Design & Simulation of Voice QoS Performance in Data Network Congestion for M/D/1 Queuing Model

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***Abstract* — Voice in IP networks is transmitted as packets over IP (VoIP), the voice signals are converted to IP packets after being digitized and compressed for transmission. However, some packets can be missed in their way to the receiving side, due to network congestion. The loss of these packets degrades the speech quality in the listener side at VoIP system transmission. Since voice is transmitted is real time, the receiver cannot request a retransmission for any lost packets. Voice and data multiplexing in VoIP network always face problems when huge TCP traffic is transmitted resulting the voice packet to be stuck in the network during congestion. Therefore, VoIP packets will be delayed. Since delay and loss are the main parameters that affect the quality of a voice signal in a VoIP network.**

This paper presents a design and a simulation study of voice and data integration in a VoIP network and analyses which scenario will suite the best performance for voice packets in traffic congestion to have a high voice quality rating when using a single data TCP source and a multiple TCP sources when multiplexed with a UDP voice source. This is accomplished by using NS-2 network simulator (version 2), and M/D/1 queue type with various queuing systems such as First in First out (FIFO), Fair queue (FQ) and Deficit Round Robin (DRR), which represent the technique mechanism to serve voice and data packets in a queueing system. Then loss and delay are measured for each scenario to determine the quality of voice.

Keywords: VOIP, TCP, UDP FIFO, FQ, DRR.

I. Introduction

In home network and broadcast services, personal and streaming (broadcast) content is receiving great attention from users who would like to access it from any device such as (mobile phones, tablets, personal computers, TVs, etc.) [1][2]. This growing figure in consumer electronic devices which are accessing Internet based services originating from both home networks and broadcast networks. The dominant scenario today for such devices is to perform more efficient when it comes to Voice over IP (VoIP) and its priorities in home network content. [3][4]

Multiplexing of data and voice packet is always a challenge network planning and design, where voice packets negatively impacted always due to the reason of the greedy nature of the TCP traffic in network. This negative impact happens at peak on low bandwidth, to give a reason that pushes many to research in this field; many researchers are working on different scheduling techniques and mechanisms to overcome this issue. [5]

They predict that VoIP traffic will soon be a significant fraction of the total telecom traffic moved around the world; consequently, networks are being built for high-capacity packet switched infrastructure. As VoIP evolves in modern world and everyone is using it daily, a VoIP call traditionally works on UDP protocol because the voice call transmission is real time and if some packets are dropped retransmission has no bandwidth benefit, which is completely opposite to TCP [6],[7].

The major issue faced by voice packet is when huge TCP traffic is transmitted resulting the voice packet to be stuck in the network at congestion points in the network, or any other purposes causing high Pervasive developmental disorder (PDD) delay for the call. If the delay is above acceptable voice delay levels the user hangs up the call. This is the reason why different scheduling techniques are used for the transmission of traffic so that voice packet is transmitted as soon as possible. This also does not make life easier for giving priority to voice which may result in high voice traffic only causing a TCP transmission delay in the network.

For different cases using one UDP source with TCP sources different scheduling techniques such as FIFO, FQ and DRR are being used and designed to solve VoIP and data problems depending on the R-factor formula for voice quality rating. When there is transmission of some TCP application such as a user is uploading large size file to the internet and in the other hand a VoIP call is established, a dramatic impact on VoIP call will occur specially on small bandwidth network such as home DSL. [8]

Our contributions can be summarized as follows:

- data interaction in the network and simulate a voice data multiplexing using NS2 simulator, and give a comparison between the different scenarios of voice packets being multiplexed and served in the network;

- Find the best parameters that can be used for delay and loss in nodes / buffers, by using the R-factor which will determine the degree of having a great voice quality not affected by the data packets;
- Improve the performance of the networks and enhance the voice quality in terms of loss and delay;
- Study the Queue of packets in the buffer, and reduce the congestion through the networks and minimize the packet loss through a VoIP networks.

II. Background

Voice over Internet protocol commonly as (VoIP) is the most widely used technology for transmission of voice and data over the Internet protocol. This technology completely revolutionized the world replacing the old methods of telephony over circuit switched networks. In the beginning there was POTS (Plain Old Telephony Service). In POTS cost was a big factor its consumers had to bear it in order to maintain their network. Routing in POTS was also much less dynamic as compared to VoIP. For example, if a line is down the call could not get through but in VoIP, multiple calls can establish multiple routes. In VoIP the data is transmitted over the packet switched network, which gives the users, a good performance and more importantly it increased the capacity of the system providing opportunities to service providers to accommodate more users. VoIP not only enables to send voice data but also it allows the transmission of video and streaming applications through it. It provides the user with many useful features such as email and web surfing [9], [10], [11]

VoIP enables a user to communicate with friends, employees etc. on the move by using a wireless IP phone or similar devices. In VoIP as the number of user's increases it is very difficult to maintain the quality of the content delivery especially in the case of voice as it is delay sensitive and it requires careful steps so that other types of traffic (e.g. TCP) do not interfere with it.

Problems with VoIP and TCP: Unique problems occur when other traffic are multiplexed with voice data. As voice traffic is very jitter sensitive it requires special arrangements. For example, in the case of TCP as from its greedy nature it tries to occupy the whole of the bandwidth and the voice packets will be stacked behind many large TCP data packets [12], [13]. [14]

Moreover, as TCP is greedy it ramps up all the available bandwidth, which causes starvation for the voice traffic. In order to cope with these problems Priority Queuing and non FIFO-schedulers are used in order to protect voice traffic from different types of greedy traffic.

