Efficient Multimedia Service Delivery Over State of the Art Network Architecture

Thesis Submitted to
The Superior College, Lahore

In Partial fulfillment of the
Requirement for the Degree of

Master of Science in Computer Science

By
Farooq Haider

Roll No. MSCS-F16-005

Session: 2016-2018

Registration No: MSCS-F16-005

The Superior College (Faculty of Computer Science & IT), Lahore
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DEDICATION

This work is dedicated to my Supervisor, family and friends who encouraged me to face any challenge.

ACKNOWLEDGEMENT

In the name of Allah Almighty, the most Beneficent and the most Merciful

First and foremost, we like to give thanks to almighty ALLAH who blessed us with guidance and helped us for making this short period of educational journey in a reality.

We would like to thank project co-coordinator Dr. Muhammad Hasanain Chaudary for accepting us and giving opportunity to do our research project, His encouragement and excellent advice give us fuel to drive our thesis in a right direction. Also thank him again for his excellent intellectual advice and encouraging guidance regarding our thesis which helped us to think carefully in the research field.

We also want to extend our gratitude to our beloved parents & family for their heartless love, support and encouragement during our thesis work, in particular our mother for her kind love and encouragement advice.

We are pleased to thank our friends for their support and their entertaining fun during our research work that helped us to feel relax.

Last but not the least; we are very thankful to Superior University & all the members of this department for their lovely guidance and providing best quality of education which will help us to drive our knowledge in the future of our life.

Farooq Haider

ABSTRACT

Over the last years, we have witnessed a rapid deployment of real-time applications on the Internet and many research works about Quality of Service (QoS) in particularly IPv4(Internet Protocol version 4). The inevitable exhaustion of the remaining Ipv4 address pool has become progressively evident. As the continuing evolution of Internet Protocol (IP) goes on, deployment of Ipv6 QoS is underway. With respect to different QoS architectural models such as Best Effort Service (BE), Differentiated Service (DiffServ) and Integrated Service (IntServ) have been introduced by the Internet Engineering Task Force (IETF) for QoS performance enhancement in the IP network. The BE model is not capable to provide the guaranteed QoS for real-time applications. IntServ model provide end to end delivery services but need a resources booking along its path. In contrast, DiffServ does not need reservation of resources and provide packets classification & vice versa in an efficient way. From the evolution of Ipv4 QoS solutions, the integration of DiffServ and MPLS TE is known to satisfy the guaranteed QoS requirement for real-time applications. The goal of this research project is to evaluate the Ipv4/Ipv6 QoS performance of real-time multimedia applications such as video streaming which operates OverDiffServ in Ipv4/Ipv6 networks. Further we also study the interaction of Expedited Forwarding (EF), Assured Forwarding (AF) traffic aggregation, link congestion, and the effect of various performance metrics such as end-to-end delay, delay variation, queuing delay, throughput and packet loss. The effectiveness of DiffServ integration in Ipv4/Ipv6 network is illustrated and analyzed by using different tools and techniques. The research project shows that, IPV6 abilities against IPV4 and give confidence to ISP’s to deploy this version of IP to provide their customer quality of services support.

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CHAPTER I

1. Introduction

1.1 Multimedia:

MULTIMEDIA content exchanged by mobile devices is increasing dramatically in terms of both number of streams and their quality as the expectations of users also increase. Mobile devices including smartphones and tablets are overtaking classic devices such as desktops in terms of the amount of multimedia content they store, process and share. For instance, the mobile video traffic accounted for 55 percent of total mobile data traffic in 2015 and it is estimated that will reach 75 percent by 2020 [1]. At the same time, cloud computing is already supporting a wide range of flexible innovative applications and services, many multimedia-based.

The highly popular social networking services for example are seeing an increased number of users sharing with peer's multimedia content either originating from their mobile devices or previously received from media servers. Mobile users of such rich media communication-oriented applications possess increasingly sophisticated and capable portable devices, in terms of connectivity, processing and graphical display capabilities. Additionally, most mobile devices are already equipped with multiple wireless interfaces which allow them to connect simultaneously to multiple wireless networks Using different wireless communication technologies (e.g., Wi-Fi, LTE, etc.), enabling them also to form ad-hoc networks. Although not yet available on the market, mobile devices equipped with multiple interfaces on the same technology (i.e., Wi-Fi) are already discussed and designed both in the academia and industry [2], targeting an even better mobile inter-connectivity. These devices have limited battery resources will handle diverse data types and will operate in enviornment that are insecure , unplanned and shows different characteristics.

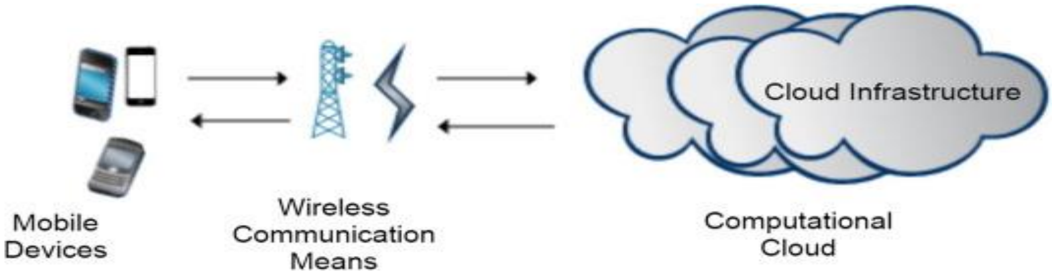

Figure 1: Mobile Multimedia Cloud Computing

1.2 Wireless Communication:

Mobile computers require wireless network access, although sometimes theymay physically attach to the network for a better or cheaper connection. Wireless communication is much more difficult to achieve than wired communication because the, noise and echoes. As a result wireless connections have a lower quality than wired surrounding environment interacts with the signal, blocking signal paths and introducing connections: lower bandwidth, less connection stability, higher error rates, and, moreover, with a highly varying quality. Three key problems in networked wireless multimedia systems are 1) the need to maintain quality of service (throughput, delay, bit error rate, etc) over time-varying channels, 2) to operate with limited energy resources, and 3) to operate in a heterogeneous environment.

1.2.1 Quality of Service –

Considerations of energy efficiency are fundamentally influenced by the trade off between energy consumption and achievable Quality of Service (QoS). To deal with the dynamic variations in networking and computing resources gracefully, both the mobile computing environment and the applications that operate in such an environment need to adapt their behaviour depending on the available resources including the batteries.

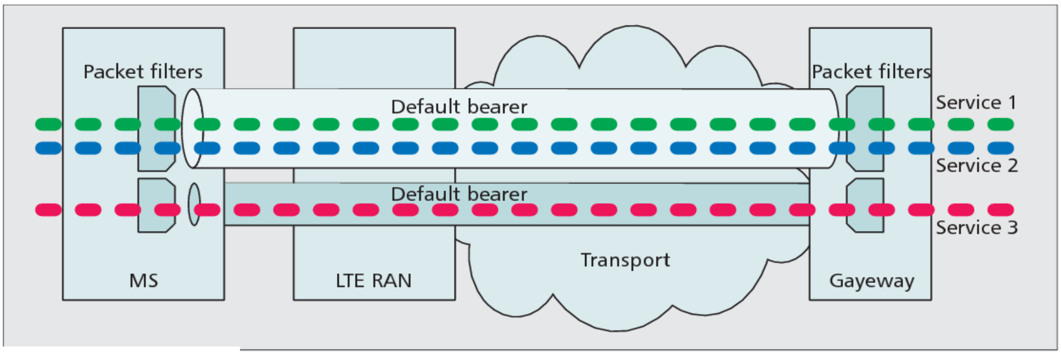

Fig:2 *edicated bearers of a terminal (MS) in the LTE QoS framework.*

1.2.2 Energy Efficiency The wireless network interface of a mobile computer consumes a significant fraction of the total energy of a mobile computer. More extensive and continuous use of network services will aggravate this problem. Energy efficiency can be improved at various layers of the communication protocol stack. Adaptability of the
protocols is a key issue is achieving this.

1.2.3Heterogeneity – In contrast to most stationary computers, mobile computers more heterogeneous network connections. As they leave the range of one network transceiver they switch to another.
Heterogeneity makes mobile computing more complex than traditional networking.

1.3 Mobility

Mobile systems will have a set of challenges arising from the diverse data types with different quality-of-service (QoS) requirements they will handle, their limited battery resources, their need to operate in environments that may be unpredictable, insecure, and changing, and their mobility resulting in changing set of available services. The following are the key technological challenges.
1.3.1 Battery Life– As the current portable computers have shown to be capable of assisting mobile users in their daily work, it is becoming increasingly evident that merely increasing the processing power and raising raw network bandwidth does not translate to better devices. Weight and battery life have become more important than

pure processing speed. Energy consumption is becoming the limiting factor in the amount of functionality that can be placed in portable computers like PDAs and laptops.

1.3.2 Infrastructure – The design of mobile systems cannot be done in isolation. The mobile system of the future is likely to be designed to operate autonomously, but it is also very likely that it relies on an external infrastructure to access information of any kind. The mobile will likely encounter many, very diverse environments and various network infrastructures.

1.3.3 Adaptability – Wireless mobile systems face many different types of variability in their environment in both the short and the long term. Mobile systems will need the ability to adapt to these changing conditions, and will require adaptive radios, protocols, and codecs and so on. Adaptive error control and adaptive compression are Examples of such techniques.

1.3.4 Reconfigurability – To combat a higher degree of variations in operational environment than is possible with adaptable systems, reconfigurable architectures can be used that allow new software and hardware functions to be downloaded. Thus rather than changing parameters of algorithms to current conditions, an entirely new set of protocols and algorithms can be used.

1.3.5 Security – When computers become more involved in people's personal and business activities security i.e. confidentiality, privacy, authenticity and nonrepudiation become important concerns.

1.3.6 User interfaces – Traditional keyboards and display based interfaces are not adequate for the mobile systems of the future because of the required small size and weight of these system. Instead, intrinsically simpler interfaces based on speech, touch, pen and so forth are more likely to be used and more adequate to the small form factors of these systems.

1.4 Mobile Systems Today –

The research community and the industry have expended considerable effort toward mobile computing and the design of portable computers and communication devices.

These devices now support a constantly expanding range of functions, and multiple devices are converging into a single unit [6]. Personal computers are becoming an integral part of daily life, as portable appliances such as wristwatches and cellular phones have become over the last few years.

Current mobile systems can be classified into the following categories based on their functions and form factors.

1.4.1 Laptops – Laptops are not really mobile systems since they are too large and too heavy. In essence they are just battery operated small desktop machines. Wireless communication is generally based on WLAN products that can be plugged in as a PC-card.

1.4.2 Pen tablets – Pen tablets can be viewed as laptops without keyboards. Interaction with the pen tablet is through pen input. In most cases, the pen replaces the mouse as pointer device. Some tablets have an internal radio modem, whereas others require an external radio modem. Generally spoken, these terminals are no different from the average desktop.

1.4.3 Virtual books – Recently several products have been introduced that replace paper as the medium for reading and browsing a wide variety of material [13][59][65].

These systems have good quality displays, and a rather conventional architecture. User input is limited to a few buttons, and a pe

1.4.4Handheld Personal Computers (HPC) – Systems of this category are basically miniature laptops. They are characterised by a reduced form factor keyboard and a half-VGA resolution display. They usually run reduced versions of Windows applications, including word processing, presentation, and scheduling software.

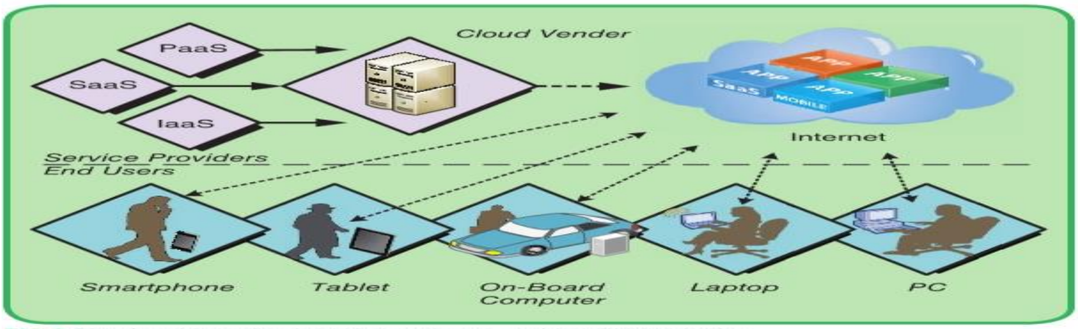

FIG 3 Cloud entities: cloud vendor, end user, and services provider.

1.4.5Personal Digital Assistants (PDA) – the PDA is generally a monolithic device without a keyboard (although some have small sized keyboards) and fits in the user's hand. As such, pen input is the norm, and handwriting recognition is common. Communication abilities involve a docking port or serial port for connecting to and synchronising with a desktop computer, and possibly a modem.

1.4.6.Smart phones – Although cellular phones may have several peripheral functions like a calculator, date book, or phone book, they are foremost a communication tool. Combination devices like the Nokia 9000 are essentially PC-like devices attached to a cellular phone.

1.4.7 Wireless terminal – These systems are basically nothing more than the wireless extended input and output of a desktop machine which acts as the server. These systems are designed to take advantage of high-speed wireless networking to reduce the amount of computation required on the portable.

1.5 Multimedia Data Characteristics:

The systems that are needed for multimedia applications in a mobile environment must meet different requirements than current workstations in a desktop environment can offer. The basic characteristics that multimedia systems and applications needs to support are [17]:

1.5.1 Continuous-media data types – Media functions typically involve processing a continuous stream of data, which implies that temporal locality in data memory accesses no longer holds. Remarkably, data caches may well be an obstacle to high performance and energy efficiency for continuous-media data types because the processor will incur continuous cache-misses.

1.5.2 Provide Quality of Service (QoS) – Instead of providing maximal performance, systems must provide a QoS that is sufficient for qualitative perception in applications like video.

1.5.3 Fine-grained parallelism – Typical multimedia functions like image, voice and signal processing require a fine-grained parallelism in that the same operations across sequences of data are performed. The basic operations are relatively small.

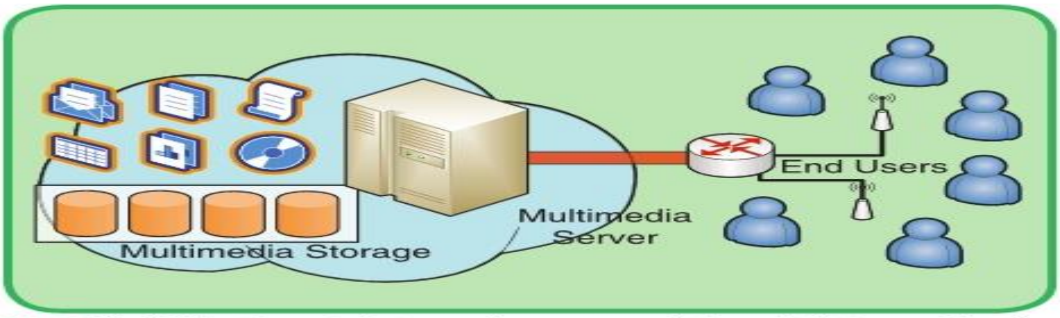

FIG 4 The link bottleneck between the server and the switch for multimedia delivery.

1.5.4 Coarse-grained parallelism – In many applications a pipeline of functions process a
single stream of data to produce the end result.

1.5.5 High instruction reference locality – The operations on the data demonstrate typically high temporal and spatial locality for instructions.

1.5.6 High memory bandwidth – Many multimedia applications require huge memory bandwidth for large data sets that have limited locality.

1.5.7 High network bandwidth – Streaming data – like video and images from external sources – requires high network and I/O bandwidth.

CHAPTER II

2. Related Work:

2.1 What is Quality of Service (QoS)?

The quality of service is a set of services that must be provided to functional an application efficiently in the network. By having QoS, it is possible to ensure proper information delivery, giving priority to critical performance applications that simultaneously share the network resources with other non-critical applications. Implementing QoS in a network manages network performance in a more predictable way and uses bandwidth in quite good manners.

2.1.1 QoS Metrics: There are different metrics proposed to measure the services provided on a network with QoS. Most are defined by the working group IP Performance Metrics, which can include variables such as bandwidth (bandwidth), amount of data transmitted per second (throughput), delay (delay), delay variation (jitter), cost and probability of loss among others.

2.1.1.1 Bandwidth: The digital bandwidth represents the amount of data that can be transmitted in a time unit. The total delay experienced by a packet is the sum of various types of delay:

2.1.1.2 Delay: Is the time that takes to put all the packet bits in a particular link. Among the types we have **propagation delay**: the time that takes a bit to pass through a link;

Processing delay: the elapsed time to process a packet in a node;

Queuing delay: The time-out for a packet in the queue before being transmitted.

2.1.1.3 Delay Variation (jitter): The delay variation measured the delay experience bt packets that come across the same route network.

2.1.1.4 Packet Loss: Packet loss is measured by the number of packets transmitted that are received at the destination, against the total number of packets transmitted.

2.2 Quality of Service Mechanisms:

To facilitate true end-to-end QoS on an IP-network, the Internet Engineering Task Force (IETF) has defined two models: Integrated Services

(IntServ) and Differentiated Services (DiffServ).

2.2.1 Integrated Services (IntServ): Integrated Services was the first QoS model developed by the IETF in the early 1990s. It was a standard that best suited long transmissions with guaranteed throughput before the existing the multimedia applications.

2.2.1.1 How IntServ works (in general)? The application will first characterize its traffic and what resources it will need. Then, the network will use a reservation protocol (RSVP) to reserve the specified bandwidth in each router along the way. Each router, or hop, will check whether it can guarantee the required resources, and hold that reservation for as long as it was asked to by the reservation request. Once all the hops have been set up, the sender can begin its data transfer, knowing that the data will get to the destination in time, in order and in good timing.

2.2.1.2 Resource Reservation Protocol (RSVP): The central part of Integrated Services is the Resource Reservation Protocol (RSVP). When a sender wants to transmit to a receiver (via unicast or multicast), the sender sends a PATH message toward the receiver(s). A PATH message contains several things; it passes information to the receiver about the traffic source, it passes on characteristics of the network path, and finally it installs the necessary state for the soon to come RESV ("Reserve") message to find out how to reach the senders from the receivers. Once the receiver receives the PATH message, it returns a RESV message along the exact reverse path that the PATH message travelled. RESV messages actually reserves the needed bandwidth in the routers along the path. As the sender receives the RESV message, it will start its transfer.[63]

2.2.1.3 Messages Types used by RSVP:

i) **PATH Messages:** The PATH messages (part of RSVP) are sent from the sender to the receiver along a unicast or multicast path, like any other data package.
ii) **RESV Messages:** The other message involved in a path set up is the RESV message.

2.2.1.4 **Reservation Styles:**

The information carried with the RESV message is not always the same since

Different reservation styles are available.

2.2.1.4.1 Wild-card-filter (WF) style: A WF implies a shared reservation. All receivers share a common reservation.

2.2.1.4.2 Fixed-filter (FF) style: The opposite of the Wildcard style. FF implies a "distinct reservation and explicit sender selection" meaning everyone will have to set up a separate reservation of their own.

2.2.1.4.3 Shared explicit (SE) style: This third style uses a shared reservation but explicit sender selection, meaning that there will be several senders in one reservation.

2.2.1.5 Offered Services:-

2.2.2.1.5.1 Best Effort:

2.2.2.1.5.2 Guaranteed Service: GS is the best choice if the data stream is very error-intolerant. This service guarantees that for instance no more than X packets will be dropped or delayed more than Y seconds. The benefit is that you get strict worst-case boundaries which are good for various real-time applications.

2.2.2.1.5.3 Control Load Service: CL does not provide any quantitative guarantees on delay boundaries or bandwidth capacity. Instead tries to emulate a lightly loaded network, providing good performance but not without occasional jitter, delays and packet drops.[72]

2.2.1.6 Issues with IntServ: **-** The main problem with the IntServ/RSVP architecture is scalability. The model does not scale well in the Internet core primarily because,

1. Huge storage and processing overhead is placed on the routers since the amount of state information in the routers increases proportionally with the number of flows,

2. The requirements on routers is very high, each router must implement RSVP, admission control, classification and packet scheduling.

2.2.2 Differentiated Service (DiffServ):

The IETF completed the Request for Comments (RFCs) for DiffServ toward the end of 1998. As stated in the DiffServ working group objectives, "There is a clear need for relatively simple and coarse methods of providing differentiated classes of service for Internet traffic, to support various types of applications, and specific business requirements. The differentiated service approach to providing quality of service in networks employs a small, well-defined set of building blocks from which a variety of aggregate behaviors may be built.

A small bit-pattern in each packet, in the IPv4 ToS octet or the IPv6 traffic class octet, is used to mark a packet to receive a particular forwarding treatment, or per-hop behavior, at each network node. In order to deliver end-to-end QoS, this architecture (RFC-2475) has two major components-packet marking using the IPv4 ToS byte and PHBs.

