Performance Study of Channel-QoS Aware Scheduler in LTE Downlink Using NS3

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Abstract— **Recent mobile technologies allow users to interact with each other through voice and video by using smart-phones or tablets. The applications available on the Internet are mostly using a similar protocol which is best-effort data network by service provider. Best effort service does not guarantee data to be delivered within the allocated time. Real-time applications compete equally for network resources in best effort, hence real-time application perform poorly using best effort services. In this paper, two classical packet schedulers and two channel-aware schedulers for Orthogonal Frequency-Division Multiple Access (OFDMA) are studied. The schedulers are simulated using real-time packets with a various channel condition of user equipment (UE) in NS3 simulator to test the performance of each scheduler. The Channel-Quality of Service (QoS) Aware scheduler outperforms all other schedulers for real-time traffic.**

Keywords-VoIP, Real-Time, Video, NS3, simulation, LTE.

I. INTRODUCTION

Long Term Evolution also known as LTE was introduced by 3GPP earlier in 2004 but it was only finalized and approved in 2008 [1][2]. LTE network consists of LTE devices and System Architecture Evolution (SAE). SAE is an evolution from third generation mobile internet (3G) packet core network. LTE has become the new standard for mobile network. The downlink and uplink peak rate of LTE are able to achieve 300Mbit/s and 75Mbit/s respectively. Latency of LTE is controlled by the Quality of Service (QoS) that permits less than 5ms latency in the Radio Access Network (RAN) [3].

LTE is able to cater fast moving mobiles as well as to support both Frequency Division Duplexing (FDD) and Time Division Duplexing (TDD) [4]. Although TDD seems to be overall a better choice, FDD is more widely implemented due to earlier technologies. LTE consists of a user plane called evolved Universal Mobile Telecommunications System (UMTS) Terrestrial Radio Access Network (eUTRAN) and a control plane called evolved packet core (EPC). In the eUTRAN, there is an evolved NodeB (eNodeB) which handles radio resource management (RRM) [5].

Real-time traffic has a strict requirement for delay. QoS ensures high-quality performance for critical applications. Traditionally network traffic uses best effort services which do not guarantee any reliability, delay, jitter, or other performance characteristics. On the Internet Protocol (IP) based network, the Integrated service (Intserv) and differentiated service (Diffserv) are able to provide preferential treatment to specified traffic.

The outcome of this paper is to analyze the capability of channel-QoS aware (CQA) in a mixed traffic environment. As a result of the simulation, CQA scheduler is able to allocate resources efficiently when considering the delay, channel condition, and guaranteed bit rate. We simulate Voice over IP (VoIP), video and File Transfer Protocol (FTP) traffic using NS3 to test the capability of schedulers as shown in Figure 1. The CQA scheduler is good for real-time traffic as it is able to provide better QoS as compared to other schedulers [6][7].

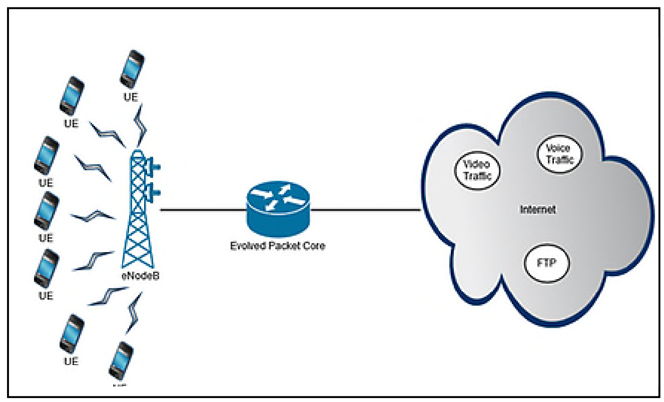

Figure 1. NS3 LTE Simulation Scenario.

This paper focuses on the downlink scheduler of the LTE network. The main focus is the QoS performance as it plays an important role in determining the improvement of QoS in LTE. This includes the networks throughput rate, delay, fairness and packet loss. The rest of the paper is organized as follows: Section II briefly describes several related works. Section III discusses the schedulers, while Section IV explains and elaborates on the methodology. Section V concludes the paper and indicates future works.