One of the problems associated with hard thresholds in IP networks is when a TCP connection loses a packet because of the network congestion, it enters a slow-start phase and starts reducing load on the network, and alleviates congestion. TCP is greedy and jumps dramatically to take off all the bandwidth that is available.

In a network congestion with a field buffer, any incoming packets are discarded, affecting many TCP connections, leading to a period of under-utilization followed by a substantial load increase after the slow start, which will lead to a global synchronization cycle [15] [16], [17].

III. Modelling and Simulation

A. Network Simulator (NS2)

There are many types of network simulators, and the most used in research papers are NS2/NS3, OPNET, Packet Tracer and NetSim. In this paper NS2 will be used. Network Simulator the second version, is known by NS2, which is an object-oriented simulation tool, discrete event driven network simulator [18]. UC Berkely occupied this development, which is written in OTcl and C++ languages. NS is useful for simulation of wireless, wired networks and protocols (e.g., UDP, TCP, routing algorithms,). In general, it can be said that this simulation tool can provide users with a specification of network protocols and simulating their corresponding behaviors [15] [16].

B. M/D/1 Design

M/D/1 is a good queueing model. It is used in this paper for it has a fixed service rate and a memoryless arrival rate and one server. Therefore, M/D/1 can give a better performance and simplicity for this design. M/D/1 is used mainly due to its arrivals are determined by Poisson processes. Therefore, in this case it represents voice while using UDP. Poisson distribution fits the voice distribution. The basic characteristics of Poisson distribution are that the packets will arrive with average of λ in a particular interval. The inter arrival time between two packets is exponentially distributed. The arrival rate λ is defined in the OTCL code, and the inter-arrival time between two packets is set to be exponential. The generation of Poisson traffic can be made using On-Off source, setting Ton to zero generates one packet after the off period.

M/D/1 experimental design is introduced with these factors to represent an M/D/1 queue type. By considering a simple topology of three nodes, a Poisson traffic source, UDP protocol and a Droptail buffer, is shown in Fig. 1.

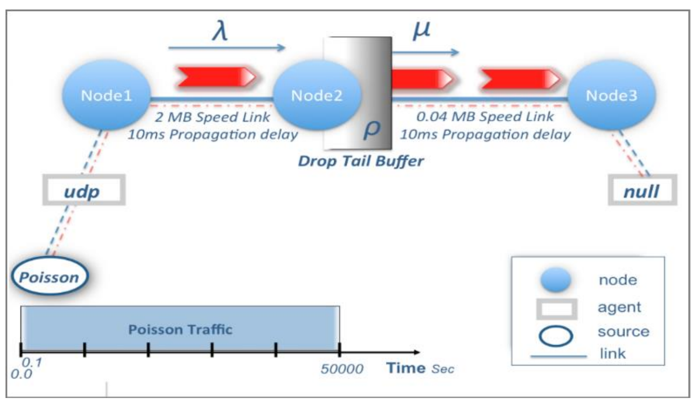

Figure 1: network topology of M/D/1 design

As shown in the Figure above, the traffic is generated from Poisson traffic source with specific parameters and attached to node1, using UDP agent; where the agent defines the type of traffic if its TCP or UDP. Therefore, node1 will generate the packet flow sending it via a link of 2Mb with a propagation delay of 10ms to node 2. This represents FIFO / Droptail queue, connected to the last node that represents the sink via a link of 0.04Mb with a propagation delay of 10ms, given a bottleneck shape to study the congestion after the buffer. To study the buffers reaction and the flow of packets in M/D/1 queue type, specific parameters where set for M/D/1 and NS2.

C. Parameters used in M/D/1

For the simulation of M/D/1, these two parameters are set in the topology:

- Packet size 1000 bytes.
- Link bandwidth of 400Kb.

On the basis of these parameters' M/D/1 parameters can be set. First the service rate is calculated as:

$$Service\ \ rate = \frac{link\ bandwithe}{paket\ \ size} = 5\,pkt/\,\mathrm{SEC} \qquad (1)$$

As a purpose study in this paper, an assumption of various values of the load for the buffer are used as initial values to measure the arrival rate in different arrivals for the study of the buffer statues and the M/D/1 queue.

When setting the service rate to 5pps, and using assumed values for the load, the rest of parameters can be shown in Table I. This simulation tool can provide users with a specification of network protocols and simulating their corresponding behavior's [15] [16].

Table I. Parameters of M/D/1 & NS2

Parameters	Values	Required for
Link bandwidth	0.04 mb	M/D/1
Packet size	1000 bytes	NS2
Service rate	5 pps	M/D/1
Load	0.1, 0.3, 0.5, 0.7 and 0.9	M/D/1
Scheduler	Droptail	NS2
Propagation delay	10 ms	NS2
Buffer size	100 pck	NS2
Sampling time	0.2	M/D/1
Simulation time	50000s	NS2
Source type	Poisson	M/D/1
Burst time	0 s	NS2
Off time	1/(arrival rate)	NS2

D. Topology design

A queuing system of an M/D/1 queue type is used in this case, where for the voice traffic a Poisson source is used to generate voice packets with a UDP protocol, for it is a real time protocol and this source is attached to node '0', where in the other hand node '1' is attached to a FTP application source with TCP, to generate TCP packets. Both node '0', '1' will generate two flows of traffic TCP and UDP and multiplexed in node '2' which represent the buffer FIFO / FQ / DRR in our case. Figure. 2 Shows the used network topology in this design.

