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Optimal Control of Multicasting Multimedia Streams in Cloud Data Centers Networks

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***Abstract* - The objective of this paper is a thorough study on the optimal control of multicasting multimedia streaming over the components of data center networks. Such a network control includes finding the best locations of traffic multicasting, assigning the quality of service (QoS) while multicasting, and the right network devices for this objective. Since streaming media accounts for a large portion of the traffic in networks, and its delivery requires continuous service, it is essential to analyze the traffic sources and investigate the network performance. IP multicasting technology adds tremendous amount of challenge to a network as the streamed media delivered to users must be multiplied in volume. In this paper, we first demonstrate a study of the complexity and feasibility of multicast multimedia networks by using different multicast protocols and video sources. In our proposed method, we then create peer-to-peer (P2P) and data center topologies in order to analyze the performance metrics. The implementation and evaluation of the presented methodology are carried out using OPNET Modeler simulator and the various built-in models. Further, we implement performance tests to compare the efficiency of the presented topologies at various levels. At the end of the paper we analyze the optimal locations for multicasting multimedia streaming traffic.**

Keywords - video streaming; cloud data centers; multicast; multimedia; performance evaluation; video codecs; IP; MPLS.

I. INTRODUCTION AND BACKGROUND

In [1], an efficient video-based packet multicast method for multimedia control in cloud data center networks was presented. The current paper extends that work and conducts a thorough study toward several other aspects, such as Comparison of throughput and delay at switching nodes on the control of multicasting multimedia streaming over the components of data center networks.

Over the years, major development in the industry have been involved in the integration of various multimedia applications. Delivery of streaming media (video on demand), e-learning with minimum delay and highest quality has been one of the major challenges in the networking industry. Video service providers, such as Netflix, Hulu are constantly changing the architecture in order to service these needs. These service providers face stiff competition and pressure to deliver the next generation of streaming media to the subscribers. The next generation media can be divided into categories: real-time and non-real time. Examples of real time can be live streaming and video conferencing and non-real time can be e-learning and video on demand [2]. The next generation of streaming media [3] involves a large number of subscribers whose delivery is closer aligned with the latest protocols than with the traditional systems. In such cases, it is required that the service providers upgrade their infrastructure and support them [4].

One of the main challenges in the multimedia industry that motivates us to look into it in this paper is multicasting the video streams. IP Multicast is one of the major techniques that can be used for efficient delivery of streaming multimedia traffic to a large number of subscribers simultaneously. Group membership, unicast and multicast routing protocols are mainly required for multicast communications [5]. *Inter Group Membership Protocol* (IGMP) utilized in our study maintains one of the most commonly used multicast protocols at user facility site. IGMP is used to obtain the multicast information in a network. Unicast routing protocols can be either *distance vector* or *link state*; the latter being preferred due to the dynamic reaction of these protocols to changes in topology. Multicast routing protocols can be integrated with the unicast routing protocol or can be independent of them. Protocols, such as the *Multicast Open Shortest Path First* (MOSPF), depend on the underlying unicast protocols used, whereas protocols such as *Protocol Independent Multicast* (PIM), are independent of the type of unicast routing protocols used. A combination of IGMP, MOSPF and PIM in sparse mode or dense mode can be used for successful implementation and efficient delivery of multimedia traffic in networks [6].

Multicast routing enables transmission of data to multiple sources simultaneously. The underlying algorithm involves finding a tree of links connecting to all the routers that contain hosts belonging to a particular multicast group. Multicast packets are then transmitted along the tree path from the source to a single destination or a group of receivers belonging to a multicast group. In order to achieve the multicast routing tree, several approaches have been adopted. Group-shared tree, source based tree and core based tree are some which are explained here.

a. *Group-based tree*: In this approach a single routing tree is constructed for all the members in the multicast group;
b. *Source-based tree*: This involves constructing a separate routing tree for each separate member in the multicast group. If multicast routing is carried out using source-

based approach, then N separate routing trees are built for each of the N hosts in the group [7]; and

c. *Core-based tree*: This is a multicast routing protocol, which builds the routing table using a group-shared tree approach. The tree is built between edge and core routers in a network, which helps in transmitting the multicast packets.

MOSPF and PIM use one of the above mentioned approaches in the transmission of packets. As PIM is the multicast routing protocol used in the implementation, we discuss the working of PIM.

PIM is a multicast routing protocol that is independent of the underlying unicast routing protocols used [8]. PIM works in two modes dense mode and sparse mode. In the former mode the multicast group members are located in a dense manner and the latter approach has the multicast group members distributed widely. PIM uses Reverse path forwarding (RPF) technique in dense modes to route the multicast packets. In dense mode, RPF floods packets to all multicast routers that belong to a multicast group whereas in a sparse mode PIM uses a center based method to construct the multicast routing table. PIM routers which work in sparse mode send messages to a center router called rendezvous point. The router chosen to be rendezvous point transmits the packets using the group based tree model. As seen in Fig. 1, the rendezvous point (RP) can move from a group-based tree model to a source-based approach if multiple sources are specified [9][10].

A broadband network is a communication infrastructure that can provide higher bandwidth services. Regional networks are connected to such broadband backbone networks to form the Internet. Internet Service Providers own the regional broadband networks. A broadband network is required to support the exchange of multiple types of information such as, massive data storage access, voice over IP (VoIP), video streaming, and live multicasting, while satisfying the performance requirements of each application. In short, the delay and jitter should be minimum for better performance of these applications.