II. RELATED WORKS

The increase in the popularity of LTE technology system worldwide has gained more interest in the design of LTE packet scheduling algorithms. There are various downlink schedulers that were proposed by researchers. The advanced

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technology of high speed wireless network has caused the trend of VoIP calling to grow. Apart from having VoIP, video conference is also a need in this era. VoIP and video conferencing is categorized under real-time traffics and has a strict delay requirement in order for it to perform well. However, classical schedulers are not designed to be aware of QoS requirement for real-time traffic.

Various studies have been performed in OFDMA systems. For example, Proportional Fair (PF) scheduling algorithm allocates resources accordingly in order to achieve fairness among users [8][9]. However, PF can neither provide the best throughput nor provide the best fairness. The basic scheduling algorithm Round Robin (RR) was also studied in trying to optimize the capabilities of LTE [5][6][10]. PF and RR are not able to handle the priority for type of traffics [6]. Next, Priority Set Scheduler (PSS) was studied. PSS was designed by combining frequency domain (FD) and time domain (TD) scheduler that aims to provide a defined target bit rate to all users [11]. PSS is able to achieve part of QoS requirement by using guaranteed bit rate (GBR) as the observable matrix. However, it does not consider delay requirements, making it perform less efficiently. Another scheduler CQA was proposed by [12], whereby it is designed to improve the resource allocation for real-time voice traffic. CQA sorts the traffic's priority according to the channel condition, head of line (HOL) delay, and GBR. Thus, VoIP through LTE structure improved significantly using CQA scheduler [12].

III. DOWNLINK SCHEDULERS

A. Proportional Fairness

PF in NS3 works by allocating resources to User Equipment's (UE) when the UEs' channel quality is instantaneously high even though the average channel condition over time is low.

Using the PF scheduler, UEs are allocated to different Resource Block Group (RBG). Channel conditions and the throughput value of previous transmission is used to calculate the metrics value [9][13].

B. Round Robin

RR can be considered as the simplest scheduler. The scheduler works by dividing the resource blocks (RB) between the flows with non-empty queues. The scheduler method is to divide all the available resources to active traffic flows if the resource quantity is able to serve all the incoming traffic flows [8][9].

RR will not be able to allocate the resource to all flows if the traffic flow is bigger than the RB. RR will only allocate resources based on time to interval (TTI) and continue to allocate resources in the next sub frame and start from the last unallocated flow.

C. Priority Set Scheduler

PSS is a scheduler that combines FD and TD. It targets to provide fairness to UEs by using a specified Target Bit Rate (TBR). PSS works by selecting UEs which Radio Link Control (RLC) buffer is not empty [1]. The UEs are then divided into two according to their TBR. By dividing the UEs according to their TBR, the scheduler is able to decide based on the priority of serving those UEs [11].

UEs with the highest priority metric are forwarded to the FD scheduler. Then, the RB for each UE is allocated by using PF scheduler to calculate the metric. In the case of having a minimal number of UEs, FD scheduler provides a weight metric to control the fairness.

D. Channel-QoS Aware (CQA)

The CQA works by taking HOL delay, channel quality, and GBR parameter into consideration. CQA also utilizes TD and FD scheduler where it depends on channel quality and QoS requirements to allocate resources. This allows the attainability of a higher amount of spectral efficiency while satisfying the traffic delay requirements [12].

Similarly to PSS, CQA scheduler divides UEs according to their priority in the TD scheduling. The CQA scheduler then groups the UEs into flows and ensures that the FD scheduler allocates resources starting with flows which consists the highest computed metric. The metric is calculated using HOL, channel quality, and GBR.

The channel quality can be calculated using two methods which are PF or frequency selective fading. The GBR is specified in EPS bearer when simulating the CQA. In other words, CQA is aware of its channel conditions as well as its QoS parameters. Hence, a minor requirement on the networks performance is made available which can be in the form of guaranteed amount of data, etc. [14].