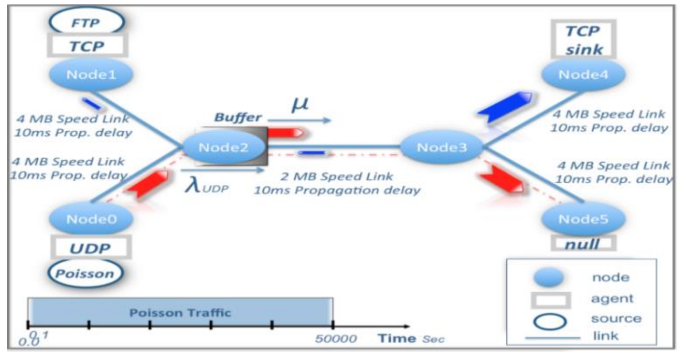

Figure 2. Network topology design for voice and data multiplexing

After simulating this topology, the study area will be between node '3' and '5', and measuring the delay of voice packets, as for the queue size of the buffer the area between node '2' and '3' will be in study, hence a link of 2 Mb is used between the nodes to give a less service time. In one case a FIFO scheduler is implemented in the network and the other case is a FQ and DRR scheduler.

E. Parameters used in simulation

For the simulation of voice and data packets, these two parameters are set in the topology:

- Voice packet size 100 bytes.
- Link bandwidth of 2Mb.

On the bases of these parameters the service rate is calculated, as shown:

$$Service\ \ rate = \frac{link\ \ bandwithe}{paket\ \ size} = 2500\,pkt/\,\mathrm{SEC} \qquad (2)$$

Table II. Parameters of voice &data multiplexing design and NS2 simulator

Parameters	Values	Required for
Link bandwidth	2 Mb	UDP
UDP packet size	100 bytes	UDP
TCP packet size	1000 bytes	TCP
Service rate	2500 pps	UDP
Application	FTP	TCP
Window size	20	TCP
Scheduler	Droptail / FQ/DRR	NS2
Propagation delay	10 ms	NS2
Buffer size	100 pck	NS2
Sampling time	0.2	UDP
Simulation time	10000s	NS2
UDP traffic source type	Poisson	UDP
TCP traffic source type	FTP	TCP
Burst time for UDP	0 s	NS2
Off time for UDP	1/(arrival rate)	NS2

IV. RESULTS AND DISCUSSION

A. Validation of M/D/1 simulation

The design in M/D/1 was explained in the previous section, when executing the NS2 code, it will give results of number of packets per minute and the queue size in the buffer. Starting with the initial values for the load in the buffer, load with values of 0.1, 0.3, 0.5, 0.7 and 0.9 are used for studying the queue system within these values and observing the output. These values of load represent almost all of the periods that the buffer will be queued with packets, where load is equal to 0.1 means that the buffer is loaded 10% (ratio arrival rate over service rate), load equals to 0.5 means 50% load, and when the buffer is in the heavy traffic period is when the load equal to 0.9, 90% loaded. Where the results are generated as data in the format of a trace file in NS2, then being plotted via Matlab, using the statistical distribution tools in Matlab and codes these results are given.

1. Poisson arrival rates

The number of packets per minutes in NS2 gives the throughput. The given results are distributed in shape of Poisson distribution, while the traffic is generated as Poisson distribution. Fig. 3. Shows the results of number of packets per second for different values of load with a fitting of theoretical Poisson distribution over the simulation results of arrival rate for each load.

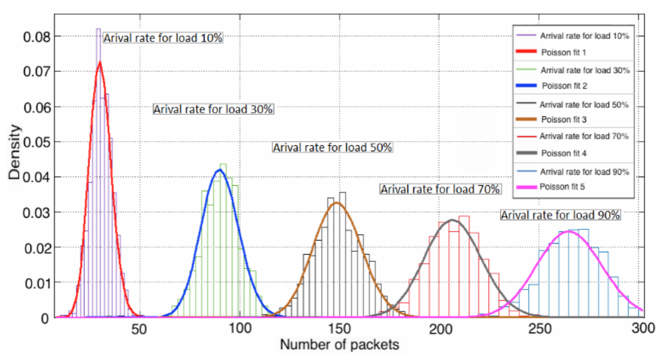

Figure 3. Fitting Poisson distribution to arrival rate (No. pkt/min) for all values of load

2. Average Queue size

A Queue is generated when more than one packet is waiting in the buffer to be served to its destination Figure 8 shows the PMF of queue size results from NS2 for each load. A log scale plot of the queue length for all the various periods of the load is used to compare between their values as shown in Fig.4.

