Interactive Multi-Media Applications: Quality of Service Guaranteed under Huge Traffic

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ABSTRACT
The internet has brought about revolution in the telecommunication system. The use of computer applications has changed with easiness and low cost. Interactive Multimedia IMM applications such as Voice over Internet Protocol VOIP and video conferencing are being produced. They offer beneficial services to academicians, officers and other users. But these services suffer from performance degradation in the today's high speed Wireless Local Area Network WLAN. However, guaranteed Quality of Service QoS remains the bottleneck in the network which becomes a great challenge to improve. This work reviewed many approaches attempted to improve the QoS for these applications. Here we considered mapping a QoS class parameter i.e Quality of Servive Class Identifier-to-Differentiated Services Code Points QCI/DSCP to the upstream and downstream data flowing in the core of the network that improves its overall performance. This is achieved by mapping QCI to DSCP and then mapping again the QCI/DSCP to the IMM traffic. This gives the QoS bearer packets highest priority and a strong signal. The results obtained after simulation in QualNet shows that our proposed mechanism produced better performance of the network in comparison to the default. This is measured in terms of three network performance metrics (average delay, average jitter and throughput). The overall average end-to-end delay is decreased by 34%, while overall average jitter drops by 24% and the throughput rises slightly by 4.6%.

Keywords
Interactive Multimedia IMM, Voice over Internet Protocol VOIP, Wireless Local Area Network WLAN, Quality of Service QoS, Quality of Service Class Identifier QCI, Differentiated Services Code Points DSCP.

1. INTRODUCTION
The so called IMM was accomplished in 1876 when ring down circuit was used. Two devices were connected with a wire and the transmission is one-way. The advancement in wired networks brings about bi-directional transmission, the genesis circuit switched technology. The original communication system was analog infrastructure which incorporates a lot of noise. later on, the digital communication evolve which uses repeaters to amplify the signal after some certain distance and reduce the noise by polishing the signal to its original condition. For example, the old Public Switched Telephone Network (PSTN) samples the voice stream at 8 kHz and transmits the digitized voice at a rate of 64kbps [1].

A satisfactory QoS has been enjoyed by PSTN. This is due to its dedicated circuits to each call with a constant connection between the two nodes until the call is finished. The modern telecommunication applications such as the VOIP are having it difficult to maintain the same QoS like that of PSTN because its architecture integrates the voice, video and data in the same channel, although TCP/IP is being used to overcome such drawbacks.

VOIP is now widely accepted telecommunication service that uses internet as the transmission medium. It transmits packets or signals for voice communication. VOIP is now used on PCs, hand phones, PDAs etc. The decrease in the cost of internet also decreases the cost of transmitting voice packets. The primary reason for its growth is its low service cost and support under WiFi, because WiFi doesn't require any cellular or other networks, and can be use simply on portable devices.

However, the quality of the VOIP communication remains a bottleneck for its service. The throughput of the VOIP is practically low compared to its wired counterpart and again Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) based medium access mechanism used in the Wi-Fi networks wastes a lot of time in collision avoidance thereby causing more delay and leads to voice quality degradation [1].

Video conferencing is also another multi-media application which requires QoS through its transmission and end to end performance. Users often tell their assessment which is so called QoE but it could be biased. But if presented realistically, it can make the designer to improve on his network. Video QoE effects are caused by the QoS problems such as bandwidth, jitter, delay, loss and throughput. Customers often assess the network based on cost, availability, reliability, usability, and fidelity [1].

Quality of Service (QoS) is an important parameter to be dealt with in any networking and communication system. It has been implemented in the old Public Switched Telephone Network (PSTN) where it is guaranteed because of the dedicated circuit to each call. It is also implemented in the Next Generations Network (NGN) where it is not guaranteed because single circuits are not dedicated to single calls and resources are been shared among users. Many researchers consider the configuration of QoS at layer 3 and the extension to layer 2. However, it is very essential to ensure better QoS for the efficiency of these communicating networks.

Wireless local area network (WLAN) is aimed at providing connectivity over a local and remote area with high speed. The use of WLAN has been increasing, and this does not only apply to computers (laptops and desktops) but also mobile devices, smart phones, game consoles, internet-enabled Television set, notebooks.

