MINT 709

CAPSTONE PROJECT

VoLTE SIMULATION

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1. INTRODUCTION

VoLTE is a standardized system for transferring voice traffic over the LTE network. LTE is designed as an all IP-network for supporting various types of packet data services. LTE provides seamless connectivity of user (user equipment) and Packet Data Network. LTE has the potential to achieve higher bandwidth, better Quality of Service and reduced latency. As the term VoLTE describes that it is the mechanism of transferring voice traffic over LTE Packet Data Network, and also enables end users to experience best performance, reliability and interoperability. LTE uses IP Multimedia Subsystem for advanced applications such as Push to Talk (PTT), content services, gaming, video traffic, voice services, and applications functions.

1.1 PROJECT SCOPE AND PROBLEM DESCRIPTION

Besides the fact that LTE provides higher bandwidth, reduced latency, seamless connectivity, and support for various kinds of packet data services; achieving best QoS for voice services in an LTE network is still challenging. In our project, we will be simulating VoLTE calls with one other data service (video streaming in our case) to determine the effect on VoLTE calls even when other types of traffic is present in the network. We will test our design in both congestion and non-congestion scenarios. Later, we will find out the behavior of VoLTE with respect to different kinds of QoS parameters and determine whether or not our design meet VoLTE specifications for optimal QoS.

2. IP MULTIMEDIA SUBSYSTEM AND VoLTE SERVICE DESIGN

IP Multimedia subsystem is a framework for delivering multimedia services over wireless networks. IMS has a major role in VoLTE design and deployment. There are various components which make up the IMS network ready for use. Before discussing the details of IMS components, let's first have a look at LTE network and its components.

2.1 THE LTE NETWORK

2.1.1 The User Equipment (UE)

User equipment in an LTE network is always a SIP (Session Initiation Protocol) User Agent which is used for sending SIP messages out to the LTE network. The UE contains a Universal Integrated Circuit Card (UICC) which has many different applications like IPMPI (IP Multimedia Private Identity) which is used as a global identity for the device and is assigned by the operator*, and* IPMPU (IP Multimedia Public Identity) for communicating with other users.

2.1.2 The Mobility Management Entity (MME)

The LTE network can have multiple MME, they are selected for operation based on geographical location of the UE. MME is used for tracking UE locations, authentication of UE into the LTE network, setting up radio access bearers between eNodeB and the UE, security, and attaching/detaching the UE to the EPC (Evolved Packet Core).

2.1.3 Serving Gateway (S-GW)

The key function of a serving gateway is to act as an anchor which can be either local or mobile. S-GW creates a connection to the different eNodeB(s) in the network, so whenever a mobile moves from one eNodeB to a different eNodeB the S-GW is responsible for maintaining the connection, and that is the reason we call it as a local or mobile anchor. The S-GW can be any device like a router that receives packets from one device and forwards to another (usually from eNodeB to P-GW and vice versa).

2.1.4 Packet Data Network Gateway (P-GW)

P-GW is the gateway to the packet data network. P-GW connects up to many different external networks like internet, IMS or any other service networks. The P-GW has an important role of assigning IP addresses to the UEs and acts as an IP anchor. P-GW is used for enforcing QoS policies on users as received by PCRF which is a separate node in an IMS network.

2.1.5 Home Subscriber Server (HSS)

HSS is a database for home subscriber. HSS is a node that sits between the LTE and the IMS network. HSS has the same role as that of an HLR (Home Location Register) in a UMTS and GSM. It has all the information related to user profiles, authentication, service authorization, subscriber's location and IP information.

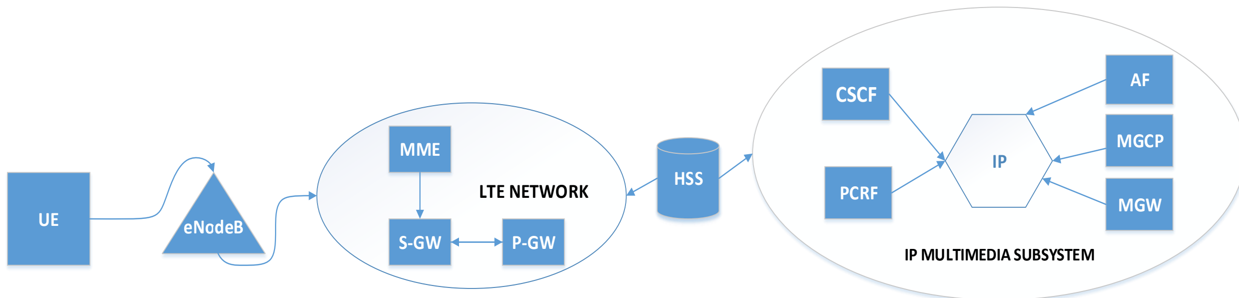


Figure 1: The LTE Network

2.2 IP MULTIMEDIA SUBSYSTEM (IMS)

IMS core can be divided in to five basic functions:

2.2.1 Call Session Control Function (CSCF)

CSCF is responsible for establishing, monitoring, supporting and releasing multimedia sessions. CSCF does all the setup processing required in establishing a call. It has a similar role that an MSC has in a circuit-switched voice call. Moreover, it directly connects to HSS for getting information about whether a user is allowed to have access to a particular service or not. The CSCF is sub divided into three separate nodes.

Proxy-CSCF (P-CSCF)

P-CSCF is the first point of contact for all UE SIP messages and also in the whole IMS network. UE sends all the SIP messages through the LTE network to the P-CSCF, and it's the P-CSCF's role to route the messages further down in the IMS network typically to S-CSCF or I-CSCF. P-CSCF can be either in home or in visited networks. It may include a Policy Control Function which is responsible for enforcing QoS policies.

Serving-CSCF (S-CSCF)

S-CSCF is typically the brain of the IMS core operation. It's the node that has subscriber profile downloaded to it. It knows about the services that a user is allowed to initiate, it's the node that has to decide how to forward the user request to the particular destination node (application servers or MGW). S-CSCF is always in home network.

Interrogating-CSCF (I-CSCF)

I-CSCF has much smaller role as compared to the S-CSCF and P-CSCF. I-CSCF is typically used in S-CSCF selection during the initial IMS registration.

2.2.2 Home Subscriber Server (HSS)

As we've already discussed above about the HSS in LTE network, the functionality of HSS in an IMS network is the same. It's totally up to the equipment manufacturer and operator to have a separate HSS for both LTE and IMS network, or to share the same.

2.2.3 Application Functions (AF)

Application functions is a distributed services node that an IMS network provide. Different servers may be responsible for providing different applications. Examples include Push to Talk (PTT), content services, gaming, video etc.

2.2.4 Media Gateway (MGW)

Media Gateway is responsible for converting VoLTE calls to PSTN networks or some legacy networks e.g. 3G, 2G etc. This is because the language of VoLTE calls is not the same as that of PSTN or legacy phone networks. The S-CSCF is responsible for handing over all non-VoLTE calls to MGW.

2.2.5 Media Gateway Control Function (MGCF)

Media Gateway Control Function is the partnering node of MGW. The role of MGCF is to control the MGW, codecs conversion, and conversion of media between RTP used in IMS and PCM used by circuit-switched network.

3. IMS IN THE CONVERGENCE OF FIXED & MOBILE IP NETWORKS

Fixed mobile convergence or FMC integrates wireless, cellular and wireline technologies and its services to create a single network that enable telecom service providers to provide all multimedia and voice services to its clients whether being wireless, wired, or cellular. FMC allows users/clients with seamless connectivity to access different multimedia and voice services regardless of their location and what device they are using.

IP based multimedia subsystem (IMS) defines a highly reliable and flawless architecture for supporting multimedia and voice services for both fixed and mobile network architectures. IMS is not only standardized for VoIP services but also a platform that can support all types of multimedia services quiet efficiently. The most common types of IMS multimedia services include text, images, instant messages, conferencing, presence, and video telephony etc. IMS provides SIP and IP based solution to handle multimedia signaling over the wireless domain. As IMS is based on SIP signaling therefore it requires SIP user agents and SIP servers for the whole communication. Different application servers are used for different types of communication in IMS based network. Examples are SIP AS, IM-SSF, OSA SCS etc. These application servers are basically open service modes of the IMS that provide the capability of application service to securely use network resources.