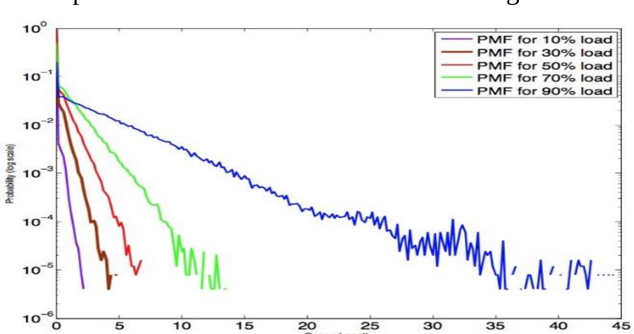

Figure 4. Log plot of queue size PMF for all values of load in log scale

3. End to End Delay

In networks, the time that a packet takes to be transmitted from its source to its destination is referred to as 'end to end' delay. This means how much delay a packet experiences during his transmission period. The 'end to end' delay in M/D/1 simulation results are shown in Table III, They are plotted in Figure 5. The delay for the packet in its transmission period in the case of load (10 – 30) % is low when compared to the load at 90 %, where it reaches a delay value seven times greater than it value at 10% load. This is due to the waiting period in the buffer. As seen previously the mean queue length in 90% load is high, given an increase in the time period for the packet waiting in a queue to be served.

Table III. End-to-End delay simulation results

Parameters	**UDP source**				
Load	**0.1**	**0.3**	**0.5**	**0.7**	**0.9**
Arrival rate λ UDP (pkt/sec)	0.5	1.5	2.5	3.5	4.5
E-2-End delay NS2, d (sec)	0.2323	0.26514	0.32309	0.43373	0.93934

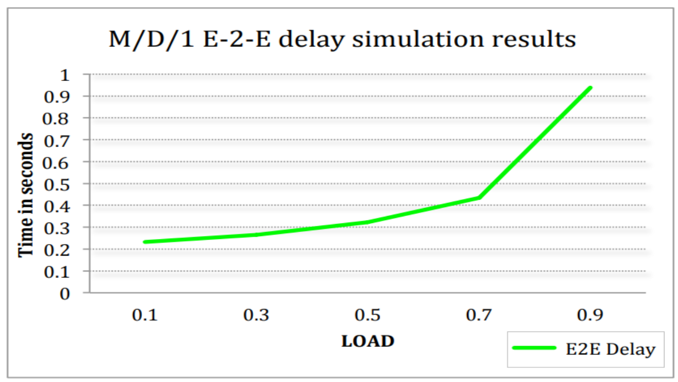

Figure 5.End-to-End delay simulation results

4. Calculation of Results and comparison

From M/D/1 equations, it is possible to find the theoretical values for the queuing system. For each value of the load the arrival rate is calculated for a fixed service rate, from:

$$\lambda = \mu . \rho \tag{3}$$

From each arrival rate the off time for Poisson source is calculated, and set in the NS2 to be compiled giving results. For the specified arrival rate and load, the off time is given by:

$$Toff = \frac{1}{\lambda} \tag{4}$$

The arrivals are (pkt/sec) converted to (pkt/min), giving the theoretical results for the system throughput. Meanwhile the mean queue length is calculated from M/D/1 formulas:

$$\text{Mean queue length,} \quad L = \lambda . EW \tag{5}$$

$$\text{Mean waiting time,} \quad E(W) = \frac{\rho^2}{2(1-\rho)} \tag{6}$$

Table IV. Theoretical results and simulation results

Service rate, μ =5 (pkt/sec)	**Load, ρ**				
	0.1	***0.3***	***0.5***	***0.7***	***0.9***
λ (pkt/sec)	0.5	1.5	2.5	3.5	4.5
Mean off_time, ***Toff*** (sec)	1.996	0.662	0.396	0.2817	0.2182
Arrival rate (Theory), λ (pkt/min)	30	90	150	210	270
Arrival rate (Ns2), λ (pkt/min)	30.08	90.16	149.13	207.12	265.53
Mean service time, ***E(s)***	0.2	0.2	0.2	0.2	0.2
Mean waiting time, ***E(w)***	0.0111	0.0428	0.1	0.233	0.9
Mean queue length (Theory), ***L***	0.0055	0.0642	0.25	0.8165	4.05
Mean queue length (NS2), ***L***	0.0051	0.0598	0.229	0.7170	3.17

Using a log-normal graph, the values of theoretical and simulation results for each load becomes clear. It is observed that the results match totally for the arrival rate, while for the mean queue length it is observed that the results have an excellent fit for the load values 0.1, 0.3, 0.5 and a slight deviation in the results for the load 0.7 and 0.9, as shown in Fig.6 and Fig.7.

