VOL. 9, NO. 12, DECEMBER 2014 ISSN 1819-6608
ARPN Journal of Engineering and Applied Sciences

www.arpnjournals.com

AN IMPROVED QOS IN THE ARCHITECTURE, MODEL AND HUGE TRAFFIC OF MULTI-MEDIA APPLICATIONS UNDER HIGH SPEED WIRELESS CAMPUS NETWORK

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ABSTRACT
Multi-media applications are in high demand by many users. Many telecommunication companies are into the business of providing these applications such as the Voice over Internet Protocol (VOIP) and video conferencing. Users often become unsatisfied with the services they receive from their service providers. This is as a result of lack of guaranteed QoS in today's high speed wireless campus network (WCN). The internet is the main platform on which these applications run on and therefore all credits and blame goes to the designers of the internet. Several researchers have worked towards improving the QoS of these applications and they have contributed immensely to it. However, this work reviewed previous works and comes up with a new methodology that ensures QoS of multi-media applications in the high speed Wireless Local Area Network (WLAN). The design and simulation shows a great improvement in the overall network performance. This is measured in terms of three network parameters, viz: average end-to-end delay, average jitter and throughput. We introduce a mapping technique with which we assign the highest of all priorities to the multi-media traffic. Firstly there is a mapping between the Quality of Service Class Identifier (QCI) to the Differentiated Service Code Points (DSCP), where each of it is a QoS giver. Then, the QCI/DSCP is mapped to our multi-media traffics. Our design has already excelled the default design by achieving lower end-to-end delay, regular jitter and higher throughput. We then incorporate a huge traffic in our scenario and measured the performance again; our new mechanism achieves the best result compared to all the similar scenarios with huge traffic. The overall performance of the network is improved with no less than 50%.

Keywords: quality of service (QoS), interactive multimedia (IMM), wireless campus network (WCN), quality of service class Identifier (QCI) and differentiated services code points (DSCP).

1. INTRODUCTION
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 and 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. A satisfactory QoS is enjoyed by PSTN 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 whose effort is appreciated.

Multi-media applications are now widely used in educational sectors for the purpose of conversation and conferencing within and among users. In the field of e-Learning, several interactive web-based learning systems are developed by many institutions of learning, where they integrate rich media streaming which may compromise network performance. Typical example of e-Learning system involves a combination of text, images, audio and video, and their quality of service is based on the combination of all of these rather than any individual component and should be considered mandatory for the successful usage of the e-learning applications. The adaptation of QoS in real time applications like video conferencing ensures quality conferencing and improves scalability. Because QoS technologies are still object of intensive development and fine tuning, modeling and simulations are highly required in this filed.

The so called IMM is accomplished in 1876 when the devices are wired and analog. It is a one way communication. Later on, the system is digitized and make the channel becomes two ways. The former devices are only meant for conversation like the PSTN but today's multi-media applications integrates a lot of services in it perhaps this is the genesis of having guaranteed QoS in the applications. The VOIP applications is one of the multi-media applications which its popularity is increasing everywhere. It is an application for voice conversation, video call and chat. The reasons for its popularity are: cost, integrated services, highly scalable and easy updates, disaster recovery, security and fax over IP.

Video conferencing is another multi-media application which requires QoS in its transmission and end to end performance. Users often tell their assessment which is so called QoE but this 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

VOL. 9, NO. 12, DECEMBER 2014 ISSN 1819-6608
ARPN Journal of Engineering and Applied Sciences

the following; cost, availability, reliability, usability, and fidelity (Singh *et al*., 2014).

(i) Cost; since connection to internet is cheap, this also makes using the multi-media applications also cheap. For example 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.
(ii) Availability: users always want to get access to their applications any time they need so. As long as there is internet connectivity, multi-media applications are always ready for use. However, usage may only be limited to insufficient funds to be charged by the network.
(iii) Reliability: users always have hope on such applications for performing the intended purpose. This includes have all the calls reached unimpeded.
(iv) Fidelity: both network designers and service providers measure the IMM applications based on its correctness in delivering service.

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 this classifies the IMM traffic with priority, 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.

This paper is organized as follows: it starts with an abstract, then introduction and followed by a comprehensive and critical literature review. Then it gives an overview of the QoS and its Architecture. The next section is the Design of the Proposed Mechanism. The subsequent section is the Enhanced Prioritized Traffic Model. The Experiment section follows as well as Results and Performance Evaluation in this order. Lastly the paper ends up with a conclusion.

2. LITERATURE REVIEW

VOIP application is reviewed, evaluated and improved by (Singh *et al*., 2014). The authors evaluated the network thoroughly based on the network factors, viz: networking conditions, coding processes, speech content and error correction schemes. Their objective model developed is more realistic in assessing the network. However, it is found that VoIP throughput is low compared to its wired counterpart and the CSMA/CA mechanism wastes a lot of time causing more delay and leads to voice quality degradation. (Sarkar et al., 2012) study the performance of nodes in voice conferencing through simulation of voice traffics. Their work proved better performance with DSR over AODV in terms of coverage area and their result achieves lower delay and loss. However, the authors only looked at routing protocols whereas there are many simulation parameters in the simulation environment that could be considered.

Video and voice over Internet Telephony (VVOIP) are the most sensitive traffic (called delay sensitive traffic). The performance of the VVOIP can be evaluated by the end users subjectively or a by using a system objectively. Most network operators do not rely on the subjective technique evaluation. This leads the authors in (Calyam, Ekici, Lee, Haffner, and Howes, 2007) 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 their work and are dealt with by the new offline automated model which can measure the network conditions in terms of network factors such as delay, loss, bandwidth and jitter. Whereas the VVOIP is estimated in terms of “Good”, “Acceptable”, or “Poor” (Calyam, Ekici, Lee, Haffner, and Howes, 2007).

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 (Singh *et al*., 2014).

QoE is another parameter that could be evaluated to enhance the performance of multi-media applications. (Kim and Choi, 2013) evaluate the QoE of streaming video through an objective model. They are successful in having a better correlation between the QoS and QoE of the video. Their model provides reliable multi-media services. This can also help network operators to prevent the unnecessary investment for enlargement, maintenance and repair of the network. However, the QoE of streaming services becomes difficult to measure using this model because a feedback is needed from the other end before a conclusive result could be reached.