IV. LTE IN NS3

A. Voice Traffic

Simulation of VoIP traffic in NS3 is characterized by two periods; ON and OFF. ON is for the time when the users spend on talking whereby constant packets are transmitted at regular intervals. The OFF time is the time where the user stops from talking and packets are not transmitted.

Parameters and details for the traffic simulation are illustrated in TABLE I. ON and OFF time are given as 0.352 and 0.650 seconds respectively [15]. Actual bit rate depends on the codec used and the packetization time. In [15], G.711 codec is used. G.711 does not perform any compression and ensures the best voice quality. Due to the absence of compression rules, bandwidth requirements will be high.

TABLE I. VOICE TRAFFIC CONFIGURATION

Voice Traffic	Details
ON	0.352 seconds
OFF	0.650 seconds
Codec	G.711
Output	64 kbps
Bandwidth per voice call	200 bytes
Bandwidth at IP layer	80 kbps per call
Average bandwidth	28.1 kbps

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OnOffHelper is used to generate the traffic. In Figure 2, it is an example of the script to simulate voice traffic with OnOffHelper module.

```
OnOffHelper P1dlVoIPonoff=OnOffHelper("ns3::UdpSocketFactory",InetSocketAddress (ueIpIface.GetAddress (u), pdlPort));
P1dlVoIPonoff.SetAttribute ("OnTime", StringValue("ns3::ConstantRandomVariable[Constant=0.352]"));
P1dlVoIPonoff.SetAttribute ("OffTime", StringValue("ns3::ConstantRandomVariable[Constant=0.65]"));
P1dlVoIPonoff.SetAttribute("PacketSize", UintegerValue(VoIPdlDataSize));
P1dlVoIPonoff.SetAttribute("DataRate",DataRateValue(VoIPdlDataRate));
ApplicationContainer P1dlVoIPapp = P1dlVoIPonoff.Install (P1remoteHost);
P1dlVoIPapp.Start (Seconds(1.0));
P1dlVoIPapp.Stop(Seconds(simTime));
```

Figure 2. Voice ON OFF Helper.

B. *Video Traffic*

The simulation of video traffic requires Evalvid module in NS3. The Evalvid module simulates video by tracing the frame of the video. The behavior of this simulation is according to real time services such as video conferencing. In this simulation, the module uses st_highway_cif.st as the trace file for video traffic.

C. *Best Effort Traffic*

To simulate best effort application in NS3, a UDP echo module is used. The UDP echo is setup using echo server and echo helper. There are 3 attributes that are maximum packet transmitted, the time interval between packets, and the packet size.

D. *Simulation Setup*

The number of UEs starts from 5 UEs to 80 UEs with the interval of 5. The parameter is configured using the command line attribute. It is set within the script to enable the arguments to be parsed in command line. An example of parsing the command line argument is presented in Figure 3. The node for eNodeB and UE is configured with LTE stack protocol. All UEs are attached to the eNodeB and Radio Resource Control (RRC) connection is created between them.

SrsPeriodicity sets the maximum number of allowable UEs to be attached to eNodeB. Radio Network Temporary Identifier (RNTI) is generated to address the UEs. Collision requests from UE to eNodeB cause some RNTIs to be unused. Therefore, to enable the simulation to run more than 30 users, the srsPeriodicity needs to be configured to 160. This will enable the simulation to run without error. Figure 4 is the script to set the srsPeriodicity. The Internet is then created using PointToPointHelper which the attribute of the DataRate is set to 100Gb/s. The mobility model is also setup to position the UEs in different areas. The scheduler is setup using SetSchedulerType and the attribute of path loss model is set to Friis Spectrum Propagation Loss Model. LTE device is installed to the nodes and all the traces are enabled. The simulator is configured to run for 50ms before it stops and it is destroyed.

```
// Command line arguments
CommandLine cmd;
cmd.AddValue("numberOfNodes", "Number of eNodeBs + UE pairs", numberOfNodes);
cmd.AddValue("simTime", "Total duration of the simulation [s])", simTime);
cmd.AddValue("distance", "Distance between eNBs [m]", distance);
cmd.AddValue("interPacketInterval", "Inter packet inJavaScriptterval [ms])", interPacketInterval);
cmd.Parse(argc, argv);
```

Figure 3. Parsing Command Line Argument.

```
static ns3::GlobalValue g_srsPeriodicity ("srsPeriodicity",
                                          "SRS Periodicity (has to be at least "
                                          "greater than the number of UEs per eNB)
                                          ns3::UintegerValue (160),
                                          ns3::MakeUintegerChecker<uint16_t> ());
```

Figure 4. Setting SrsPeriodicity.