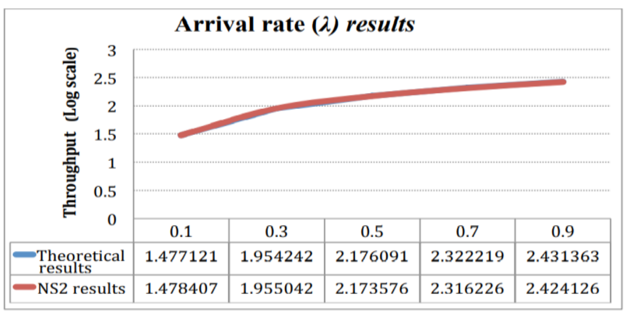

Figure 6.Theoretical and simulation results comparison of arrival rate

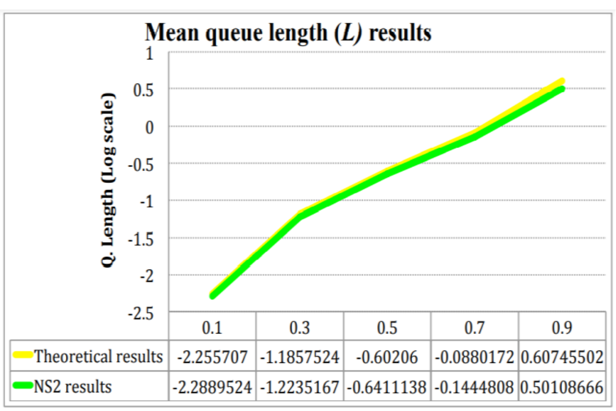

Figure 7.Theoretical and simulation results comparison of mean queue length

From the PMF results it is noticed that the queue size probability has an exponential distribution shape. By using Matlab tools a comparison between PMF queue size simulation results and theory as an exponential distribution fit is shown in Fig. 8.

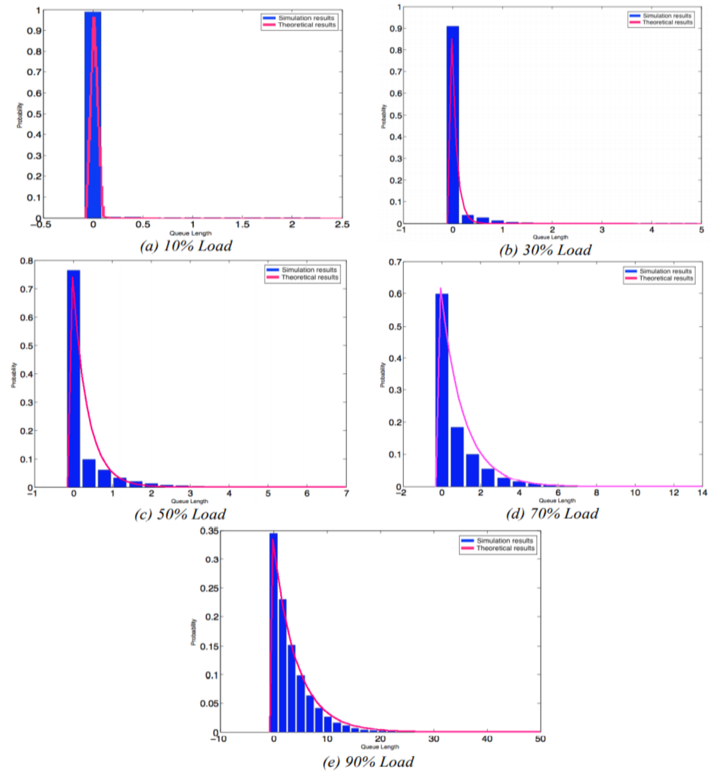

Figure 8.Comparison of simulation & theory results for queue length probability

5. *Discussion of M/D/1 results*

The average arrival rate of packets in the graphs for Poisson traffic fits with Poisson distribution for all buffer loads. Different load values were chosen for an overall study of all queuing periods. In the first period it is observed that a small number of arrivals λ = 30 (pkt/min) and increasing dramatically as the load starts to increase, ending with a value of λ = 270 (pkt/min) for a higher load of 90 %. The theoretical results match perfectly with simulation results.

The probability distribution graphs show that the highest probability is represented for a period when the buffer is empty of packets. This means the queue size is a very law figure. At a load of 10% this probability is very high of more than 95% due to its low load. The probability changes in a small portion of its values at loads of 30% and 50% with a small decrease in the high probability of 5% and 10% respectively at the low values of queue size. On the other hand, it is observed that a load of 70% shows a drop of probability at low queue size and a negative exponential shape for inter-arrivals of packets in the buffer. This case has more chance to fill the buffer than previous values and ranges up to 12 packets. While the load starts to increase the buffer backlog increases. At 90% load, the probability of low queue sizes drops to almost 35%, whereas it rises higher when the buffer backlog range reaches 25 packets given a high number of queue size, meaning it is starting to fill up the buffer.The average queue length has been chosen on five loads, as 10%, 30%, 50%, 70% and 90%. The theoretical values are given by

M/D/1 model and plotted in a lognormal graph with the simulation results. It shows a very less deviation between theoretical value and simulation value at high load and an excellent match at low load. This proves that M/D/1 queue is perfectly working, as the building block of this project. As it is working properly, it can say that it would perfectly suit voice and data multiplexing condition. The average values of delays in M/D/1 rises up as the load starts increasing; it tends to create a backlog. It is observed that the packets will take longer time to be served with high average delay values as the loads increase.

B. Simulation of Voice & Data multiplexing

1. Results of voice and data multiplexing

Voice and data multiplexing design was explained in the previous section. In this simulation, the most important is the voice quality where the voice packets are multiplexed with data packets representing a home network scenario or other networks. Fig. 9 represents the voice and data multiplexing design

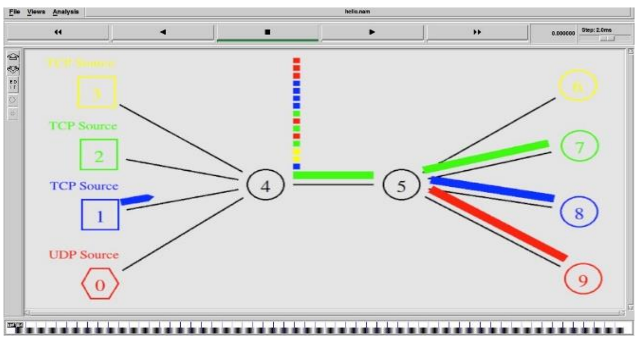

Figure 9. NS2 animation output for voice & data multiplexing design

In this design node '4' will represent the buffer in each case and observing its results for all three mechanisms, focusing on delay and loss and the relationship between them, which is represented by R factor to measure quality of voice.