The Institute of Electrical and Electronics Engineers (IEEE) 802.11 standards never guarantee an upper limit of packet loss or delay, therefore call jitter and call drop may occur, especially when there is high traffic load that affects voice quality. Even though, the IEEE 802.11e is introduced, which assign some priorities to the IMM traffics; it still suffers from the problems of jitter, large delay and low throughput especially if channel is so busy. 802.11g and 802.11n are the IEEE wireless protocol standard which provides connectivity in computing devices and applications. 802.11g/n protocol infrastructure is widespread and being used for a number of applications and because it is cheap and has integrated chip-sets, it then popularly become the choice for many devices, that includes smart-phones and low-cost consumer devices such as net-books and hand-held game consoles and/or computer peripherals [2].

A very challenging issue regarding the QoS is its design and management. There are two main architectures for QoS, viz; Integrated Services (IntServ) and Differentiated Services (DiffServ). These two architectures follow different philosophy as they follow the name of the topic from different perspective. The protocol associated with these architectures is the Reservation Protocol (RSVP) although it’s complicated in operation [3].

The choice of architecture for the QoS within the IP network follows either of these two. The Intserv architecture is deemed unsalable while the Diffserv has survived for many years and it classifies packets into batch of flow aggregates at the edge of the network instead of per flow state in IntServ. There is always a standard for the QoS from the International Telecommunication Union – Telecommunication Sector (ITU-T) and the European Telecommunications Standards Institute (ETSI) which comes along with a high volume of signaling traffic within IP transport network, but the signaling traffic is negligible and is treated with best effort [4].

This paper is organized as follows, it start with an introduction, followed by the related works. Then the QoS and its parameters followed. The next talks about the performance evaluation, under it there is explanation on the performance metrics, i.e simulation parameters setup and discussion about the simulation results. Finally the paper concluded and references followed.

2. RELATED WORK

Video and voice over Internet Telephony (VVOIP) are the most sensitive traffic (called delay sensitive traffic). The authors in [1] cited [13] as they make a review on the delay and bandwidth utilization for voice traffic over the Internet. They suggest that priority queues could lead to best approach to solve the issue of the bandwidth utilization and delay minimization. The analysis and simulation in [14] showed that when there is high aggregation delay the maximum number of quality calls decreases. 150ms on the end-to-end delay was used as a limit in order to ensure quality of calls and ignoring the effects of loss of packets, whereas the effect of packet loss is defined. Low delay alone is not sufficient to guarantee acceptable voice quality.

There are zero idle slots in MAC layer transmission assumed in a similar work. The scenario is a high traffic, the nodes count the idle slots simultaneously, but, these idle slots are presented at higher load. The AP is got to transmit half of the packets. However, the work is only suitable for single hop and not for multiple hops WLAN because each packet is relayed multiple times by intermediate nodes before reaching the AP. Also the effects of channel noise dejitter buffer and capture effect are ignored [15].

The performance of the VVOIP can be evaluated by the end users subjectively or objectively. Traditionally, voice quality is subjective because it users are allowed to rate the quality of the service they enjoyed. Of course, there can be some elements of biasness from the users’ side. In order to have a justifiable rating, the ITU-T introduced an objective E-Model that can presents the voice quality based on the User Perceived Quality (UPQ). The E-Model works by mapping network impairment factors to psychological satisfaction levels based on assumptions [14].

The authors of [16] proposed a capacity measurement model with interference mapping and carrier sense factor so that the residual capacity can be known. In WLAN nodes can join and leave freely, therefore this behavior of the nodes causes the measurement information to traffic overload, and also limits the usability of the model and in the design phase of the network as well. The work of [14] looked into voice quality assurance and with considering the network and codec parameters for a single hop WLAN, and then extended to multiple hops WLAN by estimating the capacity analytically. The parameters modeled are loss and delay and other real world factors all to estimate the VOIP call capacity.

In [5], multi-media applications are measured in both subjective and objective way. The video is first measured in a subjective way based on the ITU-T Recommendation. A number of users were gathered in a room and asked to evaluate what they saw. For the objective measurement, a software is used which analyzed the strength of the signal called Peak Signal to Noise Ratio (PSNR) and present the result as the evaluation of the video performance.