2.2.2.1 Packet Marking:

Unlike the IP-precedence solution, the ToS byte is completely redefined. Six bits are now used to classify packets. The field is now called the Differentiated Services (DS) field, with two of the bits unused (RFC-2474). The six bits replace the three IP-precedence bits, and is called the Differentiated Services Codepoint (DSCP). With DSCP, in any given node, up to 64 different aggregates/classes can be supported (2^6). All classification and QoS revolves around the DSCP in the DiffServ model.[72]

2.2.2.2 Per Hop Behaviors:

Now that packets can be marked using the DSCP, how do we provide meaningful CoS, and provide the QoS that is needed? First, the collection of packets that have the same DSCP value (also called a codepoint) in them, and crossing in a particular direction is called a Behavior Aggregate (BA). Packets from multiple applications/sources could belong to the same BA. In more concrete terms, a PHB refers to the packet scheduling, queuing, policing, or shaping behavior of a node on any given packet belonging to a BA, and as configured by a Service Level Agreement (SLA) or policy.

To date, four standard PHBs are available to construct a DiffServ-enabled network and achieve coarse-grained, end-to-end CoS and QoS.

2.2.2.2.1 The Default PHB (Defined in RFC-2474)

The default PHB specifies that a packet marked with a DSCP value (recommended) of `000000' gets the traditional best effort service from a DS-compliant node. Also, if a packet arrives at a DS-compliant node and its DSCP value is not mapped to any of the other PHBs, it will get mapped to the default PHB.[74]

2.2.2.2.2 Class-Selector PHBs (Defined in RFC-2474)

To preserve backward compatibility with the IP-precedence scheme, DSCP values of the form `xxx000,' where x is either 0 or 1, are defined. These codepoints are called class-selector codepoints. Class- PHBs retain almost the same forwarding behavior as nodes that implement IP-precedence based classification and forwarding. For example, packets with a DSCP value of `110000' (IP-precedence 110) have a preferential forwarding treatment (scheduling, queuing, etc.) as compared to packets with a DSCP value of `100000' (IP-precedence 100). These PHBs ensure that DS-compliant nodes can co-exist with IP-precedence aware nodes, with the exception of the DTS bits.

2.2.2.2.3 Expedited Forwarding PHB (Defined in RFC-2598)

Expedited Forwarding (EF) PHB is the key ingredient in DiffServ for providing a low-loss, low-latency, low-jitter, and assured bandwidth service. Applications such as VoIP, video, and online trading programs require a robust network-treatment. EF can be implemented using priority queuing, along with rate limiting on the class (formally, a BA). Although EF PHB when implemented in a DiffServ network provides a premium service, it should be specifically targeted toward the most critical applications, because if congestion exists, it is not possible to treat all or most traffic as high priority. The recommended DSCP value for EF is `101110' (RFC-2474).

2.2.2.2.4 Assured Forwarding PHB (Defined in RFC-2597)

The rough equivalent of the IntServ controlled load service is the Assured Forwarding PHB (AFxy). It defines a method by which BAs can be given different forwarding assurances. For example, traffic can be divided into gold, silver, and bronze classes, with gold being allocated 50 percent of the available link bandwidth, silver 30 percent, and bronze 20 percent. The AFxy PHB defines four AFx classes: AF1, AF2, AF3, and AF4. Each class is assigned a certain amount of buffer space and interface bandwidth, dependent on the SLA with the Service Provider/policy. Within each AFx class, it is possible to specify 3 drop precedence values. If there is congestion in a DS-node on a specific link, packets will get dropped according to their precedence. In our example, packets in AF13 will get dropped before packets in AF12, before packets in AF11. This concept of drop precedence is useful, for example, to penalize flows within a BA that exceed the assigned bandwidth. Packets of these flows could be re-marked by a policer to a higher drop precedence. [72]

DROP Precedence	**Class #1**	**Class #2**	**Class #3**	**Class #4**
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Table. DiffServ Assured Forwarding Code point Table

A DS-domain is made up of DS ingress nodes, DS interior nodes (in the core), and DS egress nodes. An ingress or egress node might be a DS boundary node, connecting two DS domains together. Typically, the DS boundary node performs traffic conditioning. A traffic conditioner typically classifies the incoming packets into pre-defined aggregates, meters them to determine compliance to traffic parameters (and determines if the packet is in profile, or out of profile), marks them appropriately by writing/re-writing the DSCP, and shapes (buffers to achieve a target flow rate) or drops the packet in case of congestion. Figure illustrates the typical traffic conditioner at the edge of a DS-domain.

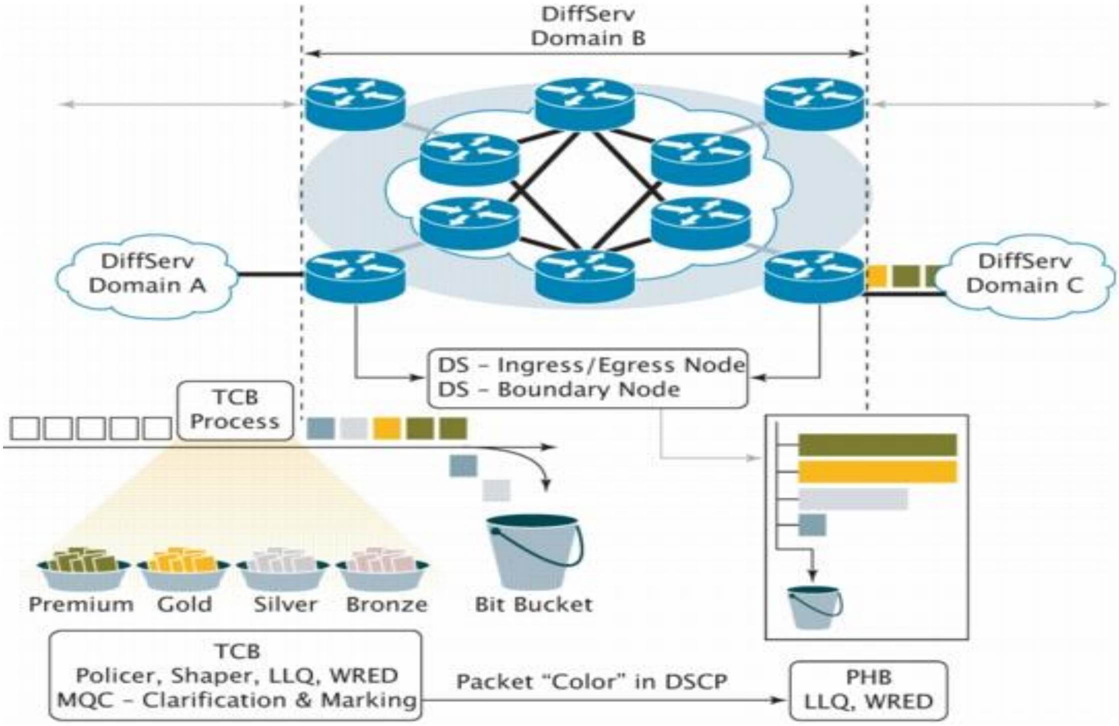

Fig 7: DiffServ domain working model

A DS Internal node enforces the appropriate PHB by employing policing or shaping techniques, and sometimes re-marking out of profile packets, depending on the policy or the SLA.

2.2.2.1 *DiffServ Issues-The Challenges:*

DiffServ enables scalable and coarse-grained QoS throughout the network and has some drawbacks. Some of the challenges for tomorrow and opportunities for enhancements and simplification of QoS delivery in an Internetwork are:

i) **Provisioning**-Unlike RSVP/IntServ, DiffServ needs to be provisioned. Setting up the various classes throughout the network requires knowledge of the applications and traffic statistics for aggregates of traffic on the network. This is very time consuming.

ii) **Billing and Monitoring**-Management is still a big issue. Even though packets/sec, bytes/sec, and many other counters are available via the class-based Management Information Base (MIB), billing and monitoring are still difficult issues.

iii) **Loss of Granularity**-Even though QoS assurances are being made at the class level, it may be necessary to drill down to the flow-level to provide the requisite QoS. For example, although all HTTP traffic may have been classified as gold, and a bandwidth of 100Mbps assigned to it, there is no inherent mechanism to ensure that a single flow does not use up that allocated bandwidth. [71]

iv) **QoS and Routing**-One of the biggest drawbacks of both the IntServ and DiffServ models is the fact that signaling/provisioning happens separately from the routing process. There may exist a path (other than the non-default Interior Gateway Protocol [IGP], such as OSPF, ISIS, EIGRP, and so on or Exterior Gateway Protocol [EGP], such as BGP-4, path) in the network that has the required resources, even when RSVP/DiffServ fails to find the resources. This is where Traffic Engineering (TE) and MPLS come into service. True QoS, with maximum network utilization, will arrive with the combination of traditional QoS and routing.[64]

2.3 Adaptive Multimedia Delivery

Many research efforts have been devoted to utilize cloud infrastructure to re-design the system architecture, and improve the QoE of media services. For instance, cloud offloading can enhance the capability of mobile devices; mitigating traditional video services to the cloud infrastructure can greatly reduce the operation cost; video over the future networks has already shown great potential of improving Quality of Experience. Time-series analysis techniques can be utilized for the prediction of a server bandwidth demand, while P2P content delivery can support the process in video-on-demand (VoD) services [11]. For the prediction of the future population of each video channel, Box-Jenkins models [12] are possible to exploit with input on the population of the video channel in the past. The auto-regressive

Moving average (ARMA) model and seasonal auto-regressive integrated moving average (ARIMA) model can also be exploited to avoid periodicity.

In this framework, social media networks play a significant role in multimedia content delivery by providing ways of interactions among users that can lead to lightning fast spread of content [13]. A probabilistic resource provisioning approach is suggested in [14] that uses standard models developed for epidemiology spreading to represent sudden and intense workload overflow in the VoD delivery process. Epidemic model spreading in scale-free networks has also been intensively studied in [15] with the main effort on the spread of computer viruses, which really resembles the epidemic spread of human diseases.

In [16], it is investigated a novel framework to distribute video files with the aid of other proximity devices based on D2D communication. Mobile users are clustered into different groups and heads to minimize the expected D2D communication transmission distance. Extensive simulation results demonstrate the efficiency of proposed scheme.

In [17] it is proposed an SDN-enabled cloud mobile distribution architecture and developed a joint video placement, to improve user experience and reduce the system operational cost. This problem is formulated into a mixed integer programming problem and solved by dual decomposition. Simulation results show that the proposed strategy can effectively cut down the total cost and guarantee user QoS.

LQA performs the adaptation by adjusting the number of video quality layers transmitted to the viewers and consequently their expected perceived quality levels depending on their available network bandwidth resources. Cross layer adaptive video delivery methods are more efficient in terms of delivery-related information gathering and processing and tend to achieve higher user perceptual quality for the remotely watched multimedia content. A good survey of these solutions can be found in [18].

An analytical model for end-to-end rich multimedia services delivered in network virtualization environments that can be used to determine end-to-end bandwidth and delay performance bounds in virtual network has been presented in [19]. This paper [20] has introduced a novel Mobile Multi-source High Quality Multimedia Delivery Scheme (M3QD) for multimedia content distribution to mobile users over hybrid ad-hoc and infrastructure-based wireless networks. The proposed solution is based on a suite of algorithms which support high quality content delivery while enabling user mobility. M3QD and its algorithms are evaluated using both simulations and subjective tests. M3QD's performance is compared with that of a single source multimedia delivery scheme in different scenarios when delivering various multimedia content clips.

2.3 Adaptive Multimedia Provision

In terms of achieving high levels of user satisfaction with the multimedia delivery service, of paramount importance is supporting high multimedia quality levels.

Follow and match the available network bandwidth is required. If short term variations can be overcome by using buffering techniques [3], for long time-scale network dynamics, rate adaptation techniques are among the most efficient solutions. Initially, basic loss and delay-based adaptive streaming solutions were proposed at network transport layer including the TCP-Friendly Rate Control Protocol (TFRCP) [4] and the enhanced Loss-Delay-based Adaptation algorithm (LDA+) [5]. These solutions present a reasonable performance in terms of Quality of Service (QoS), but their major drawback is a poor correlation with the actual end-user perceived quality. Later on, more advanced adaptive delivery techniques which succeeded to maintain high levels of user perceived quality were developed at the application layer. Such a solution with good performance in terms of user perceived quality is the Layered Quality Adaptation algorithm (LQA) [6].

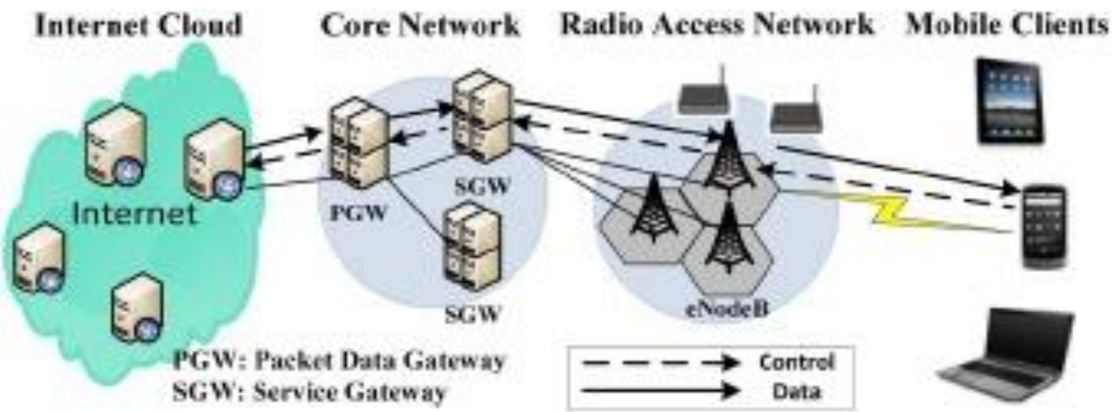

Fig 6: Media Entities of Network Environment

LQA performs the adaptation by adjusting the number of video quality layers transmitted to the viewers and consequently their expected perceived quality levels depending on their available network bandwidth resources. Cross layer adaptive video delivery methods are more efficient in terms of delivery-related information gathering and processing and tend to achieve higher user perceptual quality for the remotely watched multimedia content. A good survey of these solutions Can be found in [7].

The Quality Oriented Adaptation Scheme (QOAS) [8] involves user perceived quality estimations in the feedback based multimedia adaptation process. As it is quality-oriented, QOAS shows significant improvements in end-user perceived quality when used for streaming multimedia content in both wired and wireless networking environments. Diverse techniques were proposed for adaptive multimedia transmissions over wireless access or ad-hoc networks.

Among the proposed solutions are adaptation mechanisms at the level of layers [9] or objects [10], transmission protocols [11], fine-granular scalability schemes [12] and perception-based approaches [13]. An analytical model for end-to-end rich multimedia services delivered in network virtualization environments that can be used to determine end-to-end bandwidth and delay performance bounds in virtual network has been presented in [14]. Both theoretical analysis and experimental results have demonstrated the applicability of the model for delivery of multimedia in various heterogeneous networking systems. Region of Interest (RoI)-based adaptive schemes have been proposed including the ones introduced in [15]–[17]. These solutions treat different parts of the overall image area distinctly in the adaptation process based on the user level of interest. More recently, Ruckert *et al*. [18] have proposed a quality adaptation scheme in peer-to-peer Scalable Video Coding (SVC)-based video streaming based on objective QoE metrics. The proposed adaptation strategies increase or decrease the video quality by selecting different coding layers during the video delivery in order to result in the highest QoE possible. Bit Detect [19] is a multimedia adaptation mechanism which uses objective video quality assessment metrics such as PSNR and SSIM to
recommend specific video bitrate levels. A video delivery solution which employs network selection and balances energy consumption and video quality was described in [20], whereas the video distribution mechanism described in [21] performs Energy-quality and cost trade-
off. Khan *et al.* [22] have introduced a QoE driven adaptation scheme for video delivery over wireless networks, which employs a reference-free QoE model.This model estimates user QoE impact based on encoding frame rate, sender bit rate and packet error rate and informs the adaptation process. Cao *et al.* [11] have proposed a highly innovative ant behavior-inspired solution for video delivery in wireless mobile networks based on creation and management of minicommunities. For many years, server side streaming and multimedia adaptation have been proposed by different researchers [23]. Recently, HTTP adaptive streaming (HAS) has been introduced
In different forms including Adobe's HTTP Dynamic Streaming (HDS) [24], Microsoft's HTTP Smooth Streaming (HSS) [25], and Apple's HTTP Live Streaming (HLS) [26] and is used for diverse Internet video applications such as YouTube. This new streaming approach requires the division of the video content in multiple quality level chunks, which are short video segments. The network

condition (e.g., available bandwidth) and/or buffer status or other parameters are analyzed at the client side and requests for video segments at appropriate quality levels are sent to the server, which delivers them. The existence of multiple quality video sequences enables better adaptation to the user demand and network conditions, higher bandwidth utilization and fewer interruptions in the multimedia playback. Oyman and Singh [27] have presented an overview of HAS and HAS-specific cross-layer adaptation algorithms, which rely on tight integration of the HAS/HTTP-specific media delivery with network-level and radio-level adaptation and QoS schemes to determine optimum application, transport, network and radio configurations considering link, device and content, in order to result in highest possible user QoE. A comprehensive survey on current HTTP adaptive streaming solutions and QoE of HTTP adaptive streaming is presented in [28].

2.4 Cloud-Based Multimedia Content Delivery

In the context of this paper the mobile cloud is represented by the hybrid ad-hoc and infrastructure-based wireless network and the corresponding software components part of the system architecture.

A multimedia-aware cloud-based solution was introduced in [29]. The authors perform distributed multimedia processing and storage and provide quality of service (QoS) provisioning for remote multimedia service users. The cloud-based software architecture for multimedia collaboration introduced in [30] allows users to perform video conferencing, while also viewing shared media content in real-time. Load balancing for cloud-based multimedia systems discussed in [31] considers the load of all servers and network conditions and targets optimal resource allocation and scheduling. A Personalized DTV Program Recommendation (PDPR) system deployed on a cloud computing environment is proposed in [32]. PDPR analyses the viewing pattern of users to personalize program recommendations, and to efficiently use computing resources. User QoE impact based on encoding frame rate, sender bit rate and packet error rate and informs the adaptation process. Cao *et al.* [11] have proposed a highly innovative ant behavior-inspired solution for video delivery in wireless mobile networks based on creation and management of mini communities. For many years, server side streaming and multimedia adaptation have been proposed by different researchers [23]. Recently, HTTP adaptive streaming (HAS) has been introduced in different forms including Adobe's HTTP Dynamic Streaming

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	Mobile Cloud Storage	**Audio/Video Streaming**	**Interactive Services**	**Cloud based Rendering**	**Media Analytics**
Sample applications /services enabled on mobile devices	Storing and accessing Photos, Music, Files	Streaming audio, video; Cloud DVR	Video Chat; Remote Desktop; Interactive advertisements	Mobile Gaming; Augmented Reality; Telemedicine	Differentiated Services/Billing; Personalized services
IaaS, PaaS features needed	Cloud Storage with high availability and integrity	Cloud Transcoding, Transrating, Caching	Cloud Transcoding, Transrating	Multi-core GPUs; Efficient cloud rendering;	Cloud media usage/QoE probes; Media classification engines
Advantages	Ubiquitous access from any device; Synchronizing between devices	Low capex; High scalability with demand	Easier support for multiple devices/platforms	Enables highest quality rendering; Multi-player, multi-platform	Unified analytics for media usage across devices and networks
Challenges	Ensuring content security, privacy; Additional wireless traffic	Cloud service cost; Cloud energy, cooling costs	Response time; Video quality; Cloud service cost; Cloud energy, cooling costs	Response time; User experience; Cloud service cost; Additional wireless traffic; Cloud energy, cooling costs	Data protection; privacy

Fig/Table: 7 *Classification Mechanism of different entities*

Which are short video segments. The network condition (e.g., available bandwidth) and/or buffer status or other parameters are analyzed at the client side and requests for video segments at appropriate quality levels are sent to the server, which delivers them. The existence of multiple quality video sequences enables better adaptation to the user demand and network conditions, higher bandwidth utilization and fewer interruptions in the multimedia playback. Oyman and Singh [27] have presented an

overview of HAS and HAS-specific cross-layer adaptation algorithms, which rely on tight integration of
the HAS/HTTP-specific media. A comprehensive survey on current HTTP adaptive streaming solutions and QoE of HTTP adaptive streaming is presented in [28].

2.5 Multimedia Provision

In the context of this paper the mobile cloud is represented by the hybrid ad-hoc and infrastructure-based wireless network and the corresponding software components part of the system architecture, as described in Section III. In this context, various architectures, frameworks and algorithms have been proposed to provide efficient, flexible and high quality multimedia services to end users. A multimedia-aware cloud-based solution was introduced in [29]. The authors perform distributed multimedia processing and storage and provide quality of service (QoS) provisioning for remote multimedia service users. The cloud-based software architecture for multimedia collaboration introduced in [30] allows users to perform video conferencing, while also viewing shared media content in real-time. Load balancing for cloud-based multimedia systems discussed in [31] considers the load of all servers and network conditions and targets optimal resource allocation and scheduling. A Personalized DTV Program Recommendation (PDPR) system deployed on a cloud computing environment is proposed in [32]. PDPR analyses the viewing pattern of users to personalize program recommendations, and to efficiently use computing resources.
Table I summarizes the different categories of mobile multimedia applications that already are, or can potentially be, driven by the use of the cloud, including storage, download and synchronization applications, audio and video streaming applications, interactive applications like multi-way video conferencing, interactive advertisements, and mobile remote desktop, rich rendering based applications like mobile multi-user gaming and augmented reality, and cloud based media analytics that will provide better understanding of user preferences and experiences, and drive personalized mobile services.