Using the same network topology and only changing the scheduler mechanism, with same parameters as shown in Table 2, different results were taken for FIFO, FQ and DRR cases. With an M/D/1 queue type, the service rate in this design is measured as:

$$Service\ \ rate = \frac{link\ \ bandwidth}{paket\ \ size} = 2500\ pkt/\sec \qquad (7)$$

Starting by choosing n different values for the arrival rate and observing the buffer and taking results of voice traffic as delay and loss during a simulation period of 10,000 seconds, it is possible to determine the rate of voice quality for all cases by R-factor value from equation (1) and Table II.

2. Results of FIFO

In this case first incoming packets are served first, the voice traffic packets are multiplexed with TCP traffic. From simulation of NS2 delay, loss and queue length are measured, Table V illustrates these results.

Table V. Results of FIFO

Parameters	***FIFO results***				
Arrival rate (UDP) λ_{udp} (pkt/sec)	2250	1750	1250	0750	0250
Arrival rate Theory, λ (pkt/min)	135000	105000	075000	045000	015000
End- to-End Delay NS2, d (sec)	0.04792	0.048642	0.048973	0.05583	0.0726895
PLP, NS2	0.0552	0.0409	0.0431	0.0374	0.0403
R factor	74.9497	78.6863	78.70760	79.4991	78.2715

The end-to-end delay is plotted in a normal graph against the UDP arrival rate, shown in Fig.10. From delay and loss values in Table 5, the R-factor is calculated for the measured voice quality and plotted in a bar graph shown in Fig. 11.

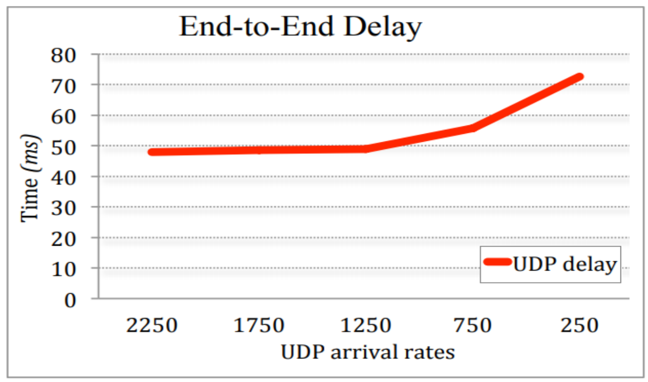

Figure 10. FIFO end-to-end delay versus UDP arrival rates

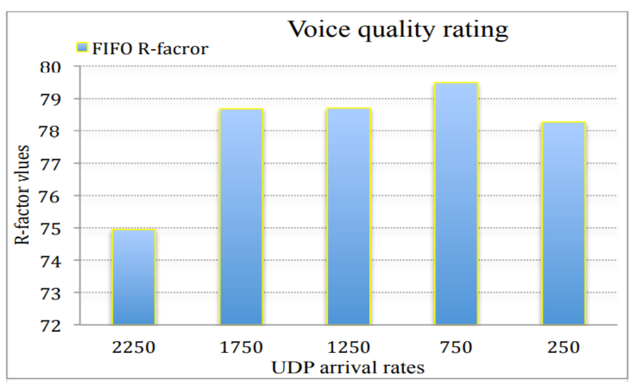

Figure 11. FIFO R factor values

3. Results of FQ

This case is different from the previous ones because both flows are given fair queuing. From NS2 it is possible to fix a parameter to take the fair queuing mechanism, setting 'secs Per Byte_' to a value of 0.09ms gives the best results for the voice traffic packets multiplexed with TCP traffic in a simulation of 10,000 second duration .

Delay, loss and queue length are measured from simulation and given in Table VI.

Table VI FQ results

Parameters	FQ results				
Arrival rate (UDP)λudp (pkt/sec)	2250	1750	1250	0750	0250
End-to-End delay NS2, *d*	0.107997	0.107933	0.107707	0.0364027	0.0330347
PLP NS2	0.09804	0.02649	0.02467	0.000032	0.000078
R factor	64.5180	81.2067	82.1747	93.3116	93.3721

The end-to-end delay is plotted in a normal graph against the UDP arrival rate as shown in Fig. 12. From delay and loss values the R-factor for FQ is calculated for the measured voice quality, and plotted in a bar graph shown in Fig.13.

