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Evaluation of Bandwidth Management Technique using Dynamic LSP Tunnelling and LDP in MPLS for Sustainable Mobile Wireless Networks

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Abstract: Fairness in bandwidth resource allocation is highly significance to the advancement of the future generation mobile and wireless technologies. It is likely that restriction of bandwidth due to the employment of some scheduling scheme would not be an appropriate option for the future development of communication systems. However, there is need to consider an implementation that would lead to good network performance and avoid unguaranteed bandwidth delivery. This paper focusses on evaluating the performance of Bandwidth Allocation using Dynamic Label Switching Paths (LSPs) Tunnelling and Label Distribution Protocol (LDP) signalling in Multi-Protocol Label Switching (MPLS) network. This will make provision for bandwidth allocation and reservation possible. An appropriate bandwidth allocation would have a positive impact on throughput as well as the delay. The results of an IP (Internet Protocol) Network without MPLS enabled is compared with MPLS model network. Furthermore, implementation of dynamic and static LSPs models are presented with about 75% decrease in packet delay variation for dynamic LSP when compared from static LSP. In addition, the models of bandwidth estimation, bandwidth allocation, delay and jitter are provided. Performance metrics used in this respect for multimedia services (Voice and Video conferencing) confirm that the modified models are improved in comparison with the baseline, having highest throughput of about 51% increment, and packet delay variation decreases drastically.

Keywords: Bandwidth Management, Resource Reservation Protocol-Tunneling Extension, Multi-Protocol Label Switching-Traffic Engineering, Label Switching Path, Label Distribution Protocol, Multimedia Services.

1. Introduction

Bandwidth management has emerged as a powerful platform for controlling the traffic volume of future mobile and wireless networks. This is as a result of using appropriate bandwidth allocation and reservation of resources for the critical applications of both sensitive and non-sensitive traffic in the networks. Many telecommunication industries have used a conventional approach to managing bandwidth to support the peak demand of the resource. However, underutilization of resources may lead to bandwidth wastage due to low demand. The same approach stated in [1] has the purpose of supplying bandwidth on a network in order to reserve capacity for users. However, the demand is low compared to the operational capacity of the network.

The main purpose of network operators is to satisfy their subscribers by providing the Quality of Service requested. This indicates that the only key to QoS is the resource management, which is made up of the decision of whether to accept the request for a net flow and then to manage flow servicing so that the QoS guarantees are met [3-5]. These two aspects of the resource allocation are termed “Admission Control” and “Scheduling”. In this respect the use of Multi-Protocol Label Switching (MPLS) technology to implement bandwidth management in the future mobile wireless network is reliable and profitable due to its valuable cost to both operators and service providers.

The aim of this paper is to perform an evaluation of the performance of Resource Reservation Protocol Tunneling Extension (RSVP-TE) and Label Distribution Protocol (LDP) in an MPLS Network model using a traffic engineering approach for proffering a solution to the next generation of mobile wireless networks. This could be achieved by the proposed design of MPLS networks to manage bandwidth efficiently as possible solution for the future mobile and wireless networks. It can be carried out by performing dynamic and static Label

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Switching Paths (LSPs) of the MPLS model network as part of Traffic Engineering (MPLS-TE). This serves as an extension to the existing MPLS Architecture.

Our main contribution in this research work is the implementation of static and dynamic LSP tunneling, which is combined with LDP signaling for the allocation of bandwidth in an MPLS model. This is followed by presenting mathematical models for bandwidth estimation, bandwidth allocation, delay and jitter. The remaining part of this paper consist of section two, which entails related work and the proposed technology to be employed; section three implements MPLS-TE on the models using performance metrics of multimedia services; simulation results can be found in section four; finally, the summary of this paper is provided in section five.

2. RELATED WORK

The existence of MPLS technology for decades can be found in the literature. However, it has become necessary to employ this mechanism for the purpose of bandwidth management to solve the critical problem of delay and packet loss. In addition, this is a technique that would utilize the available bandwidth to meet the requirement of QoS.

An effective bandwidth management system comprises of network switching devices at the core network for managing resources in the physical connection of ports [2]. Furthermore, dynamic bandwidth management can be implemented in a manner to predict future traffic of connected devices [3]. The bandwidth can be shared appropriately according to the needs of each connected device. Scheduling Algorithms are proposed in [4] for a mixture of real-time and non-real-time applications. However, this work lacked to mention the appropriate algorithm for the individual application (either for voice or file transfer protocol).

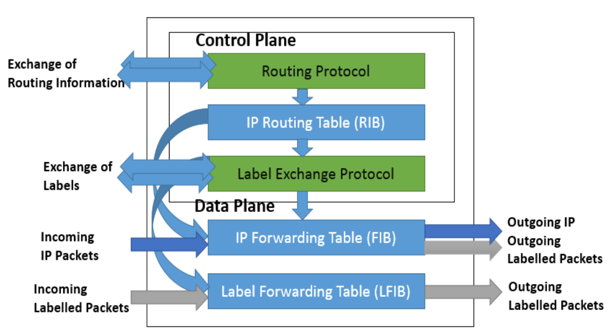


Figure 1. Existing MPLS Architecture

A stimulating idea for service providers to manage their network efficiently by improving the QoS to the customer is provided in [5]. Further issues were also mentioned as to allocate limited bandwidth with fairness to the users and the application of network management to monitor and control the traffic of multiple applications, although, there are still a lot of controversial issues yet to be resolved such as increasing network capacity and metered pricing. Gallon and Schelen [6] discuss bandwidth management in the next generation of packet networks. According to [6], there are issues surrounding the bandwidth management for next-generation voice and multimedia over packet networks. End-to-End QoS requirements for voice and multimedia service and how they might be best supported over a packet network infrastructure were investigated in [7]. However, the question of (how much bandwidth each of the multimedia services really requires) has not been answered for future generation networks.