In PSNR, every pixel error contributes to a decrease in it. Therefore Moving Pictures Quality Metric (MPQM) was proposed which is also an objective method [5]. Video Quality metric (VQM) is another objective video measurement that includes measuring the perceptual effect of video metric which includes blurring, jerky or unnatural, global noise, distortion, motion, block and color. These are combined in a single metric.

Interestingly, the correlation between the VQM and the subjective quality of video assessment has been adopted by ANSI as an objective video quality standard. It is researched that the human vision is specialized in extracting the structural information from the viewing field and not specialized in extracting the errors. Therefore, there is the need to have a structural measurement base on distortion so as to give a better correlation to the subjective impression.

Quality of Experience (QoE) is another important metric that users use to assess the quality of the video based on their perceived experience [5]. QoE is the user's satisfaction of the video. However, it is often so difficult to represent the features of the video from the bandwidth and latency time especially in an integrated network environment. The parameter QoE evaluation can be modeled and a fundamental relation between the QoE and QoS can be presented and demonstrated. However, [5] found it difficult to measure the quality of streaming services of IMM traffics from bandwidth and delay time especially in an integrated network environment with the use of QoE evaluation model because a feedback is needed from the other endpoint to calculate the QoE. Therefore, a video QoE assessment model that can assess the quality of the video by users experience is needed for network and service providers.

Most network operators do not rely on the subjective technique evaluation. This leads the authors in [12] to develop an objective technique that estimates the Quality of Experience (QoE) of the VVOIP. Video frame freezing and voice dropouts remains the bottleneck in [12] and are dealt with by the new offline automated model which can measure the network condition in terms of network factors such as delay, loss, bandwidth and jitter. Whereas the VVOIP is estimated in terms of "Good", "Acceptable", or "Poor". However, the authors here focused on the H.263 video codec at 768 Kbps dialing speed whereas other video codec higher dialing speed such as H.264 and MPEG-2 can still be considered. Also, the authors only considered network parameters viz., delay, loss, bandwidth and jitter without considering application-level parameters [1].

The parameters of QoS that shows an effect on the QoE are: packet loss (L), packet delay (D), jitter (J), and bandwidth (B). These parameters of QoS are recommended by some standards such as the ITU-T and IETF (Internet Engineering Task Force) as network-related quality elements.

3. DELAY SENSITIVE APPLICATIONS

3.1 Voice over Internet Protocol (VoIP)

Voice over Internet Protocol (VOIP) is an IMM application that evolves over the century. VOIP is a highly featured application which attracted many telecommunication companies and business firms. VOIP deals with audio conversation between two or more parties putting the internet as the medium. Many telecommunication companies are into the business of providing the VOIP application and the network it can run on. The major problem associated with it is the QoS, which conversation may be unclear or delayed so much that user may not enjoy the communication. VOIP requires no delay of transmission but requires regular and low jitter. A user will be unhappy if he waits for a minute or so before getting the reply from his partner. VOIP can only tolerate small amount of packet loss. However, this QoS is guaranteed in the old Public Switch Telephone Network (PSTN). Therefore, service providers and network designers are trying to bring back the same QoS as in PSTN.

Fig. 1: VoIP

VOIP is so beneficial to its users because of the following reasons.

1. Cost; calls via VOIP are relatively cheap regardless of the distance. It uses the internet as the transmission medium therefore no cabling cost is incurred. Voice conversations and video conferencing can all be done in a cheap rate with VOIP.
2. Integrated Services; there are more number of calls on the traditional network which VoIP network integrates with the traditional PSTN system.
3. Highly scalable and easy updates; unlike the traditional system where the increase in the number of ports add cost, VOIP maintains its original cost since it is a software based despite the number of connections added.
4. Disaster Recovery; VOIP is much more concerned with the failure in the call path since many routers and switches are interconnected together. It eliminates the failures and always provides alternatives.
5. Fax over IP; the integration of voice to the PSTN is up to fax and other traditional applications. Voice services such as caller-ID, broadcast messaging and call forwarding can be updated easily and becomes simpler to maintain.
6. Security; this is a feature called Virtual Private Network (VPN). It is to allocate a certain amount of bandwidth to the internet whereby information is encrypted to prevent public access [1].