Chapter III

3. Methodology

3 Start-up Lines:

Internet has become an integral part of our lives; we can observe it in any aspect of our lives business, education, entertainment and much more. Businesses use Internet and web related technologies to help streamline business processes and development of new business models. Behind all this success there is Internet Protocol (IP). Increasing popularity of IP change the paradigm from “IP over everything” to “everything to IP”. In order to support and manage multitude of applications such as streaming video, Voice over IP (VoIP), e-commerce, Enterprise Resource Planning (ERP), and others, to handle this a network requires Quality of Service(QoS) more than best effort. Different application needs different requirements in order to throughput, delay, reliability and jitter. For Example VoIP requires very low jitter and need guaranteed bandwidth, on the other hand any application depend on ftp does not suffer from jitter but packet loss will be consider very important.

3.1 The Mechanisms:-

DiffServ use some mechanisms in order to implement PHB’s. with the use of these mechanisms, DiffServ works in very efficiently manners. That mechanisms are briefly explained.

3.1.1 Classification: A classifier is a process in the border node where traffic enters (ingress node), which looks at each packet’s header and decides what kind of forwarding class it should be assigned to. There are two kinds of classifiers:

i) **The *Behavior Aggregate classifier* (BA)**:, which only looks at the DSCP value (Differentiated Services Code Point. This is the case if the packets have already been classified by perhaps a Differentiated Services-enabled application.

ii) **The *Multi-field classifier* (MF),** looks at several header fields in the IP packet. It usually uses a combination of fields like port number, protocol ID, Source/destination addresses to decide what class it should belong to.

3.1.2 Metering: Normally, the customer will have a *Service Level Agreement* (SLA) with the ISP. The SLA describes how much traffic that will be allowed guaranteed passage. Any traffic more than the agreement will be dealt with, for instance given the standard best effort attributes. The customer might transmit data in a higher speed than agreed, for example transmitting video-streams with 500 kbps instead of the agreed 400. That means that One out of five packets is "out of profile" and must therefore be dealt with.

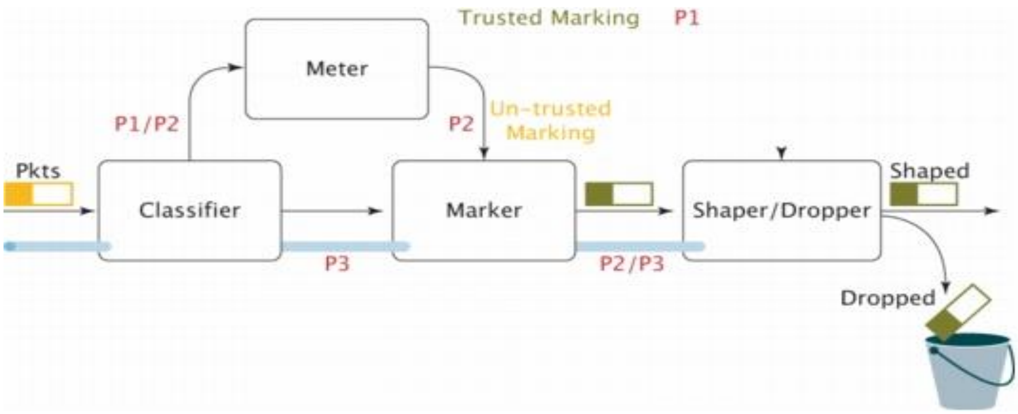


Fig 8: Classification mechanism of DiffServ Domain

3.1.3 Marking: After packets get classified and metered at the ingress node, they are ready to be assigned a forwarding class. The procedure is called marking and is necessary to enable the internal nodes of a DS domain to know what kind of packet it is, along the principle of Differentiated Services. The marking is saved in the DS field in each IP packet.The DSCP is the binary representation of a *Per Hop Behavior, PHB.* These are the rules that tell the router what kind of treatment the packet needs. The DSCP is the binary translation of a certain PHB, for instance (this is in binary) <100 100> is the DSCP for the PHB called "AF42", Assured Forwarding class 4 drop precedence 2. These mappings are assigned according to a local standard. It is up to anyone to decide which values to use, but it is recommended that the IETF standard mapping is used.

3.1.4 Conditioning and Shaping: The result of an input overflow could be to still forward traffic as requested, but to raise the customer's bill accordingly, or to assign it a lower class, or to simply drop it. These actions are called **conditioning**. An alternative to conditioning, the same thing actually, depending on where or how you look at it, is called **shaping**. Whenever packets enter a device faster than they can exit it, such as with speed mismatches, then a point of congestion, or bottleneck, can occur.

3.1.5 Queuing Algorithms: Devices have buffers that allow for scheduling higher-priority packets to exit sooner than lower priority ones, which is commonly called queuing.

i) **Low Latency Queuing (LLQ),** which provides strict priority servicing and is intended for real-time applications such as VoIP;

ii) **Class-Based Weighted Fair Queuing (CBWFQ),** which provides bandwidth guarantees to given classes of traffic and fairness to discrete traffic flows within these traffic classes.

3.1.6 Congestion Avoidance Mechanisms: Selective dropping of packets when the queues are filling is referred to as **congestion avoidance**. Congestion avoidance mechanisms work best with TCP-based applications. Congestion avoidance mechanisms are complementary to queuing algorithms. Queuing algorithms manage the front of a queue while congestion avoidance mechanisms manage the tail of the queue. These mechanisms provide rules for sensitive traffic have a certain priority over other traffic. Techniques to avoid congestion, on the other hand, monitor the flow of network traffic in order to anticipate and minimize its effect.

i) **Random Early Detection (RED)***:* It monitors the size of the queue and when it reaches a certain threshold, it randomly selects individual TCP flows which drop packets in order to indicate the sender must reduce the rate of transmission.

ii) **Weighted Random Early Detection (WRED):** It combines the capabilities of the RED algorithm with IP precedence.

3.2 Comparison in terms of QoS: Quality of service is handled in a similar manner. Originally, IPv4 quality of service was defined through the use of the type of service and IP precedence bits in the IP header. These bits went largely unused until they were redefined as differentiated service bits, with all eight bits available for quality of service definition instead of being split into two fields.

Differentiated service is now implemented in a greater number of IPv4 networks than in the past because of the increasing penetration of voice over IP (VoIP) and other quality of service-sensitive technologies. In IPv6, a similar IP header field, known as traffic class, is slated to be used in an identical manner to the current differentiated service bits.

Although some IPv6 proponents will note the existence of a new flow label field in the IPv6 header, this field should have little, if any, impact on quality of service functionality. At present, the flow field's utility is unclear and may be utilized on a supplier-by-supplier basis or not used at all. Some rfc'c try to define to this field but many are contradict. RFC 3697 define the general rules for it and some specifications.

i. **Flow Label Evaluation:** New field in IPV6 header define how traffic is handle or identified. Traffics identification using flow label allows router to identify and provide special handling for the packets that belong to same flow. Flow is sequence of packets move toward a particular direction.
ii. **Flow Label definition:** Flow label is a field in the IPv6 main header that has been designed in order to facilitate per-flow QoS treatment. General rules for the Flow Label field were proposed in RFC 3697, but specific use cases have not been described yet. It mainly has the following specifications.
iii. **Flow Label Features:**
    a. The 20-bit Flow Label field is used by a source to label packets of a flow and the zero value is used to indicate packets that are not part of any flow.
    b. The Flow Label value set by the source must be delivered unchanged to the destination node.
    c. Each established flow should expire in 120s when the flow is idle in order to improve the performance of routers.
    d. A flow can be uniquely identified with source address, destination address and Flow Label.
    e. Flow state needs to be established on all or a subset of the IPv6 nodes on the path from the source to the destination(s).

3.3 Status of the IPV6 Specific QoS Standardization:

The flow label specification in RFC 3697 has only defined general rules, some of which may lead to scalability problem in practical implementation. Up to now, various proposals (6436,6437) have been made to the IETF to define the 20bits of flow label field. These proposals make some modifications on the initial definition in RFC 3697 to some extend. In literature, authors mainly focus on applying FL to some specific technology. But now latest RFC on this field is 6436 clarify about specification and add some flexibility also it’s update is 6437 in which Flow Label specifications and minimum requirements for IPV6 nodes labeling flows, IPV6 nodes forwarding labeled packets and flow state establishment methods are specified.

The main disputable points are listed in the following.

i) Whether flow label should be with end-to-end meaning or just with local meaning? A FL with end-to-end meaning can simplify the FL maintenance process; while a local meaning FL can be more flexible and scalable.

ii) Whether the value of flow label can be changed in the intermediate nodes? It should be in accordance with the above property of FL.

iii) Whether the value of flow label should be totally random or with some specific meanings? A totally random FL is rather simple when allocating while a FL with some specific meanings can sometimes help QoS provision.

iv) A flow should be identified by FL only or with source address, destination address and FL? One dimensional flow identification process can accelerate flow-based QoS treatment, but the 20bit-length will normally be not enough for per-flow allocation.

3.4 Advantages of the Flow Label field:

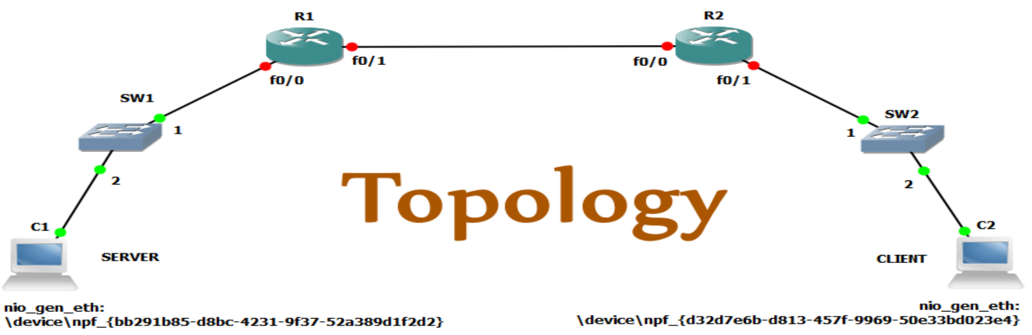
The use of the flow label field also solves the problem of violation of layer in which a router needs access to the transport-layer protocol or application to process the packets instead of using only the data from the network layer. The main advantages of using the flow label field for packet classification are as follows:

i) The use of the flow label field reduces the average processing load of the routers in the network, and therefore, reduces end-to-end delays of the packets.

ii) The reservation of resources through the Flow Label reduces the problems caused by frequent route changes.

iii) The flow label field has the potential to facilitate the implementation of mechanisms for flow routing based on QoS.

3.5 IMPLEMENTATION PHASE: MAKING TOPOLOGY IN GNS3



3.5.1 IMPLEMENTATION OF DIFF SERVE ON ROUTER CONFIGURATION FILE

!	!
!	!
version 12.4	version 12.4
service timestamps debug datetime msec	service timestamps debug datetime msec
service timestamps log datetime msec	service timestamps log datetime msec
no service password-encryption	no service password-encryption
!	!
hostname R5	hostname R5
!	!
boot-start-marker	boot-start-marker
boot-end-marker	boot-end-marker
!	!
!	!
no aaa new-model	no aaa new-model
memory-size iomem 5	memory-size iomem 5
no ip icmp rate-limit unreachable	no ip icmp rate-limit unreachable
ip cef	ip cef

!	!
no ip domain lookup	no ip domain lookup
!	!
multilink bundle-name authenticated	multilink bundle-name authenticated
!	!
archive	archive
log config	log config
hidekeys	hidekeys
!	!
ip tcp synwait-time 5	ip tcp synwait-time 5
!	!
class-map match-any AF43	class-map match-any AF43
match ip dscp af43	match ip dscp af43
class-map match-any gold	class-map match-any gold
match ip dscp af11 af12 af13	match ip dscp af11 af12 af13
class-map match-any AF42	class-map match-any AF42
match ip dscp af42	match ip dscp af42
class-map match-any AF41	class-map match-any AF41
match ip dscp af41	match ip dscp af41
class-map match-any EF	class-map match-any EF
match ip dscp ef	match ip dscp ef
class-map match-any AF12	class-map match-any AF12
match ip dscp af12	match ip dscp af12
class-map match-any AF21	class-map match-any AF21
match ip dscp af21	match ip dscp af21
class-map match-any AF13	class-map match-any AF13
match ip dscp af13	match ip dscp af13
class-map match-any AF31	class-map match-any AF31
match ip dscp af31	match ip dscp af31
class-map match-any AF23	class-map match-any AF23
match ip dscp af23	match ip dscp af23
class-map match-any AF32	class-map match-any AF32
match ip dscp af32	match ip dscp af32

class-map match-any AF11	class-map match-any AF11
match ip dscp af11	match ip dscp af11
class-map match-any AF22	class-map match-any AF22
match ip dscp af22	match ip dscp af22
class-map match-any AF33	class-map match-any AF33
match ip dscp af33	match ip dscp af33
class-map match-any bronze	class-map match-any bronze
match ip dscp af31 af32 af33	match ip dscp af31 af32 af33
class-map match-any platinum	class-map match-any platinum
match ip dscp af41 af42 af43	match ip dscp af41 af42 af43
class-map match-any silver	class-map match-any silver
match ip dscp af21 af22 af23	match ip dscp af21 af22 af23
class-map match-all best-effort	class-map match-all best-effort
match ip dscp default	match ip dscp default
class-map match-any premium	class-map match-any premium
match ip dscp ef	match ip dscp ef
!	!
!	!
policy-map STREAM	policy-map STREAM
class premium	class premium
priority 500	priority 500
class gold	class gold
bandwidth percent 33	bandwidth percent 33
class silver	class silver
shape average 320000	shape average 320000
bandwidth percent 22	bandwidth percent 22
class bronze	class bronze
bandwidth percent 12	bandwidth percent 12
class best-effort	class best-effort
police 56000 1750 1750 conform-action set-dscp-transmit 0	police 56000 1750 1750 conform-action set-dscp-transmit 0
policy-map SETDSCP	policy-map SETDSCP
class EF	class EF

set ip dscp ef	set ip dscp ef
class AF11	class AF11
class AF12	class AF12
set ip dscp af11	set ip dscp af11
class AF13	class AF13
set ip dscp af12	set ip dscp af12
class AF21	class AF21
set ip dscp af21	set ip dscp af21
class AF22	class AF22
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class AF31	class AF31
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set ip dscp af33	set ip dscp af33
class AF41	class AF41
set ip dscp af41	set ip dscp af41
class AF42	class AF42
set ip dscp af42	set ip dscp af42
class AF43	class AF43
set ip dscp af43	set ip dscp af43
!	!
interface FastEthernet0/0	interface FastEthernet0/0
ip address 1.1.1.2 255.0.0.0	ip address 2.1.1.2 255.0.0.0
duplex auto	duplex auto
speed auto	speed auto
service-policy input SETDSCP	service-policy output STREAM
!	!
interface FastEthernet0/1	interface FastEthernet0/1
ip address 2.1.1.1 255.0.0.0	ip address 3.1.1.1 255.0.0.0

duplex auto speed auto service-policy output STREAM ! router rip version 2 network 1.0.0.0 network 2.0.0.0 no auto-summary ! ip forward-protocol nd ! ! no ip http server no ip http secure-server ! access-list 101 permit udp any any range 16384 32768 access-list 102 permit tcp any any eq tacacs access-list 104 permit tcp any any eq www access-list 105 permit ip any any access-list 108 permit tcp any any eq telnet access-list 109 permit tcp any any eq smtp access-list 110 permit tcp any any eq ftp ! control-plane ! line con 0 exec-timeout 0 0	duplex auto speed auto service-policy input SETDSCP ! router rip version 2 network 1.0.0.0 network 2.0.0.0 network 3.0.0.0 no auto-summary ! ip forward-protocol nd ! ! no ip http server no ip http secure-server ! access-list 101 permit udp any any range 16384 32768 access-list 102 permit tcp any any eq tacacs access-list 104 permit tcp any any eq www access-list 105 permit ip any any access-list 108 permit tcp any any eq telnet access-list 109 permit tcp any any eq smtp access-list 110 permit tcp any any eq ftp ! control-plane ! line con 0

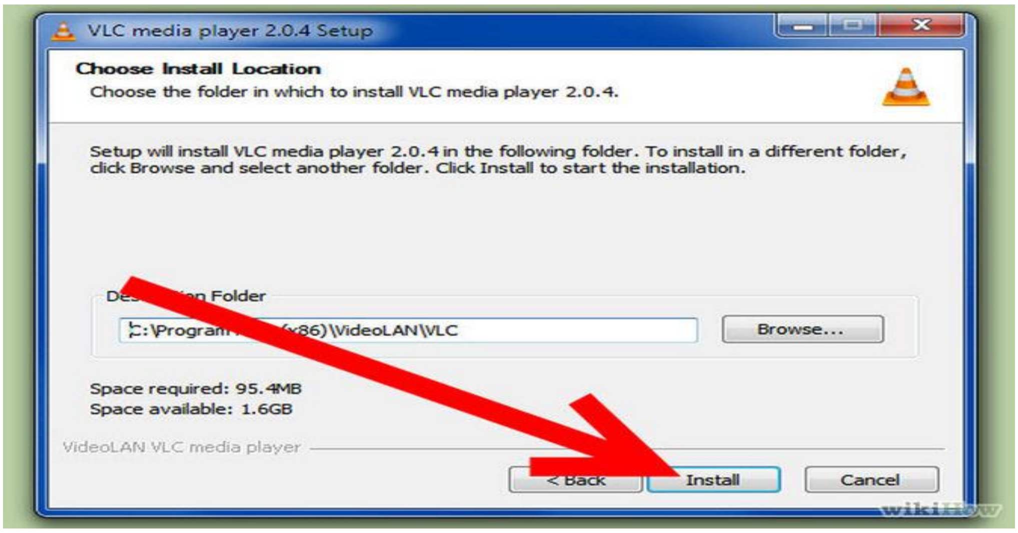
privilege level 15 logging synchronous line aux 0 exec-timeout 0 0 privilege level 15 logging synchronous line vty 0 4 login ! end	exec-timeout 0 0 privilege level 15 logging synchronous line aux 0 exec-timeout 0 0 privilege level 15 logging synchronous line vty 0 4 login ! end

3.5.2 How to Use VLC to Stream Audio and Video to Multiple Computers on Your Network

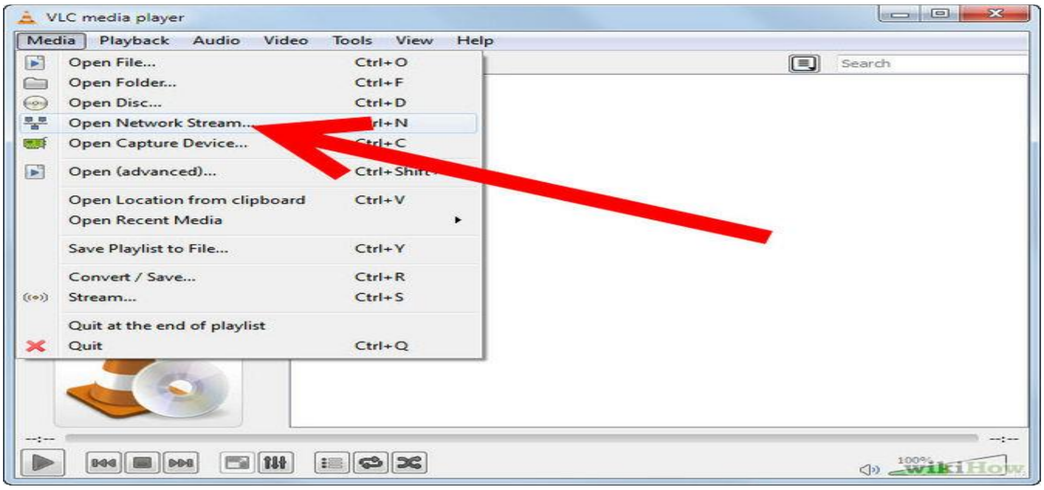
VideoLan media player (VLC) is an incredibly versatile media player available for Windows, Linux and other *Nix clones. It's also available for Mac, and gives you powerful options for advanced media controls and display. Using VLC makes it easy to stream audio and video using Multicast.

Steps

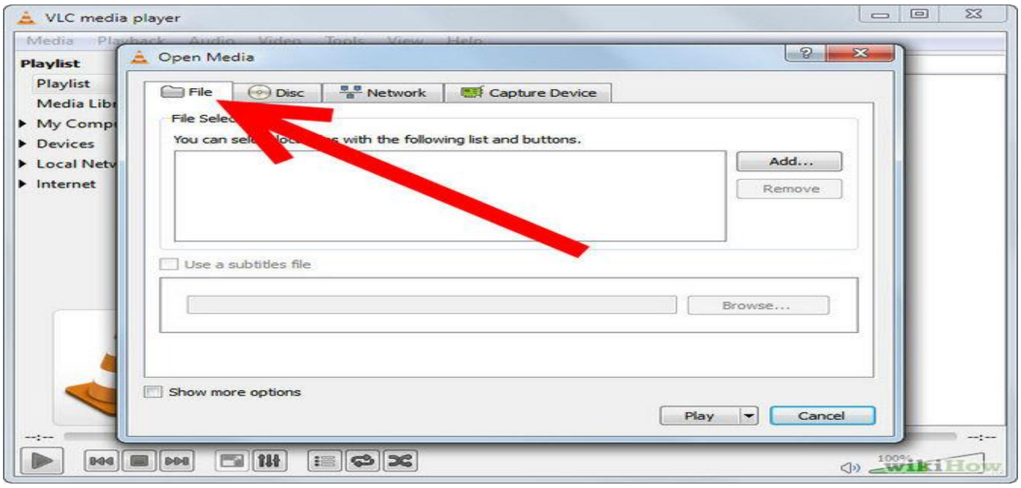
i) **Install VLC media player with full features.** When installation is complete, open the program.