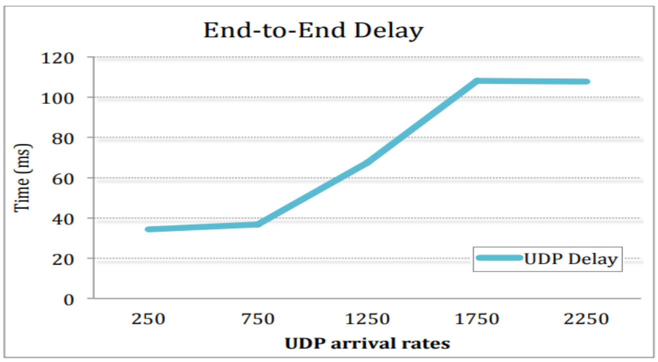

Figure 12. FQ end-to-end delay versus UDP arrival rates

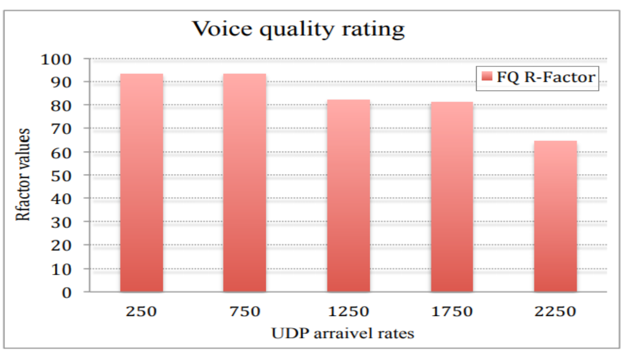

Figure 13.FQ R-factor values

4. *Results of DRR*

This case is different from the two previous cases because here both flows are given a high and low queueing priority; the voice is given a high priority and low priority for data. In NS2 DRR parameters are chosen to set queueing mechanisms as :

- 'buckets_', set to '10' where they indicate the total number of buckets to be used for hashing each of the flows .
- 'blimit_ ', is set to '1000' which represents buffer size in byte .
- 'quantum_ ', is set to 10,000 which is the number of bytes for each flow can send during one round .
- 'mask_,' is set to'0', which means a flow consisted of packets with same port, node Ids .
- Results for the voice traffic packets multiplexed with TCP traffic obtained from a simulation of 10,000 second duration. Delay, loss and queue length are measured from simulation and given in table VII.

Table VII. DRR results

Parameter	DRR results				
Arrival rate λ_{udp} (pkt/sec)	2250	1750	1250	0750	0250
End-to-End delay NS2, *d*	0.03427	0.03425	0.03423	0.03425	0.03437
PLP, NS2	0.000025	0.000021	0.000028	0.000016	0.000078
R factor	93.3661	93.9681	93.3656	93.3705	93.34

From the results in Table VII the end-to-end delay is plotted in a normal graph against the UDP arrival rate, shown in Fig. 14. From delay and loss values in same table, R-factor for DRR is calculated for the measured voice quality, and plotted in a bar graph shown in Fig. 15.

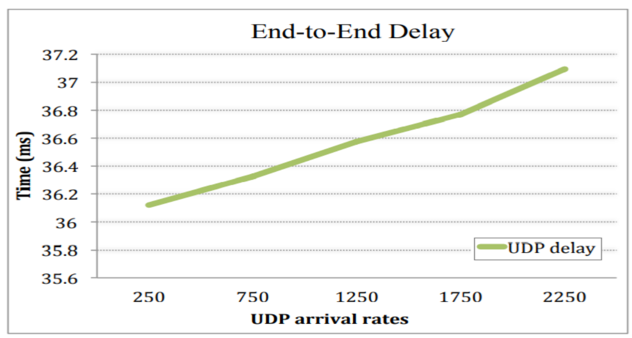

Figure 14. DRR end-to-end delay versus UDP arrival rates

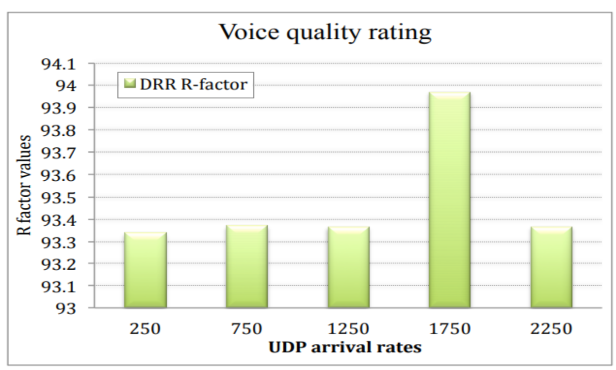

Figure 15. DRR R-factor values

5. Discussion & comparison of multiplexing Results

In simulation processes UDP and TCP are multiplexed together in three different scenarios, where in the first two scenarios FIFO and FQ case, TCP was the dominant traffic flow creating changes in the network when compared to the M/D/1 simulation.

The first case of FIFO, presents how voice packets jostle and contend with the data packets to manage to reach its end point. The queue size probability distribution shows that the buffer has the same queue length shape distribution for all UDP arrival rates. The highest probability is for the queue size of high values and low for low values for all arrival rates. As for the delay, it is represented by end-to-end delay showing that voice packets have a high delay when TCP traffic have greater arrivals than UDP traffic, ($\lambda TCP > \lambda UDP$) and a decrease in the delay for the opposite case ($\lambda TCP < \lambda UDP$) where UDP packets start flowing with high arrival rate in the network.