3.2 Video

Video is another multi-media application which provides the user with a variety of services such as the video conferencing, video play back and video on-demand and so on. A continuous streaming of the video is highly required for the video application. Jitter and loss disturbs the video so much while delay does not affect the playing of the video. If jitter is irregular, the video may appear blurred but if loss is high the video may be distorted. And if both jitter and loss are high, the video will be unclear, blurred and distorted [5]. The delay is not so much effective in the video because its frames are played sequentially while some of the frames may be in motion while some may not play so fast. Therefore, video also requires a guaranteed QoS as that of the PSTN for its streaming [6].

4. QUALITY OF SERVICE

QoS generally refers to the applications' quality (e.g voice and video) as perceived by the user. That is, the responsiveness of interactive voice, the presentation quality of the video. While from the network perspective, QoS may refer to the service quality itself or the service level that the network offers to multi-media applications and or users in terms of network QoS parameters.

Quality of Service is an important metric when quality is needed in the IMM applications. However it has become a bottleneck for the IMM applications. User always becomes unhappy when the quality of their network is poor especially when he pays more and receives less. It is very important to have continuous communication when conversation starts. However, during a telephone conversation, a user cannot wait for one minute to receive response from his counterpart.

Likewise, the video movie will not be enjoyable if the streaming is played in breaks and slow. This attracts both the service providers, network designers and researchers to pay attention and try to look for ways of improving the QoS for the IMM applications in order to have an end user satisfaction. Academicians also place their interest of having collaborative learning between colleges and universities. This is why the QoS for IMM applications becomes a focus area for many researchers.

In general, to have improvement in the quality of applications and network, QoS has to be enabled in the system and extended where necessary (as extending from layer 3 to layer 2) [7]. This gives an improvement in the network and application's quality although different researchers follow different technique to achieve this.

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When talking about QoS, some parameters need to be addressed which includes: bandwidth, packet loss, latency or delay, jitter, some policies introduced by some devices like firewall and Network Address Translation (NAT), and most importantly the throughput [2].

4.1 QoS Parameters in IMM Application

Whenever guarantee, quality and efficiency are needed in the IMM applications, some issues (called the QoS parameters) need to be addressed. The so called QoS parameters are always used to evaluate the performance of any network. The most common QoS parameters includes; bandwidth, delay or latency, jitter, packet loss, throughput and some other protocols.

I. Bandwidth; this is the gateway for the packets; it must be enough for all the packets to get through unimpeded. It is symmetric in the sense that both ends requires to transmit and receive at the same link speed. The bandwidth should be able to accommodate the number of packets being sent at a moment [8].

II. Packet loss; this refers to the number of packets that fails along the transmission channel so they do not arrive at the destination. This is due to insufficient bandwidth, transmission errors or high latency. In order to achieve an understandable speech, the packet loss should always be between 0%-0.5% for good quality speech and conferencing, while 0.5%-1.5% is still acceptable [9]. The packet loss parameter is always opting to be minimized as much as possible in order to obtain optimum throughput.

III. Delay; when packets are sent, they are sent through encoding and decoding process, so latency is the amount of time a packet travels from the origin node to the destination node. It is considered that the maximum acceptable level of delay should be not more than 400ms as a standard set by the ITU-T and other standard organizations. Delay or latency is also opting to be reduced at all times [1].

IV. Jitter; this is the average of the time variation between the received packets. It is the difference in the arriving time of two consecutive packets. i.e t1 – t0, t3 – t2 and so on. Multi-media applications require packets to be delivered in a regular interval in order to have a good quality and continuous speech. However, when packets do not arrive in a regular interval, it seriously affects the quality of the IMM applications. The value for the jitter should be less than 1ms [10].

V. Firewalls and Network Address Translation (NAT) controller policies; these devices and others introduced some policies to hide or protect network elements from the wider internet. These are mostly used in a subnet of a bigger network where a node needs to ask for permission before communicating to other nodes in the network [8].

5. PERFORMANCE EVALUATION

This section describes the performance matrices (the network parameters) used to evaluate the performance of our network. The simulation scenarios are discussed in detail and also the simulation results are presented.

5.1 Performance Metrics

A quantitative measure is done on three performance metrics viz; averge delay, average jitter and throughput. This is to analyze and evaluate the performance of the network before and after mapping some differentiated priorities. We looked at the effects of increasing traffic when the transmission rate is increased by reducing the interval rate.