An attempt by (Ahmed and Sarma, 2013) was made to improve the QoS of the IEEE 802.11e standard and it is somehow successful. The QoS here is improved by having some fair and dynamic Access Point (AP) and Stations (STA). AP is responsible for connecting stations while IEEE 802.11e WLAN was simulated with single Quality of service AP (QAP). The multi-media traffic under Enhanced Distributed Channel Access (EDCA) received differentiated service for priority traffic as

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compared to the traffic under Differentiated Code Points (DCP). The results show that adjusting the Contention Windows (CWs) will slightly improve the QoS and QoS management can lead to efficient utilization of radio resources.

However, to have an overall improvement on the QoS, Admission Controller needs to be flexible to maintain QoS and it is seen that When traffic increases, multi-media traffic begin to loose packets which affects their QoS as well as QoS for the best effort traffic. If this continues, there will reach a state when there will be no multi-media traffic present while best effort traffic has a lower channel priority access. Another thing is to resolve Quality of Service Access Point (QAP) load in balance.

(Romdhani and Mohamed, n.d 2013.) proposed a technique of distributed layered mapping in order to improve video QoS by mapping video layers over EDCA by an algorithm. The algorithm helps in improving the performance, as well as decreasing packet lost and proves that canonical mapping is better. Having some simulations, it proves the accuracy of the optimized mapping performance. However, this algorithm then has to be implemented with considering the channel sensitivity, network resources and protocol optimization feedback information, and after that only then can be implemented in the university wireless network.

There is an interesting work by (Prasad and Kannan, 2013) considering the QoS for the future WLAN. This work introduces three influencing factors:

a. The Application Specific Threshold; this ensures that the Mobile Host (MH) maintains a threshold value that corresponds to the current application so that unnecessary handoffs would be avoided especially when the MH is far apart from the AP.
b. Decision Maker Function; this function is implemented in the application layer which ensures that the handoff procedure does not go beyond the maximum delay time of the sensitive applications, otherwise more handoffs and scanning will be generated.
c. Load balancing; the MH should neither be more than the APs that will be associated with nor will the AP be overloaded. If any of this happens, the QoS of the MH get reduced. Therefore the load at the target is considered before associating with it. Hence the implementation of Innovative handoff mechanism (IHM) aids in decreasing the overall delay greatly while improving the network throughput and performance (Prasad and Kannan, 2013).

An algorithm was introduced by (Singhrova and Prakash, 2012) to improve the QoS by reducing vertical handoffs. They explained three approaches they follow:

A. Classical; it's very basic and simple and evaluation of handoff decision is based on Reduced Signal Strength (RSS).
B. Fuzzy; handoff decision is based on few parameters, high ping-pong, low throughput and minimum complexity.
C. The proposed; this has 6 input parameters, reduces ping-pong, improves End-point Service Availability (ESA) and throughput. It achieved its result by varying the simulation time which reduces the vertical handoffs for the proposed Vertical Handoff Decision Algorithm (VHDA). Also the reduced number of unnecessary vertical handoffs save resources and time and reduce the number of calls being dropped.

QoS is improved in the performance of multi-media applications by adapting QoS in the multi-media application based on Class Based Weighted Fair Queue - Low Latency Queue (CBWFQ-LLQ) which classifies and prioritizes the multi-media traffic. This is done with OPNET IT Guru simulator. It shows that the applications of CBWFQ-LLQ for video traffic gives better performance to video traffics only, while the applications of CBWFQ-LLQ for voice traffic gives better performance to all types of traffic (Badr and Darwesh, 2011). The simulation involves three scenarios:

a) The first experiment was conducted on a First Come First Serve (FIFO) without enabling any QoS technique. This gives privileges to non-real time applications e.g Hyper Text Transfer Protocol (HTTP), File Transfer Protocol (FTP) etc as they travel conveniently fast while having negative effect on the multi-media applications i.e. voice and video as they are queued long and began to drop packets.
b) The second scenario enables QoS at the routers using CBWFQ-LLQ for video traffics. This enables strict priority, low jitter for delay sensitive applications. The result favors video traffic only while affecting other traffics including voice.
c) The third is for voice traffic, but it didn't prevent any traffic from flowing, instead it improves the overall performance of the network.

However, this work prioritizes voice and video traffics, there is need to test this model on other traffics required in collaborative systems such as e-learning system. Also QoS has to be extended to layer 2 to achieve even better quality in the performance of multi-media applications. Again applying congestion management techniques will improve its overall performance (Badr and Darwesh, 2011).

(Barakovic *et al*., 2010) tries to configure priority level for signaling service class and simulate the scenario. Some few achievements were obtained as follows; the neighboring domains of Layer-3 are able to recognize the original Differentiated Services Code Points (DSCP) encoding before remarking it. It is observed that QoS objectives for signaling service class is very essential for the efficiency of standardized Next Generations Network (NGN) approaches for QoS control.

However, a lot of recommended future work remains for this research. Viz; Layer-2 QoS mechanisms are not visible at all to the neighboring networks. Uncoordinated suboptimal packet treatment occurs when local policies are enforced and options are remarked. Another thing is that other networks may not be able to refer back to the source behavior encoding and adopt their re-marking accordingly. And lastly but not the least, there are no cross- layer QoS mappings standardized between, e.g., Ethernet Priority Code Point (PCP) marking and IP DSCP marking (Barakovic *et al*., 2010).

Interestingly, (Bouras *et al*., 2008) published a paper in which he solves one the limitation of (Barakovic *et al*., 2010) which applies QoS at layer-2. The QoS at layer-2 is derived from layer-3 of the network application layer protocol. Premium traffic streams benefitted from the application of QoS at L-2. Routers' configuration can be altered to provide marked flows traffic and Switches' queues can be set to provide QoS. The activation of QoS at layer-2 also benefits the overall result that is produced by only Layer-3 QoS in Greek Research and Educational Network (GRNET). Extensive experiments in laboratory and production environments proved the proper configuration and operation of the Layer 2 QoS service (Bouras *et al*., 2008).

So many factors could guarantee the QoS of multi-media applications. In the work of (Bin, Latif, Rashid and Alam, 2007), jitter is considered as the metric that could improve the QoS of IMM applications. This work considers varying the Super Frame (SF) and Contention Free Period (CFP) so as to attain lower jitter values. The first scenario of the simulation was SF versus CFP where SF was varied from 10ms to 60ms while CFP was kept constant. The result shows an increase in the jitter value by 0.1ms while the SF is between 40ms to 50ms. The second scenario varies the CFP while SF is kept constant; there is drastic drop in the value of jitter when the CFP is at 50% out of the range of 30% to 50% (Bin *et al*., 2007).