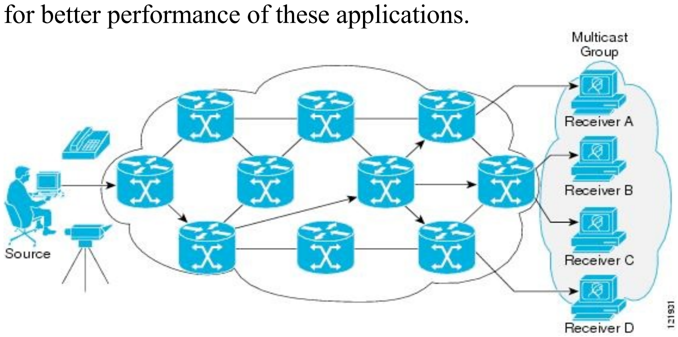

Figure 1. Sample diagram of multicast routing

The above-stated requirements are satisfied by employing broadband routers and switches, fiber optic cables, and tunneling mechanisms. Tunneling makes the communication faster in broadband networks compared to normal routing mechanism. *Multiprotocol label switching* (MPLS) is networking infrastructures used in a high-speed backbone network to provide a better quality of experience for broadband applications such as, video streaming and other real-time applications.

In any MPLS network, the routers at the edge of the network are the most complex ones. In edge routers, user services such as policies, rate limiters, logical circuits, and address assignment are created. In a certain connection between a pair of users, edge routers keep a clean separation between a complex edge and create services while routers in the middle mainly do basic packet forwarding by switching MPLS packets from one interface to the other. The MPLS header (also known as the label) is imposed between the data link layer (layer 2) header and network layer (layer 3) header.

MPLS adds some traditional layer 2 capabilities and services, such as traffic engineering, to the IP layer. The separation of the MPLS control and forwarding components has led to multilayer, multiprotocol interoperability between layer 2 and layer 3 protocols. MPLS uses a small label or stack of labels appended to packets and typically makes efficient routing decisions. Another benefit is flexibility in merging IP-based networks with fast-switching capabilities.

In MPLS networks, any IP packet entering the MPLS network is encapsulated by a simple header called a *label*. The entire routing is therefore based on the assignment of labels to packets. The compelling point of MPLS is that this simple label is always processed for routing instead of the IP header in the network. Note that labels have only local significance. This fact removes a considerable amount of the network-management burden.

The rest of this paper is organized as follows: Section II provides a detail of our architecture and its functionality and Section III presents a performance analysis of the designed architectures. Finally, Section IV concludes the paper.

II. NETWORK ARCHITECTURE

Network architecture has been designed from service provider's and user's perspective. Network service providers are concerned with the available bandwidth and utilization of resources whereas end user's main concern is with the delivery of streaming media with lowest time and maximum efficiency. In order to obtain the various parameters that are required for the best design of multimedia network, two network models were implemented and analyzed.

A. *Implemented Peer to Peer (P2P) Network Design*

Peer to peer network model is a distributed architecture where the application is transmitted between source and destination through peers. Applications such as music sharing, file sharing use peer-to-peer network model for transmitting the data. A peer-to-peer network was built using the values as shown in Table I. The network architecture shown below represents an organizational division where the admin department is the source of multimedia traffic, which is simultaneously streamed to the remaining departments namely the HR, finance and IT. The topology contains two backbone routers connected back-to-back, a video streaming source is configured and stored in the admin department, where the video frames are encoded with a H.264 codec and generating a frame rate of 15-20 frames per sec. The backbone routers are configured with PIM-DM as the multicast protocol that is responsible to carry multicast packets.

TABLE I. CONFIGURATION PARAMETERS FOR A PEER TO PEER NETWORK DESIGN

Link speed (in Mbps)	Frame size	Frame interarrival rate	Video Codec	Multicast protocol used
100	128x120	10 fps	H.264	PIM-DM
100	128x240	15fps	H.264	PIM-DM
1000	352x240	30 fps	H.264	PIM-DM

B. Implemented Data Center Topology

The data center and its network testbed topology [12] implemented in this paper are shown in Fig. 2. A data center contains certain facilities for computing, data storage, and other technology resources, as shown by "server racks." In the network testbed topology, the interconnections among the switches are regular. The figure shows a data center network using four layers of switches as two layers of *core switches,* one layer of *aggregate switches,* and one layer of *edge switches*. An edge switch, also called top-of-rack (ToR) switch, directly connect to end servers at server racks, and core switches directly connect to the routers such as Router1 and Router2 that are attached to the outside of the data center or directly to the Internet. In this figure, two groups of 6 destinations are considered in the testbed: group A destinations and group B destinations.