Bandwidth allocation to each class type and provision of bandwidth protection and QoS can be implemented using admission control [8]. There are three "Bandwidth Constraint Models (BCM)", which have been experimental [13] to control bandwidth allocation/protection within the Differentiated Service Traffic Engineering (DSTE) framework. It is illustrated that with the implementation of the constraint models, Russian Dolls Model (RDM) can yield poor results since the pre-emption is not enabled. In the case of analysis and simulation results of Maximum Allocation with Reservation (MAR) and Maximum Allocation Model (MAM) bandwidth constraint models, the MAR bandwidth constraints model perform better than the MAM bandwidth constraints model [13,15]. RDM, MAR, and MAM are the three BCM proposed by the Internet Engineering Task Force (IETF) for supporting DSTE.

3. MPLS MODEL SCENARIOS

MPLS begins with the label forwarding at the ingress edge router called Label Edge Router (LER) in which the label is assigned and imposed by the IP routing process. Therefore, Label Switched Path (LSP) form the basis for labelled packets forwarding (label swapping) while Edge Label Switching Router labels IP packets, which are forwarded into the MPLS domain, or labels are removed and forwards IP packets out of the MPLS domain. The current MPLS architecture is illustrated in Figure 1. The setup of dynamic LSPs is configured manually to establish and propagate LSP information to other LSRs in the network. When the signaling protocols are enabled across the LSRs, the LSP information is transmitted throughout network. More resource utilization obtained because of the exchange and process of packets and instructions done in LSRs by dynamic LSPs than static LSPs. Static configuration requires to explicitly configure every LSR in an LSP manually with no signaling protocol enabled. The procedure of how to configure dynamic and static LSPs is depicted in Figure 2.

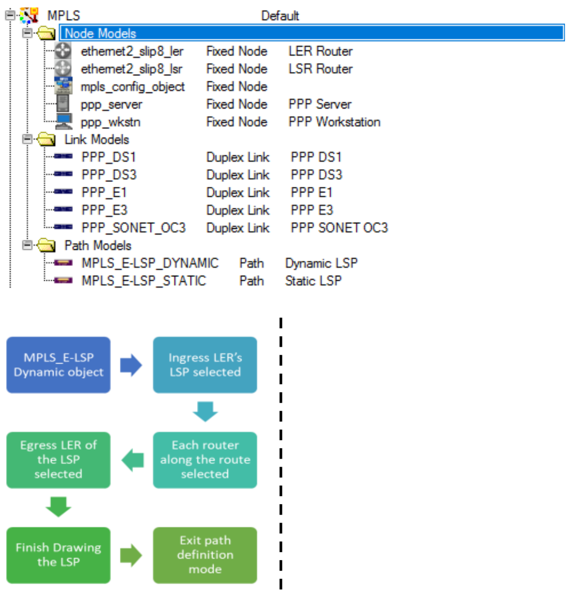


Figure 2. MPLS modules and flow charts of Dynamic and Static LSP models.

A. Dynamic and Static Label Switch Paths

The OPNET tool is used to design and simulate the performance of MPLS network model. It provides a virtual network environment for the entire network models, which include its routers, switches, protocols servers and individual applications [9]. The goal of the simulation is to obtain results and gain an insight into other model systems by evaluating the results. The MPLS model consists of configuration modules and connectivity of the nodes to generate packet switched data transmission from point-to-point. It is designed to support the availability of resources by providing multimedia services that are sensitive to transmission in order to meet the requirement of the (QoS). These modules are Application Definition, Profile Definition, IP (QoS) Attribute Definition, MPLS Attribute Definition and Worldwide Interoperability for Microwave Access (WiMAX).

Table I gives the parameters of voice and video as obtained from the OPNET. In the past, packet switch networks have been supporting multimedia applications such as audio, video, and data. There are two different approaches developed to provide adequate QoS: Integrated services and Differentiated services. The RSVP uses the integrated services approach as stated in [10-12], which is a state-establishment protocol that will enable the Internet to support real-time and multimedia applications, such as teleconferencing and videoconferencing applications [12, 13].

TABLE I. VOICE &VIDEO PARAMETERS

VOICE		***VIDEO***	
Attribute	***Values***	***Attribute***	***Value***
Encoder scheme	G.711	Frame per second	30
Voice Frame per packet	1	Frame size (B)	352x240 pixels
Type of Service	Interactive voice	Type of Services	Interactive video
Data rate (kbps)	120	Data rate (Mbps)	30