The second case in this paper is the FQ, which presented how fairness is done between voice and data packets in the buffer. The queue size probability distribution shows that the buffer has different queue length distributions for all UDP arrival rates. At low UDP arrival rates the queue looks more likely like a Poisson distribution at λUDP = 250 pps, 750 pps. When the arrivals rise higher, the queue takes another shape and the rest arrivals look like a steady state period. This is due to the fact that the system takes same number of packets from each flow. The delay is also represented by end-to-end delay showing that voice packets have a low delay when TCP traffic has greater arrivals than UDP traffic, ($\lambda TCP > \lambda UDP$) and increase in the delay when its arrivals change to ($\lambda TCP < \lambda UDP$) where UDP packets start to congest in the network.

The last DRR case is the important case in this paper for it presents a design for the best mechanism of priority for voice traffic when multiplexed with several data traffics. Voice packets flow in the system with a ratio of 10:1 taking in each serving round '10' UDP packets and '1' TCP packet. The queue size probability distribution shows that the buffer has high probability for small queue size values when λUDP = 250 pps and the queue is distribution is close to Poisson for the rest UDP arrival rates. The end-to-end delay for the voice packet in this case takes approximately similar performance to the M/D/1 simulation for they are less affected by TCP traffic then in FIFO and FQ. The results show that voice packets have a low delay when TCP traffic has greater arrivals than UDP traffic ($\lambda TCP > \lambda UDP$) and an increase in the delay when the arrivals change to ($\lambda TCP < \lambda UDP$) where UDP packets get served in fast technique in the network.

Voice quality is the main issue in this study and from the R-factor it is possible to rate how good or bad the voice quality is based on delay and loss where delay does not define the quality in its own; on the other hand, the acceptable range of delay in the network for most user applications is from '0' to '150' milliseconds. From these results all cases had a delay within this range. DRR had the lowest end-to-end delay when compared to the other schedulers.

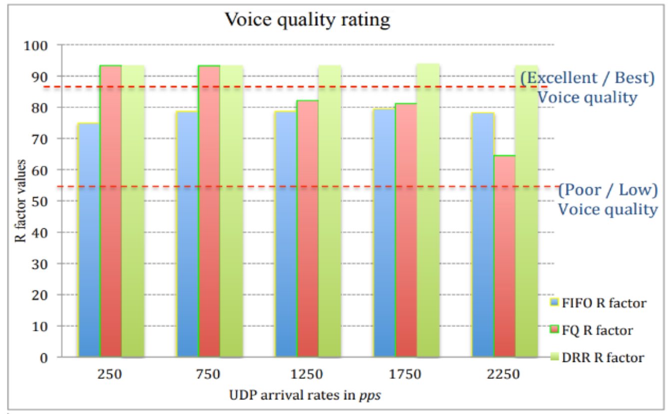

Figure 16: comparison of voice quality rating between FIFO, FQ and DRR

From Fig. 16 above, when comparing all cases and scenarios for voice quality, it shows that values of R-factor are oscillating between '95' and '65', passing through all rating periods; where for FIFO voice quality takes its highest rate at '79' for a Poisson arrival rate λUDP = 750 pps, and takes its lowest value when λUDP = 2250 pps, for an R- factor value of '74'.

While the FQ mechanism acts similarly, only the quality of voice drops to a poor rating when λUDP = 2250 pps with an R-factor value of '64', and its highest value of '93', given best performance in the low UDP arrival rates. As for DRR it levels at a value of '93', which is in the best rating range for all different arrival rates. In general, DRR gives the best and excellent voice quality when compared to FIFO and FQ, having a standard of excellent rate to all arrival rates. This is due to the priority technique to enhance the voice quality in the network, as shown in Figure 26. In the figure FIFO is swinging in the same fair quality of voice rating, which can be acceptable for some users, while in FQ it peaks in values of excellent voice quality rate dropping to poor rating for higher UDP arrival rate.

V. Conclusion

This paper has focused on studying the voice packets when multiplexed with data packets. It was found that TCP packets are really a greedy traffic flow which ramp into the buffer and starts to fill it up causing congestion to voice packets such as in low bandwidth networks, and this affects the quality of voice. This can be avoided or prevented from happening if the incoming traffic flows to the network are controlled in a certain mechanism. For voice, given the priority technique to UDP flows can enhance the voice quality in the receiving end, and by using M/D/1 queue type to serve packets in a fixed rate can make it easier to prevent voice congestions.

Given priority to one flow in the expense of another can give excellent results only if the lower priority flow is non real-time protocol. In this case, TCP is put to a lower priority to give voice a better performance in the network for its real time protocol.

DRR can be a priority mechanism when given a value in bytes how much a flow can send in a round robin technique, and voice packets are smaller than data packets in size lets the voice flow be served in each round more than data. DRR gives the best solution for VoIP problems when compared to Droptail or fair queueing mechanisms, which resulted an excellent quality of service for the voice, having the best voice quality rating. While the fair queueing mechanism FQ, resulted to good results for a certain period, which is low arrivals for voice packets, the results change when a rate of packets increases.

Using R-factor to rate the voice quality made it easier to understand how to improve and design techniques in network modelling and design; the use of R factor in this project gave a kind of simplicity in finding the quality of voice rating which is the quality of experience. The main important conclusions are therefore:

- The enhancement of voice quality is done when given priority to UDP packets, and keeping TCP in a lower priority level.
- This mechanism not only enhances voice quality but also can increase the network efficiency for a certain type of flow.
- TCP traffic is the dominating traffic in networks when schedulers are not used for priority queueing.
- M/D/1 queue type is very good for voice and data multiplexing.

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