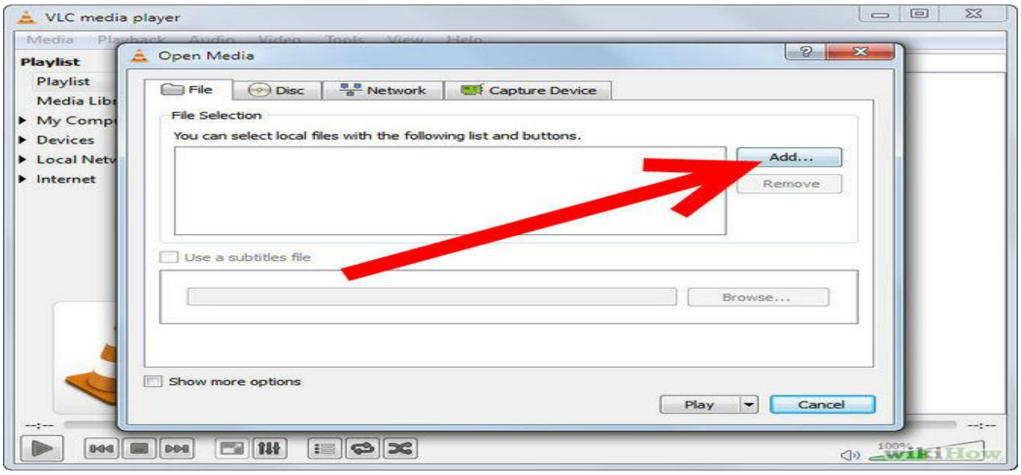
ii) In the Menu bar, click “Media” and “Open Network Stream”.



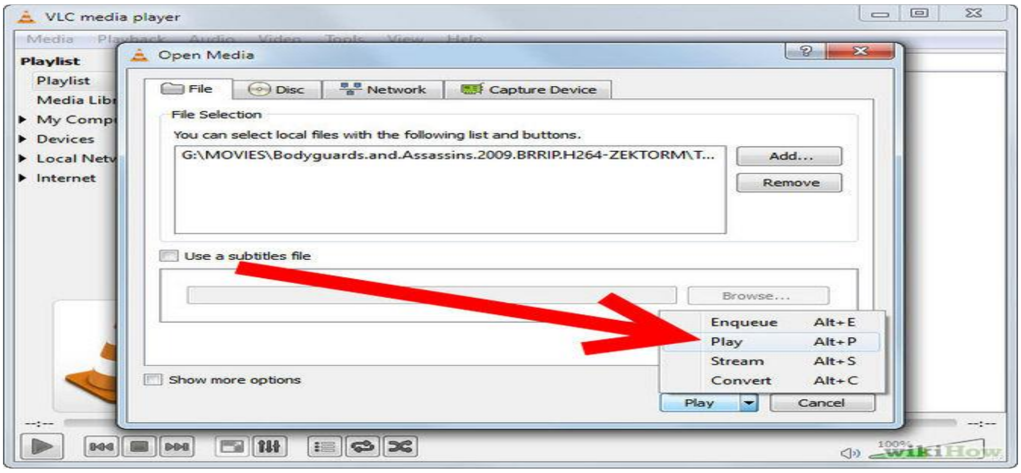
iii) In the Open Media windows, click on “File”.



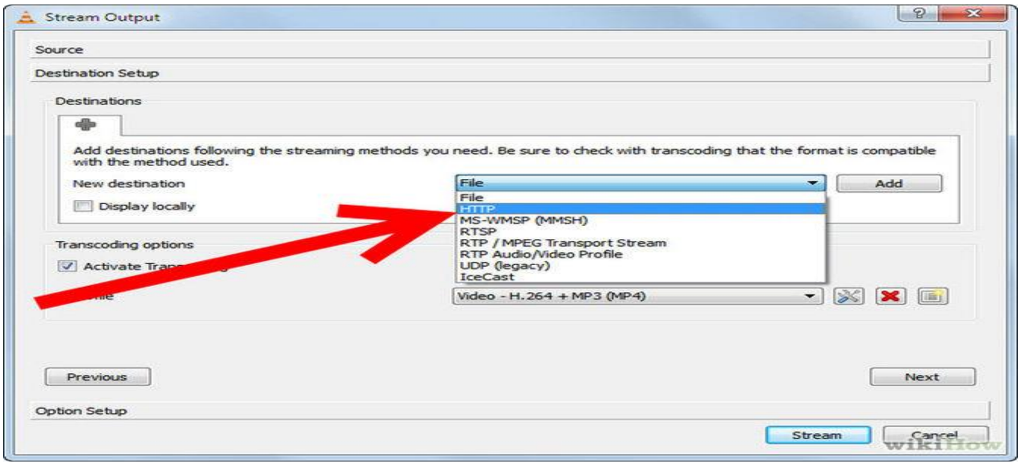
iv) Click "Add" and select the file you want to stream. Near the bottom of the screen, click the drop arrow next to "Play" and select "Stream."



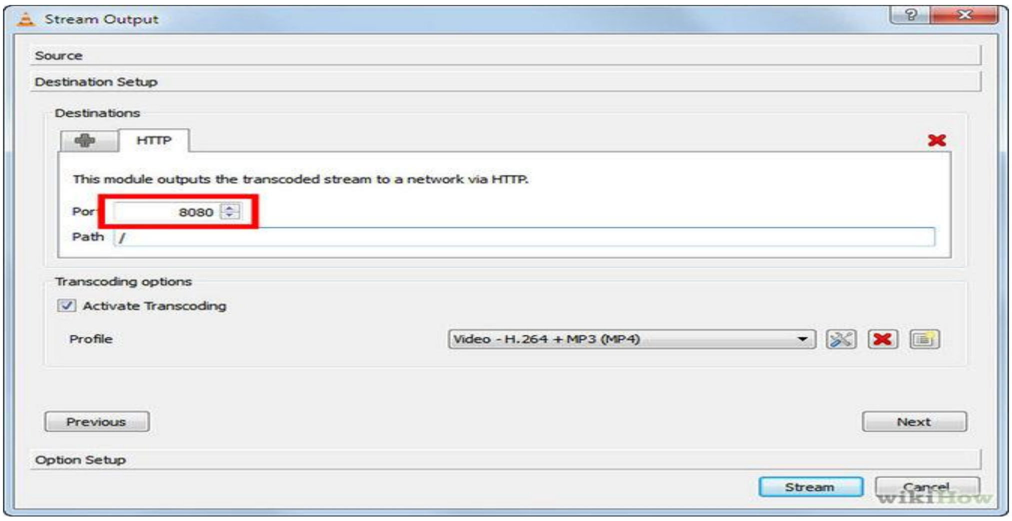
v) **Click on "Next".**



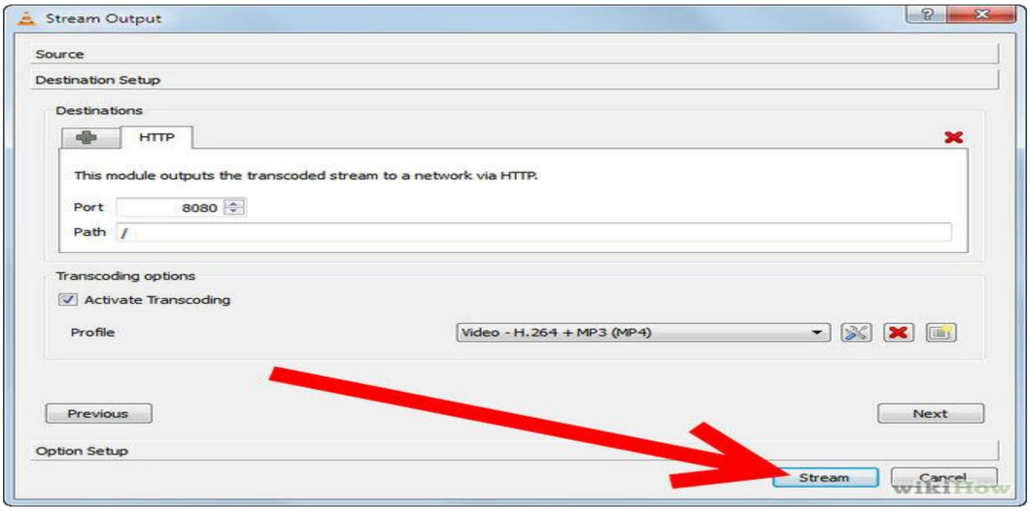
vi) **In the Destinations box, click on the drop menu and chose “HTTP.”** Click on “Add".



vii) **In the stream output window, make sure port number is 8080.** Check that no other software uses the port 8080.



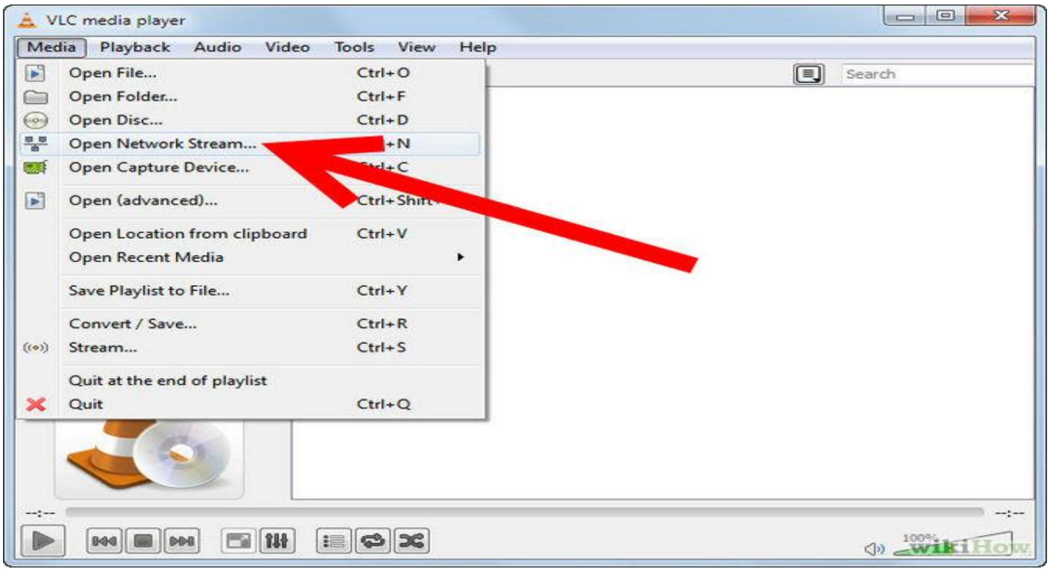
viii) **Click "Stream".**



ix) VLC Steaming is ready now.

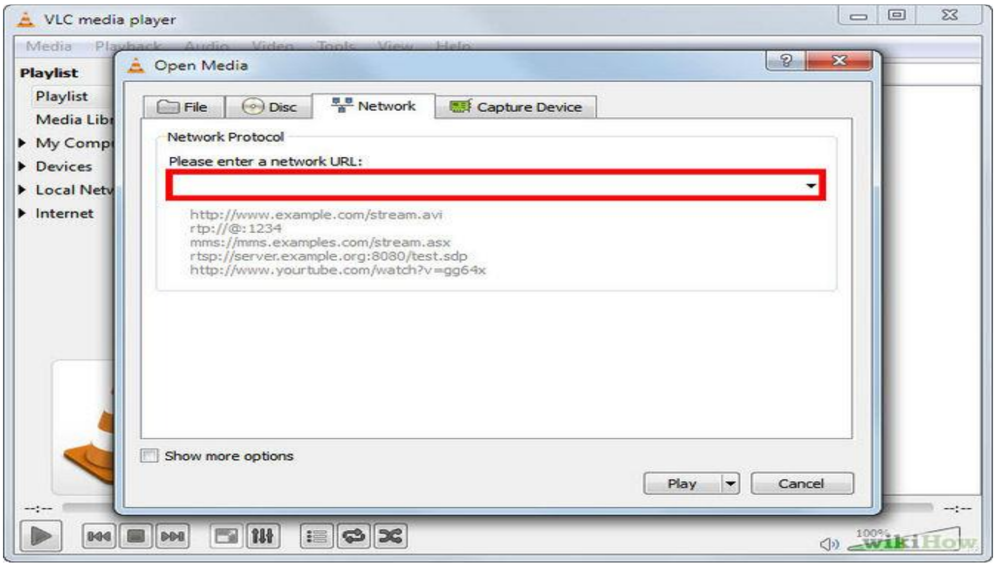
Streaming on a Network Client

i) **Open VLC media player, click on "Media," and select "Open Network Stream".**



ii) **In the "Network" tab, enter the IP address of the media server, as well as the port number.** Click "Play."

iii)



QoS Network Performance Measurement Network QoS is a measurement of how well the network operates and a means to define the characteristics of specific network services (B. Kalonov, 2002). My aim here is to analyse how efficiently IPv4 and IPv6 behave under these performance parameters below:

Delay

End-to-end transit delay is the elapsed time for a packet to pass from the sender through the network to the receiver. The higher the delay between the sender and receiver, the more insensitive the feedback loop becomes, and therefore, the protocol becomes less sensitive to short term dynamic changes in the network.

Jitter

The variation of end-to-end transit delay is called jitter. In packet switched networks jitter defines the distortion of the inter-packet arrival times compared to the inter-packet times of the packet transmission.

Throughput

Throughput is a measurement of the rate at which data can be sent through the network. It is similar to the concept of Speed-which is usually measured in bits per second (bps).

Packet loss

Normally when a device/path is overloaded and cannot accept any incoming data at a given time then packet loss occurs. Heavy packet loss has a great impact especially on video quality in either continuous or real time perspective.

Measurement

We tested the streaming over both protocols. We used IPv4 protocol and then IPv6 protocol. For both cases we captured packets from the server and client sides with the open source network packet analyzer called Wireshark1. Furthermore, We had to install WinPcap2 which is the standard tool for link-layer network access in the Windows environments: it can be used to capture and transmit packets.

By using the raw data we plotted graphs for analyzing my two protocols behavior. we used captured packets in our second phase for calculating throughput, packet delay and jitter. In the following section, We will be presenting out our results in detail.

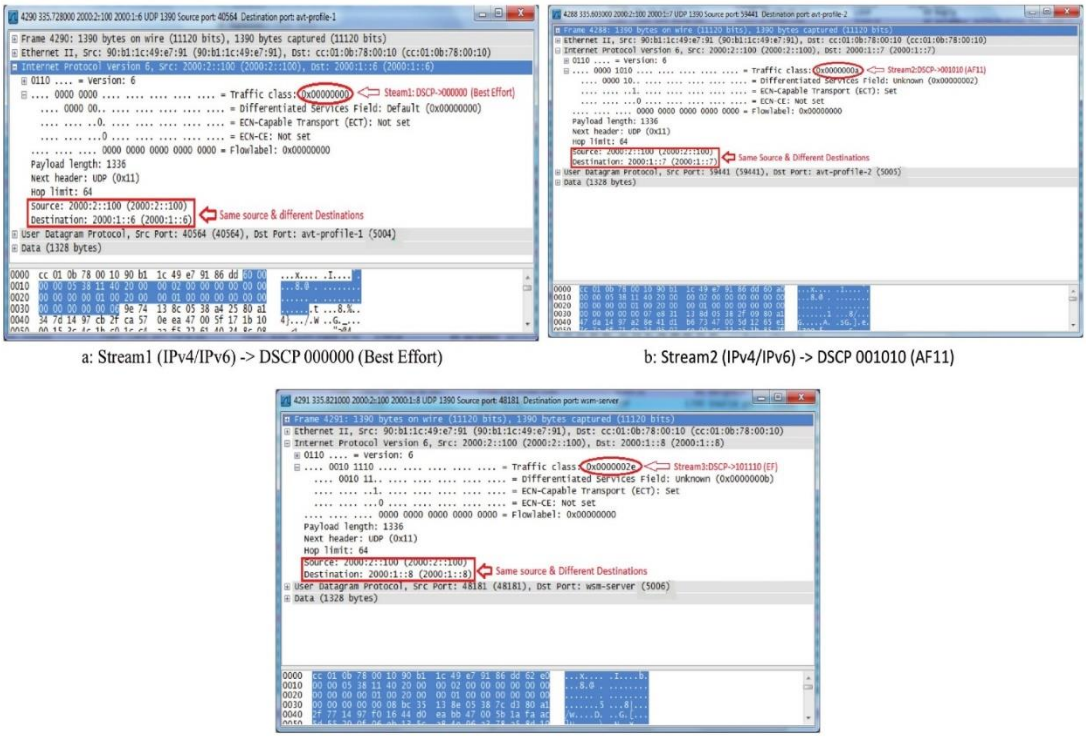
Observation

After doing the experiment over two test-bed configurations, we observed that the IPv6 performance was significantly better over the IPv4 network.

Delay

We measured the delay for both IPv4 and IPv6 protocol during the simulation. My main concern was to know how delay differentiates between these two protocols. The results are as follows:

3.5.3 Measurements in Wireshark(Packet Sniffing Tool)

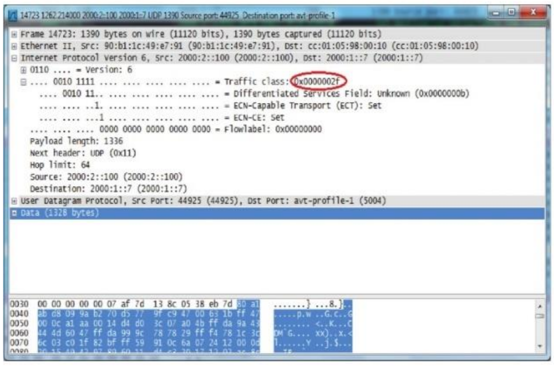


a: Stream1 (IPv4/IPv6) -> DSCP 000000 (Best Effort)

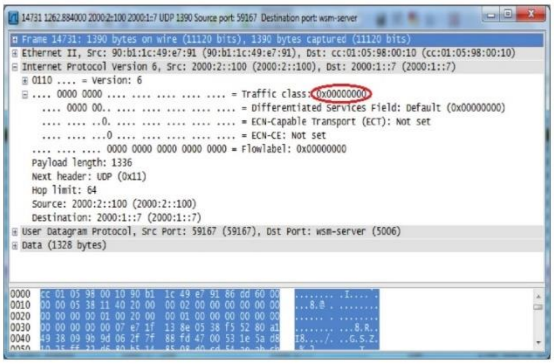
b: Stream2 (IPv4/IPv6) -> DSCP 001010 (AF11)

c: Stream3 (IPv4/IPv6) -> DSCP 101110 (EF)

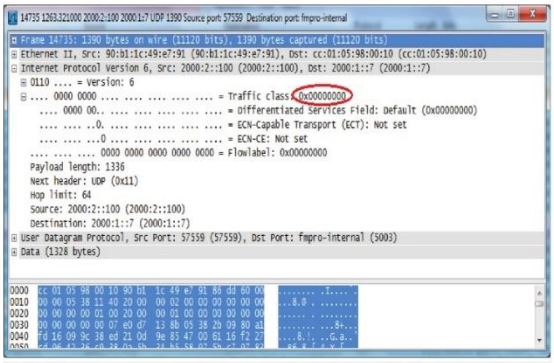
Figure7. Header Mappings



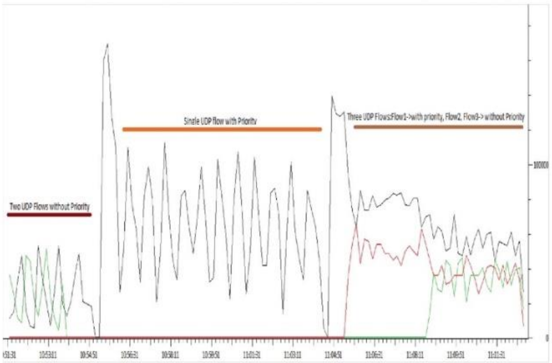
a: : One UDP Stream with Quality of Service (QoS)



b: One UDP Stream without Quality of Service (QoS)



c: One UDP Stream without Quality of Service (QoS)



d: One UDP Streams with Quality of Service (QoS) and two UDP Streams without Quality of Service (QoS) Result

Figure 6 Hybrid UDP Streams with & without Quality of Service (QoS)