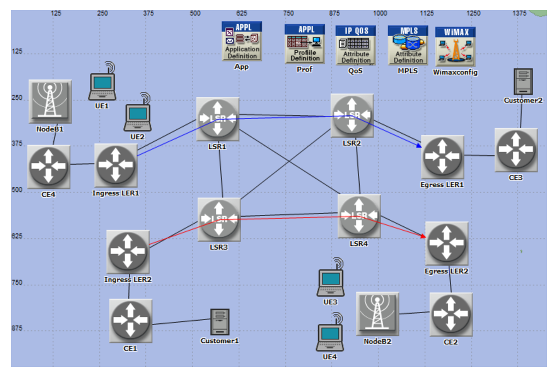


Figure 3. Static MPLS LSP

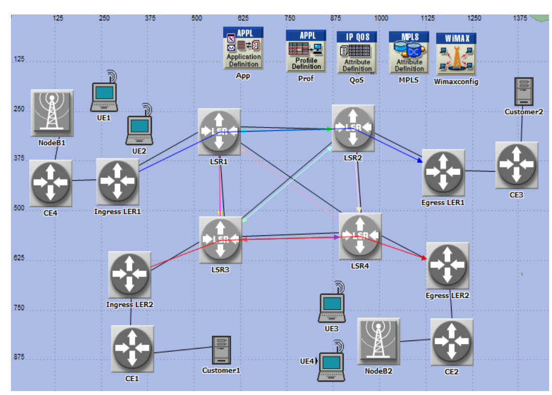


Figure 4. Dynamic MPLS LSP

All the routers (LERs and LSRs) along the route are defined by the LSP using MPLS_E-LSP_DYNAMIC object to provide the linkages. Then, an update of the LSP details is obtained before the simulation. This simulation uses the signaling protocol RSVP-TE to establish an LSP from source to destination. Also, a network model is employed for the static LSP configuration of the MPLS with the LSPs created from ingress LER1 to egress LER1 and from ingress LER2 to egress LER2. It is then compared with the scenario of the dynamic LSP configuration as shown in Figures 3 and 4 respectively. Each connection request has a unique LSP identity (ID) assigned by either the ingress LER1 or ingress LER2.

In the LDP, there are LSR discovery mechanisms, which implies that the protocol will initially discover the LSRs in the surrounds through the LSR mechanisms [27]. It is used between nodes in an MPLS network to establish and maintain the label bindings. For MPLS to operate correctly, label distribution information needs to be transmitted reliably, and the LDP messages pertaining to a particular Forwarding Equivalence Class (FEC) need to be transmitted in sequence. This is shown in Figure 5.

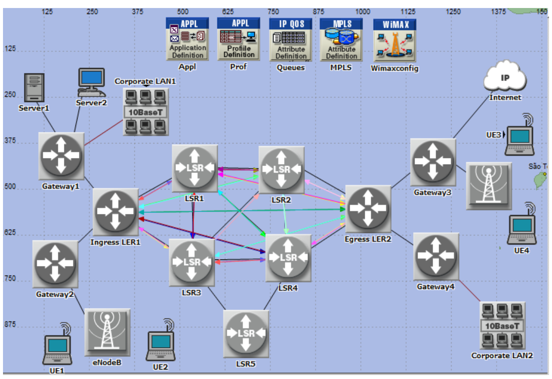


Figure 5. Implementation of MPLS with LDP between nodes

Figure 6 illustrates the traffic flow in MPLS from the ingress point to the egress point of the network. This shows the view of MPLS LSP configuration with the allocation of bandwidth on LSPs created from Ingress LERs to LSR1, LSR2, LSR3, LSR4, and egress LERs. In other words, the LDP configuration leads to the distribution of bandwidth on logical links of the LSRs. The design of MPLS models for bandwidth management using OPNET tools such process, node and network models can be found in [22, 23].

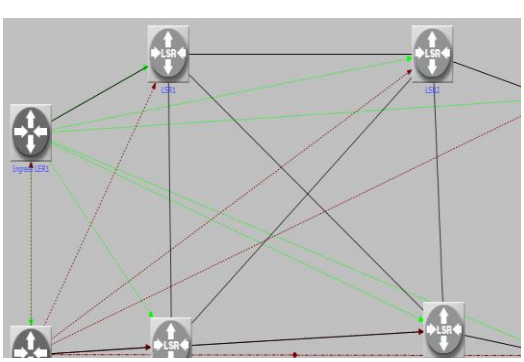


Figure 6. Traffic Flow using LSP in MPLS

B. Analysis of Packet Processing Algorithm

Let G = (N, E) be a graph depicting the physical topology of the network. Then, N is the set of nodes in the network and E is the set of links; Let H = (U, F, d) be the induced MPLS graph, where U is a subset of N representing the set of LSRs in the network, F is the set of LSPs, and d is the set of demands [26]. All the set of routers, in accordance with MPLS network formation, can be categorized into two subsets:

In an MPLS network, finding a solution to routing issues in terms of flow models is necessary in order to calculate one or a multitude of LSPs between a pair of edge "sender-receiver" nodes and define the sequence of the set intensity of traffic distribution between them [24, 25].

$N^+ = \{U_r^+, r = \overline{1, m_{LER}}\}$ - A Subset of LERs.
$N^- = \{U_j^-, j = \overline{1, m_{LSR}}\}$ – A Subset of LSRs.

U_r^+ −: $r - LER$ at which k-traffic arrives into the MPLS Network.

U_e^+ −: $e - LER$ at which k-traffic leaves the MPLS Network.

K_r^s - : Multitude of s is in Class of Services (CoS), arriving into $r - LER$.

$I^{k_r^s}$ –: Intensity of $k_r^s -$ traffic with servicing class, arriving into $r - LER$.

$p_{ij}^{k_r^s}$ −: Routing variable, which characterized the intensity of k_r^s- traffic in $(i,j) \in E$ link for every $r - LER$ and $k_r^s \in K_r^s$.