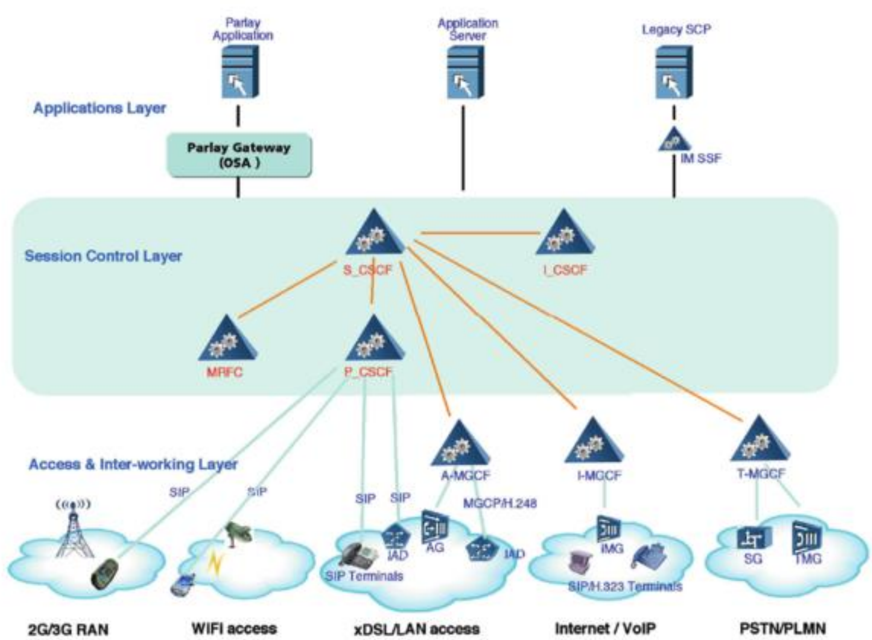
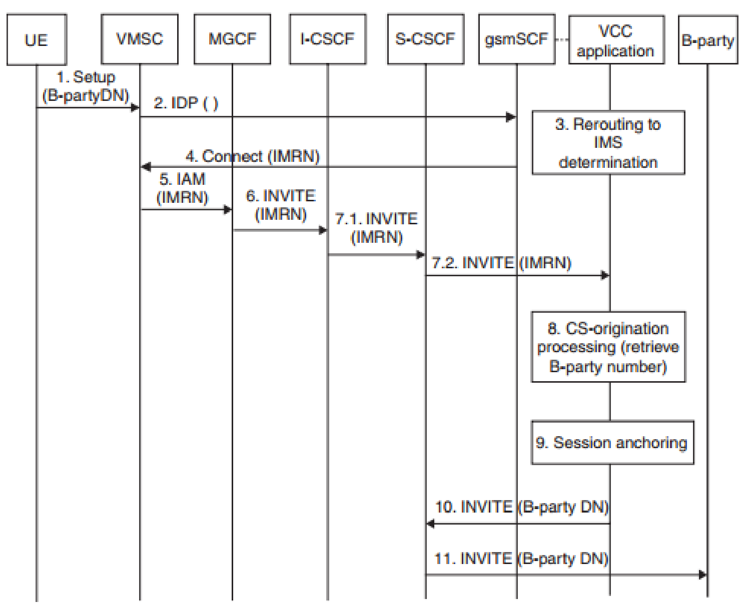


Figure 2: IMS Architecture

There are many advantages of adopting SIP as the control layer for IMS. As SIP is an open standard that supports multiple media formats, provides session control in a way that service provisioning and session control are both separated; provides dynamic registration, location management and redirection mechanisms. SIP also features presence, fork subscription, facilitating the deployment of new services, featured a great potential for extension.
IMS has the ability to offer the complete service delivery environment to IP endpoints. IMS can entertain both IP-based clients as well as circuit-switched client(s) via Media Gateways. Media Gateways are responsible for translating all IP-based calls to legacy/PSTN networks. Media Gateways have to perform extensive functions like codecs conversions, conversion of media between RTP used in IMS and PCM used by circuit-switched networks. The convergence that is taking place should be transparent to the end users in the properly implemented Fixed Mobile Convergence solution. Switching between packet-switched to a circuit-switched network should be seamless in a well-implemented FMC solution.

For seamless Fixed Mobile Convergence, 3GPP defined Voice Continuity Call (VCC) standard. VCC assumes that IMS is the central core for both VoIP and circuit-switched calls. Therefore to achieve this, the VCC-application server is deployed in the IMS network. To utilize the use of VCC approach, the subscriber handsets should be multimode handsets to support both VoIP and circuit-switched calls.

Figure 3: A simple VCC call Setup



4. SESSION BORDER CONTROLLERS

A session border controller (SBC) is a dedicated hardware or software element that control real-time session traffic at the signaling, call-control, and packet layers as they cross a notional packet-to-packet network border between networks or between network segments. SBCs are highly used for Quality of Service, IP to IP Gateways, protocol interworking such as H.323 to SIP, H.323 to H.323, SIP to SIP. An SBC can also serve as a firewall for session traffic, applying its own quality of service (QoS) rules and identifying specific incoming threats to the communications environment.

IMS complaint architecture have CSCF (Call Session Control Function) which acts as Session Border Controllers. CSCF is responsible for establishing, monitoring, supporting and releasing multimedia sessions. CSCF does all the setup processing required in establishing a call. It has a similar role that an MSC has in a circuit-switched voice call. CSCF is the first point of contact for all UE SIP messages and also in the whole IMS network. UE sends all the SIP messages through the LTE network to the P-CSCF, and it’s the P-CSCF’s role to route the messages further down in the IMS network.

4.1 VIRTUALIZED SESSION BORDER CONTROLLER

Session Border Controller provides traffic control functions, security, call control overload protection, interworking and media handling. SBC is a translator that speaks different kinds of media languages so networks can share voice video and data together with different networks and devices. Session Border Controller is usually a hardware device that manages all the above activities in a Service Provider Network.

With the evolution of cloud computing and virtualization, we have seen a major trend in the industry about virtualizing servers, networks, storage and other networking applications. Nowadays SBC’s are also available in software or virtualized editions. Many major vendors have developed or, are in the process of developing software based Session Border Controllers.

4.2 COMMERCIAL SESSION BORDER CONTROLLER AVAILABLE IN SOFTWARE VERSIONS

There are various commercial virtualized SBCs available which will discussed in this section. Vendors like Oracle, Acme Packet and Sonus have launched their commercial software based Session Border Controllers. Let’s have a look at Oracle’s Feature offering about SBCs:

Oracle Enterprise Session Border Controller Features	
Feature	**Capabilities**
Security	• Granular access control • IP address and SIP signaling concealment • Layer three through five topology hiding and signaling overload controls • IP telephony spam protection • Stateful deep packet inspection • Signaling and media encryption
Reliability	• Stateful signaling and media failover • Quality of service (QoS) marking, virtual local area network (VLAN) mapping, access control • Registration storm avoidance • Call rate limit enforcement • Trunk load balancing • Stateful session routing • QoS-based routing • Stateful signaling and media failover
Cost management	• Least cost routing • Codec renegotiation
Regulatory compliance	• Media forking • Signaling and media encryption • Session prioritization for emergency services • Internet Engineering Task Force (IETF) standard SIP Recording (SIPREC) interface • Federal Information Processing Standard (FIPS) 140-2 certified • Call detail records (CDRs) with local or remote storage via RADIUS
Interoperability	• SIP message normalization • Response code translation • Session Description Protocol (SDP) and Dual Tone Multi-Frequency (DTMF) manipulation • Number and uniform resource identifier (URI) manipulation • Signaling message header manipulation • Signaling interworking (SIP, H.323) • Protocol interworking: Transmission Control Protocol (TCP), User Datagram Protocol (UDP), Stream Control Transmission Protocol (SCTP) • Encryption interworking: Transport Layer Security (TLS), Mutual TLS, Secure Real-time Transport Protocol (SRTP), IP Security (IPsec) • Network address translation (NAT) and firewall traversal • IP address translation: private/public, IPv4/IPv6 • Transcoding • Avaya Personal Profile Manager proxy