To simulate traffic from the Internet to eNodeB and to UEs, multiple remote hosts are setup. It acts as the host to each type of traffic. Each remote host is given an IP address and route using routing helper to the default gateway for eNodeB.

This simulation runs using Waf, which is a python-based framework for configuring, compiling and installing applications. A shell script is created to execute the Waf program in order to run the simulation. The simulation is performed multiple times by looping the command in the shell script.

The simulation is monitored by a module called flow monitor. The flow monitor is installed in UE and remote host. While monitoring the flow of the packet, the flow monitor records the data which can be called using a pointer. The flow monitor automatically maps the packets with all the data using flow ID. The throughput and delay can be calculated using the flow monitor. The trace file is loaded into Octave to compute the fairness index, packet loss ratio, throughput, and delay then plot the graph, as in Figure 5.

```
Logger << "Flow ID: " << iter->first << " Src Addr " << t.sourceAddress << " Dst Addr " << t.destinationAddress << std::endl;

Logger << "Tx Packets = " << iter->second.txPackets << std::endl;

Logger << "Rx Packets = " << iter->second.rxPackets << std::endl;
Logger << "Throughput: " << iter->second.rxBytes * 8.0 / simTime / 1024 / 1024 << " Mbps" << std::endl;
Logger << "Packets Loss Ratio: " << (( iter->second.lostPackets * 100) /iter->second.txPackets) << [illegible] << std::endl;
plr << (( iter->second.lostPackets * 100) /iter->second.txPackets) << std::endl;
```

Figure 5. Flow Monitor

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The entire flow diagram for configuring the simulation for NS3 is shown in Figure 6.