Since the introduction of WLAN, guarantee QoS for huge traffic has become a bottleneck for the network. IMM applications (i.e voice and video conferencing) can only tolerate maximum time delay of packets up to 400ms (for one way end-to-end delay) and less than 1ms for jitter for good quality speech and conferencing [1, 10, 11], while packet loss should be between 0% - 0.5% for better quality speech and conferencing, while 0.5% - 1.5% can be acceptable as well, but no more than 1.5% [1, 9]. This works then considers the maximum acceptable end-to-end delay to be 50ms.

5.2 Simulation Parameters Setup

In this simulation work, we assume that all nodes have the required hardware, software and codecs. The software supports multi-party conferencing using QualNet network simulator. We chose some parameters that favor IMM traffic. In the Physical layer (PHY) the radio type chosen is IEEE 802.11b, while at the MAC layer is IEEE 802.11e standard, Medium Access Control (MAC) protocol and Distributed coordination function (DCF) is employed as well because it is mainly used for classifying and differentiating IMM traffic. The type of scheduler and queue in the network layer is diffserve which is the main architecture for classifying and identifying IMM traffic, where the frame scheduler is First-In-First-Out (FIFO).

We employ the Ad hoc On Demand Distance Vector (AODV) routing protocol so as to have a very good path for the IMM traffics, because AODV has less overhead by carrying only the destination address unlike other routing protocols such as Dynamic Source Routing (DSR) that carries the full routing information. Also AODV is more adaptable to highly dynamic networks [9]. The design area is restricted to 1000m2 X 1000m2. The scenario contains one AP and twenty four (24) other stations all of which are connected in a range not more than 30 meters from the AP. The applications used are voice, video, best effort and background.

The voice and video traffic are of 512bytes packet size but the size was increased to 1024 & 2048 bytes in the subsequent simulations. The interval is set as 50ms which sends 20 packets per second, and this interval is reduced to 20ms in the subsequent simulations in order to increase the traffic, and then observe the performance of the network scenario under this situation. Then the simulation time is just 100sec before it is later increased to 200sec, 300sec, 400sec and 500sec, all for the purpose of observing the behavior of the network under different simulation times. The pause time is 0 so that nodes will continue to be motion. All these nodes are allowed to move from the random waypoint mobility model. The remaining parameters used in the simulation are summarized in the table below.

Table 1: Simulation Parameters.

Parameter	Value
Area	1000m2 * 1000m2
No. of nodes	25
Routing Protocols	AODV
Types of traffic used	Video, Voice, BE & BK
Packet size	512 bytes
Simulation Time	100 sec
Traffic connection	8
Scheduler and Queue	Diffserv
Frame Scheduler	FIFO
Priority	QCI/DSCP
Interval	50ms
Bandwidth	11Mbps
Radio Type (PHY)	802.11b
MAC Protocol	802.11e
Mobility Model	Random waypoint

5.3 Simulation Results

First of all, the scenario is run without any priority being assigned to all the traffics generated (i.e voice, video, background and best effort), then Quality of Service Class Identifier / Differentiated Services Code Points (QCI/DSCP) is mapped to the voice and video traffic. This mapping proves that the network performs better after mapping QCI/DSCP to the IMM applications. The mapping and priority assigned increases the throughput obtained, and decreases the delay and jitter.

5.3.1 Average Delay

The average end-to-end delay is the amount of time packets are delayed before reaching to its destination in the network path. The total delay is taken as the average of all the delays on the individual nodes in the network scenario.

It is observed that before the mapping, an average value for end-to-end delay is obtained but after QCI/DSCP is mapped to the delay sensitive traffic (i.e voice and video) at a data rate of 512 kbps, the overall average end-to-end delay drops from 47ms to 31ms thereby giving a percentage drop of 34%. This shows that mapping QCI/DSCP reduces the overall delay.

