑: Group
Jitter	☑☑			

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

QUESTIONS?

Reference

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