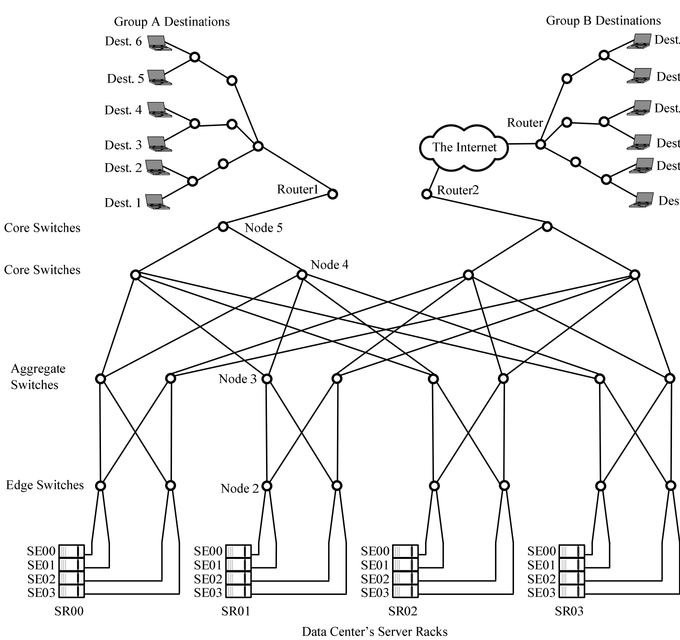

Figure 2. Data center topology as the testbed for multimedia streaming

The topology has been implemented taking into account redundancy at all levels, and responds dynamically to failures at link, path and device level. Scaling the number of nodes, both horizontally and vertically has been considered in order to analyze the performance metrics of the network. Streaming media content stored at the servers are configured for varying bit rates and varying frame sizes.

The OPNET simulator has been used as the simulation tool for implementing and testing multicast multimedia traffic. The detailed metrics that are used for data center implementation has been shown in the Table II.

TABLE II. CONFIGURATION PARAMETERS FOR A DATA CENTER

Number of servers per rack	2
Number of TOR switches used per rack	2
Number of distribution switches per rack	2
Number of core switches per rack	1
Total number of servers	8
Total TOR Switches	8
Total distribution switches	4
Total Core switches	2
Link speeds in data centers	1000 Mbps
Link speeds to WAN	PPP DS3
Video Application and codec used	Video streaming, H.264
Frame sixe	Constant (5000)
Bit rates	Constant (10 fps)

III. PERFORMANCE ANALYSIS

The configuration parameters for used for the performance evaluation are shown below in Table III. Since backbone routers are majorly involved in the transmission of traffic over the internet, Ethernet load across these links has been considered. As the frame size increases load across the backbone links increases, which leads to increase in the delivery of media to destination.

TABLE III. VIDEO CONTENT CONFIGURATION PARAMETERS

Test Name	Frame Size (in bytes)	Video Codec Used	Frame Inter-arrival Time	Ethernet Load Across the Link (packets/sec)
Video1	15,360	H.264	10 Frames/sec	280
Video2	5,000	H.264	Exponential	530

A. Study on Ethernet Load

In order to reduce the end to end delay, latency and prioritized traffic using quality of service (QoS) was implemented. Opnet simulator has various built-in QoS profiles, some of them being WFQ, FIFO, priority queueing. Differentiated services code point based QoS is being used in this implementation wherein based on the priority of traffic delivery, a certain level of service is configured depending on which resources are allocated along the path of delivery.

Now, we present the Ethernet load test – a performance metric which determines the amount of data packets that are carried by the network. Although each link in the network carries data packets WAN / core routers are chosen for analysis. In peer-to-peer topology mentioned earlier, the links connecting the backbone routers are considered, whereas in a data center topology core router links/WAN links have been chosen.

The variation in the graph can be explained as follows. In this case the bit rate and frame size s, both have been kept as exponential increasing functions. From Fig. 3 it can be observed that in a two-node network since there is a single link connecting the backbone routers, Ethernet load across these links is considerably higher than that of a multi node model where, PIM builds a tree structure (source based, or center based) for sending the multicast packets. As a result, the load is distributed across various links thereby reducing the failure percentage. One more alternative that can be used is port-channel can be configured to distribute the load across the links connecting the routers. Over the time considered it was observed that the load was higher in a two-mode network and lesser in a multi node network.

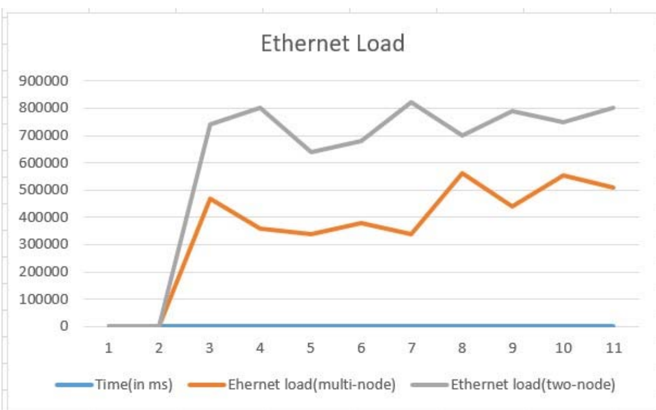

Figure 3. Comparison of Ethernet load between two nodes and multi-mode cases

Our next experiment is concerned with the queueing delay which is the amount of time that a packet waits in the router's queue before being sent onto the network. This is one of the most important parameters for multicast networks as an increase in the queueing delay can cause significant delay in the transmission of packets across the network. Queueing delay can be due to many factors, such as buffer size in a router, router's processing capacity, link speed used, number of hops from source to destination. In this analysis, the queueing delay has been analyzed for a two node and a multi-node environment. From the graphs shown in Fig. 4, it can be observed that although in a two node network links of higher speed were used, when packets of multiple applications arrive, a two-node network experienced significant queueing delay which led to the delay in the transmission of packets. Since no QoS was configured all the packets were serviced based on packet arrival times. The graph for a 2-node network shows peaks of highest queueing delay and lows of least queueing delay. This is due to the fact that when packets related to multiple applications arrive there has been peaks of high queue delay and when packets related to single applications arrive less queueing delay has been experienced. In order to have less queueing delay priority traffic can be classified based on QoS policies which helps in serving these packets better.