QoS is also enhanced with regard to ensuring fair channel utilization in terms of time among wireless stations. A Fair QoS Agent (FQA) algorithm is proposed which is simulated using NS-2 simulator (Park, Kim, Choi, and So, 2007). The simulation setup consists of N wired nodes, one router, one AP, and N wireless stations. The wired nodes are connected to the router with a link, its capacity and propagation delay is 10 Mb/s and 30ms, respectively. The link capacity between AP and router is 100Mbs. Three types of traffic are then generated; voice, video and data. The data traffic is generated by a greedy File Transmission Protocol (FTP) application over a Transmission Control Protocol (TCP) connection. The video traffic is generated by a Constant Bit Rate (CBR) application over an Uninterrupted Data Protocol (UDP) connection while the voice traffic is by on/off traffic. Packets for voice are at a constant rate of 64kb/s which is the rate for VOIP. The average burst time and idle time are both set to 3sec. The packet sizes for the data, video, and voice traffic are set to 1000 bytes, 1000 bytes and 100 bytes, respectively. For each traffic type, average values of throughput (TH [bit/s]), packet loss rate (L [%]), end-to-end delay (D [s]), and jitter (J [s]) are taken as performance indices. J is defined as the standard deviation of delay (Park *et al*., 2007).

However, the model presented by (Park *et al*., 2007) is flexible, but a common drawback is that they need to be implemented on STAs as it was at the AP in order to regulate uplink traffic. This is noticeably inconvenient in practical deployments with legacy stations(Cacace, Iannello, Vellucci, and Vollero, n.d.). Then the proposed scheduling schemes in (Ferng and Liau, 2009) does not only provide better QoS but also excel in achieving better flow-level and station-level fairness, while FQA provides per-station fairness with per-class QoS only.

(Jesús A Pérez *et al*., 2006) considers voice and video as the most sensitive type of traffic and prioritizes it through a model. This model collects all the incoming traffic, classifies it according to the level of their sensitivity, then gives voice and video traffic the highest priority and other traffic which are delay tolerant the least priority. The total bandwidth is then allocated to the prioritized traffic allowed to be sent directly to the destination without passing through the congestion avoidance technique. While the remaining traffics follows and share the bandwidth according to some specific set of policies.

A network infrastructure is employed and a default configuration of routers, the traffic is allowed to flow while opening a videoconferencing between two end points. The performance and the quality of the data is observed and the same experiment is repeated with prioritized traffic flowing. It is clearly noticed that the quality and performance of the voice and video traffic is increased by 70-130% when it is being prioritize. This prioritization is done in order to make sure that the most sensitive data keep flowing at all times. However, this model cannot control traffic from outside sources. Traffics from outside includes those generated by an unknown user (e.g. a call from an unknown caller). Despite the improvement brought by this model, it is still important to come up with a rich model that can control traffics from outside sources.

In a similar work from the same authors, they presented a prioritization scheme similar to that in which can be represented as a general model which the traffic follow in order to get prioritized. As the traffic flows through the scheme, firstly it is marked according to our needs with special conditions (such as maximum delay) to be met while considering delay sensitive applications. All traffics are classified according to its own class. Special treatment or strict priority is given to the delay sensitive data i.e. voice and video traffics as the most important traffics using LLQ so that they will keep on flowing all the time i.e. goes straight to the output without passing through the Congestion Avoidance Mechanism (CAM) while the other traffics are considered as delay tolerant and shares the remaining bandwidth according to the policies configured for each data class. Once all conditions are met and all policies are applied, the marked and prioritized traffic being sent, guarantees end-to-end QoS (Jesús Arturo Pérez *et al*., 2006). However, as the QoS is so much improved in the outgoing traffic, it is equally important to let the QoS enabled as well in the incoming traffic and to be able to control the traffic from outside source.

However, the proposed 802.11e comes with so many challenges which include the handling time varying network conditions, the adapting to varying application profiles, and the managing link layer resources. 802.11e gets to its limit when it supports inaccurate flow reservation, varying flow requirements, and congestion in contention- based access. As a result, algorithm was developed that will dynamically associate traffic flows appropriately to the two medium access modes Polling Based (PB) and Contention-Based (CB) and adjusting the duration of access in each mode. It improves by decreasing the delay and increasing the throughput of the QoS of the multimedia applications (Ramos, Panigrahi, and Dey, 2006).

Looking far back at the work of (Iera, Molinaro, Ruggeri, Tripodi, and Mediterranea, 2005), the effect of dynamic traffic priority assignment to contention-based channel access mechanism of IEEE 802.11 in single and multi-hop WLANs is analyzed. A dynamic priority assignment algorithm is proposed that is able to automatically allocate access priorities to applications traffic, which match users' requirement. The proposed method works properly in the IEEE 802.11e.

However, IEEE 802.11e still suffers from performance degradation. This is due to the increased number of users and the remote land area is now getting connected to the WLAN. This makes the network suffer because of high utilization; it is requested to give more than what it can deliver.

A management tool for the service of QoS is designed and implemented in the high speed backbone network of WLAN by (Ramos, Panigrahi, and Dey, 2005). The supported QoS services include the IP Premium that tries to eliminate packet loss and maintain minimum delay and jitter which is less than the Best Effort service. There is also a management tool for the service that allows managing QoS requests. (Ramos *et al*., 2005) tries to enable comprehensive QoS in the 802.11e and to understand the 802.11e, the application and the higher layer QoS schemes. Also to create a new Highest Common Factor controller framework that can support the approaches presented.

A research survey was carried out by (H. U. A. Z. Hu, Ing, Hlamtac, and Rabhakaran, 2004) in which QoS schemes in the upper and lower layers of the network, all of the MAC and physical layers are surveyed. The survey is classified into three: i) Link adaptation in the physical layer. ii) Channel access coordination in the MAC layer. iii) Admission control strategies in MAC and higher layers. When issues are combined i.e. admission control and resource allocation, differentiating flows and coordinating the order of channel access extend the concepts of QoS guarantee in 802.11 and make it better suited for today's network.