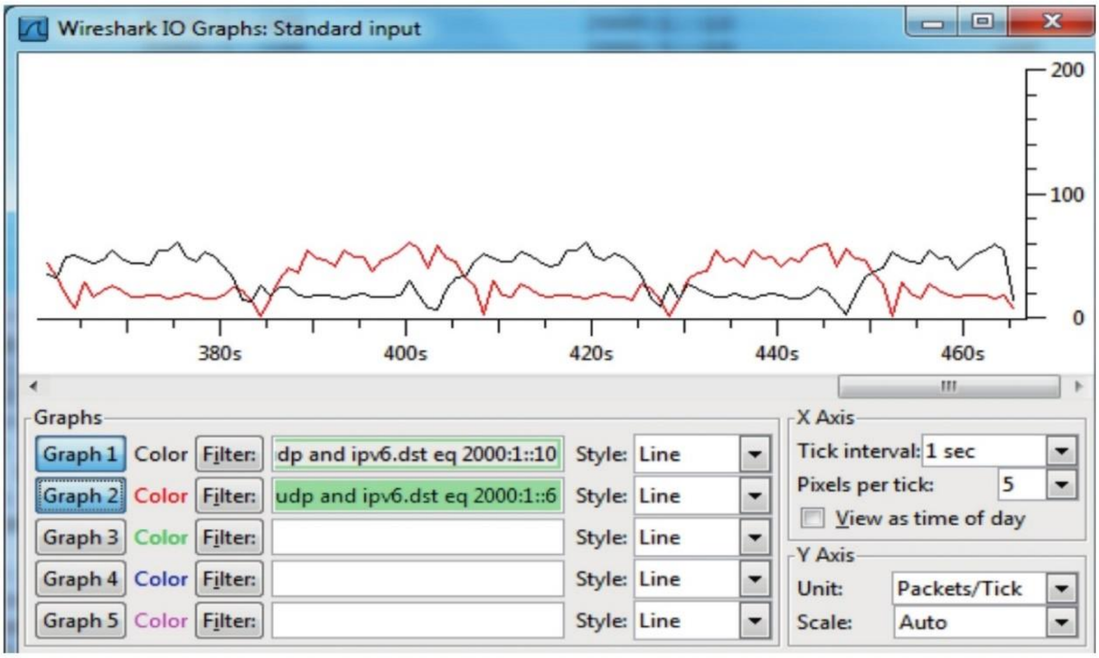


Figure 5: Quality of Service (QoS) Enabled UDP Streams Result

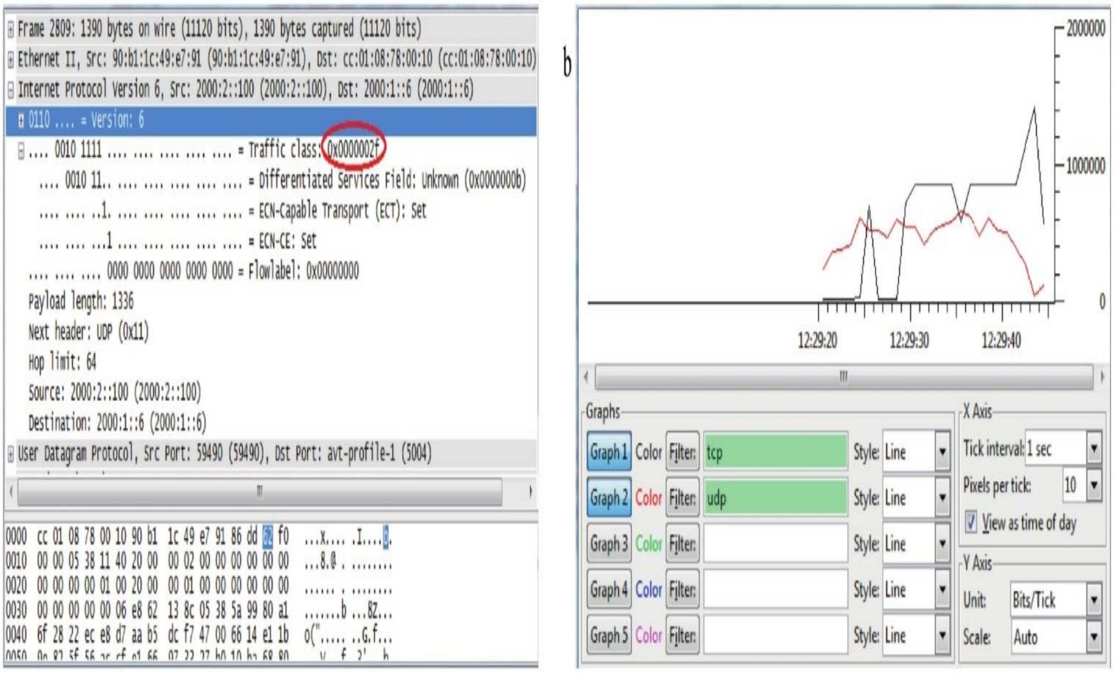


Figure4. With Quality of Service (QoS) Result

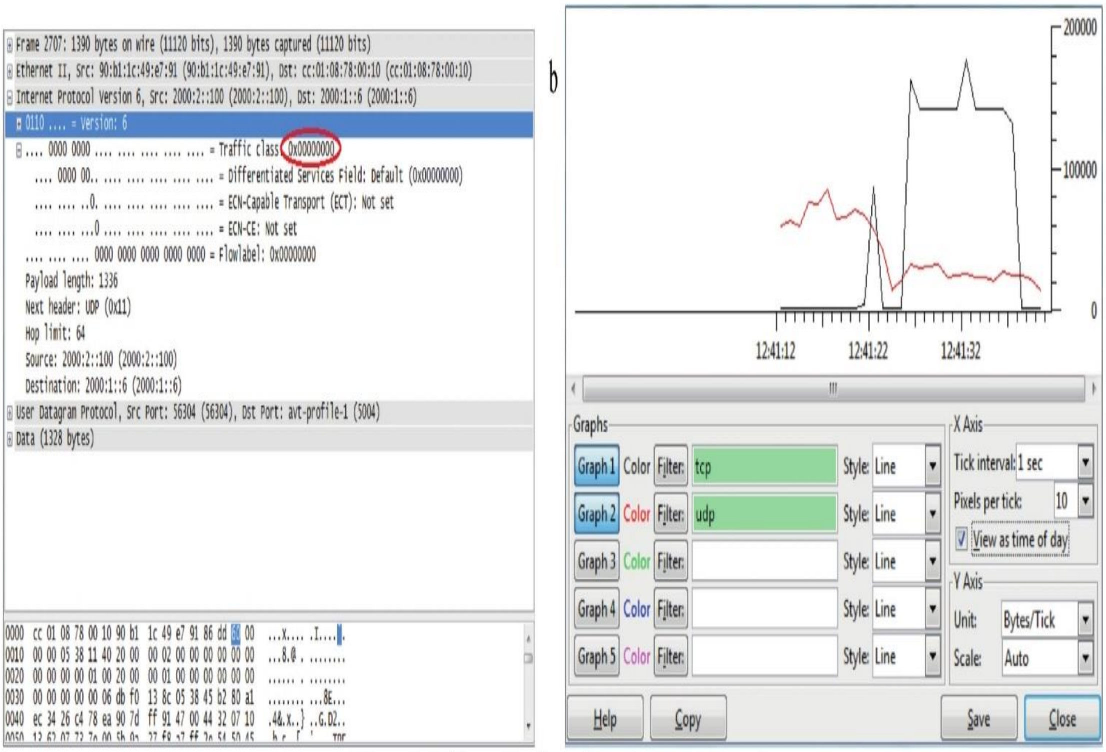
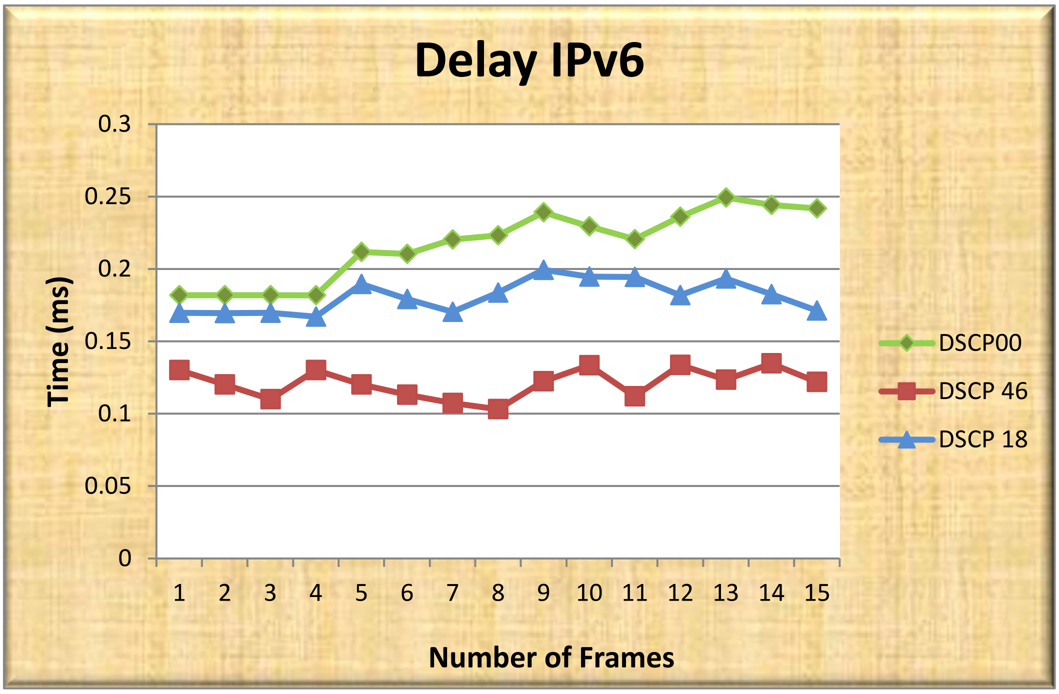
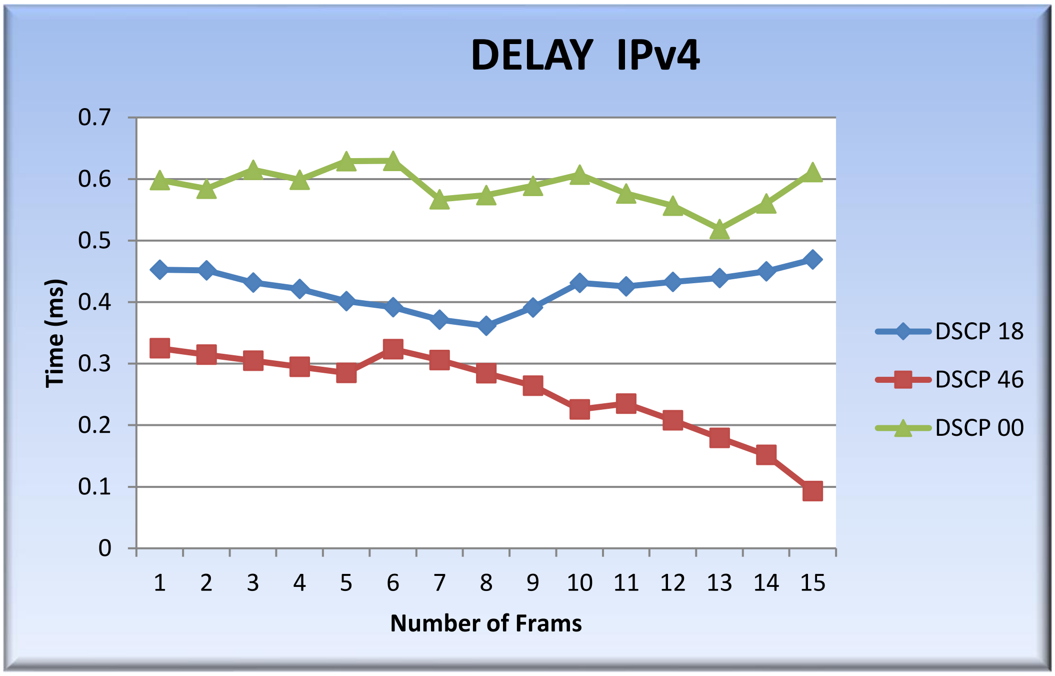


Figure3. Without Quality of Service (QoS) Result

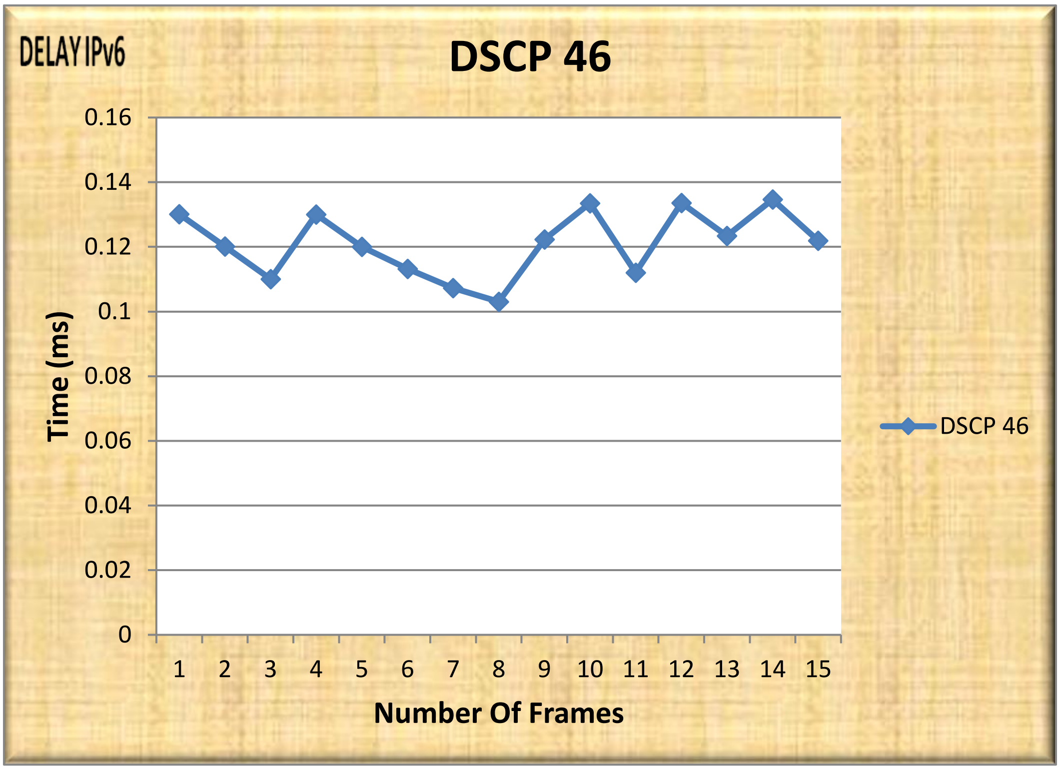
3.5.4 Comparative Delay Graphs of IPv4 & IPv6