Figure 4: Oracle Enterprise Session Border Controllers

Below is the comparison of Oracle's various Session Border Controllers and their specifications:

Feature	Oracle Enterprise Session Border Controller 3820[2]	Oracle Enterprise Session Border Controller 4500[3]	Oracle Enterprise Session Border Controller 6300[4]	Oracle Enterprise Session Border Controller, Virtual Machine Edition[5]
Chassis	1U, rack mount	1U, rack mount	3U, rack mount	Host platform dependent
Architecture	Dedicated, purpose built	Dedicated, purpose built	Dedicated, purpose built	Software based
Session capacity[6]	Up to 8,000 simultaneous sessions	Up to 32,000 simultaneous sessions	Up to 80,000 simultaneous sessions	Up to 250 simultaneous sessions
Memory	2 GB for active configuration and logs 256 MB internal flash memory for runtime image and backup configurations	4 GB for active configuration and logs 256 MB internal flash memory for runtime image and backup configurations	600 GB disk for CDRs, log files, and other storage 16 GB memory for configuration files and OS storage	Host platform dependent
High availability	Active/standby systems (1:1 redundancy) with checkpointing of signaling, media, and configuration state for no loss of service	Active/standby systems (1:1 redundancy) with checkpointing of signaling, media, and configuration state for no loss of service	Active/standby systems (1:1 redundancy) with checkpointing of signaling, media, and configuration state for no loss of service	Active/standby systems (1:1 redundancy) with checkpointing of signaling, media, and configuration state for no loss of service
Network interfaces	Four 10/100/1000 Mbps Ethernet ports (fiber or copper)			Host platform dependent
Two-level encryption acceleration hardware	Session setup: IPsec tunnel and TLS sessions Traffic encryption/decryption: IPsec and SRTP			Not applicable
Management interfaces	SNMP agent, XML configuration files, Syslog, SFTP			

Figure 5: Oracle Enterprise Session Border with specifications

As we can observe from the above chart of specifications between different SBCs offered by Oracle, the major concern in virtualized version is the session capacity. The maximum session capacity is 250 in the Virtual Machine Edition, which is too less for a service provider environment.

However, there are other products offered by other vendors that can provide unlimited session capacity. Sonus SBC SWe (software edition) provides from as low as 25 to an unlimited number of sessions. Service Providers have the choice to deploy it on industry standard servers in virtualized environments or as a hosted service. The features that Sonus SWe provides are as follows:

Media Services

- Transcoding G.711, G.722, G.723, G.726, G.729A/B and iLBC
- Wireline, wireless, wideband and clearchannel codec pass through
- VAD, Silence Suppression, Dynamic Jitter Buffer, DTMF/Tone Relay/ RFC2833/RFC4733 interworking
- NAT/NAPT on media
- DTMF Trigger Detection and Notification
- Generic audio codec relay
- Tones & Announcements
- Local Ring Back Tone (LRBT) support with centralized PSX Policy Server
- RTP inactivity monitoring
- Video codec relay

Redundancy

- 1:1 Redundant Instances for Service High before Availability

Management Capabilities

- Graphical based wizards for ease of configuration
- Secure embedded web-based management GUI
- Sonus CLI, SSH
- Centralized support by Sonus Insight EMS
- SNMP V2 status and statistics
- Local logging of events, alarms, and traps; Call trace
- Sonus DSI Level 0 support for storing CDRs; RADIUS accounting records
- Live Software Update (LSWU)

Signaling

- Back to Back User Agent (B2BUA)
- SIP, SIP-I/SIP-T, SIP/H.323
- SIP protocol normalization/ protocol repair; SIP message manipulation
- NAT/NAPT on signaling

Protocol Support

- IPV4, IPV6, IPV4/IPV6 interworking
- SSH; sFTP
- SNMP; NETCONF; NTP
- HTTP/HTTPS
- RTP/RTCP
- UDP, TCP
- DNS, ENUM

Routing/Policy

- Embedded policy/routing engine
- Centralized support by Sonus PSX Policy/Route Server using DIAMETER+
- Screening, blocking, routing, presentation, call type filters
- Route prioritization
- Leading digit routing; international routing; URI based routing
- Digit/parameter manipulation
- E911 support; Priority Call handling

Security

- Session aware firewall; Topology Hiding
- Line rate DoS/ DDoS, and Rogue RTP protection
- Line rate malformed packet protection
- TLS, IPSec (IKEV1) for signaling encryption
- Secure RTP/RTCP for media encryption

Quality of Service (QoS)

- Bandwidth management
- Call admission control (CAC) per trunk group, per zone
- Per call statistics
- TOS/ COS packet marking

Packet Network Time Source

- Network Time Protocol (NTP) per RFC-1708

Minimum Requirements

- VMWare ESXi 5.1 or higher
- 1 virtual CPU; 2GHz or higher
- 5GB of RAM
- 4 virtual NICs (1 - MGMT, 1 HA and 2 packet ports)
- 65GB hard disk size

Figure 6: Sonus Software Edition SBC features

5. TOOLS REQUIRED FOR SIMULATION

5.1 COMPONENTS AND MODULES

The Omnet++ modules which we used in the VoLTE simulation are INET2.2 and SIMULTE.

5.1.1 INET

INET is an open-source communications networks simulation module for OMNET++. The INET Framework contains models for several wired and wireless networking protocols, including UDP, TCP, SCTP, IP, IPv6, Ethernet, PPP, 802.11, MPLS, OSPF, and many others.

5.1.2 SIMULTE

SIMULTE is a LTE data plane simulation module. SIMULTE supports eNodeBs, RLC, MAC Layer, and UEs used in LTE networks. SIMULTE does not support LTE control plane.

5.2 INET COMPONENTS USED IN THE SIMULATION

Below are all the INET components used in the VoLTE simulation:

5.2.1 IPv4 Network Configurator

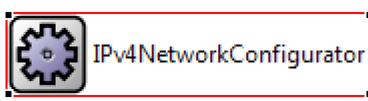


Figure 7: IPv4 Network Configurator

The IPv4 network configurator used in the simulation assigns IP addresses to the UEs, routers, S-gateway, P-Gateway and eNodeB. The configurator supports both manual and automatic address assignment and their combinations. This module performs subnet-based routing and sets up static routing for an IPv4 network.

5.2.2 Channel Control

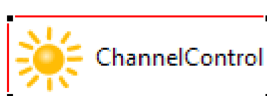


Figure 8: Channel Control

Channel Control used in the simulation tracks location and movement of the nodes (UEs) and determines which nodes are within the communication or interference distance. Channel control has one instance in every network that has mobile or wireless nodes.

5.2.3 Routing Table Recorder

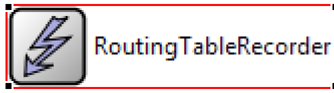


Figure 9: Routing Table Recorder

The routing table recorder used in the simulation records changes in the routing tables and interface tables of all hosts and routers. The filename has to be specified in the routing log-file configuration option that this module registers.

5.2.4 IPv4 Router

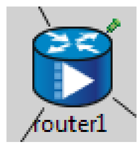


Figure 10: IPv4 Router

IPv4 router supports wireless, Ethernet, PPP and external interfaces. There are three IPv4 routers used in the VoLTE simulation.

5.2.5 Voip Server

Figure 11: Voip Server

The voip server used in the simulation is basically a Standard Host module which supports TCP, UDP and SCTP protocols. The Voip functionality implemented in this Standard Host is implemented through a VoIPReceiver udpApp located in lte.apps.voip in the SIMULTE module.

5.2.6 Video Stream Server

Figure 12: Video Stream Server

The video stream server used in the simulation is basically a Standard Host module which supports TCP, UDP and SCTP protocols. The video streaming functionality in this Standard Host is implemented through an udpVideoStreamSvr located in inet.applications.udpapp in the INET module.