3. QUALITY OF SERVICE AND ITS ARCHITECTURE

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. From this point of view, practices has to foster in the Internet using MMC's applications in e-Learning services, the need to integrates public domain multicast applications for synchronous media communication arises and a system was proposed for that, it's being supervised by a middleware based QoS management framework, intending to preserve the QoS of critical parameters for e-Learning applications (Badr and Darwesh, 2011).

While from the network perspective, QoS may refer to the service quality itself or the service level that the network offers to applications and or users in terms of network QoS parameters. 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 the throughput. These are called the network parameters. Viz:

a) **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 (Jesus Arturo Perez *et al*. 2006).
b) **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 (Sarkar et al 2012). The packet loss parameter is always opting to be minimized as much as possible in order to obtain optimum throughput.
c) **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 (Singh *et al*., 2014).
d) **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 (Elahi. E, 2012).

e) **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 (Jesus Arturo Perez *et al*. 2006).

In general, to have improvement in the quality of applications and network, QoS is to be enabled in the system and extended where necessary (as extending from layer 3 to layer 2) (Bouras *et al*., 2008). This gives an improvement in the network and application's quality although different researchers follow different technique to achieve this. QoS implemented in any system should be reliable so that it can be relied on. It should be scalable so that many users could be added to a system without additional cost. So also cost effectiveness, timeliness, mobility, heterogeneity, energy sustainability and security.

Figure-1. QoS.

The architecture here is DiffServ where classes of traffic are tried to be interconnected, matched and agree to a common code point which is supported by the interconnection scheme. QoS is tried to be configured but no network is expected to deploy all possible QoS. There is need to understand the interconnection classes and code points and gain the ability to map each to its own network class and code point scheme, this is necessary when dealing with QoS (R. Geib, 2013).

A suggestion by (M. Barnes, Ed. Polycom, 2011) was to fully address fundamental issues of the open pinhole problems and phone number routing as this will pave way for the QoS to sort itself out. A new technology was introduced that will help prevent the barriers in inter-domain multimedia services. It's called Verification Involving PSTN Reachability (VIPR) (M. Barnes, Ed. Polycom, 2011). A QoS marking communities that match neighboring class set encodings for the upper and lower layers was introduced. The layers are virtually mapped. It was suggested that QoS based tunneling mechanisms are the means of choice for transporting transparent traffic in all transit forwarding cases. Some approaches for proper inter-domain QoS strategies includes; inter-domain parameter signaling, monitoring, metering and misbehavior detection strategies. All of these approaches are complex but it all gets close to guaranteed QoS (Knoll T., 2013). This is to show how QoS is a critical metric in today's network.

4. DESIGN OF THE PROPOSED MECHANISM

VoIP and Video applications are considered as important user applications in multi-media applications and therefore it needs to be delivered with the minimum delay, less packet loss and obtain the maximum throughput as well. This work considers mapping QoS classes in order to attain interconnection negotiations among all classes so that optimum differentiated services for real time applications can be obtained under the DiffServ scheme. The aim here is to provide our WiFi Access Point (AP) with a Mobile Node (MN) QoS profile so as to get the overall resources of the network more effectively utilized. The AP will map and configure the QoS policy it receives from Wireless LAN Controller (WLC) to the upstream and downstream data packets flowing in the core of the network, then priority will be given to QoS bearer packets. To enable priority in the QoS bearer packets, DSCP is associated with the IP flow in the network. The IP flows include; IP source address, source port number, IP destination address, destination port number and protocol. In order to achieve this optimally, consistent mapping of the QoS parameters to the flow values is needed. The QoS parameters that we need to associate to the flow includes; QoS Class Identifier (QCI), Maximum Bit Rate (MBR) and Allocation and Retention Priority (ARP). Every QCI is assigned a DSCP (ie is mapped to become a high priority parameter), then it becomes QCI/DSCP. By doing this, our applications will get a simple and separate QoS (Huawei, K. J. 2012).

When the traffic is allowed to flow, it is then marked and classified into classes, our new mechanism considers IMM traffic as the most sensitive, therefore mapped this class of traffic with the QCI/DSCP. The channel is allowed to be accessed and utilized by the IMM traffic first, then the other traffics are considered as best effort will then follow through the channel if the bandwidth is sufficient enough to accommodate all, otherwise the best effort traffic are not allowed to be transmitted until after all IMM traffic finished transmitting because of the QCI/DSCP associated makes it prior. This of course gives two things to the IMM traffic; high strength signal and priority of transmission. With this, we will achieve the following; guarantee QoS for IMM applications, much reduction in loss and delay, while obtaining significant increase in the throughput. The other applications guarantee is only subjected to available bandwidth.

Our model configures and maps a class based on strict priority policy. Classes are defined and each class matches a specific traffic. Then a rule is assigned that

VOL. 9, NO. 12, DECEMBER 2014 ISSN 1819-6608
ARPN Journal of Engineering and Applied Sciences

gives every class some amount of bandwidth to be use by it. For example; there is a class named VIDEO that match ‘rtp video’ which gets 256kbps of the bandwidth.

```
R1(config-QCI/DSCP-map)#class-map match - VIDEO
R1(config-cmap)# match protocol rtp video
R1(config)#policy-map Strict Priority 256

R1(config-QCI/DSCP-map)# class-map match -
VOICE
R1(config-cmap)# match protocol rtp audio
R1(config)#policy-map Strict Priority 128

R1(config-pmap-c)#class class-Best effort
R1(config-cmap)#matchprotocol rtp B/E
R1(config)#policy-map Fair Queue 56

R1(config-pmap-c)#class class-Background
R1(config-cmap)#matchprotocol rtp B/K
R1(config)#policy-map No Priority
```

After constructing the mapping code, the model can now mark all the classes created and prioritizes the most sensitive traffic through all the routers. The same policies are maintained across all routers in order to ensure consistency. After all configurations, the sniffer laptop captures and measures the differences in time for the 1-way throughput.

We can also use a switched network which allows full duplex transmission micro segmentation in order to avoid collisions totally. This method can also help us to obtain the highest possible available bandwidth, and any device that is connected to the switch so that it can benefit from the maximum bandwidth utilization. As an attempt to have additional enhancements, we can also use open switches so as to get the possibility of increasing security and reducing broadcast domains using VLANs. We can still use trunk interfaces for extending the availability of ports for other devices.