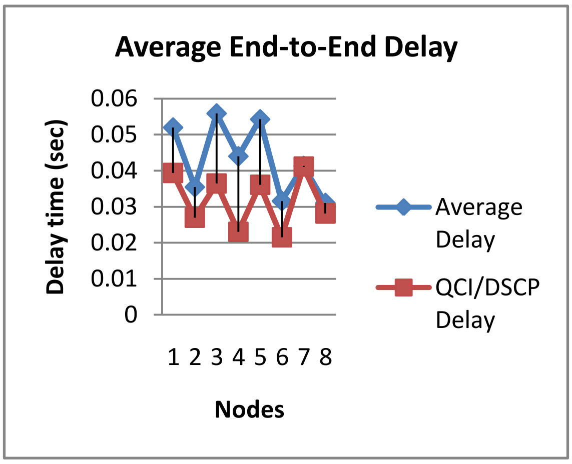

Fig. 2: Average Delay

5.3.2 Average Jitter

This is the average of the variations in the arrival rate of packets from the source to its destination. Arriving packets are expected to maintain the same interval in order to have a good quality and continuous speech. The total jitter is the overall average of the individual jitter nodes in the network scenario.
By looking at the jitter graph, it shows two lines of curves. One is the jitter values before QCI/DSCP is mapped and the other is after QCI/DSCP is mapped. A drastic decrease in the jitter is observed. i.e the first jitter value gives 0.007847sec (7.8ms) while the second jitter after mapping QCI/DSCP gives 0.00599 (5.9ms) all at a data rate of 512kbps. Thereby giving jitter decrease of 24%.

This is a good improvement in having a good quality speech and also it increases the throughput obtained.

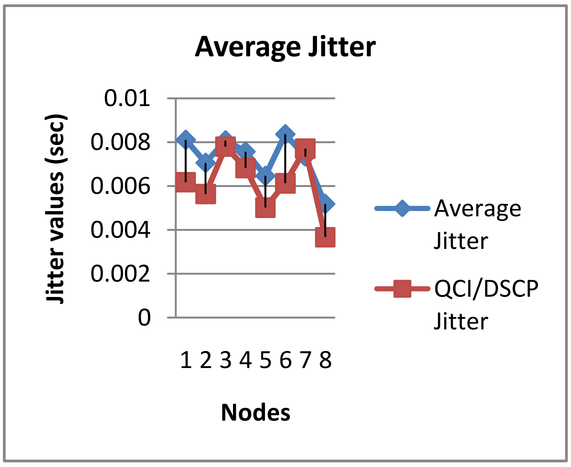

Fig. 3: Average Jitter

5.3.3 Average Throughput

This is simply the output of the packets in bytes. Throughput is the most critical performance metric. Ordinarily, throughput does not always equal the amount of data being sent. It is difficult to achieve equal throughput in any network but our aim here is to see that the throughput for IMM traffic is obtained as much as possible.

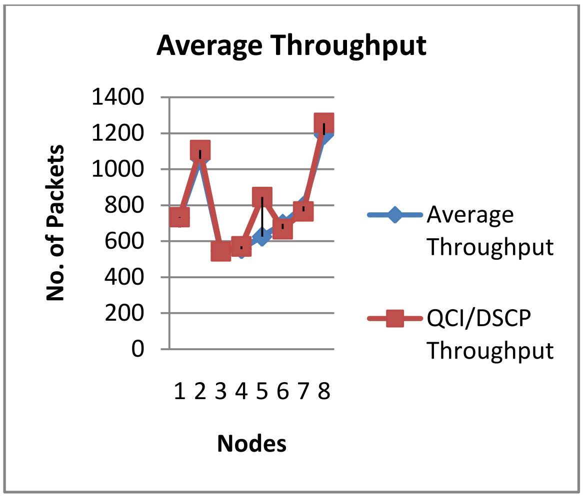

Fig 4: Average Throughput

The result shows a slight increase in the amount of throughput obtained after QCI/DSCP is mapped to delay sensitive traffic (video and voice). The values of the throughput are presented in number of packets for simplicity. At first, we obtained an average throughput value of 773.5 packets. After mapping, the average value was raised to 810.8 packets. The network experiences a slight increase of 4.6% after mapping QCI/DSCP.

This improvement is achieved by mapping QCI/DSCP to the delay sensitive traffic. Therefore, the QoS class parameter (QCI/DSCP) plays a vital role for increasing the throughput of the network scenario.