5.2.7 Thruput Metering Channel

Figure 13: Thruput Metering Channel

The ThruputMeteringChannels are used for links (connections) between routers, S-Gateways, P-Gateways, and eNodeBs. The ThruputMeteringChannel is extended with Datarate channel for calculation of throughput and data rate of nodes in the simulation.

5.3 SIMULTE COMPONENTS USED IN THE SIMULATION

Below are all the SIMULTE components used in the VoLTE simulation

5.3.1 eNodeB

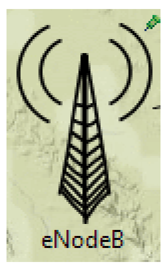


Figure 14: eNodeB

The eNodeB used in the simulation connects mobile phones (UEs) and hosts. ENodeB supports Physical, Medium Access Control (MAC), Radio Link Control (RLC), Radio Resource Control and Packet Data Control Protocol (PDCP) layers. It performs extensive traffic flow filtering based on network, uplink/downlink, and port ranges. ENodeB supports both GTP-U and Evolved Packet Core in SIMULTE.

5.3.2 LTE Binder

Figure 15: LteBinder

LTE Binder is used in the simulation for binding UEs to the eNodeBs. It stores a table of entries which has each node's ip addresses and their corresponding node IDs. Every node that sends packets to other nodes uses LTE binder to get the ID of the destination node.

5.3.3 UE

Figure 16: UE

User Equipment (UE) interfaces directly with eNodeB in the simulation. UE in SIMULTE supports all TCP and UDP applications and has mobility parameters also. The UEs can also support handovers, but handovers are not actually implemented up in eNodeB and also there is no application of Mobility Management Entity in the SIMULTE module.

5.3.4 Serving Gateway

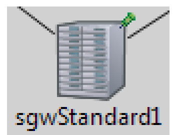


Figure 17: Serving Gateway

Serving Gateway (S-GW) in our simulation acts as the Evolved Packet Core Serving Gateway. The S-GW forwards and routes packet to and from the eNodeB and Packet Data Network Gateway (P-GW).

5.3.5 Packet Data Network Gateway

Figure 18: P-Gateway

Packet Data Network Gateway (P-GW) in our simulation acts as the interface between the LTE network and other packet data networks. Moreover P-GW performs anchoring for UE mobility, manages QoS and performs deep packet inspection.

6. SIMULATION SETUP AND PROCEDURE

In the VoLTE simulation, all the data plane elements of an LTE network are simulated including Serving Gateway, Packet Data Network Gateway, eNodeBs and UEs. There are 70 UEs in the VoLTE network with linear mobility, 50 UEs are making voice calls and 20 UEs are streaming videos from the video server. For voice calls, we have used G.711 codec whereas for video streams we have used the native VideoStreamServer from INET module which works on CBR. All the calls and video streams are performed at the same time during the simulation to determine the voice calls behavior in the LTE network even when the UEs are streaming videos.

Voice Traffic

Each voice packet is 229 Bytes and sampling rate is 20ms. The breakdown of the voice packet is below:

160 Bytes	*Payload*
20 Bytes	*IP Header*
8 Bytes	*UDP Header*
12 Bytes	*RTP Header*
29 Bytes	*LTE Information*

1(UE) x 50(packets/sec) x 229 (packet size) = 11,450 Bytes/sec (For 1 UE)
50(UEs) x 50(packets/sec) x 229 (packet size) = 572,500 Bytes/sec
50(UEs) x 50(packets/sec) x 229 (packet size) x 8 = 4,580,000 bits/sec = 4580 Kilobits /sec

Video Traffic

Each video packet is 1400 Bytes and sampling is 10ms. The breakdown of the video packet is below:

1360 Bytes	*Payload*
20 Bytes	*IP Header*
8 Bytes	*UDP Header*
12 Bytes	*RTP Header*

1(UE) x 100(packets/sec) x 1400 (packet size) = 140,000 Bytes/sec (For 1 UE)
20(UEs) x 100(packets/sec) x 1400 (packet size) = 2,800,000 Bytes/sec (For 20 UEs)
20(UEs) x 100(packets/sec) x 1400 (packet size) = 22,400,000 bits/sec = 22400 Kilobits/sec

Below are the number of devices which are used in the simulation:
IPv4 Routers: 3
Serving Gateway: 1
Packet Data Network Gateway: 1
Video Server: 1
Voice Server: 1
ENodeB: 2
Video Stream UEs: 20
Voice Calls UEs: 50

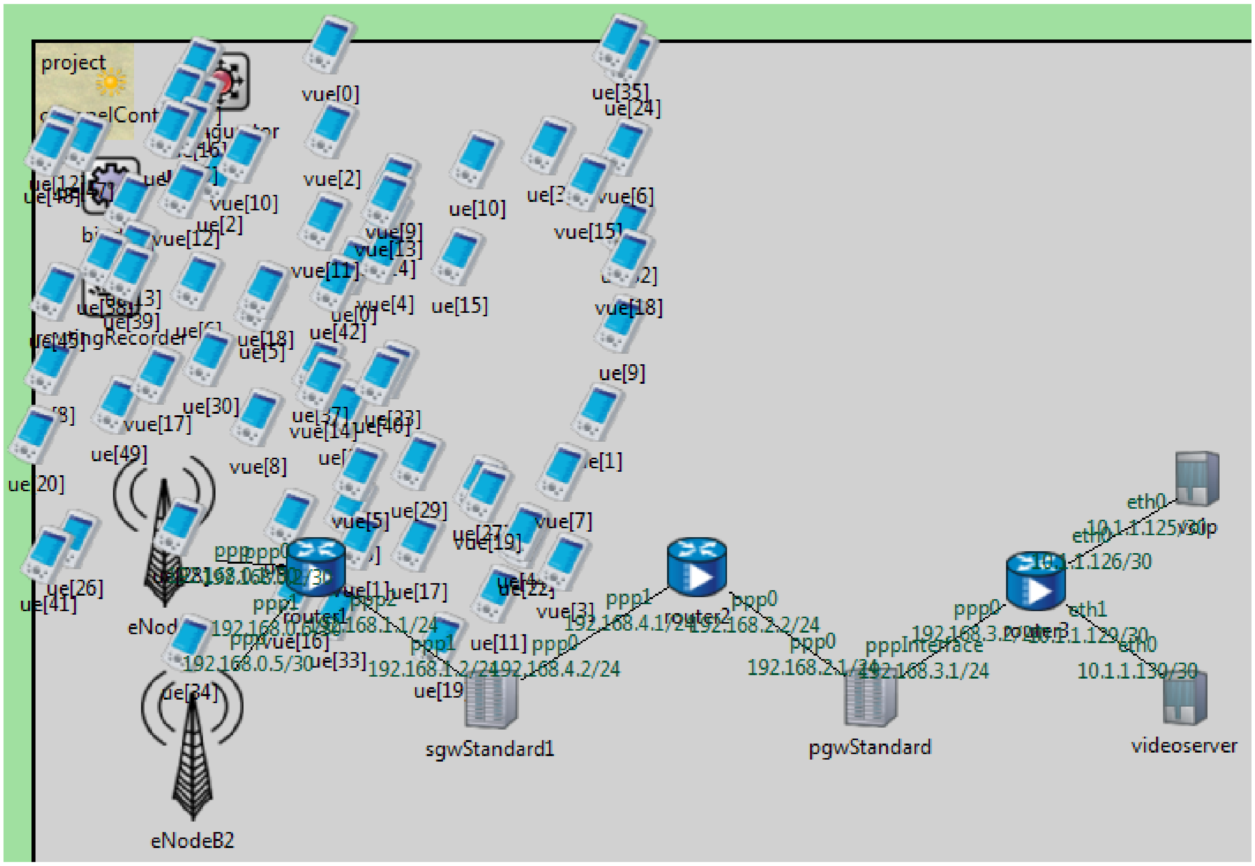


Figure 19: The Simulation Setup

6.1 NETWORK DESCRIPTION

The VoLTE simulation network is defined in Network Description (NED) file of the simulation directory.