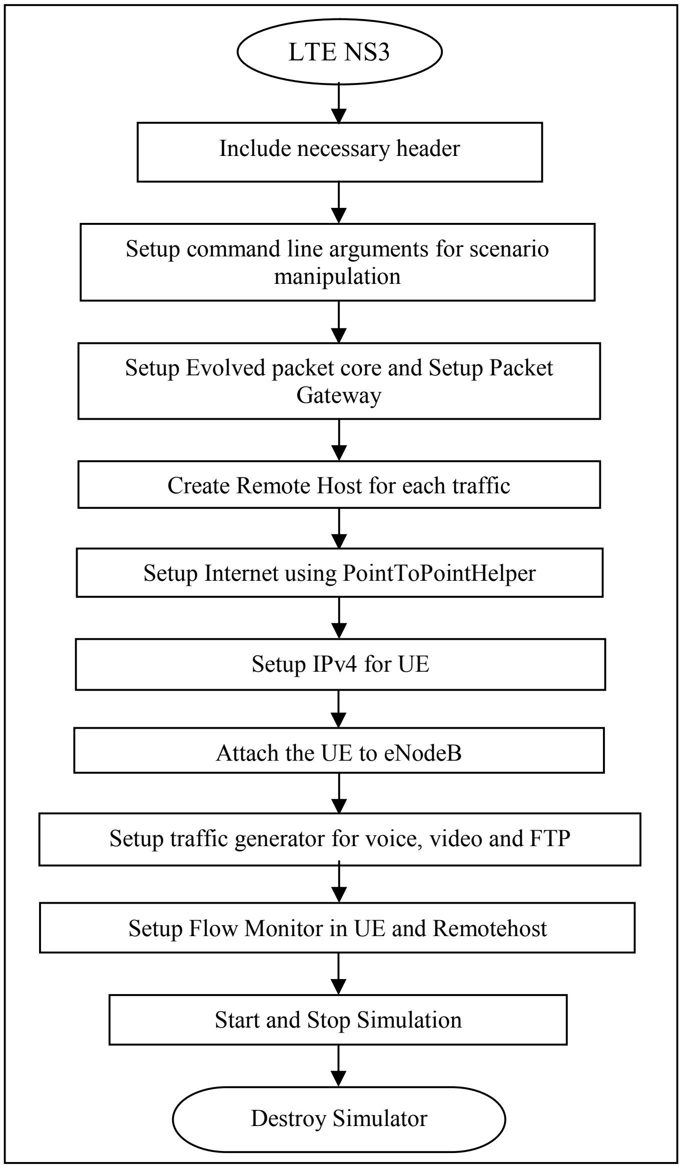

Figure 6. Flow diagram of NS3 LTE Configuration

V. Performance Evaluation

After simulating the VoIP traffic and video traffic of the LTE network in NS3, the graphs are plot using Octave. The outputs from the simulation are throughput, delay, packet loss ratio and fairness index.

A. *VoIP Result*

The CQA throughput is better than PSS where the throughput is higher starting from 30 UEs until 80 UEs. This can be observed in Figure 7. Compared to RR, throughput of VoIP is higher than CQA. This is the result of RR allocating resources without considering the channel condition of the UE. UEs with poor channel condition are able to get resources but are unable to transmit resources efficiently on time.

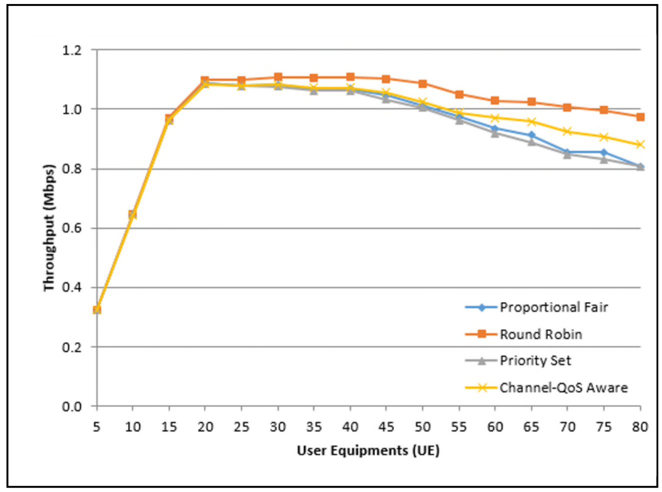

Figure 7. VoIP Downlink Throughput.

Delay for VoIP in CQA scheduler is 28.85% quicker compared to all other scheduler. This means that the voice packets are able to be transmitted in a short amount of time. The delay result is presented in Figure 8.

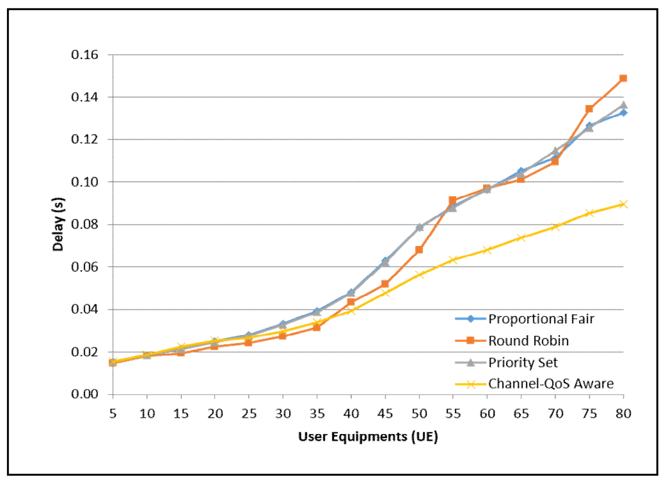

Figure 8. VoIP Downlink Delay.

B. *Video Result*

As shown in Figure 9, the video throughput for CQA scheduler is 7% lower compared to the other schedulers. This is because CQA scheduler allocates RB while considering the GBR of 64kbps for the UE to transmit the video data.