There is an exceptional case when Virtual Local Area Networks (VLANs) are incorporated in a campus network, and when the traffic which is sent by different users belongs to the same VLAN, then all the traffic sent will never pass through any router interface. If this happens, then there won’t be any need to prioritize the traffic using layer 3 policies. This is why layer two priorities need to be added in the model. The IEEE 802.1p extends from the IEEE 802.1q (VLANs tagging) standard. The 802.1q standard incorporates a tag that uses an Ethernet MAC frame. There are two parts in the tag: the VLAN ID (12- bit) and Prioritization (3-bit). The field of the prioritization was neither defined nor used in the IEEE 802.1q VLAN standard. But with the introduction of the IEEE 802.1p the prioritization field is defined. The header in the 802.1p contains a three-bit field for prioritization. This three-bit field allows all the arriving packets to be grouped and classified into various traffic classes. There are eight levels of priority established in the IEEE 802.1p. 7 is considered the highest priority and it continue to reduce the priority as the value goes down until it reaches 0 which is the best-effort (lowest) priority available. The zero value indicates the best-effort default; this is automatically invoked when no other value is been set. All the eight levels provide some sort of marking that conforms to the general prioritization model. However, hardware must also conform to this feature before it can work. The following Table shows the eight levels of priority access mapping for DSCP.

Table-1. Priorities and access class mappings of DSCP.

User priority	Designation	Access category
0	BK (Background)	AC_BK
1	BK (Background)	AC_BK
2	BE (Best-Effort)	AC_BE
3	VO (Voice)	AC_VO
4	NC (Network Controller)	AC_VO
5	(Video Excellent Effort)	AC_VI
6	CL (Video/Controller Load)	AC_VI
7	VI (Video)	AC_VI

5. AN ENHANCED PRIORITIZED TRAFFIC MODEL

As explained earlier, we propose to map the QoS class parameter (QCI/DSCP) to the marked incoming packets through the routed links of the network. We hope to tag frames using Class of Service (CoS) by allowing the devices in layer two to be able to provide QoS requirements of the packet at the data link layer. After this, we mapped the QoS of layer two (CoS) to layer three QoS (DSCP).

It is worth knowing more about the reason of choosing Distributed Services Code Points (DSCP). The DSCP is adopted from IEEE 802.11e standard which is able to provide some sort of QoS. Incoming traffic from higher layers are classified according to eight different priority class as mentioned before. These eight different classifications are further mapped into another four access categories (ACs). These ACs are a model class of our traffics, these are VIDEO, VOICE, BEST EFFORT AND BACKGROUND with indications from the ACs as AC (0), AC (1), AC (2), AC (3) respectively. Some set of parameters such as Maximum Contention Window (*CWmax*), Minimum Contention Window (*CWmin*), Arbitrary Inter Frame Space (AIFS), window’s length for a back off are assigned to each category.

For AC_i (where I is 0, 1, 2, 3 or 4), for the maximum back off window size is CWmax [i], for the first back off window size is CWmin[i] and for the arbitrary inter-frame space is AIFS [i]

For $0 \leq i \leq j \leq 3$ CWmin [i] $\geq$ CWmin [j] where CWmax [i] $\geq$ CWmax[j] and also AIFS [i] $\geq$ AIFS [j]. In

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every case, one of the conditions must be met (i.e. for AIFS [i], CWmax [i] and CWmin[i], for all I = 0, 1, 2 and 3. Whenever the initial window is smaller CWmin [i] and AIFS [i] is reduced, the traffic got better chance to access the channel, and this gives the QoS needed to the IMM applications. Let's look at the model.

Table-2. Characteristics of four access categories applications.

Application	**Voice** (VOIP)	**Streaming video**	**Web browsing**	**Telemetry emails**
Traffic class	**AC_VO conversational class** (Real time)	**AC_VI streaming class** (Real time)	**AC_BE interactive class** (Best Effort)	**AC_BK background class** (Background)
Fundamental characteristics	-It saves the time variation among the information entities of the stream. -Its pattern for the conversation gives low delay and stringent.	-It saves the time variation among the information entities of the stream. -The video application is loss tolerant to some point.	It requests the response pattern.	There is no any expected time by the destination to receive data.

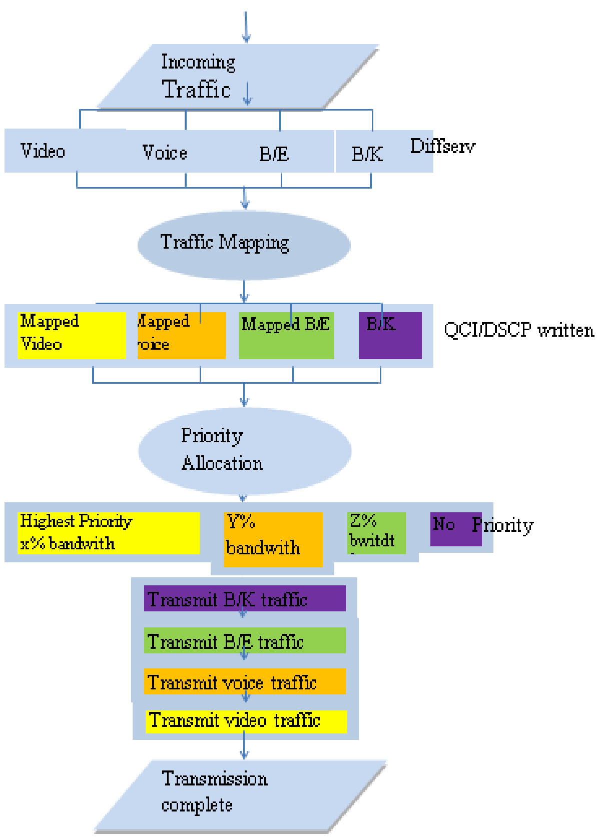

Figure-2. Traffic prioritization model.

The Algorithm for the above flow chart is given in plain language as follows:

a) Classify the incoming traffic into classes
b) Create a class to identify all the IMM traffic class (i.e. voice and video)
c) Create another class that will identify the rest of the traffic, such as FTP, HTTP, etc (i.e. non real time traffic).
d) Establish a marking policy which sets the QCI/DSCP value of the voice and video class among the traffics.
e) Create a prioritization policy which defines the amount of bandwidth to be provided to every class.
f) Apply these marking policies to the incoming traffic from the router's interface Ethernet switch.
g) Apply the prioritization policy to the outgoing traffic of the router's interface Ethernet switch.
h) Transmit the IMM traffic (i.e. voice and video) over the medium access by utilizing the full bandwidth for the IMM traffics.
i) Transmit all other traffic such as FTP, HTTP if there is remaining bandwidth to accommodate other traffics.
j) Repeat steps 8 and 9 until there is no traffic to transmit anymore in the core of the network.
k) End

6. EXPERIMENT

First, we deployed a network scenario using a default configuration (which does not have any kind of priority). After that, we generate four different types of traffic, viz: voice, video, best effort and background. We configured the routers and switches with our new model proposed in the previous section. We observe the traffic flow both in steady and transient state. We record all our results and try to configure other priority elements and see what amount of change can we obtained from our new mechanism. At the end, we compared both results in order to determine the performance improvement level obtained with our network design.