The modules imported in our simulation from other modules (INET and SIMULTE) are below:

```
import inet.networklayer.autorouting.ipv4.IPv4NetworkConfigurator;
import inet.networklayer.ipv4.RoutingTableRecorder;
import inet.nodes.inet.Router;
import inet.nodes.inet.StandardHost;
import inet.util.ThruputMeteringChannel;
import inet.world.radio.ChannelControl;
import lte.corenetwork.binder.LteBinder;
import lte.corenetwork.nodes.Ue;
import lte.corenetwork.nodes.eNBpp;
import lte.epc.PgwStandard;
import lte.epc.SgwStandard;
```

Channel Information of the links in given below:

```
channel networkConnection extends ThruputMeteringChannel
        {
            @signal[channelBusy](type="int");
            @signal[messageSent](type=cMessage);
            @signal[messageDiscarded](type=cMessage);
            @statistic[busy](source=channelBusy; record=vector?;
interpolationmode=sample-hold);
            @statistic[utilization](source="timeavg(channelBusy)";
record=last?);
            @statistic[packets](source="constant1(messageSent)";
record=count?; interpolationmode=none);
            @statistic[packetBytes](source="packetBytes(messageSent)";
record=sum?; unit=B; interpolationmode=none);

@statistic[packetsDiscarded](source="constant1(messageDiscarded)";
record=count?; interpolationmode=none);

@statistic[throughput](source="sumPerDuration(packetBits(messageSent))
"; record=last?; unit=bps);
            delay = 0.1us;
            datarate = 10Mbps;
            thruputDisplayFormat = "u";}
```

Various submodules are also used in the simulation as described below:

```
    submodules:
        channelControl: ChannelControl {
```

```
            @display("p=50,25;is=s");
        }
        routingRecorder: RoutingTableRecorder {
            @display("p=50,161;i=block/boundedqueue");
        }
        configurator: IPv4NetworkConfigurator {
            @display("p=122,25;i=block/broadcast");
            config = xmldoc("lteIpAddresses.xml");
        }
        binder: LteBinder {
            @display("p=50,94");
        }
        eNodeB: eNBpp {
            @display("is=vl;p=122,230");
        }
        eNodeB2: eNBpp {
            @display("is=vl;p=103,461");
        }
        ue[numUe]: Ue {
            parameters:
                @signal[voIPJitter](type="double");
                @statistics[sample](title="voIP jitter";
source="voIPJitter"; record=all; interpolationmode=none);
                @display("is=n");
        }
        vue[videoUe]: Ue {
            parameters:
                @signal[voIPJitter](type="double");
                @statistics[sample](title="voIP jitter";
source="voIPJitter"; record=all; interpolationmode=none);
                @display("is=n");
        }
        router1: Router {
            @display("p=184,344;i=abstract/opticalrouter");
        }
        voip: StandardHost {
            @display("p=762,286;i=device/server");
        }
        pgwStandard: PgwStandard {
            nodeType = "PGW";
            @display("p=548,430;i=device/router");
        }
        router2: Router {
            @display("p=434,344;i=abstract/opticalrouter");
        }
        router3: Router {
```

```
        @display("p=657,353;i=abstract/opticalrouter");
    }
    sgwStandard1: SgwStandard {
        @display("p=299,431;i=device/router");
    }

    videoserver: StandardHost {
        @display("p=754,430;i=device/server");
    }
```

6.2 PARAMETERS

6.2.1 Channel Control Parameters

pMax – Maximum sending power = 10W

alpha – path loss coefficient = 1

carrierFrequency – carrier frequency for all channels = 2100e+6Hz

numChannels – number of radio channels = 1

6.2.2 Mac Layer Parameters

mac.queueSize = 1MiB
mac.maxBytesPerTti -- size of the data blocks passed from the higher network layers to the radio link layer = 30KiB

6.2.3 Physical Layer Parameters

nic.phy.usePropagationDelay – Transmission Delay Recording = True

6.2.4 eNodeBs

nodeType = ENODEB

txPower – Transmission Power = 100mW

fbDelay – time interval between sensing and transmitting in TTI = 4

6.2.5 Deployer

deployer.positionUpdateInterval = 0.001s
deployer.broadcastMessageInterval = 1s

numPreferredBands = 1

6.2.6 Traffic Flow Filters and GTP Information

eNodeB

```
<!-- from GTP tunnel information to GTP tunnel information -->
      <teidTable>
            <!-- Local Address -->
            <teid
                  teidIn  ="0"
                  teidOut ="-1"
                  nextHop ="192.168.1.2"
            />
      </teidTable>
      <!-- from tftIdentifier to GTP tunnel information -->
      <tftTable>
            <tft
                  tftId   ="11"
                  teidOut ="11"
                  nextHop ="192.168.1.2"
            />
      </tftTable>
```

P-Gateway

```
<!-- from GTP tunnel information to GTP tunnel information -->
      <teidTable>
            <!-- Local Address -->
            <teid
                  teidIn  ="0"
                  teidOut ="-1"
                  nextHop ="0.0.0.0"
            />
      </teidTable>


      <!-- from tftIdentifier to GTP tunnel information -->
      <tftTable>
            <tft
                  tftId   ="1"
                  teidOut ="1"
                  nextHop ="192.168.4.2"
            />

      </tftTable>
```

S-Gateway

```
<!-- from GTP tunnel information to GTP tunnel information -->
<teidTable>
     <teid
          teidIn  ="1"
          teidOut ="0"
          nextHop ="192.168.0.1"
     />
     <teid
          teidIn  ="11"
          teidOut ="0"
          nextHop ="192.168.2.1"
     />
</teidTable>
```

7. SIMULATION OBSERVATION AND RESULT ANALYSIS

The results of the VoLTE simulation are based on two different types of scenarios. One is the congested scenario and the other is the non-congested or the normal scenario. Results are based on the characteristics like VoIP Tail Drop Loss (Packet Loss), Frame Delay, and Jitter.

The results of QoS characteristics are calculated in microseconds because SIMULTE calculates the QoS characteristics in microseconds, and moreover the link delay set on each link is also in microseconds. Therefore, for making comparisons between the theoretical delay and the simulated delay, microsecond is our consideration. However, the industry standard/acceptable calculations are in milliseconds.

7.1 CONGESTED SCENARIO

In the congested scenario, different parameters like queue size, link bandwidth/capacity, maximum bytes per transmission time interval and frame capacity of each device were reduced to analyze the behavior of our design.

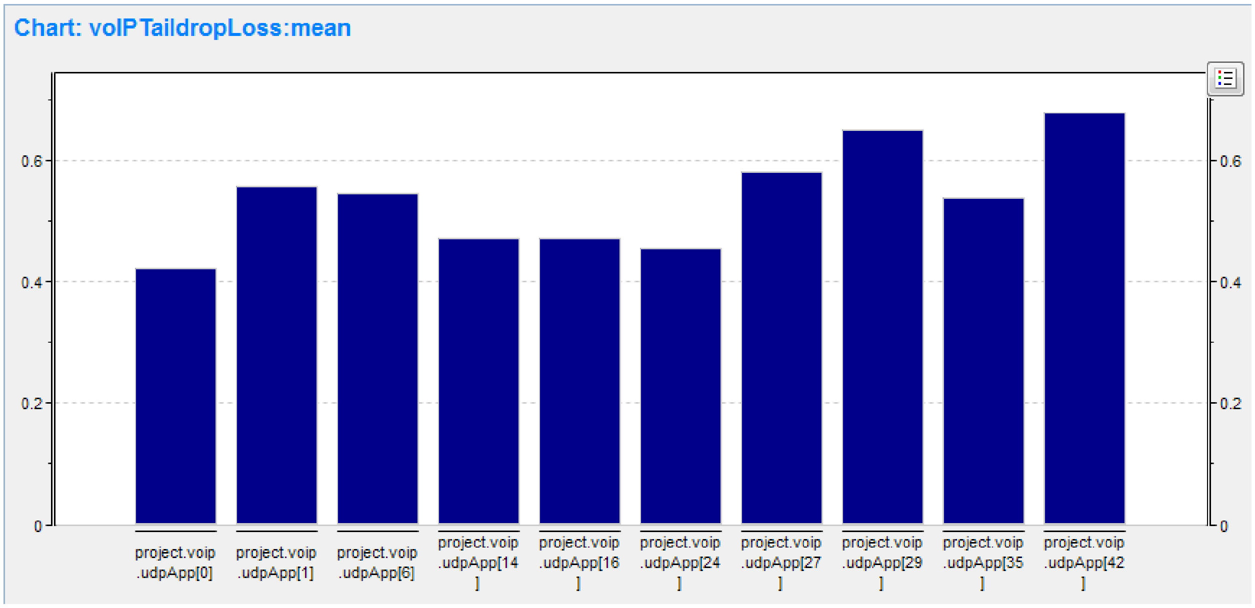
Parameters:
**.mac.queueSize = 1KiB
**.mac.maxBytesPerTti = 3KiB
datarate = 10Mbps;
**.ppp[*].queueType = "DropTailQueue"
**.ppp[*].queue.frameCapacity = 100

Delay Parameters:
Each link has a delay of 0.1µs and the total delay given is 0.6µs

Voice Bit Rate: Each voice packet is 229Bytes which is send out every 20ms by the UE and there are 50 UEs in the network, so the total bit rate for voice packets is 4580Kbps (calculation shown in *Section 6*), which utilizes 45.8% of the link.