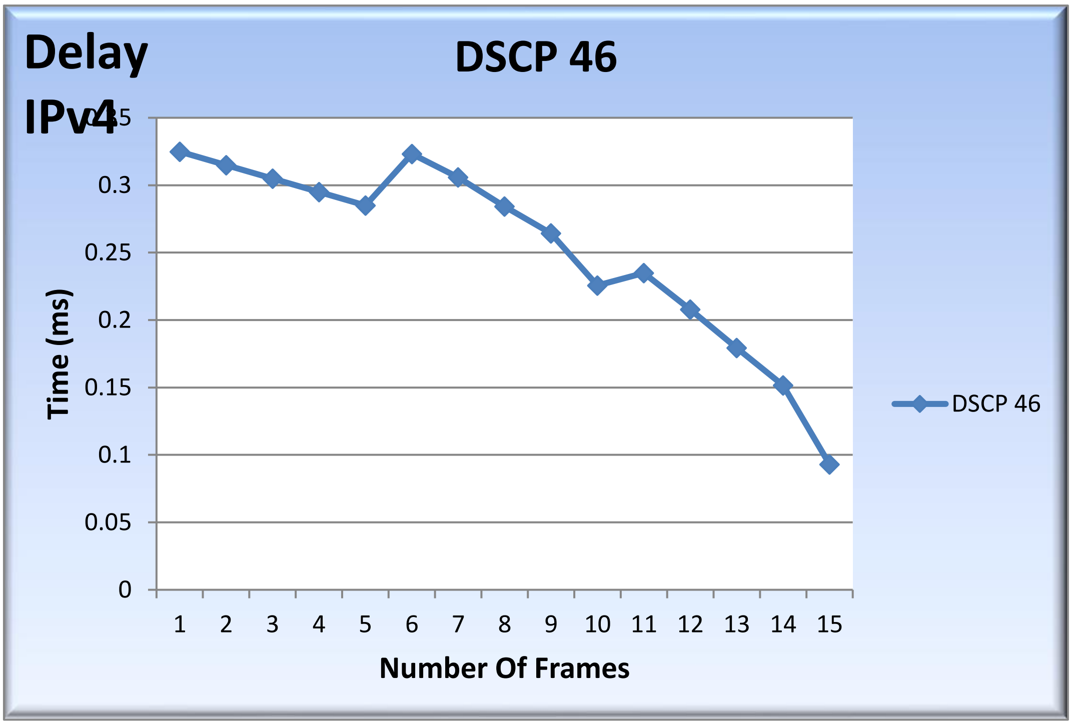




These graphs show the Delay of IPV4 and IPV6 for DSCP Values 18. 46. 00

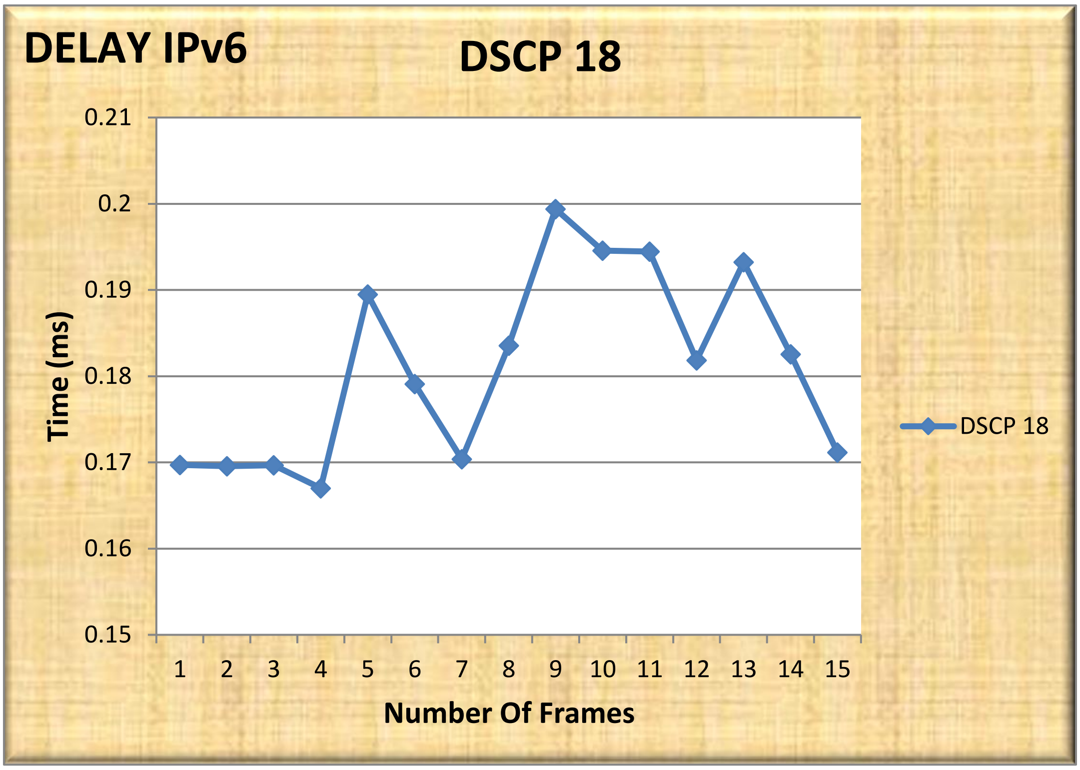
Delay For DSCP Value 46

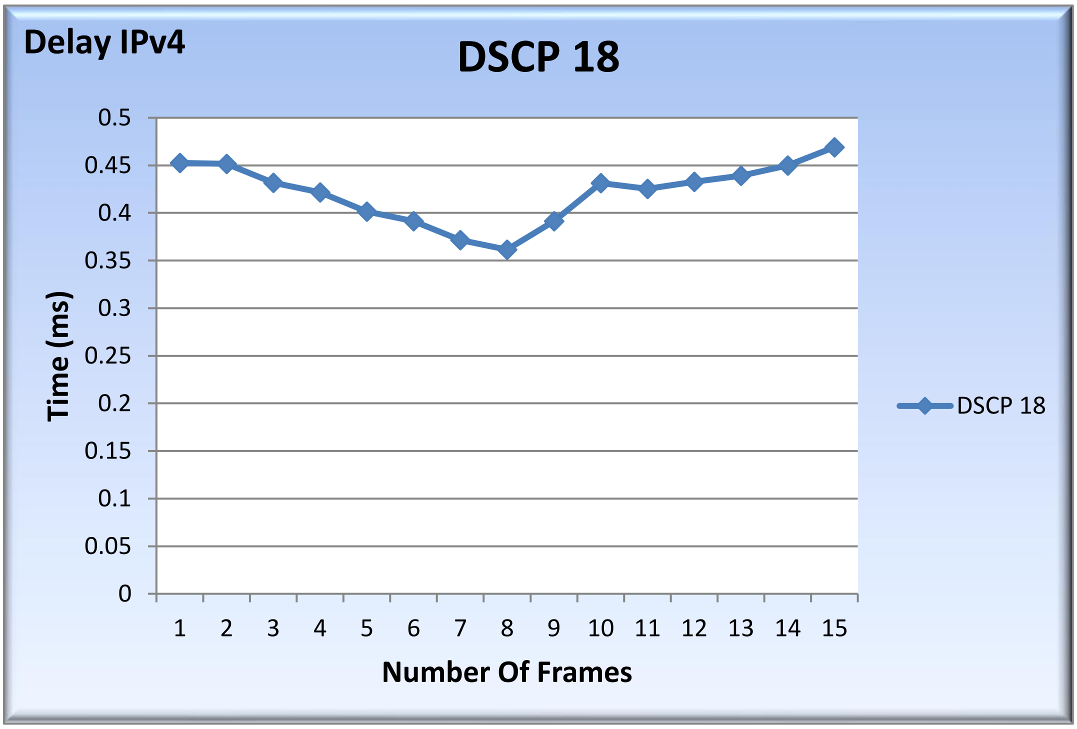




The value of 46 Guaranteed Service Shows Delay graph of IPV4 and IPV6

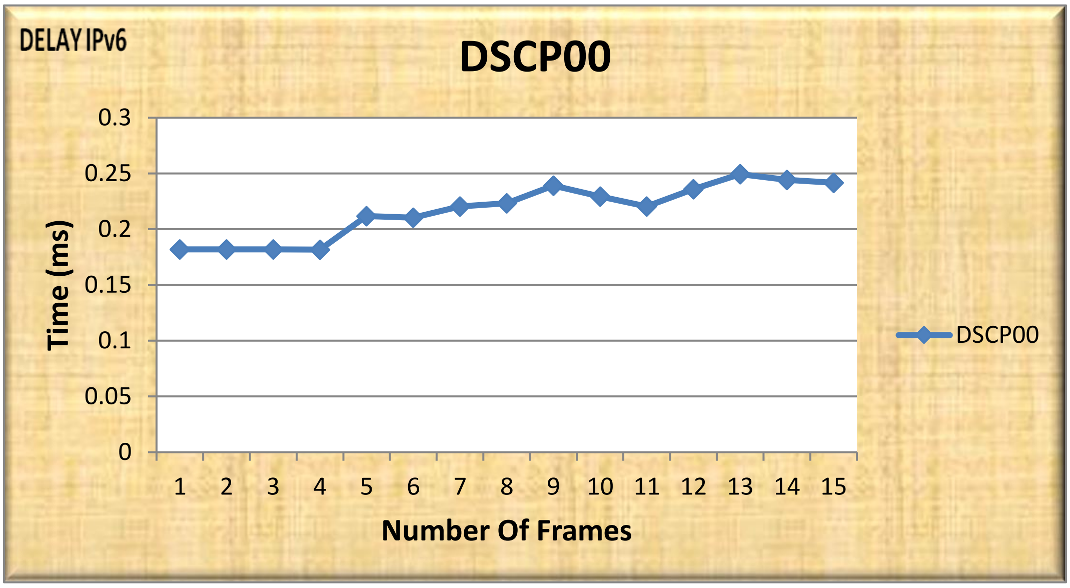
Delay For DSCP Value 18

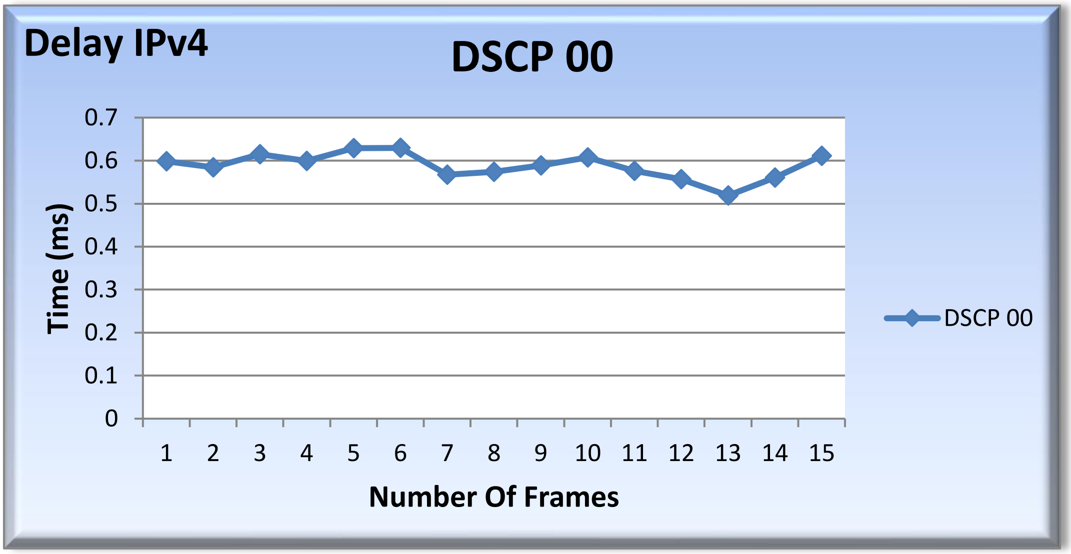




The value of 18 Assured Forwarding Shows Delay graph of IPV4 and IPV6

Delay For DSCP Value 00





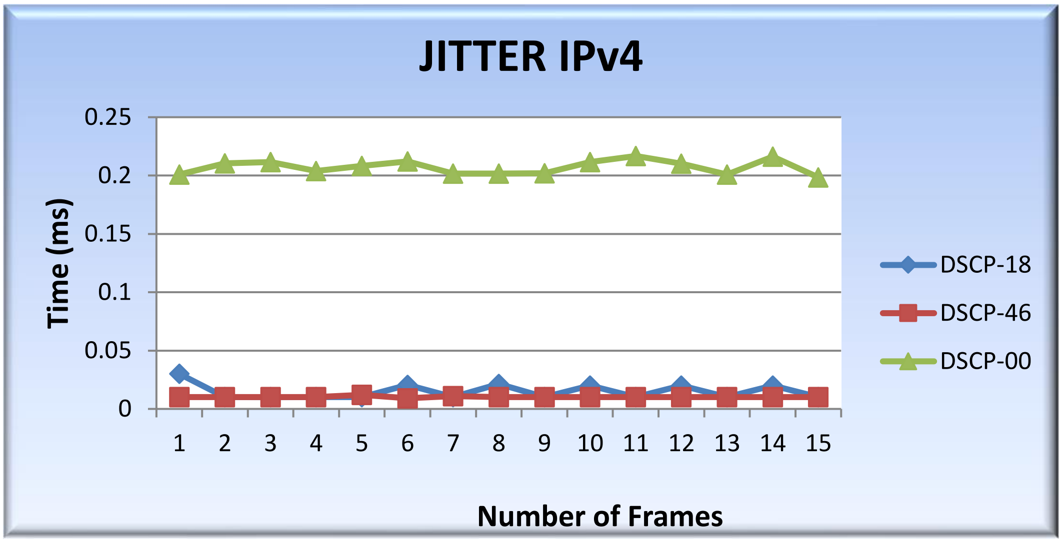
The value of 00 Best Effort Shows Delay graph of IPV4 and IPV6

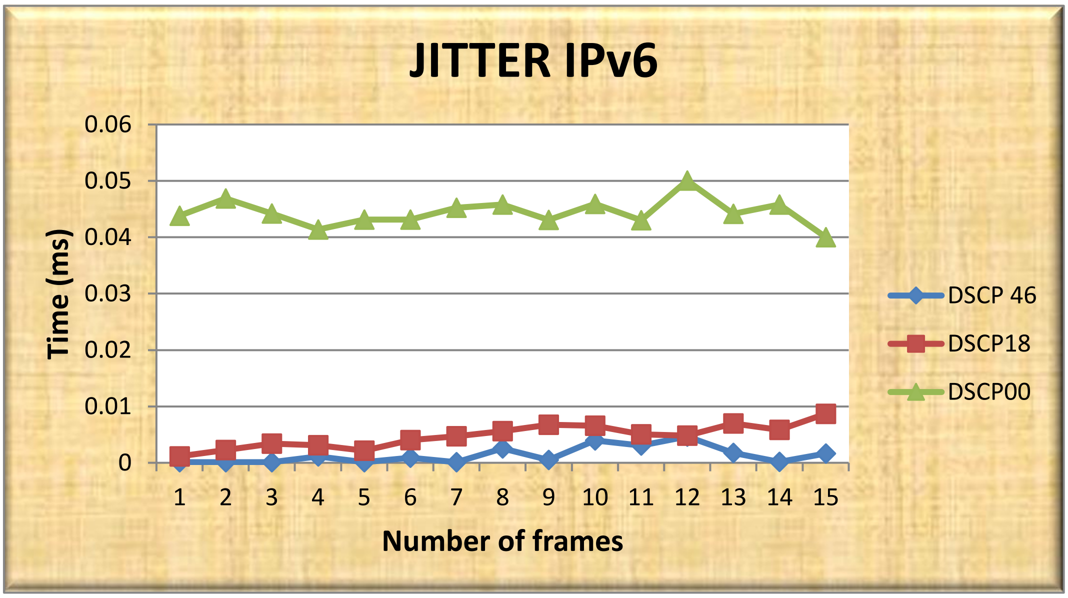
Delay Result:

Our delay analysis of both IPv4 and IPv6 networks, suggests that in general, the average delay observed in IPv6 is less than that observed in IPv4. This delay variation happens because IPv6 modifications have settled the problem of signaling redundancies of IPv4 and thus can perform better in terms of delay.

Jitter : Like delay, we measured the jitter performance for IPv4 and IPv6. The results are as follows

