Bandwidth Management Using MPLS Model for Future Mobile Wireless Networks

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Abstract. The recent surge in the development of new technologies, most especially in the field of mobile and wireless communications, requires the adequate maintenance and overall procurement of network infrastructures. This is due to a great deal of accelerating demand from Mobile users having access to real-time information such as data, voice and video services. Therefore, the operators and service providers require seamless integration of network protocols with an improved quality of service (QoS). This paper addresses the performance of multimedia services in Multiprotocol Label Switching (MPLS) nodes and network models design using a simulation approach. MPLS ensures the reliability of the communication minimizing the delays and enhancing the speed of packet transfer. It is valuable in its capability of providing Traffic Engineering (TE) for minimizing the congestion by efficient throughput. The verification of the MPLS model will be the focus of the performance evaluation. An elaborate description of MPLS and its principle of operation will be required. It will eventually address the challenges of packet loss, high latency, high operational cost, more bandwidth utilization, and poor QoS.

Keywords: MPLS network · MPLS-TE · Packet delivery fraction Multimedia services

1 Introduction

Mobile and Wireless application services form the basis for the success of the technology in the next generation of networks. Therefore, management of the bandwidth has emerged as a powerful platform for controlling the traffic volume of future mobile and wireless networks. A high priority traffic of voice and video has a large proportion of bandwidth consumption in wireless communication systems. Bandwidth is the rate at which data (frame/packet) is transmitted over the network link [1].

Bandwidth management is a generic term that describes the various techniques, technologies, tools and policies employed by an organization to enable the most efficient use of its bandwidth resources [2]. In [3, 4], bandwidth management is defined as a process of allocating bandwidth resources to critical applications on a network.

The key function of bandwidth management is to control the flow of packets on the network links to avoid traffic exceeding the capacity of the network, which would lead

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to congestion. This implies that more capacity requires more bandwidth. However, bandwidth is a very limited resource and in most markets is expensive to acquire. Therefore, there is an expectation that demand for data capacity will increase a thousand fold by 2020 [5].

Packets flow would suffer long queuing delays at congested nodes and possibly packet loss if buffers overflow [6]. To solve this problem, managing the available bandwidth would be of benefit to both the users and operators. These, in turn, play a vital role in minimizing the cost of operation rather than demanding for more bandwidth, which will be very expensive. Whilst at the same time used to monitor the effectiveness and efficiency in the performance of the network. The approach stated in [7, 26] is the purpose of supplying bandwidth on a network in order to reserve capacity for users. Nevertheless, the demand is low as compared with the operational capacity of the network.

The aim of this paper is to study the performance of multimedia services in MPLS Network model using a simulation approach. This will eventually proffer a solution to the bandwidth issues of the next generation of Mobile Wireless networks. It will be achievable by proposing design of MPLS node and core networks to manage bandwidth efficiently as possible for the future mobile and wireless networks. Implementation is by performing dynamic and static configuration of the MPLS model network as part of traffic engineering (MPLS-TE). The overall structure of this paper takes form of four sections. Section 1 gives a brief background and related work of the research carried out. Next is the Sect. 2, which entails model design, simulation, and verification of the proposed MPLS technology. Then, Sect. 3 yields the results, and lastly, Sect. 4 concludes the paper.

1.1 Related Work

Distinct bandwidth management techniques have been proposed [6–15]. Research in the area of Multiprotocol Layer Switching (MPLS) technology had been in existence for decades. However, there has been no detailed investigation of using this mechanism for the purpose of bandwidth management to solve the critical problem of delay. In addition, this is a technique that would utilize the available bandwidth to meet the requirement of QoS is required.

Available bandwidth is the maximum throughput of the communication channel without disputing any ongoing flow in the network [16]. This is view according to [17] that many of the bandwidth management techniques proposed have their pros and cons due to its appropriate in a given situation than the other. The bandwidth allocation algorithm proposed in [18] gives the same support. However, the exploration and the proper investigation of other techniques are consider for further works.

George provides a stimulating idea in [2] for the service providers to manage their network efficiently by improving the quality of service to the customer. Further issues is 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 many controversial issues yet to be resolved such as increasing network capacity and metered pricing.