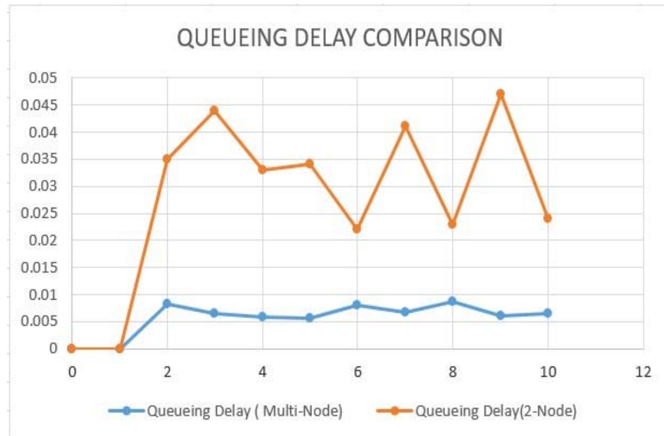

Figure 4. Comparison of queueing delay between two nodes and multi-node cases

Next, we consider a test on QoS, a mechanism which is used to analyze the performance of networks. QoS policies configured ensures traffic prioritization and reservation of resources along the path from source to destination. QoS plays a major role in multimedia networks where defining QoS policies defines the traffic priority when real time multimedia traffic and interactive media is involved. Since these types of traffic have rigid delay constraints defining QoS policies for these types can result in prioritizing them when requests for other traffic are in queue.

Simulation results of QoS implementation is shown in Fig. 5. Since real time interactive media could not be created in a simulation environment, two video sources (video1 and video2) were created and video1was configured with a WFQ QoS profile traffic group of video 1 being set to high priority and traffic group of video 2 being set to best effort with no QoS configured. From the plots in the figure, it can be observed that over a period of time when requests arrive for video1 and video2 packets requesting information, video1 is serviced with less packet delay than those packets for video2 while multicast flow is also included in the configuration. Since the IGMP convergence time was 2 min the QoS traffic servicing has started after the first few minutes.

Finally, the latency is our last performance metric to focus on. The latency is the amount of delay involved in transmitting the data from source to destination. For calculating the Latency issues in network different pixel sizes were chosen for analysis. Three different Pixel sizes were configured over a period of time with link speed and other parameters being kept constant. The link speed was defined to be 100 Mbps and pixel sizes of 352x240, 128X240 and 128X120 were defined with frame interarrival rates to be logarithmic. After several tests it was observed that the latency in the transmission of a high-quality video was more compared to the latencies of the transmission of a video of lesser resolution as shown in Fig. 6. If a video of high quality has to be transmitted in minimum time, then separate channels can be used for high definition video where source specific trees can be used for routing thereby achieving successful routing of packets.

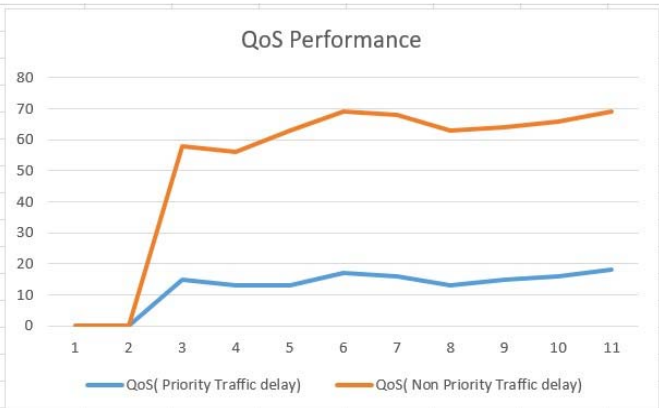

Figure 5. QoS servicing of priority and non-priority traffic

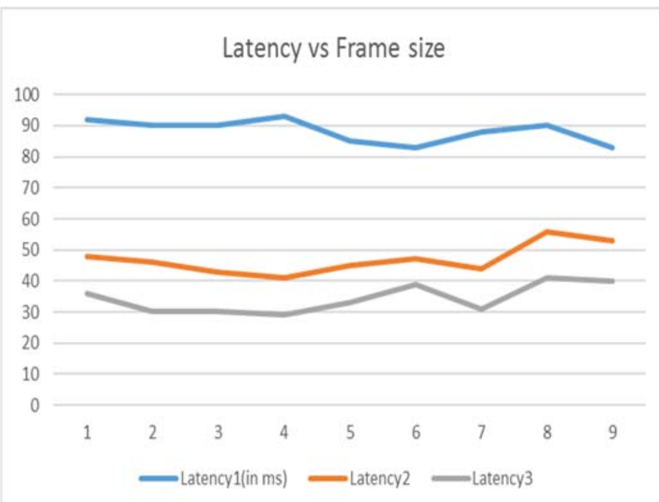

Figure 6. Comparison of latency of various frame sizes.