5.4 High Traffic Load Generation

Now, we generate more traffic in the scenario in order to see what effect they have to the overall network. We reduce the increase the simulation time from 100s to 200s, 300s, 400s, 500s and 600s. This allows more number of packets to be sent over the network i.e 4000, 6000, 8000, 10000 & 12000 every second instead of 2,000 packets in the initial setup. This traffic kept increasing until a saturation point for the network is reached. The rate here is increased by 100%, 200%, 300%, 400%, & 500% respectively which is enough to produce a change in the behavior of the network. Under this condition, we looked at the change produced in delay, jitter and throughput. Each metric is computed as the average sum of individual values obtained from its individual nodes.

We run our scenario each time we increase the simulation time, we observe the results produced for the delay, jitter and throughput. We compare the results between the default huge traffic generation scenario and when QCI/DSCP is mapped to the huge traffic generation scenario.

We can see that as the load increases, it causes more delay and variable jitter. This is a bandwidth effect that works on the principle of blocking probability whereby the bandwidth only accommodates up to its capacity and tries to block the remaining traffic by putting them in a buffer until it gets the free space that it can transmit the remaining. For this reason, the remaining traffic waited for the initial ones, this is why their end-to-end delay and jitter has to increase. Therefore we say that as the traffic continues to increase, the average end-to-end delay tends to increase as well and the average jitter tends to vary, while the throughput tends to drop.

But with our proposed mechanism, we can eliminate high delay, irregular jitter and large packets drop. The graphs below illustrated how our proposed mechanism improves the overall network performance according to these network parameters.