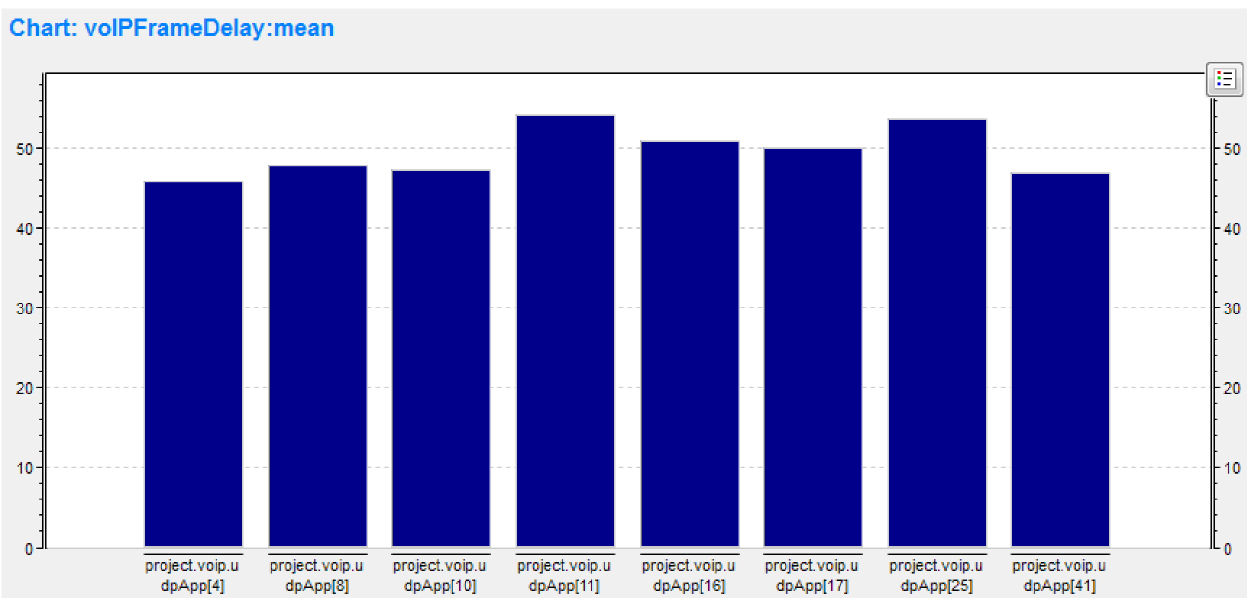
Video Bit Rate: Each video packet is 1400Bytes which is sent out every 10ms by the UE and there are 20 video UEs in the network, so the total bit rate is approximately 22400Kbps (calculation shown in *Section 6*), which utilizes 224% of the link, therefore we have named it as our congested scenario.

7.1.1 VoIP Tail Drop Loss



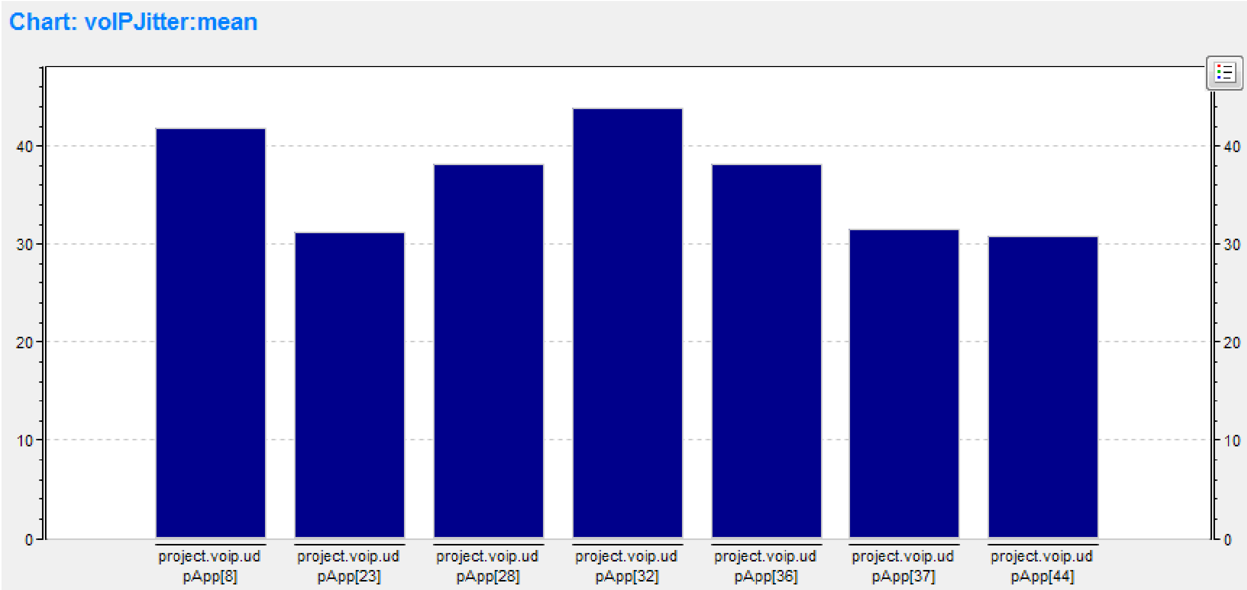
The above result shows a high number of packet losses in the different voice calls of the simulation.

7.1.2 VoIP Frame Delay



The above result shows the VoIP frame delay in the voice calls. As we can observe from the above result that the mean frame delay of the voice call is from 45µs to 54µs which differs a lot from the theoretical delay of our network i.e. 0.6µs. The delay component parameters that contribute here are *queue.frameCapacity, mac.queueSize,* and *data rate* of the channel.

7.1.3 VoIP Jitter



The above result shows high jitter values for each voice call in the simulation.

7.2 NON - CONGESTED SCENARIO

The non-congested scenario is the normal scenario in which optimal values for queue size, link bandwidth/capacity, maximum bytes per transmission time interval, and frame capacity were set in the simulation for best quality of service (QoS).