B. *Study on Network Infrastructure for Media Multicasting*

An MPLS network consists of nodes called *label switch routers* (LSRs). An LSR switches label packets according to its particular *forwarding tables*. The content of a forwarding table consists of labels to be assigned to a flow of traffic. An LSR has two distinct functional components: a control component and a forwarding component. The control component also facilitates the exchange of information with other LSRs to build and maintain the forwarding table.

A typical advanced router or switch has routing tables in its control plane and an IP forwarding table in its data plane for routing purposes. The routing table of such a device receives signaling packets such as RIP or OSPF packets and must update its IP forwarding table occasionally. An LSR has a separate forwarding table to store labels. The MPLS forwarding table interacts with the IP forwarding table in order to arrange the conversion of IP address to labels or vice versa.

Fig. 8 shows a basic comparison of streaming in IP and MPLS networks. In Fig. 7 (a), a typical IP network is shown where a source host, as host 1, connects to a destination host, as host 2. The generated IP packets enter a wide-area IP network at edge router R1 and pass over the network using all the relative routing protocols, and eventually reach edge router R2 from which they exit. In Fig. 7 (b), an MPLS-enabled network is contrasted with the IP network; any IP packet entering the MPLS network is encapsulated by a label. The entire routing is therefore based on the assignment of labels to packets. We will see in the next sections how this technique would substantially reduce the time for packet processing and routing.

Assigning labels to each packet makes a label-swapping scheme perform the routing process efficiently and quickly. In Fig. 7 (b), the edge LSRs of the MPLS network are ingress LSR1 and egress LSR2. It can be seen that a label is indeed a header processed by an LSR to forward packets. The header format depends on the network characteristics. LSRs read only labels and do not engage in the network-layer packet headers. One key to the scalability of MPLS is that labels have only local significance between two devices that communicate. When a packet arrives, the forwarding component uses the label of the packet as an index to search the forwarding table for a match. The forwarding component then directs the packet from the input interface to the output interface through the switching fabric.

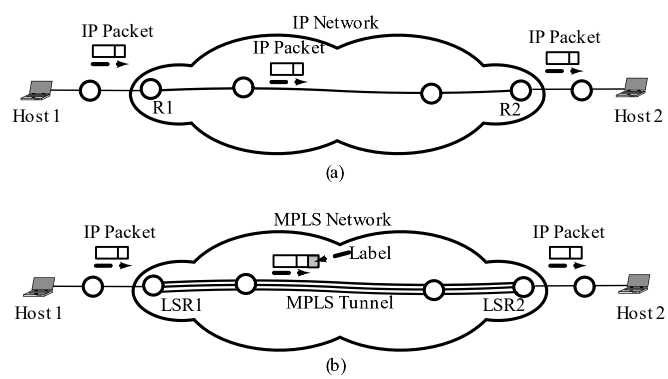

Figure 7. Operations of streaming in MPLS compared with IP infrastructures

The advance mutual agreement between two adjacent LSRs on a certain label to be used for a route is called *label binding*. Once an IP packet enters an MPLS domain, the ingress LSR processes its header information and maps that packet to a *forward equivalence class* (FEC). Any traffic is thus grouped into FECs. An FEC indeed implies that a group of IP packets are forwarded in the same manner—for example, over the same path or with the same forwarding treatment. A packet can be mapped to a particular FEC, based on the following criteria:

- Source and/or destination IP address or IP network addresses
- TCP/UDP port numbers
- Class of service
- Applications

A *label switched path* (LSP) through the network must be defined, and the QoS parameters along that path must be established. An LSP resembles a tunnel. Label usage for identifying the next hop destination instead of the IP destination address adds superior capabilities like traffic engineering to the traditional IP routing. LSRs consist of two functional components; a control component and a forwarding component. The control component uses standard routing protocols to exchange information between the LSRs and facilitates the forwarding table formation. Based on the information in the forwarding table, the forwarding component of the LSR performs the switching of label packets.

OPNET Modeler provides a global development environment capable of infrastructure networks, such as MPLS, modeling and performing discrete event simulations. Data collection and analysis, incorporated along with the design simulation, helps in evaluating and optimizing real life networking topologies. The advantages of OPNET Modeler over other modelers include a simple GUI interface and network modules integrating extensive protocol suite with queuing functionalities.

The 'System in the Loop' module included in this Modeler provides an efficient method to capture live transmission and analyze the behavior of different real-life scenarios. Applications include LAN and WAN performance modeling, network planning and protocol research.

The study on the type of network infrastructure for media streaming involves the performance evaluation of video streaming using MPLS and OPNET Modeler. The OPNET Modeler has the modules to implement the MPLS over the basic network topology. Creating dynamic and static LSPs, Traffic engineering, and Differential Service functionality could be achieved using the Modeler functionalities. The main steps involved in implementing MPLS using the simulator are:

- Configuring "MPLS config" Object
- Establishing MPLS Path Model
- Configuring MPLS parameter on MPLS routers
- Define Neighbor configuration
- Establish the Traffic mapping configuration
- Configuring IP Traffic Demand
- Configure Routing Protocol
- Update MPLS LSP details

Fig. 8 shows a global network perspective for the study on the backbone networking infrastructure used in the simulator. The network topology consists of the following elements: LER1, LSR1, LSR2, LSR3, LSR4, LSR5, LSR6, LER2, and LER2 are Cisco 7200 series routers, which support MPLS protocol along with the standard routing protocols such as, Open Shortest Path First (OSPF) and Routing Information Protocol (RIP).