φ_{ij} – : Intensity of the available link bandwidth from i to j.

p_{ij} - : traffic from i to j.

$$\begin{cases} \sum_{J:(i,j)\in E} p_{ij}^{k_r^s} - \sum_{J:(i,j)\in E} p_{ji}^{k_r^s} = I^{k_r^s}, & if\ \ i = U_r^+; \\ \sum_{J:(i,j)\in E} p_{ij}^{k_r^s} - \sum_{J:(i,j)\in E} p_{ji}^{k_r^s} = 0, & if\ \ i \neq U_r^+, U_e^+; \\ \sum_{J:(i,j)\in E} p_{ij}^{k_r^s} - \sum_{J:(i,j)\in E} p_{ji}^{k_r^s} = -I^{k_r}, & if\ \ i = U_e^+; \end{cases} \tag{1}$$

The equations in (1) imply the number of LERs and LSRs in the network system. Furthermore, it shows the process of packet forwarding in MPLS from the ingress LER (entry) through LSRs to the egress LER (exit). This is to prevent packet loss on the routers in the MPLS network [25]. The whole set of k - traffics, arriving from users (access networks), depends on which edge router this traffic comes from and according to which class it will be serviced.

$$\sum_{s=1}^{S} \sum_{k_r^s \in K_r^s} p_{ij}^{k_r^s} \leq \varphi_{ij} - \sum_{i=1}^{S} \sum_{\substack{g \in U^+ \\ g \neq r}} \sum_{k_r^s \in K_r^s} p_{ij}^{k_g^s} (r \in U^+, (i,j) \in E) \tag{2}$$

The meaning of equation (2) inequality is that the traffic, routed from r - *LER*, cannot be exceed by its intensity of the available bandwidth of the link, which remains after traffics service [25], routed from other edge routers.

C. Bandwidth Estimation and Allocation Model

Consider a network of capacity C, which is distributed by J types of connection [28]. The connections could be a voice or video conference traffic as shown in equation (3). Let nj equal the number of connections of type j = 1 ,……., J:

$$S = \sum_{j=1}^{J} \sum_{i=1}^{n_j} BW_{ji}, \quad (3)$$

This implies that:

$$M_j(s) = \log \mathrm{E}\left[e^{sBW_{ji}}\right] \quad (4)$$

M_j (s) is the properties of the log-moment generating function, which represents equation (4).

BW_{ji} is the bandwidth requirement of the i connection of type j. Also, it represents an independent random variable.

In equation (5), given C and information about the number and type of connections, the bound implies that for any $s \geq 0$.

$$\log \mathrm{P}(S > C) \leq \log \mathrm{E}\left[e^{s(S-C)}\right] = \sum_{j=1}^{J} n_j M_j(s) - sC \quad (5)$$

This is useful for the decision on whether another class of traffic k can be added and retain the QoS guarantee. If A is given to be an acceptance region or boundary:

$$A = \left\{ n \in \Re_+^J : \sum_{j=1}^{J} n_j M_j(s) - sC \leq -\gamma \right\} \quad (6)$$

This will result in,

$$(n_1, \ldots, n_j) \in A \Rightarrow \mathrm{P}(S > C) \leq e^{-\gamma} \quad (7)$$

Equations (6) and (7) show region (A) of a new connection that can be accepted, without violating QoS guarantee that $\mathrm{P}(S > C) \leq e^{-\gamma}$.

$$\sum_{j=1}^{J} n_j M_j(s) - sC \leq -\gamma \quad (8)$$

$$\alpha_j(s) = \frac{M_j(s)}{s} = \frac{1}{s} \log \mathrm{E}\left[e^{sBW_{ji}}\right] \quad (9)$$

Rewriting equation (7) becomes,

$$\sum_{j=1}^{J} n_j \alpha_j(s) \leq C - \frac{\gamma}{s} \quad (10)$$

The symbol $\alpha_j(s)$ is the estimated bandwidth of a source of class j as shown in equations (9) and (10).

The admission control simply adds the effective bandwidth of a new request to the effective bandwidth of connections already in progress and accepts the new request if the sum satisfies a limit. It is observed that there is likely to be a variation of effective bandwidth of a connection over resources of the network.

All the signalling messages generated by a request will contain the identification (ID): the reply to the signalling messages will also have this ID. It can be observed that the estimated bandwidth value of 51.498 MB is evenly distributed and still has a reservation on the links as shown in Figures 7 and 9 respectively. Similarly, the percentage of bandwidth utilisation using LDP model is shown in Figures 8 and 10 respectively.

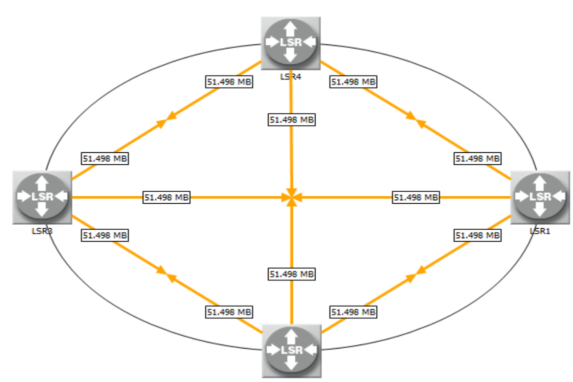


Figure 7. Circle view of bandwidth distribution using MPLS LSPs.

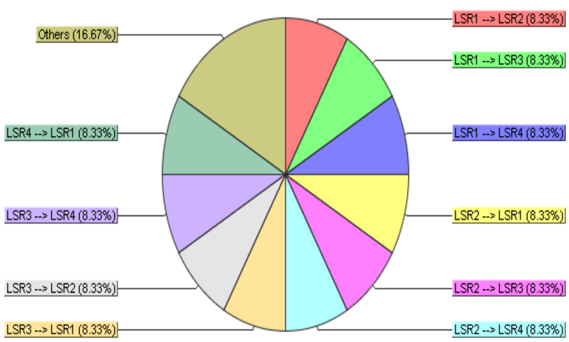


Figure 8. Percentage of bandwidth utilisation in MPLS LSPs.