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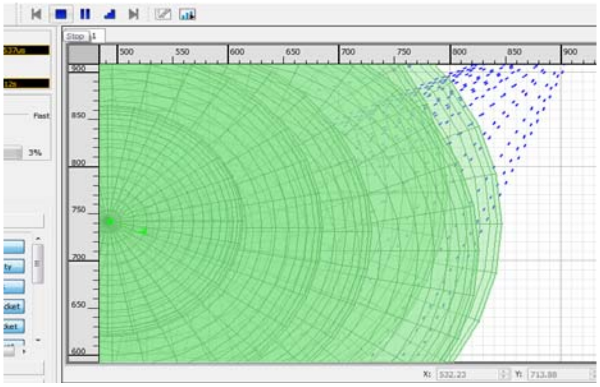

Figure-3. Simulation scenario.

The simulator software supports multi-party conferencing using QualtNet 5.1. The simulator is used by many researchers for simulating wireless networks. When setting our nodes, we assume that they all have the required hardware and software codecs. The Ad hoc on Demand Distance Vector (AODV) protocol is used from other routing protocols due to its adaptability to highly dynamic networks and carrying less overhead, this will give a good path for the IMM traffics. The 802.11b is used in the Physical layer due to its ability to convey radio signals. The 802.11e is used in the Medium Access Control (MAC) layer due to its priority ability to multi-media traffics as well as the Distributed Coordination Function (DCF). There is one Access Point (AP) and twenty four nodes in all. The applications used are basically the four access category traffic, i.e. AC_VI, AC_VO, BE and BK which can be generated from CBR, VOIP, HTTP, and FTP respectively. The voice and video traffic are of 512bytes packet size but the size was increased to 1024 and 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 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 IP addresses used are as follows: 190.160.1.1, 190.160.1.2, 190.160.1.3, 190.160.1.4, 190.160.1.5, 190.160.1.6 etc. other parameters used in the simulation are summarized in the Table below:

Table-2. Simulation parameters.

Parameter	Value
Area	$1000m^2$ * $1000m^2$
No. of nodes	25
Routing protocols	AODV
Types of traffic used	Video, Voice, BE & BK
Packet size	512 bytes
Simulation time	100 sec
Nodes connection	8
Scheduler and queue	Diffserv
Frame scheduler	FIFO
Priority assigned	QCI/DSCP
Interval	50ms
Bandwidth	11Mbps
Radio type	802.11b
MAC protocol	802.11e
Mobility model	Random waypoint

7. RESULTS AND PERFORMANCE EVALUTION

We conducted the simulation as many times as possible until the time we obtained a desired result. At the end of every simulation we checked the performance of the network by evaluating the three performance metrics; viz; average end-to-end delay, average jitter and throughput. We obtained a smaller value for the delay and jitter, while having a bigger value for the throughput.

Before we start our simulation, we opt to comply with the standard employed by some standard organizations and other researchers. For example the ITU-T has provide that the IMM applications (i.e. voice and video conferencing) can only tolerate maximum time delay of packets up to 400ms and less than 1ms for jitter for good quality speech and conferencing (Elahi, 2012) (Assem, Malone, Dunne, and Sullivan, 2013) (Singh *et al.*, 2014). However, this work sets maximum delay tolerable as 50ms. Packets loss is a critical metric among other. It 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% (Sarkar et.,al 2012) (Singh *et al.*, 2014). Let's look at simulations values obtained after the simulation with QCI/DSCP assigned.

Average End-to-End 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 individual nodes in the network scenario.

Observation: it is observed that before the mapping an average value for end-to-end delay is obtained but after the mapping a more acceptable (or lesser) value is obtained. The first value of the end-to-end delay is 0.0557327sec (55ms) while the second value for the end-to-end drops to 0.0354514sec (35ms). Therefore, the overall average end-to-end delay is reduced by 57%. This is a good improvement in having the delay being reduced by this much because it also adds up in the number of packets being received at the receiver side.

Justification: practically, when there is less traffic to send and data rate is low, the network is less congested and less interference occurs. As a result of that packet loss is also less. At this time all nodes experience almost the same delay. Therefore, its evaluation gives better performance. However, as traffic begin to increase, more delay is been experienced by stations which leads to more packet loss. In this simulation, traffic is continued to increase until when a value for the delay more than 400ms is obtained. At this point, we say the performance of the network is poor or cannot accommodate the traffic load generated. But the QCI/DSCP mapped to the IMM traffic helps in reducing the delay up to a certain traffic load in the network. Therefore, our network performs better when the traffic load is not too much beyond what the network can accommodate.

Qualitative analysis: when 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 55ms to 35ms thereby giving a percentage drop of 57%. This shows that mapping QoS class parameter to the IMM traffic reduces the overall delay.

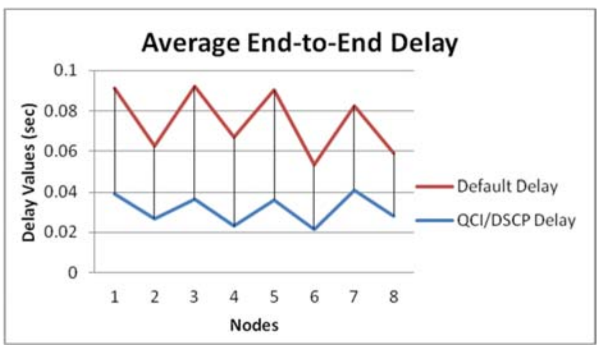

Figure-4. Average End-to-End delay.

Average jitter

This is the average of the variation 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.

Observation: 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 0.008945sec (8.9ms) while the second jitter after mapping QCI/DSCP gives 0.00678 (6.7ms). This gives a good decrease in the value of jitter which results in having a good quality speech and streaming of video and also increases the amount of throughput to be received.