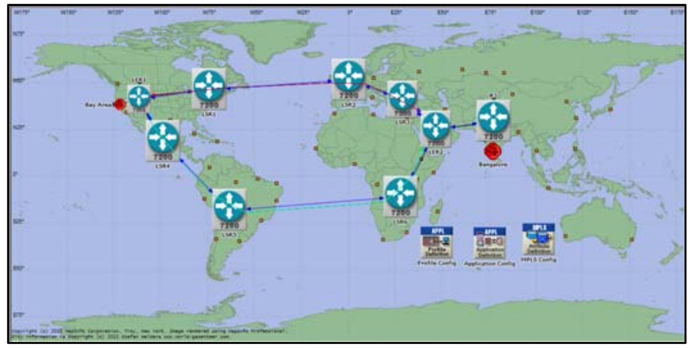

Figure 8. Global network for the study on the backbone networking infrastructure

Two subnets were considered in the simulator: San Francisco Bay area subnet and Bangalore subnet. The San Francisco Bay area subnet consists of a Cisco 4000 series router and an Ethernet workstation capable of video conferencing. The Bangalore subnet consists of a Cisco 4000 series router and an Ethernet workstation capable of video conferencing. Unique IPv4 addresses were assigned to each node in the network. All routers in the main subnet are connected via Point-to-Point DS3 links with a data rate of 4.736 Mbps. The router and Ethernet workstation in each subnet are connected using 10Gbps Ethernet links. The simulation objects used in this project are the following:

1. Application Configuration Object
2. Profile Configuration Object
3. MPLS Configuration Object
4. IP Traffic Flow Object

An application configuration object is used in the simulation to define the parameters for the video conferencing application. This object is a video conferencing application so that it receives the highest priority among all the other traffic in the network.

In comparison with the MPLS traffic, we also used an IPv4 traffic flow object that represents the IP layer traffic flow between a specified source and a destination. This object is used to create background traffic in the network. Using this object, a background traffic flow at the approximate rate of 40.5 Mb/s is created. Fig. 9 shows the sample configuration for IP traffic flow. The background traffic flow starts when the simulation starts and continues at this rate for 3600 simulation seconds. This background traffic flow is created between all nodes in the parent subnet. Thus, the background traffic flows between the nodes LER1-R3 along the two paths LER1 - LSR1 - LSR2 - LSR3 - LER2 - R3 and LER1 - LSR4 - LSR5 - LSR6 - LER2 - R3.

The profile configuration object is another object used to create a user profile named "video." This profile was then specified on different nodes in the network to generate application layer traffic. Sample configuration for the object. The video conferencing starts 150 seconds after the start of the simulation and continues until the end of the simulation. After every 60 seconds, a new video conferencing session will be created between the two workstations.