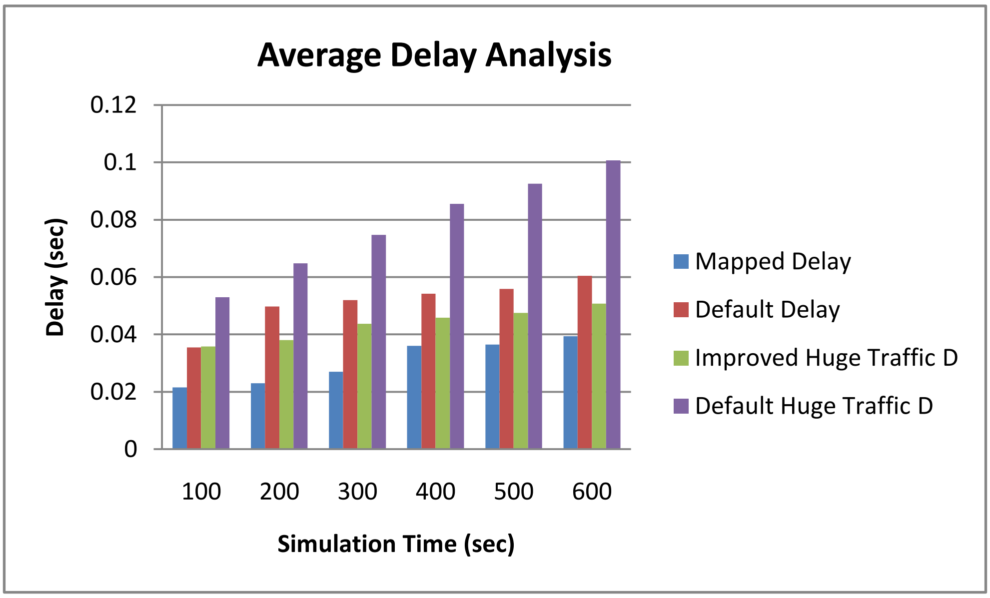

Fig. 5: Average Delay Analysis

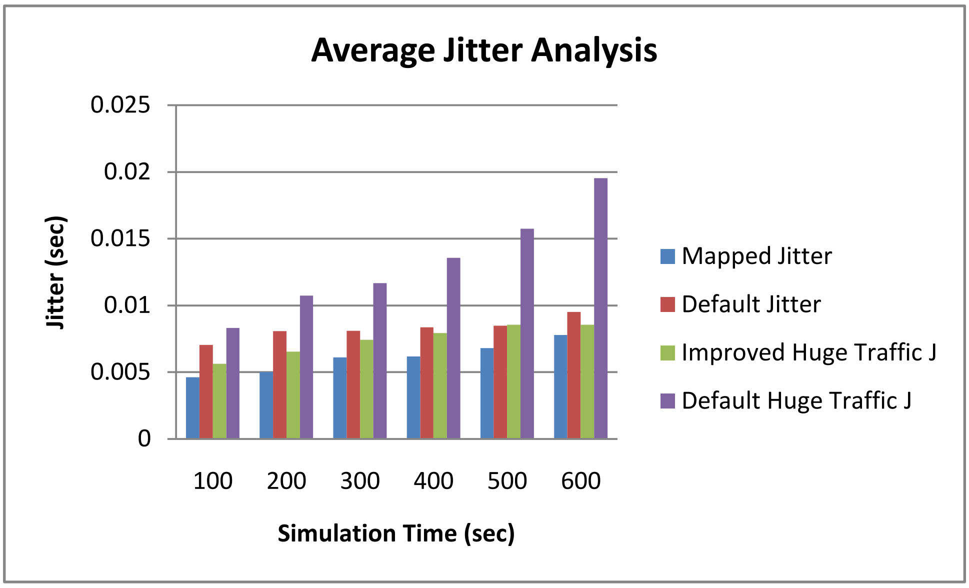

Fig. 6: Average Jitter Analysis

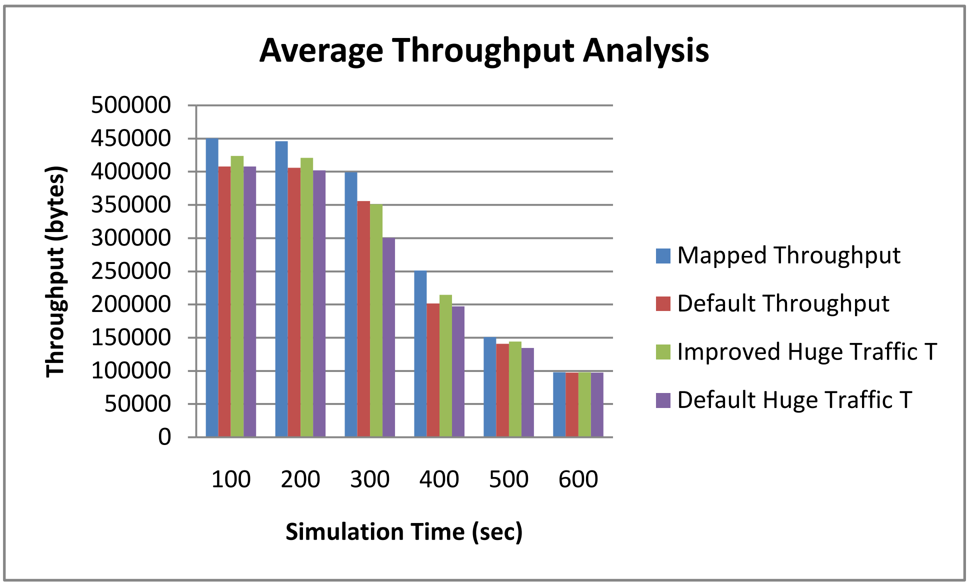

Fig 7: Average Throughput Analysis

6. CONCLUSION

We have implemented our proposed mechanism which maps a QCI/DSCP QoS class parameter to the IMM traffics. QCI is mapped to DSCP, which becomes QCI/DSCP where each of it is a QoS giver and then maps to the whole parameter to IMM traffics. We run a simulation scenario and evaluated the overall network performance based on three performance metrics, viz; end-to-end delay, jitter and throughput. Each performance metric gives a better result with our proposed technique of mapping QCI/DSCP to the IMM traffic flow. Although the default scenario (without any mapping) also shows an acceptable result but is not as good as our methodology.

We then try to compare the results obtained between our proposed technique, the default and when huge traffic is generated. In all the three states, our methodology achieves the best result by obtaining less values for overall delay and jitter while obtaining the highest values for throughput.

However, let's keep in mind that these results were obtained on a simulated network (designed network) where lots of parameters were altered and huge traffic was generated into the fast ethernet interfaces so as to flow through the serial link, thus forcing the router to apply the prioritization. Under higher data load, the values of the delay and jitter tend to rise and throughput tends to decrease.

A future direction of this research could be looking at the application-level parameters unlike the network parameters considered only here. Again, Other QoS class parameters could also be mapped such as Maximum Bit Rate (MBR), Allocation and Retention Priority (ARP) and others.

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