The CQA scheduler delay is 14.97% lower compared to other schedulers. Based on the results given in Figure 10, it shows that by monitoring the GBR and HOL delay, CQA could optimize the resource allocation and allocate the resource block efficiently.

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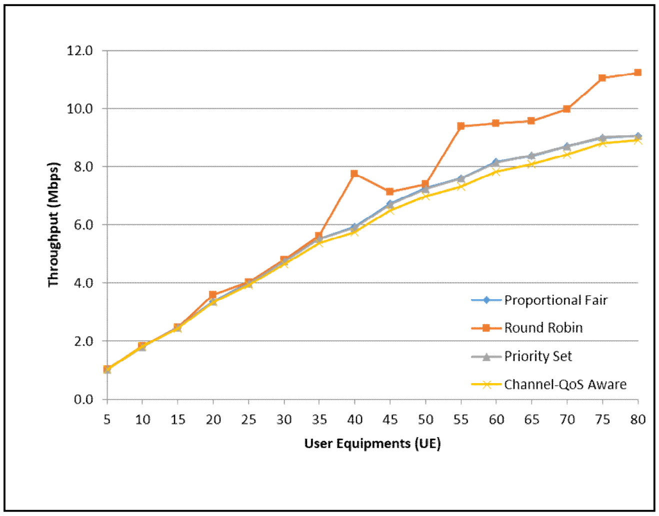

Figure 9. Video Downlink Throughput.

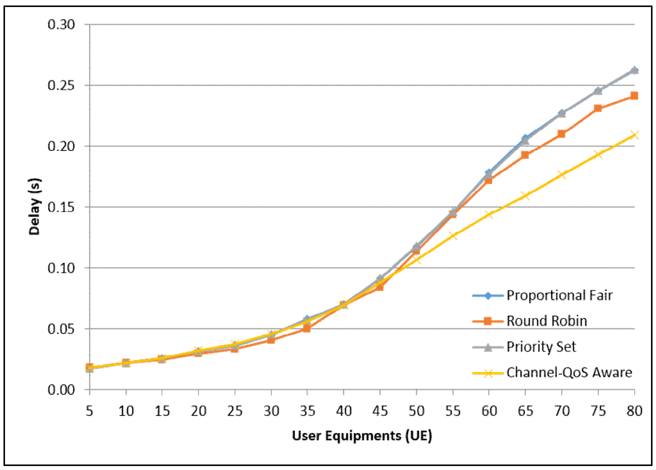

Figure 10. Video Downlink Delay.

C. *Best Effort Result*

The graph for UDP traffic that utilizes best effort technique is displayed in Figure 11. The throughput is 11.38% higher than PSS and RR. Since non-real-time traffic does not have strict delay requirement, the resource allocation prioritize real-time traffic while not jeopardizing the non-real time traffic. Therefore, the CQA is still able to allocate resources to enable the best effort traffic to be transmitted optimally.

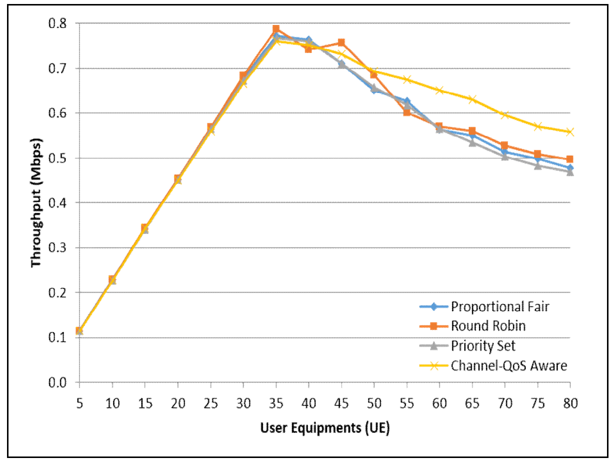

Figure 11. BE Throughput.

The delay for CQA scheduler is 14.96% higher. However, considering this is not real-time traffic, the CQA scheduler seems to fulfill its purpose. Figure 12 illustrates the best effort results for UDP traffic.