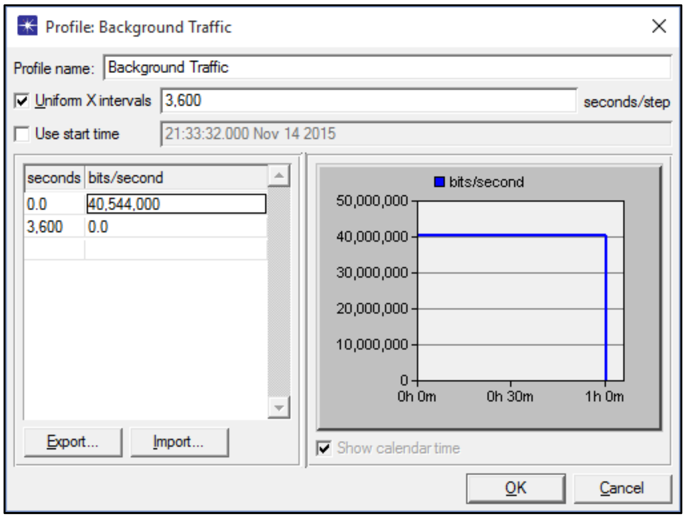

Figure 9. Traffic flow configuration

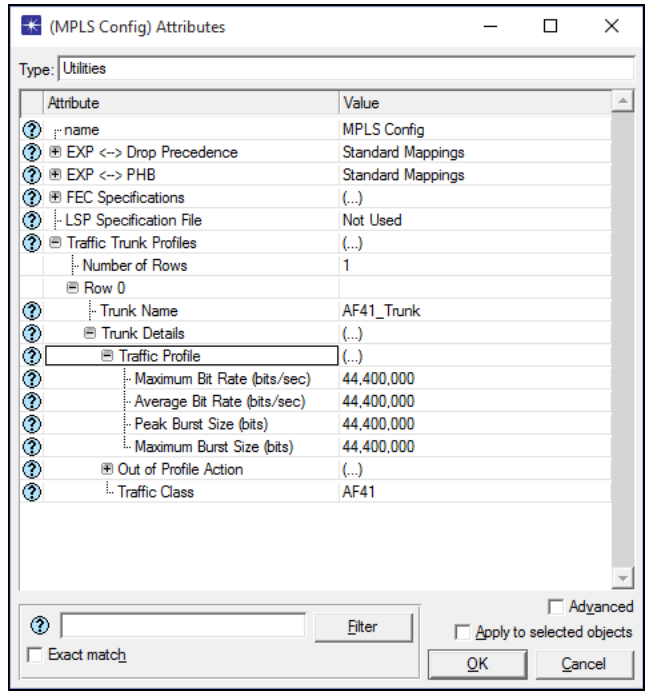

Figure 10. Traffic trunk profile

Finally, MPLS configuration object is used to define the *Forward Equivalency Class* (FEC) and traffic trunk profiles. The FEC defined for the video traffic is the Video_FEC. The *Type of Service* (ToS) or *Differentiated Services Code Point* (DSCP) is set to AF41 for this FEC. Hence, this FEC will be used only by the incoming IP datagram that has the ToS or DSCP field set to AF41. In this study, the video conferencing application has been configured in such a way that the hosts send the IP datagram for the application with DSCP bit set to AF41. Fig. 10 shows the sample configuration for the Trunk Profile.

The delay increased rapidly from 0.5 seconds to 1.05 seconds. After this point, the delay increased linearly and reached 2.3 seconds.

The comparison of delay in video conferencing in MPLS and IP networks using OSPF specifications is shown in Fig. 11. When the video conferencing is done over traditional IP networks, the delay is found to be increasing linearly with time and reached a peak value of 10.6 seconds. When the same video conferencing session is simulated over MPLS network with single LSP, the delay increased gradually to 5 seconds and remained in this range until the end of the simulation. When two LSPs are used for video streaming, there is a significant decrease in the delay. The delay remained under two seconds for the most part of the simulation.

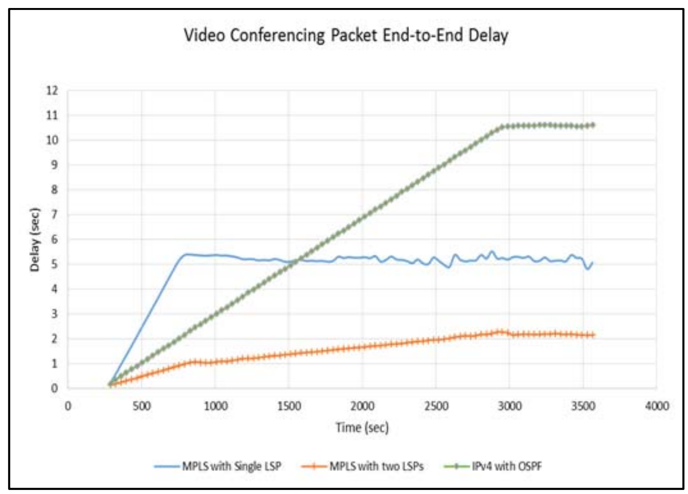

Figure 11. Comparison of end-to-end delay in video conferencing using IP and MPLS

C. *Study on the Location of Multicasting Function*

Consider again the testbed of Fig. 2. In that figure, two groups of 6 destinations are considered in the testbed: group A destinations and group B destinations. Group A destinations is directly connected to the data center network passing through Router1, while group B destination is connected to the data center network after passing through a local router, indicated by "Router," the Internet, and Router2 of the center.

In the first scenario, a video conferencing between two workstations located in San Jose and Bangalore is simulated. The routing protocol used in the scenario is IP using *Open Shortest Path First* (OSPF) specifications and is configured to run on all nodes in the network. Fig. 12 shows the end-to-end packet delay and jitter when video conferencing simulated over the network with the routing protocol OSPF/IP configured on all the nodes. Similarly, Fig. 13 shows the end-to-end packet delay and jitter when video conferencing simulated over the network with the routing protocol OSPF/MPLS configured on all the nodes indicating an improvement over the OSPF/IP case on both delay and jitter.

The Trunk profile used in this paper is AF41_Trunk. The AF41_traffic class is mapped on to this traffic trunk. The maximum bit rate and average bit rate are set to 44,000,000 bits/sec. The maximum burst size and the peak burst size are set to 44,000,000 bits.