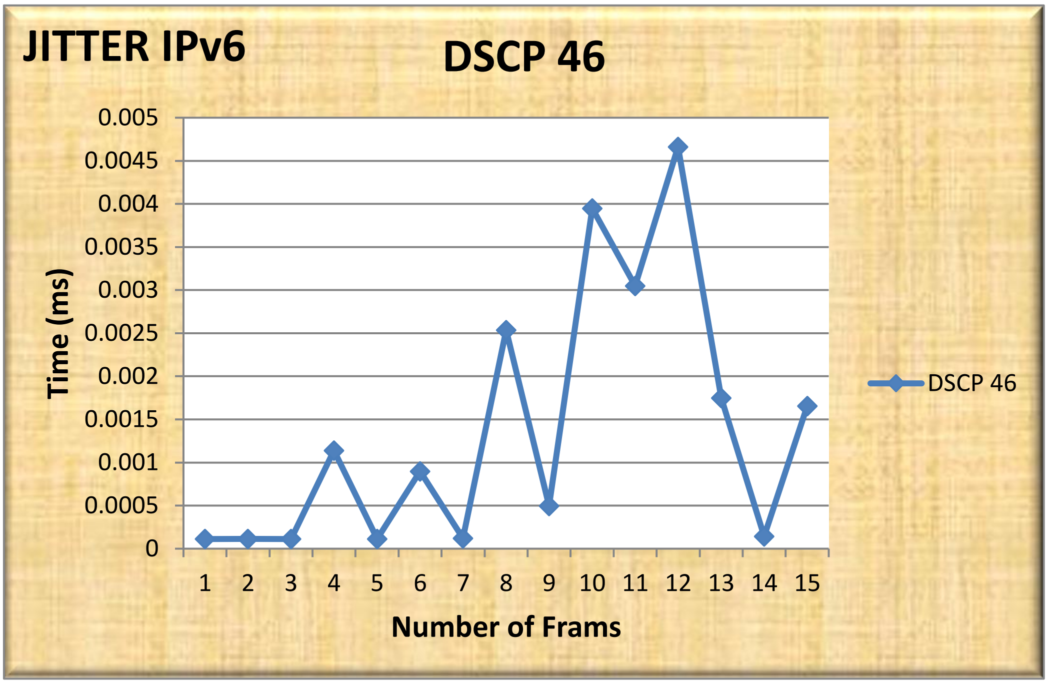
Comparative Graphs of IPv4 & IPv6

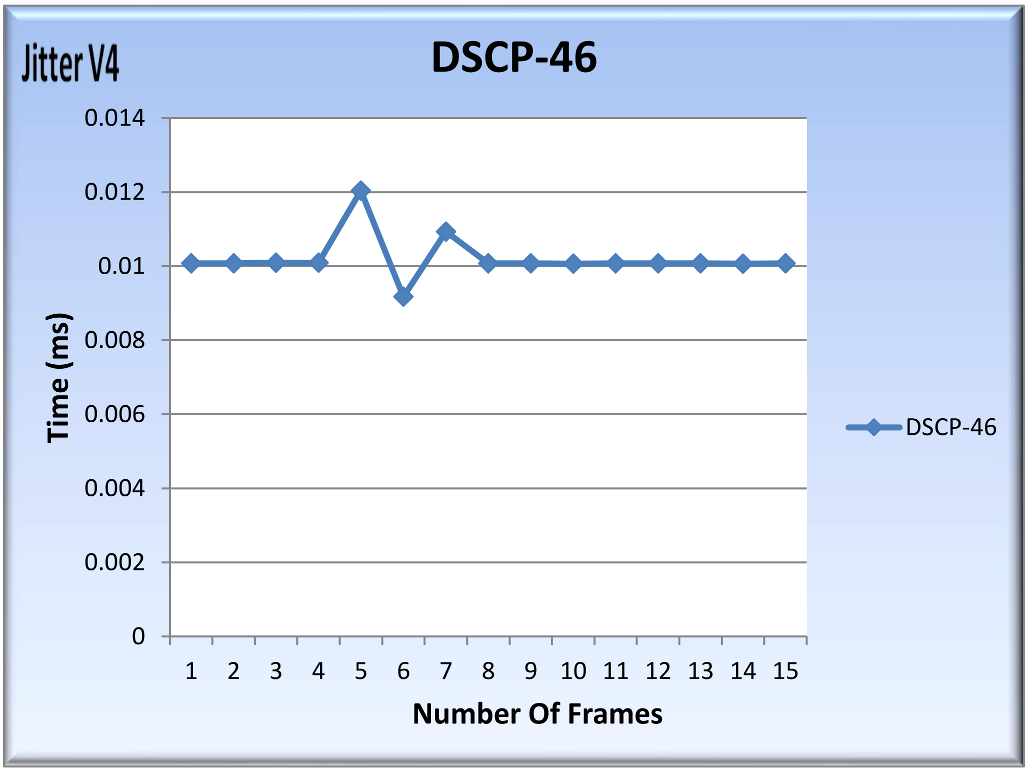




These graphs show the Jitter of IPV4 and IPV6 for DSCP Values 18. 46. 00

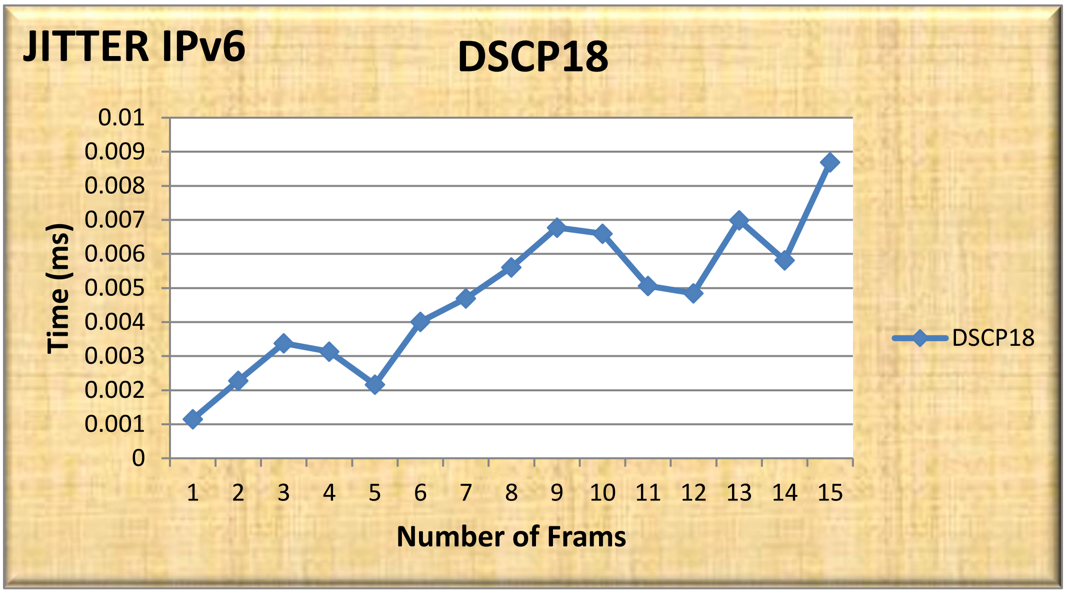
Jitter For DSCP Value 46

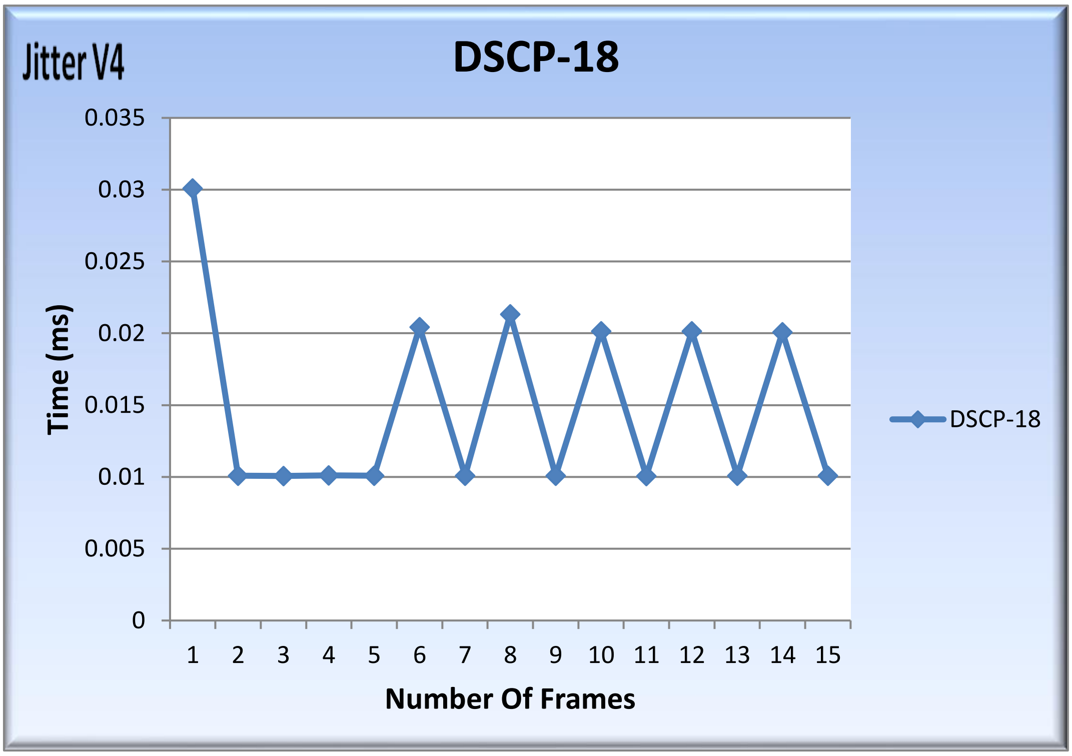




The value of 46 Guaranteed Service Shows Jitter graph of IPV4 and IPV6

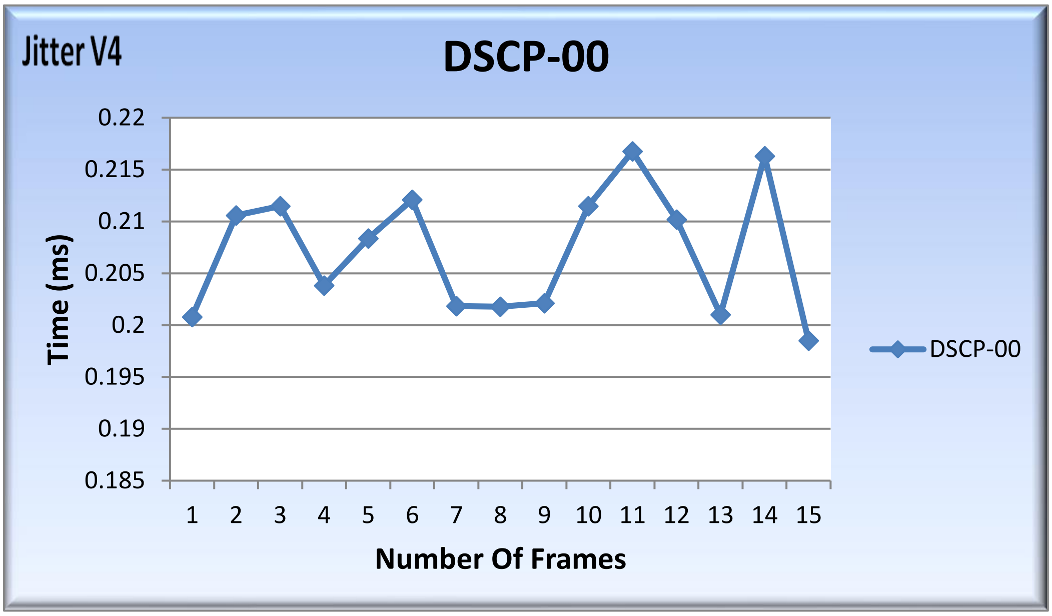
Jitter For DSCP Value 18

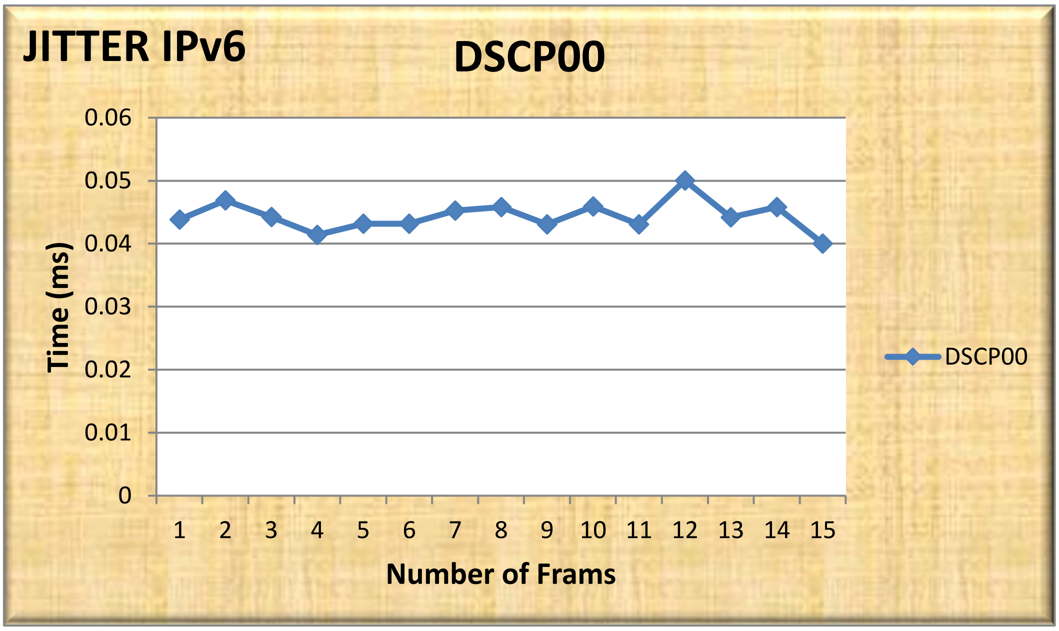




The value of 18 Assured Forwarding Shows Delay graph of IPV4 and IPV6

Jitter For DSCP Value 00





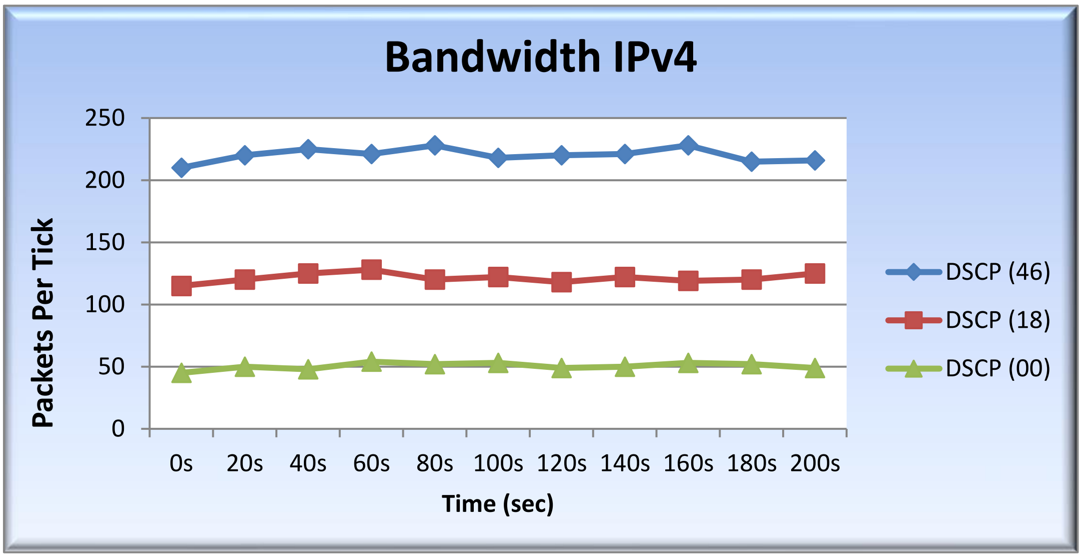
The value of 00 Best Effort Shows Jitter graph of IPV4 and IPV6

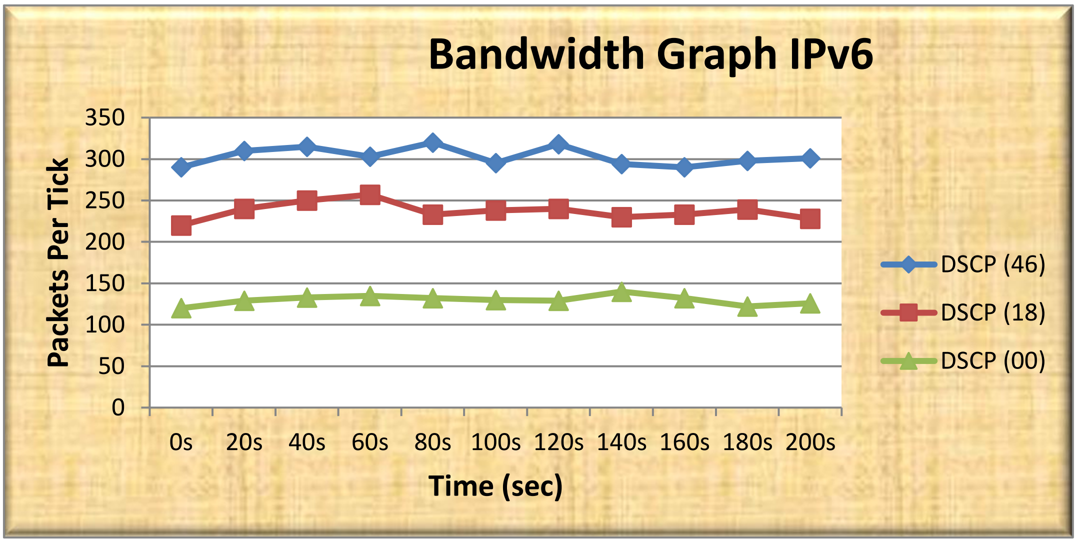
Jitter Result:

In our result, IPv6 gives significantly lesser jitter and somewhat constant jitter rate.

Throughput :

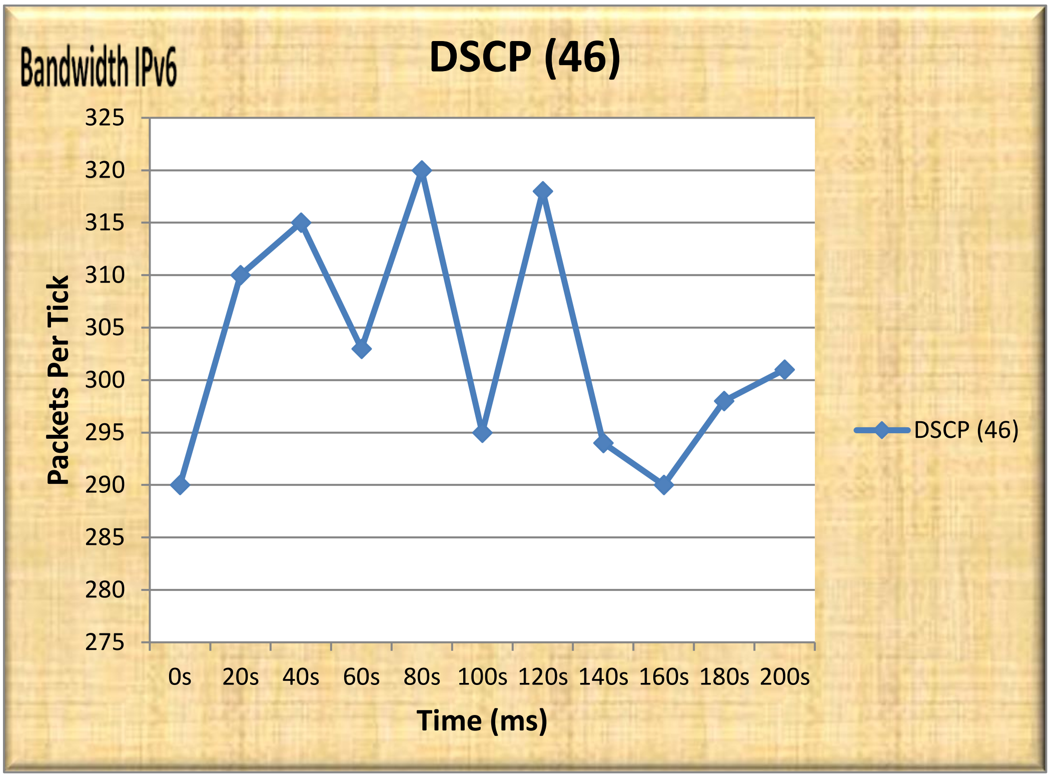
Throughput measurement has been done significantly on both server side as well as client side over IPv4 and IPv6. The results helped me to understand the relationship between IPv6 and IPv4 throughput and their overall network performance. After plotting the throughput result from experiments, the following graphs are produced:

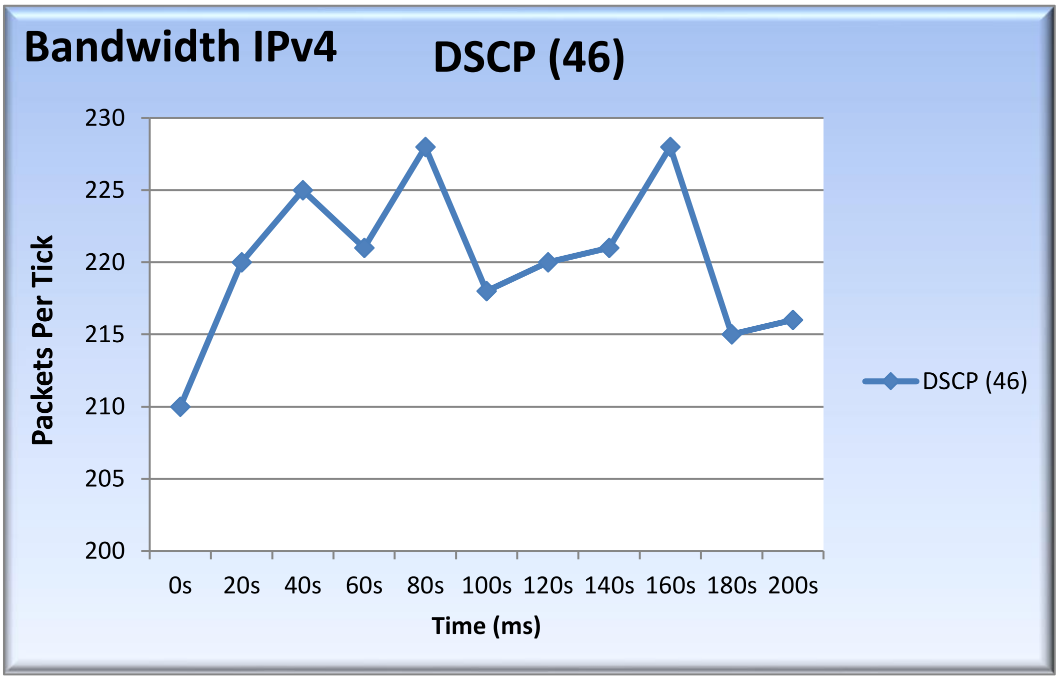




These graphs show the Throughput of IPV4 and IPV6 for DSCP Values 18. 46. 00

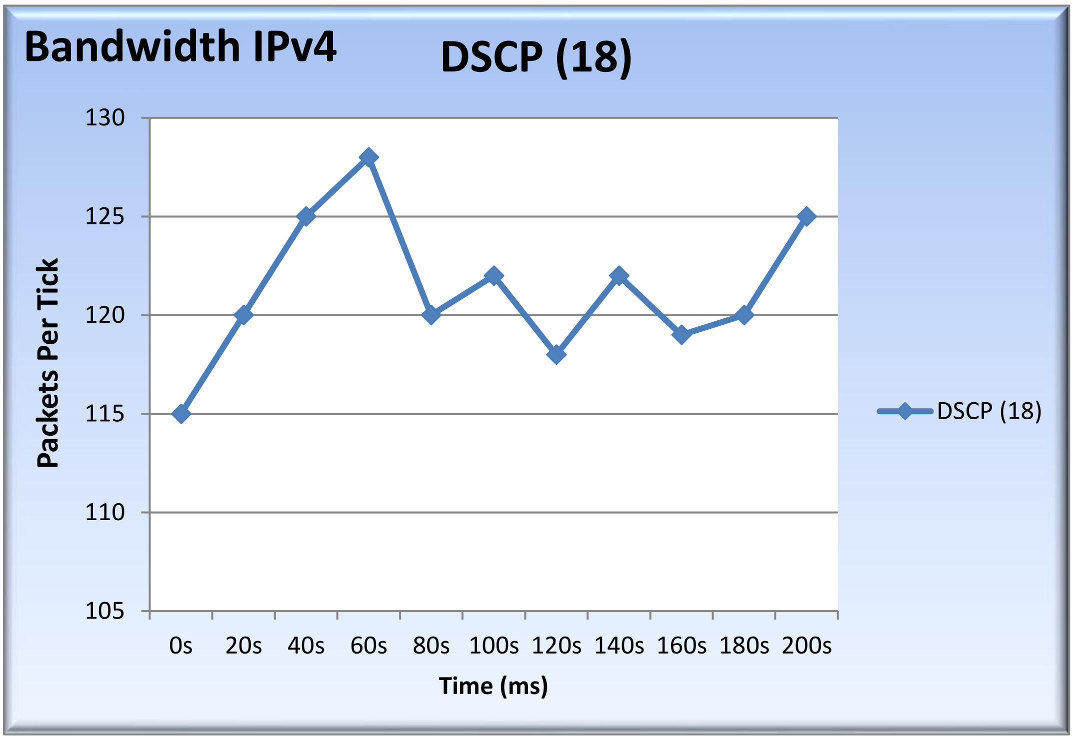
Bandwidth Graph for DSCP Value 46

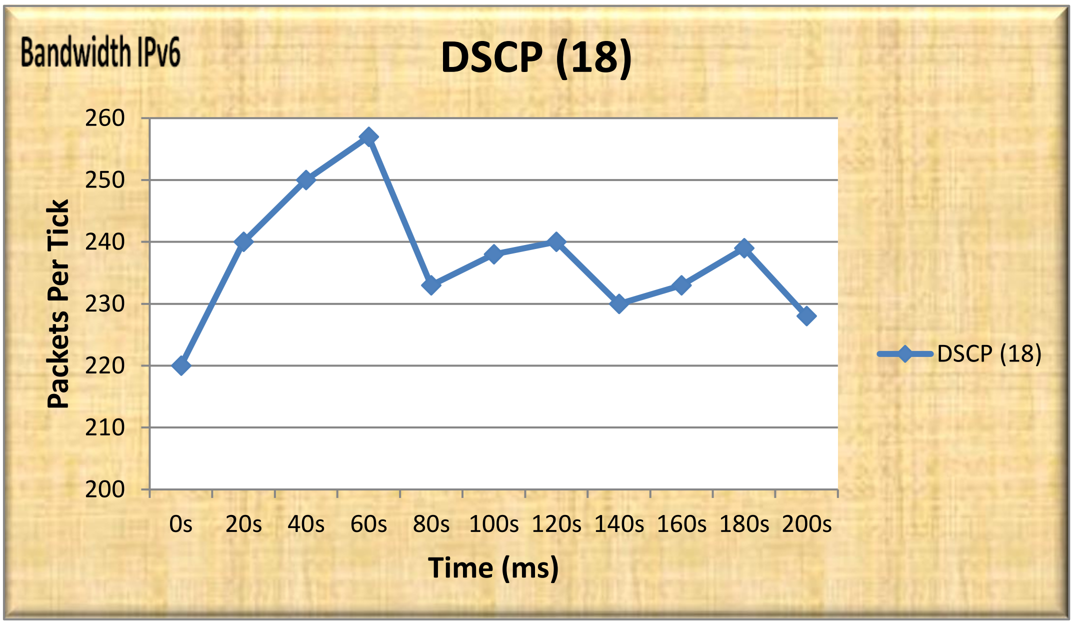




The value of 46 Guaranteed Service Shows Bandwidth graph of IPV4 and IPV6

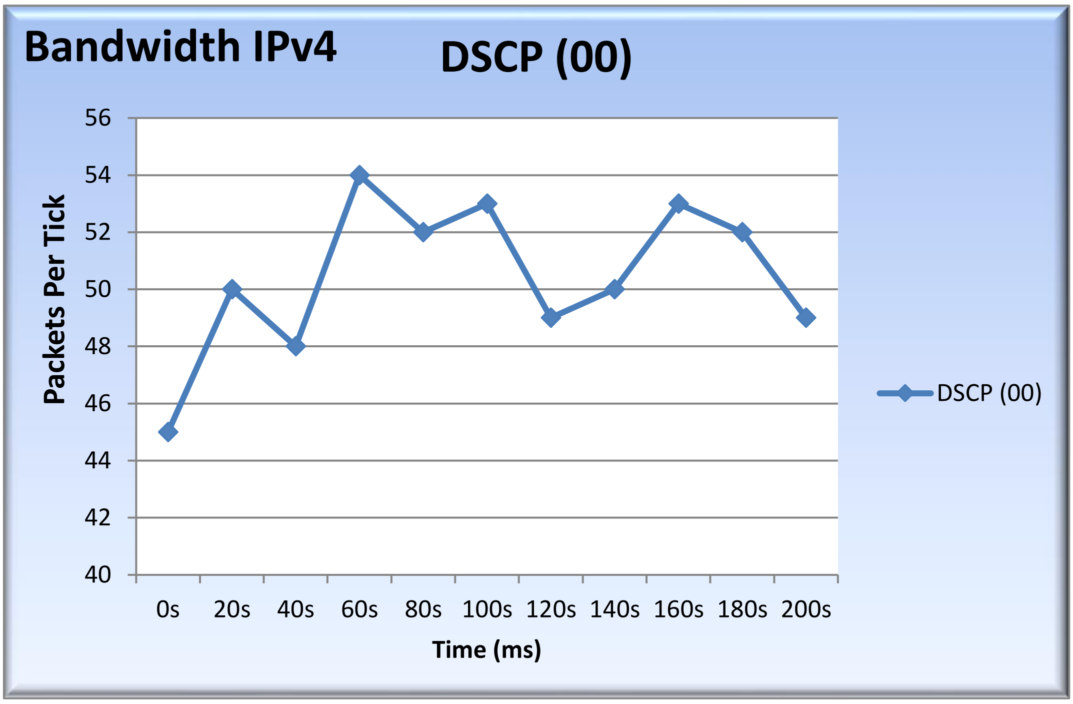
Bandwidth Graph for DSCP Value 18

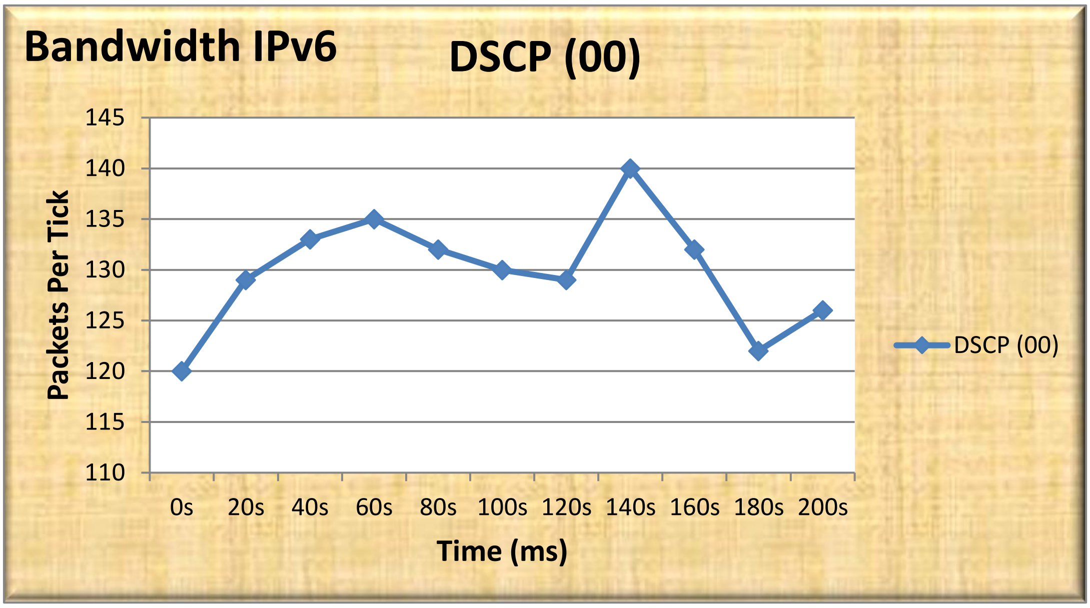




The value of 18 of Assured Forwarding Shows Bandwidth graph of IPV4 and IPV6

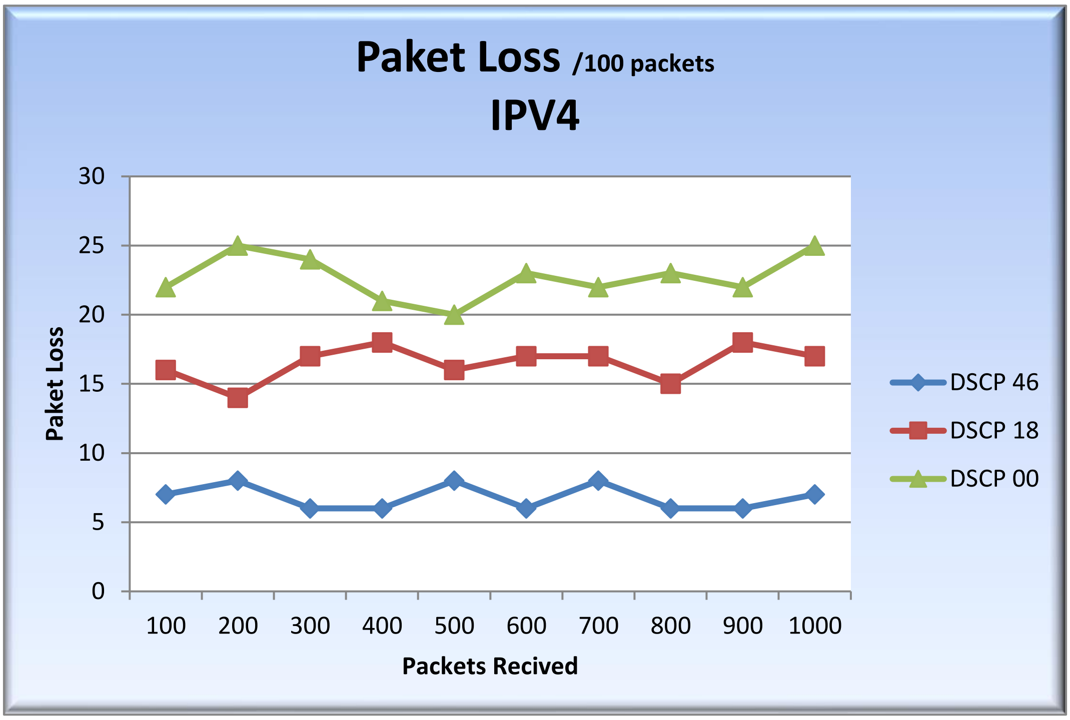
Bandwidth Graph for DSCP Value 00

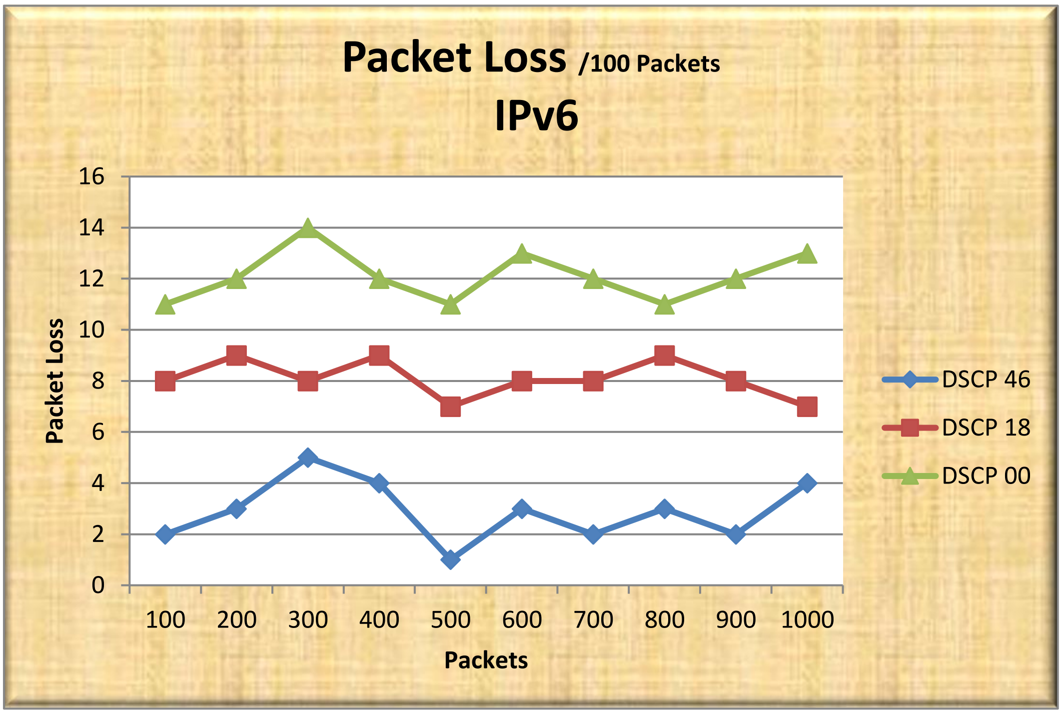




The value of 00 Best Effort Shows Bandwidth graph of IPV4 and IPV6

Packet Loss:





These graphs show the Packet Loss of IPV4 and IPV6 for DSCP Values 18. 46. 00

CHAPTER IV

4. Conclusion and Future Work

4.1 Conclusion

In our research project we comparison IPV4 & IPv6 QoS using metrics i.e. Delay, Jitter, Bandwidth & Packet Loss for different values of Differentiated Service Code Point (DSCP) . We took DSCP values of 00, 46, 18 . These values represent best effort, platinum Service and Assured Forwarding Simultaneously. In results we find IPv6 somehow better than IPv4.

Main theme was of this project to give confidence to deploy IPv6 to ISPs, about which ISP's and other stake holder are lack of confidence. This research work will give boost to their will to move toward the IPV6 as it proves itself worthy and more powerful than the traditional IPV4.

4.2 Future Works

We will measure QoS with other techniques like MPLS TE and also our main focus will be flow label field of IPv6 which is not standardized till now.

References

[1] Abnous A., Seno K., Ichikawa Y., Wan M., Rabaey J.: "Evaluation of a low-power reconfigurable DSP architecture", *proceedings 5th Reconfigurable Architectures workshop (RAW'98)*, March 30, 1998, Orlando, USA. (URL: http://xputers.informatik.unikl. de/RAW/RAW98/adv_prg_RAW98.html)

[2] Abnous A., Rabaey J.: "Ultra-low-power domain-specific multimedia processors", *VLSI Signal processing IX*, ed. W. Burleson et al., IEEE Press, pp. 459-468, November 1996.

[3] Eberle H., Oertli E.: "Switcherland: a QoS communication architecture for workstation clusters", *Proceedings ISCA '98 – 25th annual Int. Symposium on Computer Architecture*, Barcelona, June 1998

[4] Gao B., Rees D.R.: "Communicating synchronous logic modules", *21th Euromicro conference*, September 1995.

[5] Diependorff K., Dubey P.: "How multimedia workloads will change processor design", *Computer*, Sept. 1997, pp.43-45.

[6] Kozyrakis C.E., Patterson D.A.: "A new direction for computer architecture research", *Computer*, Nov. 1998, pp. 24-32,

[7] F. Haider *et al.*, "Cellular Architecture and Key Technologies for 5G Wireless Communication Networks," *IEEE Commun. Mag.*, 2014, p. 123.

[8] B. Bangerter *et al.*, "Networks and Devices for the 5G Era," *IEEE Commun. Mag.*, 2014, p. 91.

[9] Y. Kryftis *et al.*, "Resource Usage Prediction Algorithms for Optimal Selection of Multimedia Content Delivery Methods," *Proc. IEEE ICC*, 2015.

[10] J. A. P. Heesterbeek, *Mathematical Epidemiology of Infectious Diseases: Model Building, Analysis and Interpretation*, vol. 5, Wiley, 2000.

[11] D. Niu *et al.*, "Quality-Assured Cloud Bandwidth Auto-Scaling for Video-On-Demand Applications," *Proc. IEEE INFOCOM*, 2012.\

[12] A. Pankratz, *Forecasting with Univariate Box-Jenkins models: Concepts and Cases*, vol. 224, Wiley, 2009.

[13] Klasen, B. "Efficient Content Distribution in Social-Aware Hybrid Networks," *J. Comp. Sci. 4.4*, 2013, pp. 209–18.

[14] P. Gonçalves *et al.*, "Dynamic Resource Management in Clouds: A Probabilistic Approach," *IEICE Trans. Commun.*, vol. 95, no. 8, 2012, pp. 2522–29.

[15] S. Meloni, A. Arenas, and Y. Moreno, “Traffic-Driven Epidemic Spreading in Finite-Size Scale-Free Networks,” *Proc. Nati’l. Acad. Sci.*, vol. 106, no. 40, 2009, pp. 16,897–16,902.

[16] M.Chen, L.Wang, J.Chen **“***User’s media cloud assisted D2D Communications for distributed caching underlaying cellular network” China Communications, August 2016.*

[17] .Zhang, X.Jiang, H.Xi *“Joint* Content Placement and Traffic Management for Cloud Mobile Video Distribution over Software-Defined Networks” *China Communications, August 2016.*

[18] M.van Der Schaar and N. S. Shankar, “Cross-layer wireless multimedia transmission: Challenges, principles, and new paradigms,”

[19] J. Huang, C. Xu, Q. Duan, Y. Ma, and G.-M. Muntean, “Novel end-toend quality of service provisioning algorithms for multimedia services in virtualization-based future Internet,” *IEEE Trans. Broadcast.*, vol. 58, no. 4, pp. 569–579, Dec. 2012.

[20] B.Ciubotaru, C.Muntean, G.Muntean, “Mobile Multi-Source High Quality Multimedia” IEEE TRANSACTIONS ON BROADCASTING, VOL. 63, NO. 2, JUNE 2017

[21] F. H. P. Fitzek and M. Reisslein, “A prefetching protocol for continuous media streaming in wireless environments,” *IEEE J. Sel. Areas Commun.*, vol. 19, no. 10, pp. 2015–2028, Oct. 2001.

[22] J. Padhye, J. Kurose, D. Towsley, and R. Koodli, “A model based TCP friendly rate control protocol,” in *Proc. NOSSDAV*, Basking Ridge, NJ, USA, 1999.

[23] D. Sisalem and A. Wolisz, “LDA+ TCP-friendly adaptation: A measurement and comparison study,” in *Proc. 10th Int. Workshop Netw. Oper. Syst. Support Digit. Audio Video (NOSSDAV)*, Chapel Hill, NC, USA, 2000, pp. 25–28.

[24] R. Rejaie, M. Handley, and D. Estrin, “Layered quality adaptation for Internet video streaming,” *IEEE J. Sel. Areas Commun.*, vol. 18, no. 12, pp. 2530–2543, Dec. 2000.

[25] M. van Der Schaar and N. S. Shankar, “Cross-layer wireless multimedia transmission: Challenges, principles, and new paradigms,” *IEEE Wireless Commun.*, vol. 12, no. 4, pp. 50–58, Aug. 2005.

[26] G.-M. Muntean, P. Perry, and L. Murphy, “A new adaptive multimedia streaming system for all-IP multi-service networks,” *IEEE Trans. Broadcast.*,

vol. 50, no. 1, pp. 1–10, Mar. 2004.

[27] Q. Li and M. van Der Schaar, “Providing adaptive QoS to layered video over wireless local area networks through real-time retry limit adaptation,” *IEEE Trans. Multimedia*, vol. 6, no. 2, pp. 278–290, Apr. 2004.

[28] K.-A. Cha, “Content complexity adaptation for MPEG-4 audio-visual scene,” *IEEE Trans. Consum. Electron.*, vol. 50, no. 2, pp. 760–765, May 2004.

[29] Y. Cao, C. Xu, J. Guan, and H. Zhang, “QoS-driven SCTP-based multimedia delivery over heterogeneous wireless networks,” *Sci. China Inf. Sci.*, vol. 57, no. 10, pp. 1–10, Oct. 2014.

[29] M. van Der Schaar and H. Radha, “Adaptive motion-compensation fine-granular-scalability (AMC-FGS) for wireless video,” *IEEE Trans. Circuits Syst. Video Technol.*, vol. 12, no. 6, pp. 360–371, Jun. 2002.

[30] N. Cranley, P. Perry, and L. Murphy, “Optimum adaptation trajectories for streamed multimedia,” *Multimedia Syst.*, vol. 10, no. 5, pp. 392–401, 2005.

[31] J. Huang, C. Xu, Q. Duan, Y. Ma, and G.-M. Muntean, “Novel end-to end quality of service provisioning algorithms for multimedia services in virtualization-based future Internet,” *IEEE Trans. Broadcast.*, vol. 58, no. 4, pp. 569–579, Dec. 2012.

[32] J.-C. Chiang, C.-S. Hsieh, G. Chang, F.-D. Jou, and W.-N. Lie, “Regionof-interest based rate control scheme with flexible quality on demand,” in *Proc. IEEE Int. Conf. Multimedia Expo (ICME)*, Singapore, 2010, pp. 238–242.

[33] T. Nuriel and D. Malah, “Region-of-interest based adaptation of video to mobile devices,” in *Proc. 4th Int. Symp. Commun. Control Signal Process. (ISCCSP)*, Limassol, Cyprus, 2010, pp. 1–6.

[34] T. Huang, “Region of interest extraction and adaptation in scalable video coding,” in *Proc. 7th Int. Conf. Fuzzy Syst. Knowl. Disc. (FSKD)*, vol. 5. Yantai, China, 2010, pp. 2320–2323.

[35] O. Abboud, T. Zinner, K. Pussep, and R. Steinmetz, “On the impact of quality adaptation in SVC-based P2P video-on-demand systems,” in *Proc. ACM Multimedia Syst. Conf.*, San Jose, CA, USA, Feb. 2011, pp. 223–232.

[36] A.-N. Moldovan and C. H. Muntean, “Subjective assessment of bitdetect—A mechanism for energy-aware adaptive multimedia,” *IEEE Trans. Broadcast.*, vol. 58, no. 3, pp. 480–492, Sep. 2012.

[37] R. Trestian, O. Ormond, and G.-M. Muntean, “Enhanced power-friendly access

network selection strategy for multimedia delivery over heterogeneous wireless networks," *IEEE Trans. Broadcast.*, vol. 60, no. 1, pp. 85–101, Mar. 2014.

[38] R. Trestian, O. Ormond, and G.-M. Muntean, "Energy–quality–cost tradeoff in a multimedia-based heterogeneous wireless network environment," *IEEE Trans. Broadcast.*, vol. 59, no. 2, pp. 340–357, Jun. 2013.

[39] A. Khan, L. Sun, E. Jammeh, and E. Ifeachor, "Quality of experience driven adaptation scheme for video applications over wireless networks," *IET Commun.*, vol. 4, no. 11, pp. 1337–1347, Jul. 2010.

[40] G.-M. Muntean and N. Cranley, "Resource efficient quality-oriented wireless broadcasting of adaptive multimedia content," *IEEE Trans. Broadcast.*, vol. 53, no. 1, pp. 362–368, Mar. 2007.

[41] *Adobe HTTP Dynamic Streaming*. Accessed on Dec. 2016. [Online]. Available: http://www.adobe.com/products/hds-dynamic-streaming.html CIUBOTARU *et al.*: M3QD 403

[42] *Microsoft IIS Smooth Streaming Transport Protocol*. Accessed on Oct. 2016. [Online]. Available: https://msdn.microsoft.com/ en-us/library/ff469518.aspx

[43] R. Pantos and E. W. May. (Oct. 2016). *Apple HTTP Live Streaming, Informational Internet Draft*. https://tools.ietf.org/html/ draft-pantos-http-live-streaming-20

[44] O. Oyman and S. Singh, "Quality of experience for HTTP adaptive streaming services," *IEEE Commun. Mag.*, vol. 50, no. 4, pp. 20–27, Apr. 2012.

[45] M. Seufert *et al.*, "A survey on quality of experience of HTTP adaptive streaming," *IEEE Commun. Surveys Tuts.*, vol. 17, no. 1, pp. 469–492, 1st Quart., 2015.

[46] W. Zhu, C. Luo, J. Wang, and S. Li, "Multimedia cloud computing," *IEEE Commun. Mag.*, vol. 28, no. 3, pp. 59–69, May 2011.

[47] C. Gadea, B. Solomon, B. Ionescu, and D. Ionescu, "A collaborative cloud-based multimedia sharing platform for social networking environments," in *Proc. 20th Int. Conf. Comput. Commun. Networks (ICCCN)*, Jul./Aug. 2011, pp. 1–6.

[48] H. Wen, Z. Hai-Ying, L. Chuang, and Y. Yang, "Effective load balancing for cloud-based multimedia system," in *Proc. Int. Conf. Electron. Mech. Eng. Inf. Technol. (EMEIT)*, vol. 1. Harbin, China, Aug. 2011, pp. 165–168.

[49] S. Lee, D. Lee, and S. Lee, "Personalized DTV program recommendation

system under a cloud computing environment," *IEEE Trans. Consum. Electron.*, vol. 56, no. 2, pp. 1034–1042, May 2010.

[50] D. Q. Trong, H. Perkuhn, and D. Catrein, "Dynamically scaling multimedia streaming service on hybrid Telco cloud," in *Proc. World Congr. Eng. Comput. Sci. Vol I (WCECS)*, San Francisco, CA, USA, Oct. 2011, pp. 422–427.

[51] Zhenhua Wang, Qiong Sun,Xiaohong Huan, Yan Mat IPv6 END-TO-END QoS PROVISION FOR HETEROGENEOUS NETWORKS USING FLOW LABEL

[52] Ing. MSc.Octavio J. Salcedo Parra1, Ing. Angela Patricia Rios1, Dr.Gustavo López Rubio QUALITY OF SERVICE OVER IPV6 AND IPV4

[53] QUALITY OF SERVICE FOR IP NETWORKS IN THEORY AND PRACTICE **Author:** Magnus von Rosen

[54] QUALITY OF SERVICE (QOS) BETWEEN IPv4 AND IPv6 ON MPLS LINUX PLATFORM**Author:** Asyikin BT Abu Bakar

[55] QoS PROVISIONING USING IPv6 FLOW LABEL IN THE INTERNET **Authors:**Xiaohua Tang, Junhua Tang, Guang-Bin Huang and Chee-Kheong Siew

[56] QUALITY OF SERVICE IMPACT IN IPV6 DOMAIN **Authors:** ch.Bouras, A.Gkamas, D.Primpas, K.Stamos [57] PERFORMANCE EVALUATION OF THE IMPACT OF QoS MECHANISMS IN AN IPv6 NETWORK FOR IPv6-CAPABLE REAL-TIME APPLICATIONS **Authors:** Ch. Bouras, A. Gkamas, D. Primpas, and K. Stamos

[58] EVALUATING QUALITY OF SERVICE PERFORMANCE OF AUDIO AND VIDEO FOR IPV4 AND IPV6 **Author:** Durrbarry Asadullah

[59] IPV4 VERSUS IPV6 INTERWORKING WITH QOS GUARANTEES **Authors:** Mario Marchese, Annamaria Raviola, Maurizio Mongelli,Alessandro Garibbo, Vincenzo Gesmundo,

[60] IP QoS ENHANCEMENT USING DIFFSERV IN IPv6 NETWOR **Authors:**H. Zainol Abidin', N. Fisal'

[61] IMPLEMENTING AN IPv6 QoS MANAGEMENT SCHEME USING FLOW LABEL & CLASS OF SERVICE FIELDS **Authors:**El-Bahlul Fgee', Jason D. Kenney3, William J. Phillips', William Robertson' and **S.** Sivakumar'

[62] DELIVERING END-TO-END QUALITY OF SERVICE THROUGH AN INTERNET PROTOCOL BASED DIFFERENTIATED SERVICES DOMAIN **Authors:***J. C. Dullaert', M H. Rahman', and H. T. Mouftah'*

[64] COMPARISON OF QOS PERFORMANCE BETWEEN IPV6 QOS MANAGEMENT MODEL AND INTSERV AND DIFFSERV QOS MODELS **Authors:**El-Bahlul Fgee1, Jason D. Kenney1, William J. Phillips1, William Robertson1 and S. Sivakumar

[65] PERFORMANCE ANALYSIS OF QOS MECHANISMS IN IP NETWORKS **Authors:**Dix Jia, Eryk Dutkiewicz', Joe F. Chicharo

[66] RFC 791, IP Precedence field

[67] RFC 2474, Diffrentiated Service Field

[68] RFC 3697, Flow Label Specification

[69] RFC 6436,6437: Clarification of FL specifications and add some flexibility

[70] http://readanddigest.com/difference-between-ipv4-and-ipv6

[71] http://www.computerweekly.com/opinion/IPv6-offers-data-packet-security-and-quality-of-service-but-so-does-IPv4

[72] http://www.slideshare.net/shaileshpachori/master-all-home

[73] http://en.wikipedia.org/wiki/Quality_of_service

[74]http://www.cisco.com/c/en/us/td/docs/solutions/Enterprise/WAN_and_MAN/Qo_SRND/QoS-SRND-Book/QoSIntro.html#pgfId-46957

[75]http://www.cisco.com/en/US/technologies/tk543/tk766/technologies_white_paper09186a00800a3e2f.html