Justification: our setup proves good performance of the network under the traffic generated. It proves even better performance when QCI/DSCP is mapped. The decrease in the value of jitter means that packets are being transmitted continuously and with quality. However, in order to continue evaluating the network, more traffic load is increased and one way of achieving this, is by increasing the simulation time up to 500sec or so. As the traffic increase, the variation in the jitter is becoming wider and irregular, and this result in having a poor performance of the network. At this point, the performance of the network is below average. But mapping QCI/DSCP to the delay sensitive traffic (i.e. voice and video) reduces and rearranges the jitter between consecutive packets.

Quantitative analysis: the jitter values before and after mapping QCI/DSCP are 0.008945sec (8.9ms) and 0.00678 (6.7ms) respectively at a data rate of 512kbps, thereby giving jitter decrease of 32.8%. This is a good improvement in having a good quality speech and also it increases the throughput obtained.

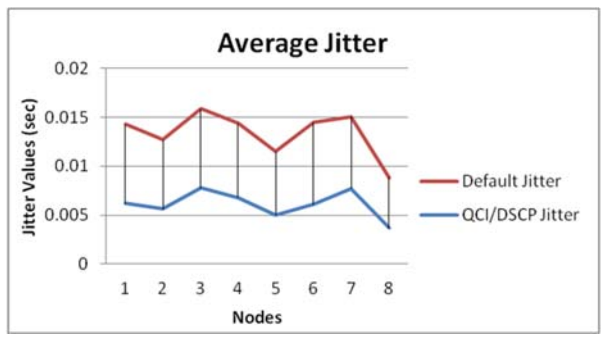

Figure-5. Average Jitter.

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.

Observation: In our network scenario, we obtained two throughputs i.e. before mapping QCI/DSCP and after mapping. The result shows a slight increase in the amount of throughput after QCI/DSCP is mapped to delay sensitive traffic (video and voice). The values of the throughput will be given in number of packets just for simplicity. At first, we obtained an average throughput value of 776 packets. After mapping, the average value was raised to 816 packets. 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.

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Justification: the throughput obtained is very important in determining the performance of our network. We both see the throughput before and after mapping QCI/DSCP and also the amount of throughput obtained with increase in the traffic in the network. It is seen that the throughput of this network scenario slightly increased after mapping the QCI/DSCP. But there is a disaster in attempting to increase the traffic load. When huge traffic is increased here, throughput begins to fall and becomes like when it is formally sent at small traffic, it becomes even disastrous when the traffic is huge enough that is beyond the network capacity. At this point, some nodes may not be able to receive any packet(s). Still the QCI/DSCP parameter being mapped increases the throughput obtained at the receiver end. Therefore, throughput loss is more notable at a high data rate.

Quantitative analysis: In terms of throughput, at a data rate of 512kbps, the network experiences a slight increase of almost 5% after mapping QCI/DSCP.

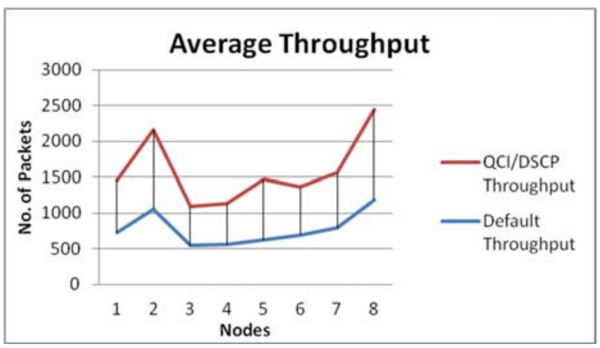

Figure-6. Throughput

Huge traffic load with variable packet size

In this sub-section we increased our traffic by varying the packet size from the least to highest. We start with 160kbps and increase to 256kbps, 512kbps, 1024kbps and 1530kbps. We test all these packet sizes in order to see its effect under both low and high traffic. We run the same scenario several times with increasing the packet size each time. Each performance metric is taken as the average sum of all the individual values of its nodes.

We setup a new simulation environment with the same parameters but now setting the packet size as focus. Voice packets can be of the size 160kbps and 256kbps and video packets can be of the sizes 512kbps, 1024kbps and 1530kbps. First, we record the values of both delay and jitter in the scenario when no priority is assigned to the traffic (i.e. the default scenario). The second scenario is when the size of all the packets is 160kbps and QCI/DSCP is mapped to the IMM applications (voice and video). The last scenario gives the values for the end-to-end delay when the load is high. So our new mechanism achieves the less end-to-end delay. We can see that as the load increases, it causes more delay.

This is a bandwidth effect. It 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 has to increase. Therefore we say that as the traffic continue to increase, the average end-to-end delay tend to increase as well. The graph below presents the results as shown.

We run the first scenario while keeping the packet size for all traffic as 160kbps. Then we recorded the average values for the jitter and delay, and then we mapped QCI/DSCP and recorded the values of jitter and delay. And then we generated more traffic and run again, we also recorded the average values for the jitter and delay as well. Then we set up another scenario keeping the packet sizes now as 256kbps. We did same for 512kbps, 1024kbps and 1530kbps. All the values are shown in the graph below.

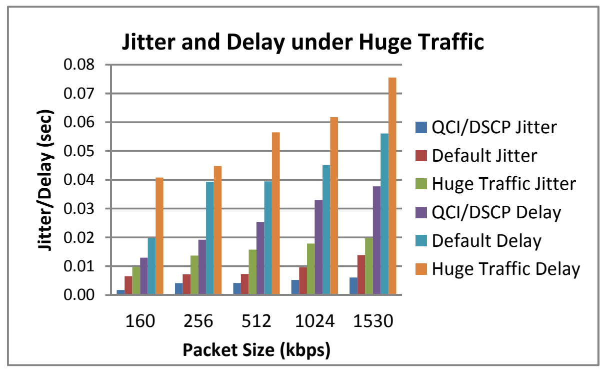

Figure-7. Jitter and delay under huge traffic with variable packet size.

For throughput, we consider the results obtained under both low and high traffic with increased size of packets. We presented the results following the same style above i.e. recording the average values for the throughput in the default scenario, then recording the values for the mapped QCI/DSCP and lastly recording the values of the throughput under high traffic generation. So also for 256kbps, 512kbps, 1024kbps and 1530kbps.