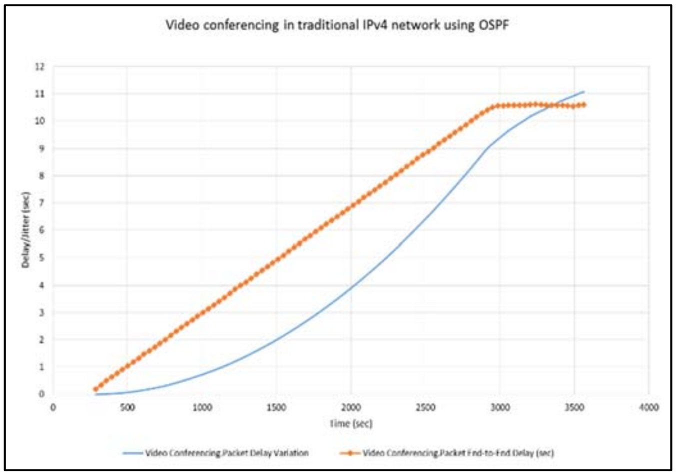

Figure 12. Delay and jitter in video conferencing using OSPF/IP

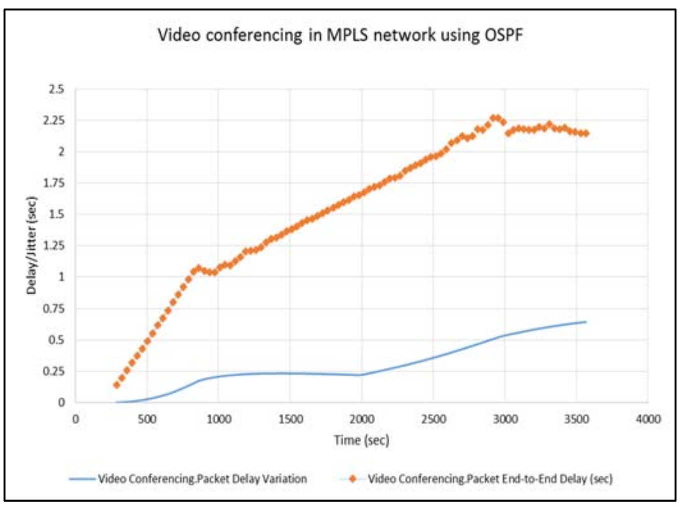

Figure 13. Delay and jitter in video conferencing using OSPF/MPLS

Fig. 14 shows the result of performance evaluation on the throughput for group A destinations. We picked three destinations, Dest. 1, 2, and 3. Then we deployed three different experiments, each considering one of nodes 2, 3, and 5 as the location of the multicasting multimedia streaming. This study clearly shows, that the throughput of multicast at the highest switching nodes (node 5) is highest with an average of 900,000 packets/sec, whereas the one for switching node 2 is the lowest with the average of 110,000 packets/sec. This experiment teaches the multicasting live multimedia must be carried out at the highest levels of the data center switching structures to return the best performance.

The results of the second experiment to determine the best location of multicasting multimedia is shown in Fig. 14. In this experiment, we consider group B destinations in Fig. 3. The study is conducted on Router2 which is attached to the cloud center and Router which is attached to the destinations. The study evidenced by Fig. 15 shows that the queueing delay (QD) at the Router is lower than the one in Router2. What this result teaches us is that the streaming must also be multicast at the closest routing stage to the destinations.

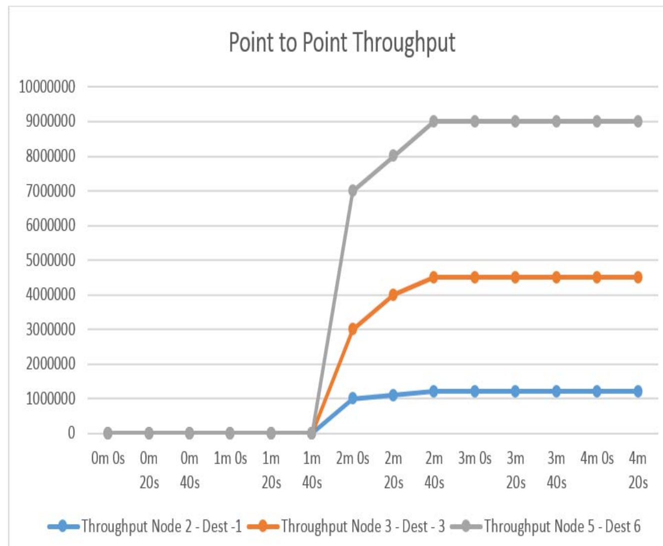

Figure 14. Comparison of throughput at switching nodes 2, 3, and 5

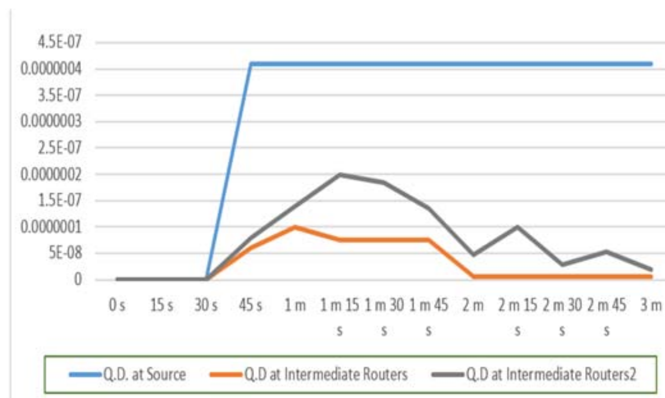

Figure 15. Comparison of queueing delay (QD) at source, and intermediate routers

IV. CONCLUSION

In this paper, we conducted a thorough study on the control of multicasting multimedia streaming over the components of data center networks. We focused on finding the best locations of traffic multicasting, assigning the quality of service (QoS) while multicasting, and the right network devices for this objective. we designed and implemented peer to peer and data center topologies under QoS restrictions and multicast requirement of streaming traffic. The two topologies were implemented for various video streaming applications such as video conferencing and video streaming. The parameters of these video sources were changed in to measure the performance metrics of the multicast networks. Parameters such as video codecs, frame size, frame interarrival rate, link speed, QoS were changed for analysis. From the analysis it was observed that a multitier architecture connected to high speed links was best suited for high end real time traffic. Further it was observed that the QoS configuration for these real time traffic reduces the packet end to end delay and the latency of these packets was also less as compared to other packets. Building a multitier not only helped in better load distribution of traffic across links but also this type of topology was better equipped to handle failures at device, links and server levels. This paper also covered the different multicast routing protocols that can be used. At the end of the paper we analyzed the optimal locations for multicasting multimedia streaming traffic. We learned that the streaming must also be multicast at the closest routing stage to the destinations.

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