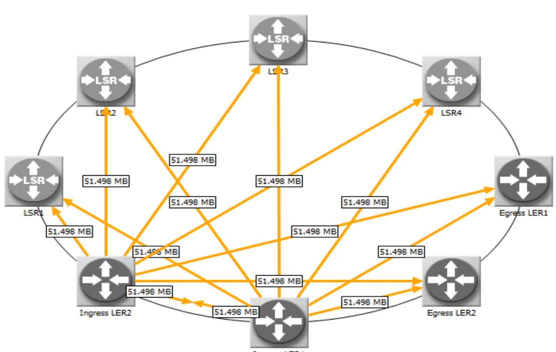


Figure 9. Circle view of bandwidth distribution using MPLS LSPs.

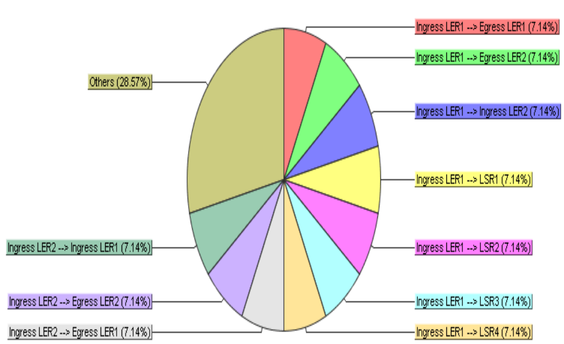


Figure 10. Percentage of bandwidth utilisation in MPLS LSPs.

Figure 10 shows improved utilisation of bandwidth with reservation of 28.57 % as compared with reservation of 16.67% in Figure 8. This indicates that moderate bandwidth utilisation can be used to control congestion in the network. However, the required standard for reservation is supposed to be 25% for moderate performance of the network. Each connection of ingress LERs to egress LERs and LSRs is allocated bandwidth of 51.498 MB with total flows of 14 (Traffic volume of 720.978MB).

D. Delay and Jitter Models

In this work, we employed the definition of jitter (J) model by IETF in [19]. This is based on the transit delay between the entry (Ingress LER) and the exit (Egress LER) nodes. Let Tj represent the delay experienced by the jth packet going through a queue. The difference of transit time between two consecutive packets of the tagged flow can be written as:

$$J_j = T_{j+1} - T_j \tag{11}$$

The average end-to-end delay jitter can be in form of expected absolute value of random variable

$$J = E\left[\left|T_{i+1} - T_i\right|\right] \tag{12}$$

We also adopt the approximate formulas for the (J) in three cases, in which the arrival rate stream is small, large and intermediate [20, 21].

$$J = E\left[\left|T_{i+1} - T_i\right|\right] = \frac{1}{\eta} \quad \text{for small arrival rate stream} \tag{13}$$

$$J = E\left[\left|T_{i+1} - T_i\right|\right] = \frac{1}{\mu} \quad \text{for large arrival rate stream} \tag{14}$$

where $\eta = \mu - \lambda$

λ: the total arrival rate

μ: the service rate

$$J = \frac{1}{\eta}\left[1 - e^{(\rho-1)/\rho}\left(\frac{1-\rho}{\rho} + e^{\rho-1/\rho}\right)\right] \text{for intermediate arrival stream} \tag{15}$$

where, Utilisation $\rho = \dfrac{\lambda}{\mu}$

Assume k represents small and large stream, then 1 – k to be part of the remaining.

Therefore, total jitter or packet delay variation is given by:

$$j = k\left(\frac{1}{\eta} + \frac{1}{\mu}\right) + (1-k)\frac{1}{\eta}\left|1 - e^{\frac{(\rho-1)}{\rho}}\left(\frac{\rho-1}{\rho} + e^{\frac{(\rho-1)}{\rho}}\right)\right| \tag{16}$$

$$= \frac{1}{\eta}\left[k\left(1+\frac{\eta}{\mu}\right)+(1-k)\left[1-e^{\frac{(\rho-1)}{\rho}}\left(\frac{\rho-1}{\rho}+e^{\frac{(\rho-1)}{\rho}}\right)\right]\right]$$

Substitute for $\eta = \mu - \lambda$

$$= \frac{1}{\eta}\left[k\left(1+\frac{\mu-\lambda}{\mu}\right)+(1-k)\left[1-e^{\frac{(\rho-1)}{\rho}}\left(\frac{\rho-1}{\rho}+e^{\frac{(\rho-1)}{\rho}}\right)\right]\right] \quad (17)$$

Further simplification gives

$$= \frac{1}{\eta}\left[k(1+1-\rho)+(1-k)\left[1-e^{\frac{(\rho-1)}{\rho}}\left(\frac{\rho-1}{\rho}+e^{\frac{(\rho-1)}{\rho}}\right)\right]\right]$$

$$j == \frac{1}{\eta}\left[k(2-\rho)+(1-k)\left[1-e^{\frac{(\rho-1)}{\rho}}\left(\frac{\rho-1}{\rho}+e^{\frac{(\rho-1)}{\rho}}\right)\right]\right]$$

$$\eta j = k(2-\rho)+(1-k)\left[1-e^{\frac{(\rho-1)}{\rho}}\left(\frac{\rho-1}{\rho}+e^{\frac{(\rho-1)}{\rho}}\right)\right] \quad (18)$$

Equation (18) implies the final total packet delay variation.