Parameters:

**.mac.queueSize = 10MiB
**.mac.maxBytesPerTti = 300KiB
datarate = 100Mbps;
**.ppp[*].queueType = "DropTailQueue"
**.ppp[*].queue.frameCapacity = 10000

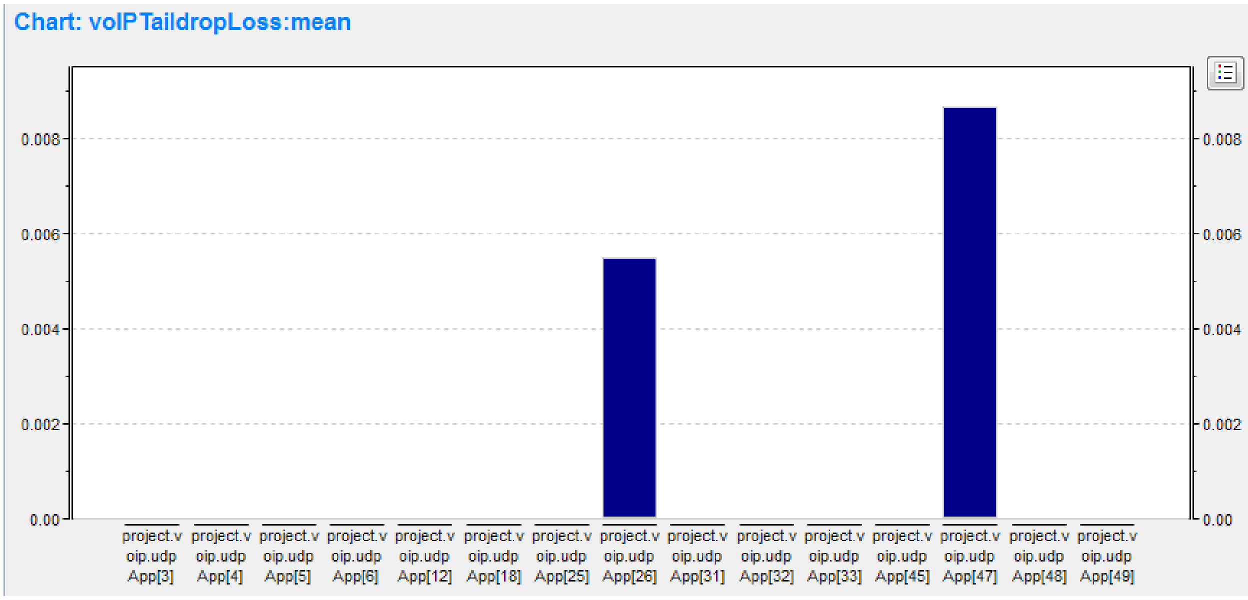
Delay Parameters:
Each link has a delay of 0.1µs and the total delay given is 0.6µs

Voice Bit Rate: Each voice packet is 229Bytes which is send out every 20ms by the UEs and there are 50 UEs in the network, so the total bit rate for voice packets is 4580Kbps (calculation shown in *Section 6*), which utilizes 4.58% of the links.

Video Bit Rate: Each video packet is 1400Bytes which is sent out every 10ms by the UE and there are 20 video UEs in the network, so the total bit rate is 22400Kbps (calculation shown in *Section 6*), which utilizes 22.4% of the link.

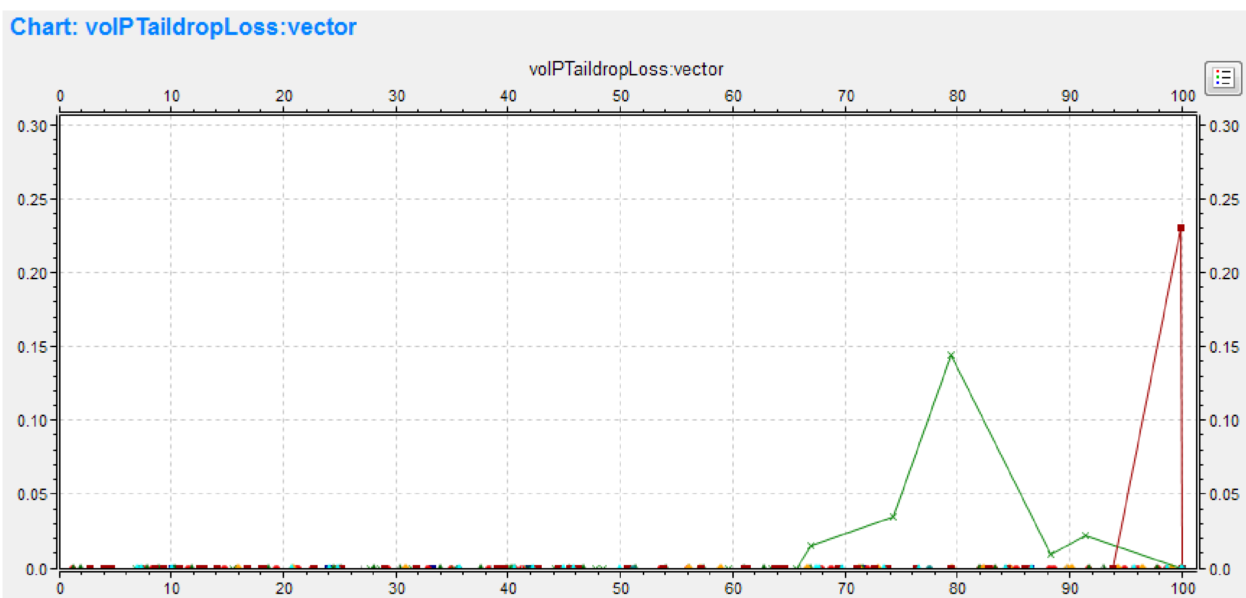
7.2.1 VoIP Tail Drop Loss

Scalar:



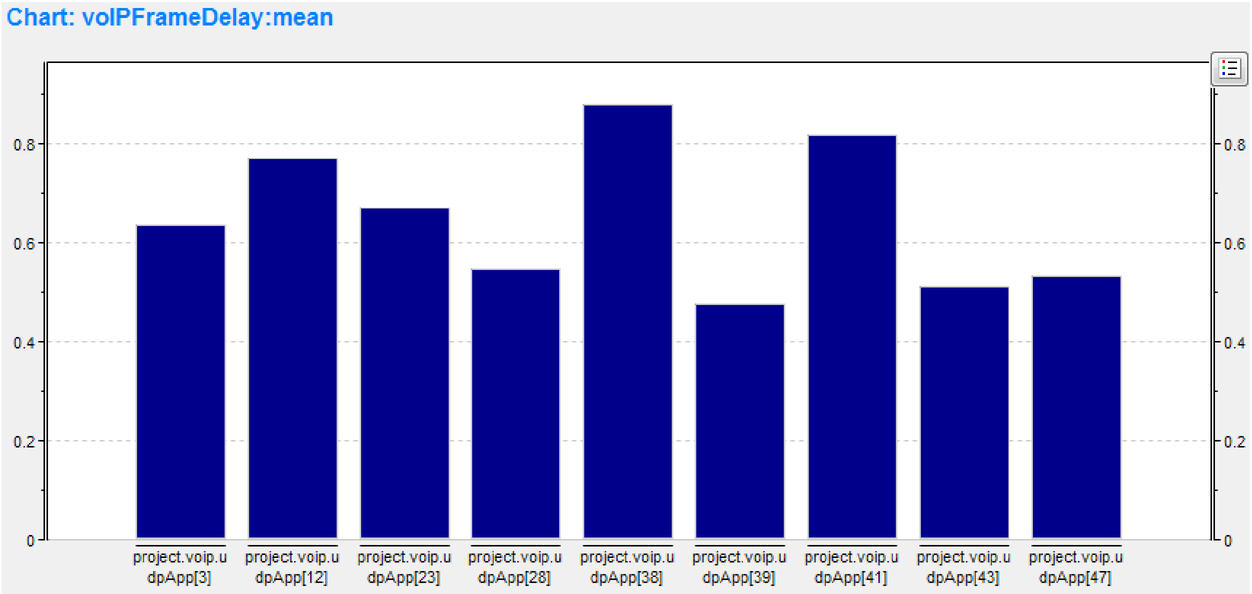
The above results shows a few packet loss in the simulation. It can be observed that the approach we took shows best quality of service (QoS) for LTE voice calls**.**

Vector Representation:



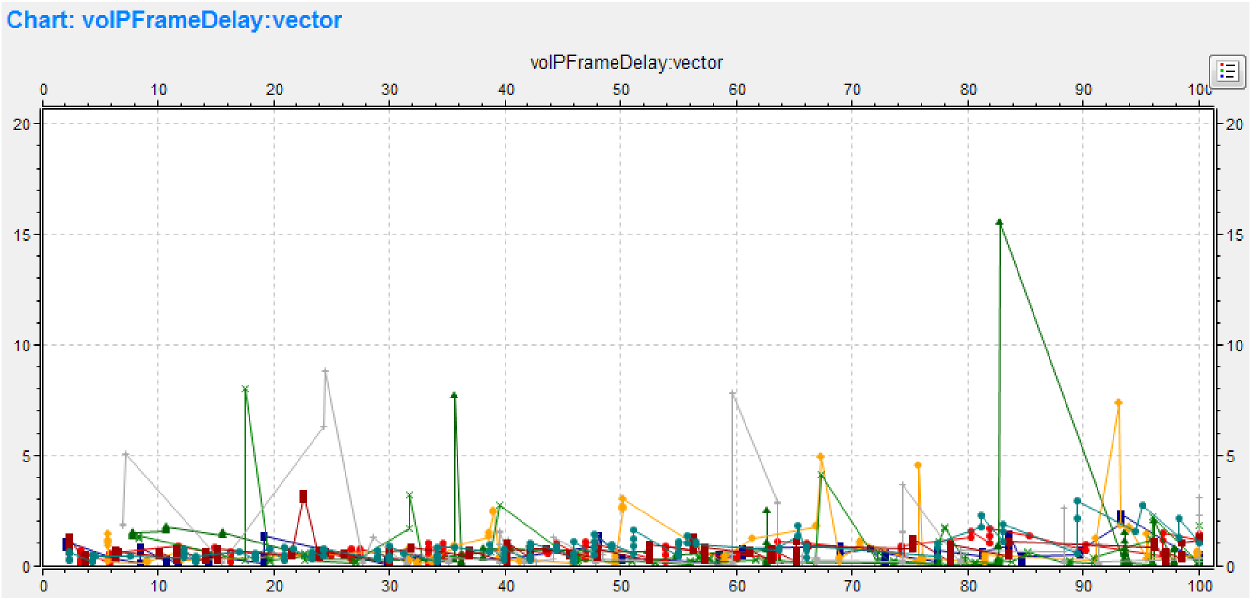
7.2.2 VoIP Frame Delay

Scalar:



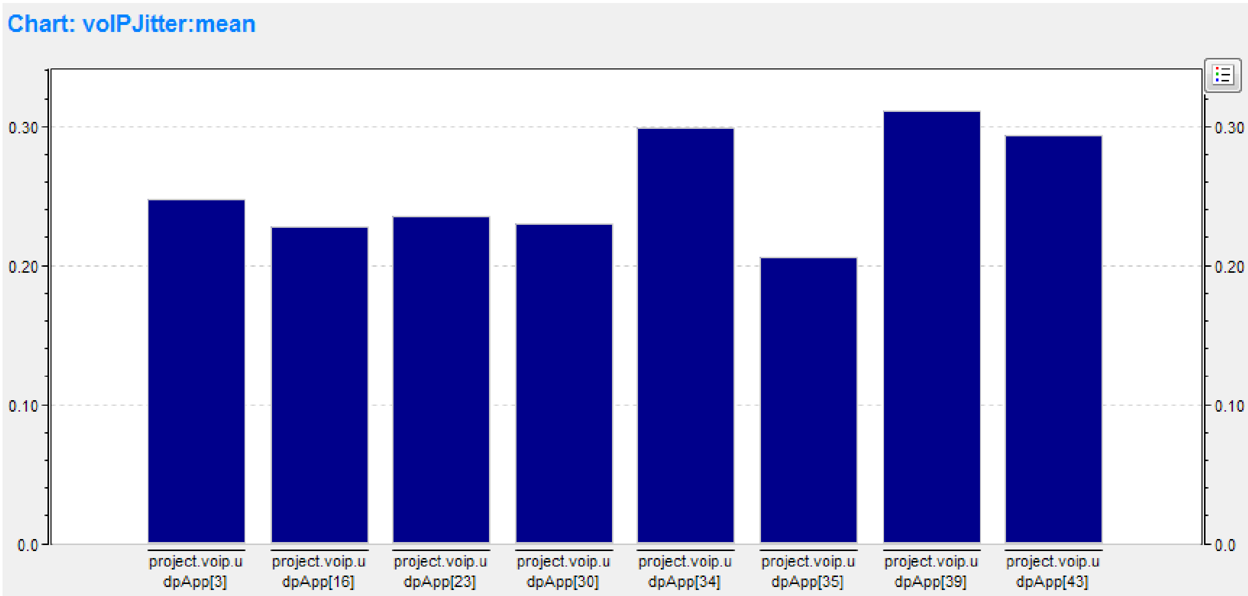
The above result shows VoIP Frame Delay for the voice calls in the simulation.

Vector Representation:



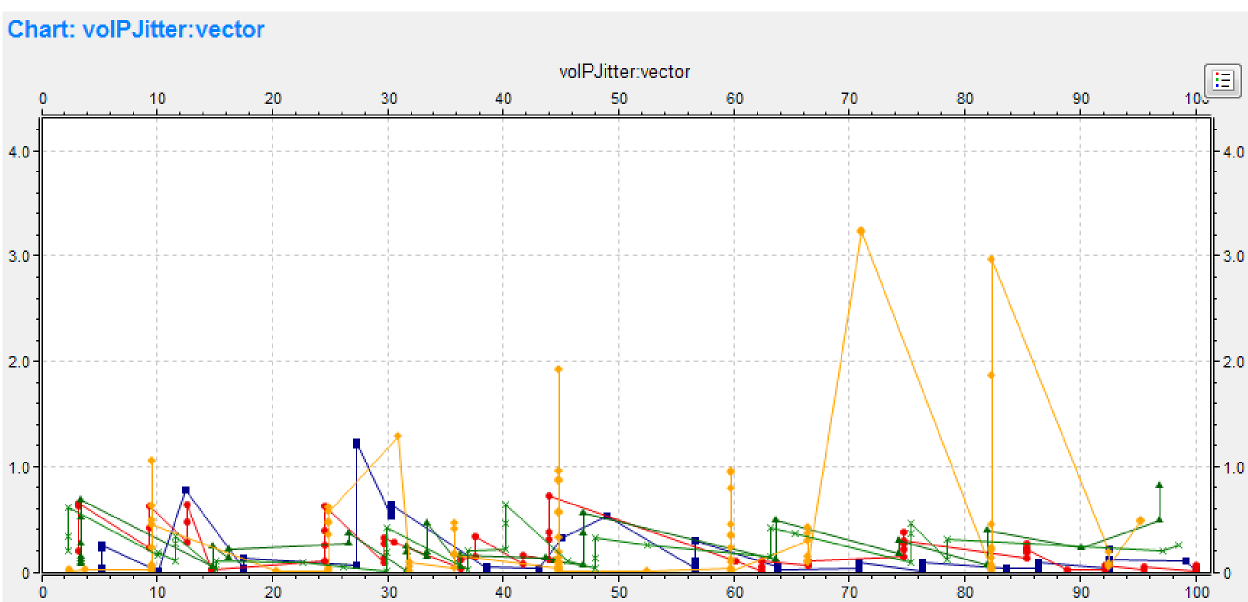
7.2.3 VoIP Jitter

Scalar:



The above result shows acceptable jitter for the voice calls in an LTE network.

Vector Representation:



8. CONCLUSION

The simulation design was tested under two different types of scenarios i.e. congested and non-congested. The results shown and discussed in *section 7* describes the clear behavior of voice calls in both scenarios even when there is other type of traffic (video streaming in our case). In the congested scenario, our results show higher number of packet loss, unacceptable jitter, and end to end delay of voice calls. On the other hand, the results from non-congested scenario shows acceptable QoS values for all the voice characteristics. The approach we took to solve the problem shows acceptable results for designing a network to meet VoLTE QoS specifications.

8.1 FUTURE WORK

In our project, we simulated voice and video traffic for VoLTE using SIMULTE. SIMULTE only supports the LTE data plane (or user plane). However, for more in depth analysis and simulation of VoLTE, control plane support should be added. Moreover, there are various application functions required for the VoLTE service design and deployment which are not present in SIMULTE. Handovers are also not implemented in SIMULTE; with handover support more accurate behavior of different flows of traffic can be observed even when a UE moves from one eNodeB to other. If support for LTE control plane, handovers and application functions can be added, then SIMULTE can be a complete framework for the simulation of all types of real world LTE scenarios.

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