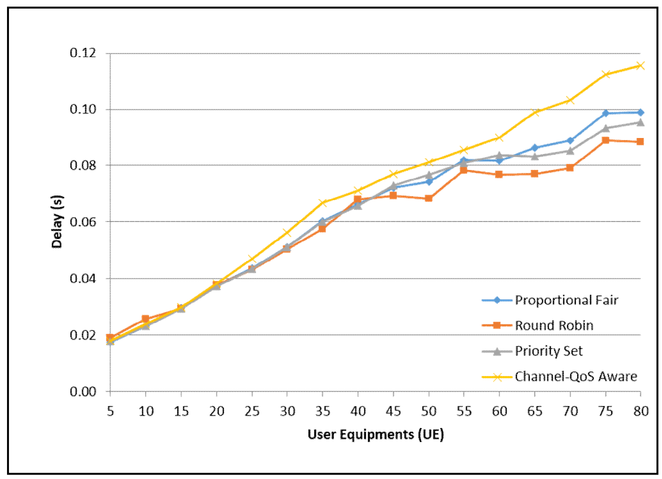

Figure 12. BE Delay.

D. *Fairness Index & Packet Loss Ratio*

The fairness index checks for the fairness in terms of resource allocation between UE. Figure 13 presents a histogram which shows that CQA scheduler performs best starting at 55 users. It is calculated that CQA is 1.94% fairer from 55 users to 80 users.

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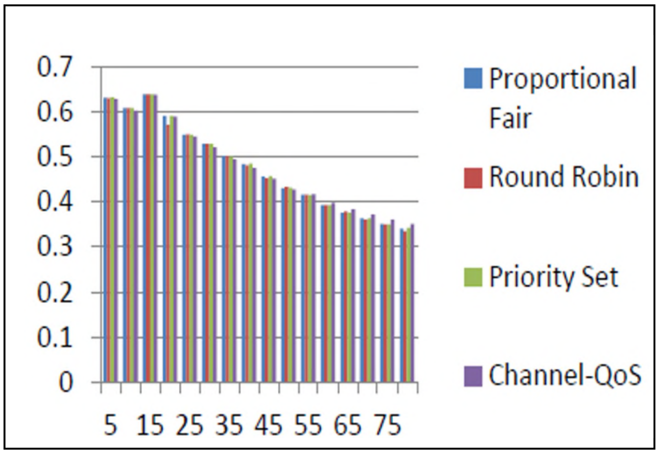

Figure 13. Fairness Index

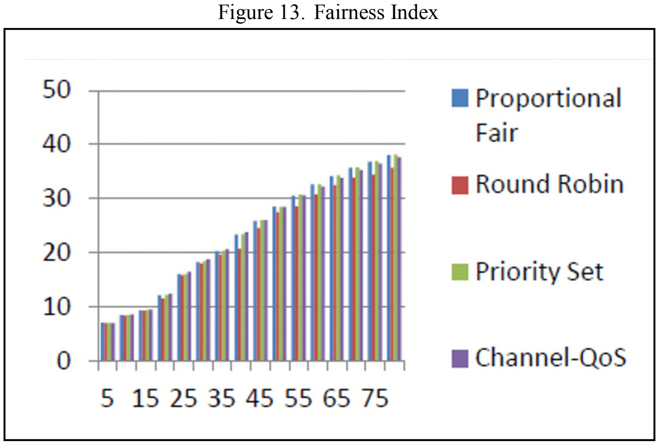

Figure 14. Packet Loss Ratio

Packet loss ratio defines the total of number for packet loss during the transmission of the packets. CQA scheduler has less packet loss ratio compared to PSS at around 0.21% lesser. This can be seen in Figure 14.

VI. CONCLUSION

Based on our study, PF and PSS have similar results. CQA scheduler is more suitable for real-time application compared to RR where delay of packets and fairness of resource allocation is an issue for RR. The delay for CQA is 15.23% better than PSS in video and 25.06% better than PSS in VoIP. Overall, we can conclude that CQA scheduler is more suitable for real-time application compared to PSS.

Further studies will be conducted on how to improve the throughput for video application while having optimum results for voice and best effort traffic.

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