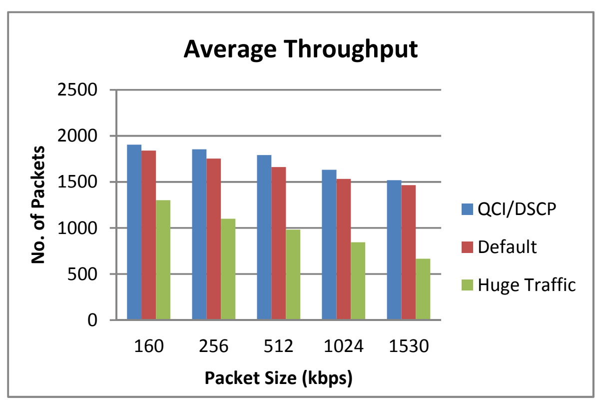

Figure-8. Throughput under huge traffic.

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Huge traffic load with increased rate of transmission

Now, we took another dimension of generating more traffic in the scenario in order to see what effect they have to the overall network. We reduce the interval for the transmission rate of packets from 50ms to 25ms, 16.67ms, 12.5ms, 10ms and 8.33ms. This gives more number of packets to be sent over the network i.e. 4000, 6000, 8000, 10000 and 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%, and 500% respectively which is enough to produce a change in the behavior of the network. Under this condition, we are going to consider the average end to end delay and average jitter first, then the throughput. Each metric is computed as the average sum of individual values obtained from individual nodes.

We have a comparison to see the difference and effect between the traffic loads. The first scenario is the default without any priority being assigned to it. The second scenario is our proposed mechanism which mapped the QCI/DSCP so as to give differentiated service to IMM applications. The third scenario is when the load is increased and here we pay our attention. From the graph below, we can see that the first scenario (default) maintains a reasonable end-to-end delay because all parameters are set to standard before QCI/DSCP is mapped to the loss sensitive applications. The second scenario is our proposed scheme, the highest priority is given to the voice and video application, which can access the channel and utilize it fully before other traffics (i.e. best effort and background). But when traffic continue to increase (in the third scenario), it becomes so difficult for the network to send and receive among its nodes in the required time.

The high delay under the huge traffic is caused due to the long queue of packets in the network. The same channel and connections are maintained but only the amount of traffic is increased. During the default and our proposed scheme set up, all the traffic is not beyond the network capacity of the channel and so can pass through so easily based on FIFO frame scheduler. But when the load is increased, it is difficult for all the traffic to pass through in the required time. For this reason, long queue is created which results in giving out an increase in the overall end-to-end delay and jitter.

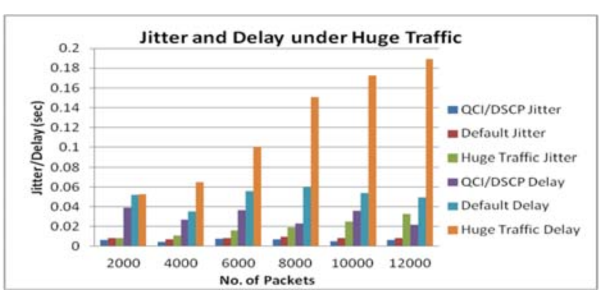

Figure-9. Jitter and delay under huge traffic with variable number of packets.

For the throughput, we observe the difference of the throughput under increased traffic scenario. Like in the former analysis the first scenario is the result of the throughput at default setup. The second scenario is the result of the throughput for our proposed scheme implementation where we achieve the best result of the throughout for our simulation after QCI/DSCP mapping. The third is the average sum of the throughput when traffic is being increased by reducing the interval of the transmission rate of packets. It is found that our proposed scheme managed to obtain the highest throughput for all the simulation setup.

But when traffic keeps increasing, throughput begins to drop drastically up to a point when some nodes could not even receive a single packet. Throughput is the most critical metric to achieve among all the performance metrics as a slight change in the setup could lead to huge loss. When the long queue occurs, many of the packets that are kept in waiting are dropped at last because they can no longer continue to wait beyond some certain amount of time. Therefore, under huge traffic, throughput keeps decreasing drastically.

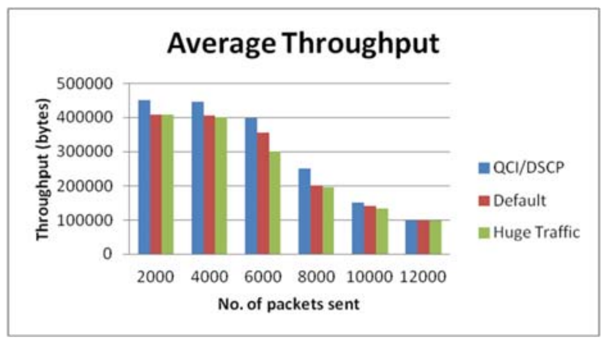

Figure-10. Throughput under huge traffic.

8. CONCLUSIONS

We have implemented a new mechanism which maps QCI/DSCP to the voice and video traffic, an enriched QoS class parameter so as to ensure a satisfactory level of QoS for our IMM applications. We run our technique under the diffserve scheme which classifies and identifies the delay sensitive traffic. We use the QualNet 5.1 simulator using the richer parameters. First, QCI is mapped to DSCP, and QCI/DSCP is mapped to IMM traffics. The QCI/DSCP mapped associates and prioritizes the IMM traffic so as to ensure voice quality speech and video conferencing. We run the simulation for several scenarios 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 for mapping QCI/DSCP to the traffic flow. Although the default scenario also shows an acceptable result but is not better than our methodology.

We look at the network performance when the traffic is increased by two ways. One is increasing the traffic by varying the size of the packets. Another way is increasing the traffic by reducing the interval at which packets are transmitted which of course increases the number of packets to be sent. In both situations, the overall network performance is appreciated with the mapping of QCI/DSCP to the upstream and downstream packets flowing in the network because lesser values for the average end-to-end delay and jitter are obtained as well as higher values for the throughput than when QCI/DSCP QoS parameter is de-mapped. 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 for being less and obtaining the highest throughput.

However, the network performance degrades under huge traffic and large packet size. But with our proposed technique, some level of interconnection negotiations is attained and an optimum differentiated service for multi-media applications is achieved and most importantly resources are being utilized effectively. We can therefore say that our campus network can be at its best with average load and average packet size. At the end of our simulation, we see that delay is reduced, loss is minimized and throughput is increased.

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