E. Mean Opinion Score (MOS) for voice

Mean Opinion Score (MOS) is presented for both theoretical based on the standard in [16] and simulation results as shown in Table II. MOS depicts the relation between rating value and users' satisfaction in term of the services provided and is a measure of Quality of Experience (QoE) for voice users in the network. While the R-factor is called rating factor which is used to measure the quality of the voice call based on parameters such as packet end-to-end delay, packet loss etc.

Id is the impairment caused due to the delay of voice signals and Ie is the impairment caused due to the packet losses in the network. The specification for theoretical MOS are given in [14-17]. Equation (19) shows the relationship between rating factor and the impairments. Equations (20), (21) and (22) are standards provided for the MOS.

$R = 942 - Id - Ie$

(19)

For $R \langle 0 \quad MOS = 1$ (20)

For $0 \langle R \langle 100$

$$MOS = 1 + 0.0035 * R + 10^{-6} * R * (R-60)(100-R) \quad (21)$$

For $R \rangle 100 \quad MOS = 4.5$ (22)

TABLE II. RELATIONSHIP BETWEEN R-VALUE AND USER SATISFACTION

R-value (lower limit)	***Theoretical MOS***	***Simulation MOS***	***User Satisfaction***
90<R<100	4.34	3.86	Very satisfied
80<R<90	4.03	3.66	Satisfied
70<R<80	3.60	2.94	Some users dissatisfied
60<R<70	3.10	2.10	Many users dissatisfied
50<R<60	2.58	1.39	Nearly all users dissatisfied

The difference between Theoretical and Simulation values of MOS can be represented by ($cm_j^T - m_j^S$) as shown in equation (23). This is the error at $m = m_j$, which is due to the delay of voice signals and packet losses in the network. The estimate value of the error e is given as follows:

$$e = \sum_{j=1}^{n}\left(cm_j^T - m_j^S\right)^2 \quad (23)$$

$$c = \frac{\sum_{i=1}^{n} m_i^T m_i^S}{\sum_{i=1}^{n}\left(m_i^T\right)^2} \quad (24)$$

R = rating factor
Id = impairment due to packet delay (s)
Ie = impairment due to packet loss
c = variable
m_j^T= theoretical MOS value
m_j^S= simulation MOS value

It is the value of *c*, which provides Least Square Fit (LSF) to the network model. The value of *c* is estimated to be 0.8120. By minimizing the value of c, the error due to delay of packets in the network model will be minimized. This is shown in the equation (24).

4. SIMULATION RESULTS

The results obtained are tentatively to improve for further research work using procedure of validation and refinement. As for the results of the implementation, the parameters of dynamic and static models for voice and video conference are used as shown in Figures 13 to 15 respectively.

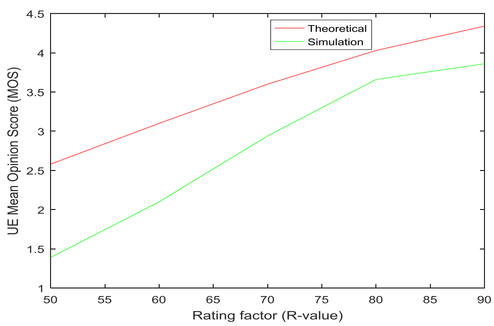
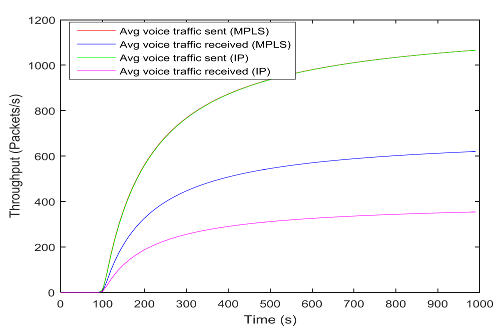
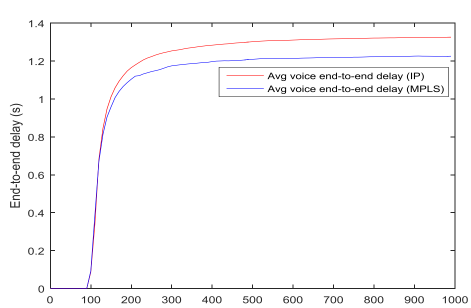


Figure 11. MOS values against Rating factor

There is a linear relationship between MOS and rating factor using theoretical and simulation results, as shown in Figure 11. This served as the qualitative technique using different level of satisfaction with QoS for voice communication. There is a deviation from 4.03 and 3.65 of MOS reaching the highest value of 4.34 and 3.84 at the R-value of 80 and 90 respectively. It is illustrated that the initial MPLS model designed was simulated in order to verify its performance using multimedia services. This served as the baseline for the subsequent change in configurations. Voice traffic sent for the baseline and the change in configuration are the same while there is variation in the received traffics as shown in Figure 12 (a). In addition, end-to-end delay in voice is shown in Figure 12 (b), indicating that delay is drastically reduced when MPLS is enabled.