Chris and Olov discuss bandwidth management in next generation of packet networks [19]. According to the two authors, there are issues surrounding the bandwidth management for next-generation of voice and multimedia over packet networks. End-to-End QoS requirements for PSTN-grade voice and multimedia service and how it might best support over a packet network infrastructure were investigated. However, the unanswered question of, what amount of bandwidth does each of multimedia services really required?

The authors in [20] proposed Software-based End-to-End Bandwidth Allocation and Reservation for Grid application. The description of the multi-layered architecture that can support the network resource reservation is given. However, there has been little discussion on its interface using networking terms instead of application terms.

In [21], Sanjeev et al. develop ideas of bandwidth management for mobile media delivery. They studied the fundamental problems of mobile broadband networks especially in 3G and 4G technologies such as packet loss and delay. In addition to that, there remains a paucity of performance analysis of the rate control algorithm.

2 MPLS Node and Network Design

Multi-Protocol Label switching (MPLS) is a fast packet forwarding and scalable mechanism that is widely used in the core network [22]. MPLS introduces the Label Switched Path (LSP) tunnel, which provides the mechanism to transport labeled data packets from the source node along the path to the destination node. There are three components in an MPLS network, ingress and egress Label Edge Routers (LER), and Label Switch Routers (LSR). LERs are located at the edges of the MPLS network.

The design of the MPLS node and network models have been implemented as shown in Figs. 2 and 3 respectively. This is in accordance with the configuration of the individual node starting with the application and profile definitions nodes, where the parameters are set out for voice and video. It therefore, seek to obtain statistics collection of the performance behaviour of the model, then running of simulations to view the obtainable results.

The first level of planning design is to create the process model of the peripheral node with define variables, macros, and transitions. A finite state machine (FSM) implements the behaviour of a process model using the states and transitions to determine the actions to be performed [23]. The peripheral nodes proceed to the central hub by point-to-point links, which can be unidirectional, or bidirectional. The main role of the MPLS node model is to simulate packets forwarding from one site node to another site node through packet switching technology. Looking at the nodes shown below in Figs. 1 and 2 indicate the linking of mesh topology using LSRs as the core network. This makes further connection to the sites through LERs both at the ingress and egress points. A source module is another essential aspect of node model, which generate the packets. The processor assign destination addresses to the packets, sends them to a node of the point-to-point transmitter, and retrieve packets arriving from the point-to-point receiver.

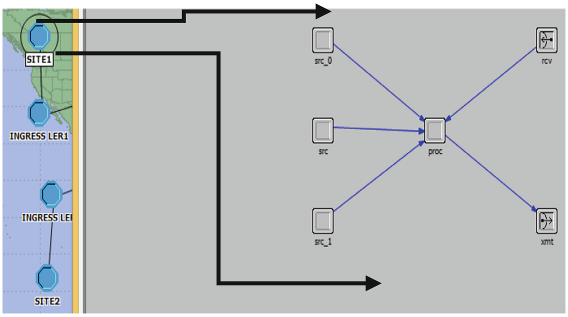

Fig. 1. Source nodes, peripheral nodes, and processor

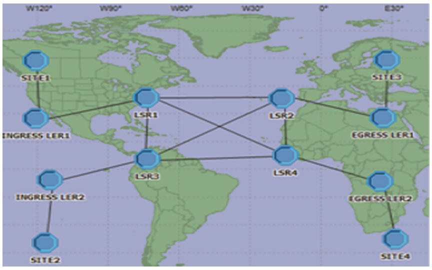

Fig. 2. Nodes model design

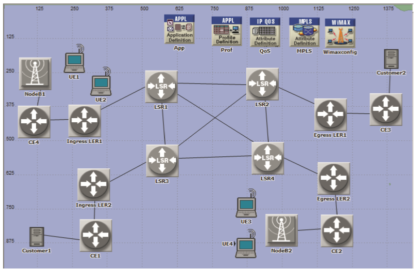

Fig. 3. MPLS network model design

The MPLS network model is shown in Fig. 3, it consists of configuration modules and connectivity of the nodes to generate packet switched data transmission from point-to-point. It is structure to support the availability of resources in order to meet the requirement of the quality of service (QoS). These modules are Application Definition, Profile Definition, IP QoS Attribute Definition, MPLS Attribute Definition and WiMAX. There exist the imposition of the label at the ingress routers LER1 and LER2 called label edge routers, then swapping of the label occurs at the core routers referred to as LSR (label switch routers). Next, is the removal of label and transformation to IP packet format at the egress routers LER1 and LER2 (label edge routers).