(a)

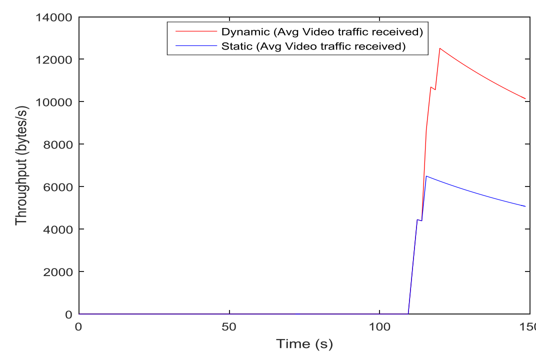


(b)

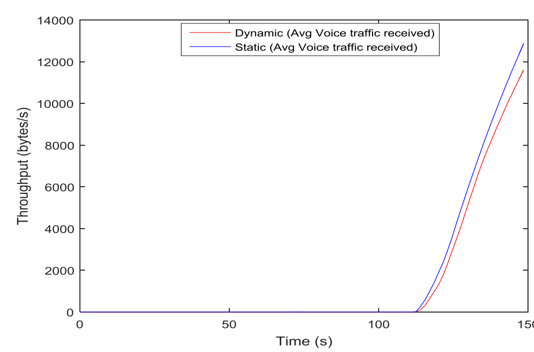
Figure 12. (a) MPLS baseline average traffic sent and received with MPLS enabled and without MPLS (IP) model (voice), (b) Average end-to-end delay in IP and MPLS (voice)

As shown in Figure 13 (a), packet delivered for dynamic model have considerable level of throughput more than static model with the application of video-conferencing application. There is a tremendous increase in the transmission of packets from one end of the site to another as a result of high throughput.

However, there exists a sharp decrease at the maximum for both. As can be seen from Figure 13 (b), the throughput received is able to increase rapidly to an average of about 13 kbps and 12 kbps for both configurations using voice. There appears to be a slight gap between dynamic and static configuration.



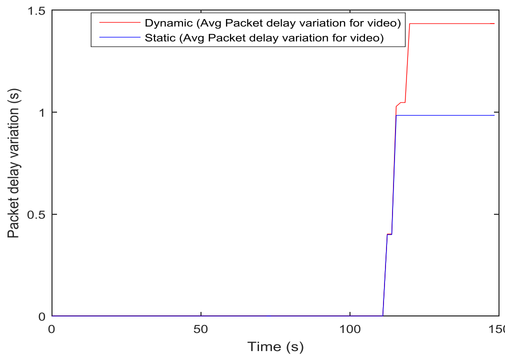
(a)



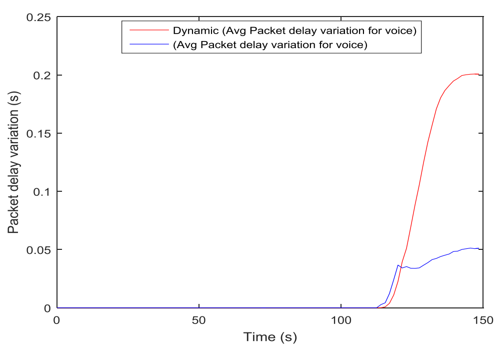
(b)

Figure 13. (a) Average Video Traffic received using Static and Dynamic LSP, (b) Average Voice Traffic received using Static and Dynamic LSP.

Figures 14 (a) and 14 (b) depict the packet delay variation (jitter) while Figures 15 (a) and 15 (b) show packet end-to-end delay for both video and voice traffics. Trending by the packet delay variation, there is an uprising to the average peak of about (0.2 s / 0.05 s) for voice and (1.4 s / 0.95 s) for video in static and dynamic models, which later remain steady. Also, end-to-end delay appears to follow the same pattern in which that of the voice has to reach up to (1.25 s / 0.65 s) and video has the peak of 3.9 s / 3.7 s) respectively. Packet delay variation is the parameter of variance in end-to-end delay among all the packets received from the user. On the other hand, end-to-end delay is the parameter that gives total voice packet delay [18].

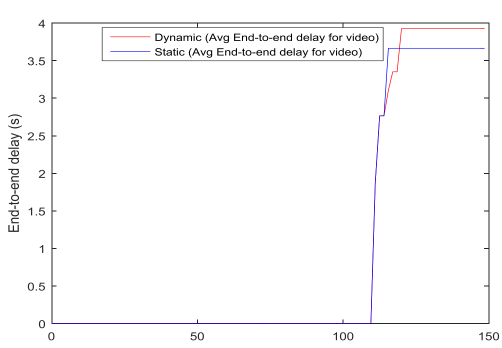


(a)

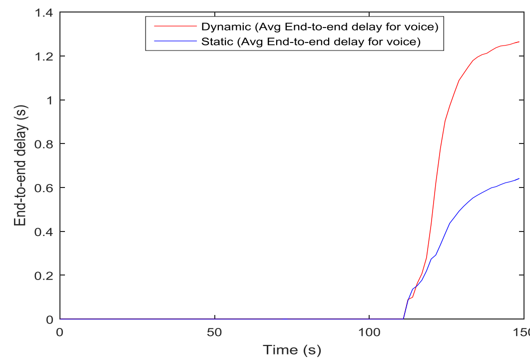


(b)

Figure 14. (a) Average Packet delay variation using Static and Dynamic LSP (video), (b) Average Packet delay variation using Static and Dynamic LSP (voice)



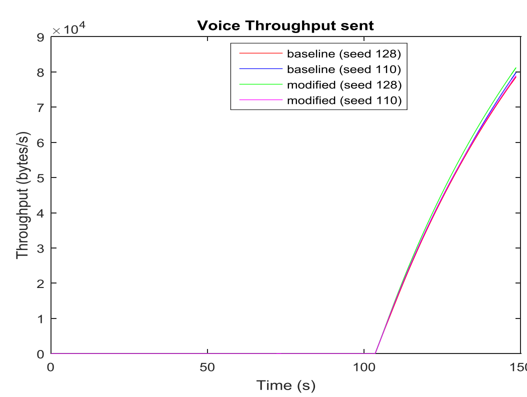
(a)



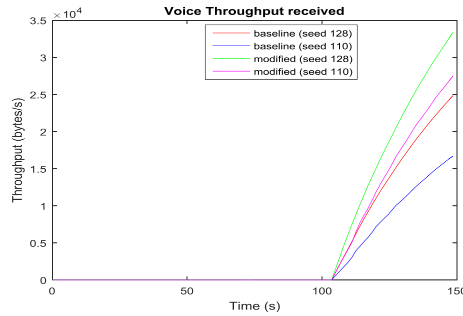
(b)

Figure 15. (a) Average End-to-End delay using Static and Dynamic LSP (video), (b) Average End-to-End delay using Static and Dynamic LSP (voice).

The performance indicates that there is an absolute packet delivery from ingress operating point to the egress endpoint. As for the results of the implementation, the MPLS baseline and modified MPLS networks with two scenarios (seed 128 and seed 110) using configurations of voice and video conference are presented, which yielded results as shown from Figures 16 to 18. A close linear relationship exists between baseline and modified model for the average voice traffic sent from the source of information (Ingress) in Figure 16 (a). There is an absolute variation in the result of traffic received at the destination (Egress) point. Higher throughput is experienced in the modified network. This is due to the LDP being configured at the core routers (LSRs) to allocate bandwidth uniformly as shown in Figure 16 (b).



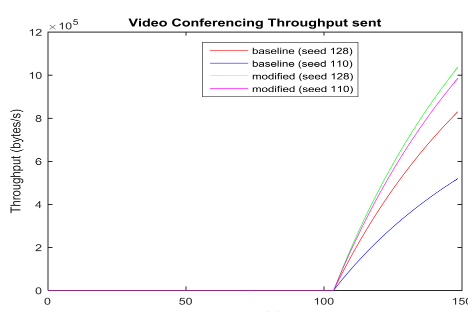
(a)



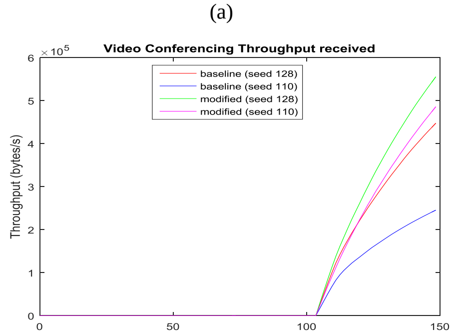
(b)

Figure 16. (a) Average voice Throughput sent (b) Average Voice Throughput received

As shown in Figure 17 (a), the video traffic sent spread out considerably with a slight difference of 30 kbps for the network with LDP configuration while a wide gap of 360 kbps can be seen on the baseline. There is a tremendous increase in the transmission of packets from one end of the ingress LER to another end of the egress LER. This indicates that there is more traffic on the distributed links in the core network. In Figure 17 (b), all the throughput received increases rapidly to an average of about 550 kbps and 490 kbps for the video conferencing configuration. There exists a considerable difference of the received traffic having average values of 440 kbps and 240 kbps respectively. This is an indication of constant traffic flows.



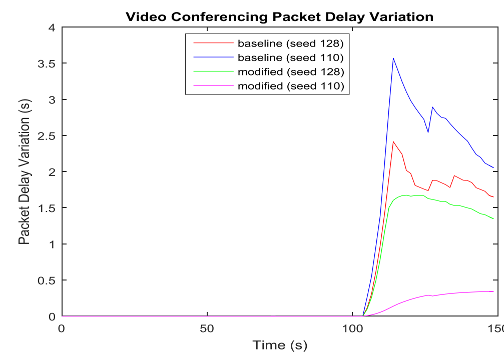
(a)



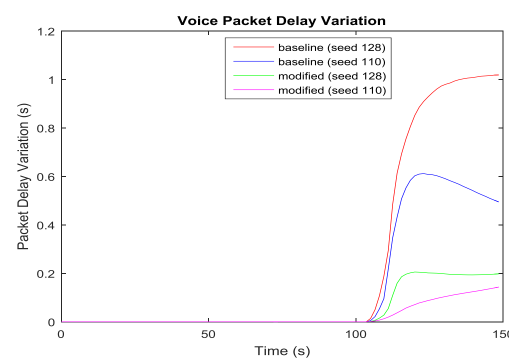
(b)

Figure 17. (a) Average Video Throughput sent (b) Average Video Throughput received.

Figures 18 (a) and 18 (b) illustrate the packet delay variation (jitter). In the packet delay variation graph, there are steady and low values resulting from the modified network with LDP of the average peak of about (0.2 s / 0.18 s) for voice and (1.4 s / 0.4 s) for video as compared with the baseline without LDP. However, the baseline result for video appears to decrease sharply.



(a)



(b)

Figure 18. (a) Average Packet delay variation (Voice) for MPLS baseline and modified (b) Average Packet delay variation (Video) for MPLS baseline and modified.

5. CONCLUSION AND FUTURE WORK

In summary, performance evaluation of the MPLS technology using static and dynamic models are presented. This would sustain the future exponential increment in user demand with adequate allocation of bandwidth. This can be justified using the theoretical and simulation results, which have moderate performance due to low indications of end-to-end delay and high throughput.

Further evaluation of the MPLS-TE in combination with Software Defined Networking (SDN) will be put into consideration for the adequate allocation and reservation of bandwidth to the next generation of mobile and wireless networks. This will provide separation of the control plane from data plane, whereby solving problem of map abstraction in traffic engineering. In addition, an appropriate QoS scheme would be required to meet the accelerating demands for adequate bandwidth requirement and specification for the future technology. More verification, validation, and refinement of the model designed would be required to meet the requirements of the data rates and minimum bandwidth specification for 5G technology.

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