Packet Delivery Fraction (PDF). Packet delivery fraction is the ratio of a total number of a data packet received to the total number of data packets sent. This is an estimation of how routing protocol is efficient and effective [24, 25]. The Table 1 shows how the transmission of packets of voice and video conference from the source to the destination. It is justified that a number of packets received are less than the number of packets transmitted. This could be the result of packet drops along the channel of communication. Table 1 illustrates the relationship between packets sent from the source and packets received at the destination using voice and video services.

Table 1. Packet delivery fraction

Multimedia services	Number of packets received (packets/s)	Number of packets sent (packets/s)	Packet delivery fraction
Voice	677	2000	0.3385
Video	14000	19333	0.7241

2.1 MPLS Model Simulation and Verification

OPNET simulator is very useful when working with complex networks with a big number of devices and traffic flows, or in networks where a little change could be critical. Prior to any change in the implementation, it is possible to predict the behaviour and to verify the configurations of the devices [23]. Generally, probability theory and statistics are appropriate for the validation and verification of the network model. As the simulation model gained an improvement, the need to verify and validate the model will be of highly considerable. Verification determines whether the model performs the intended function and meets the required specifications. The fundamental procedure of verification is testing that the OPNET tools and mathematical model are working properly.

Utilization is the ratio of the offered load to the available resource in a given period (instantaneous) [30]. There is provision for Eqs. (1) and (2) as follows:

$$\rho = \frac{\lambda.s}{\varphi} \tag{1}$$

This implies that

$$d = \lambda.s \tag{2}$$

$$\rho = \frac{d}{\varphi}$$

ρ = utilization,
λ = demand for the resource per unit time,
d = demand the resource,
s = waiting or holding time,
φ = supply of the service provided or capacity of the system.

According to [26], utilization is applicable to any resource. If ρ is less than 0.50; this is an under-utilization of the resources. As ρ increases about 0.75, the time spent waiting grows exponentially, asymptotically approaching infinity. If exceeds 1.0, then the number of entities waiting for the resource grows linearly with (d−φ). As indicated below, the mean delay is inversely proportional to the number of packets per second in Eq. (5) while Eq. (6) shows that number of sources is directly proportional to the ratio of number of packets/s and mean service requirement. In Fig. 6, the mean service utilization is inversely proportional to the number of sources. There is variation in the service supplied to the customers at different utilization percentages (25%, 50% and 75%).

Know that 1 byte = 8 bits, let pps = packets/s, bps = bits/s
Therefore, $pps * average\_packet\_size = bps$
Let the Average rate and utilisation be A_r and ρ respectively
C = buffer service capacity (bits/s)
n = number of sources generating background traffic

$$n = \frac{\rho * C}{A_r} \tag{3}$$

but,

$$C = \frac{\lambda}{\mu}(bits/s)$$
$$\lambda = \frac{no\_of\_packet}{mean\_arrival\_time} = (packets/s) \tag{4}$$

$$Mean\_delay(\bar{w}) = \frac{1}{\mu C - \lambda} = \frac{1}{\frac{C}{1/\mu} - \lambda} \tag{5}$$

$$n = \left(\frac{d}{\varphi}\right) * \left(\frac{1}{A_r}\right) * \left(\frac{\lambda}{\mu}\right). \tag{6}$$

3 Results

There is presentation of results of the initial node and process models designed in Fig. 4. There is a constant delay of 0.001 ms and the throughput is increasing rapidly from zero to the level of approximately 16000 packets/s. As for the network designed, the application and profile configurations of voice and video conference are used which yielded results as shown in Figs. 5 and 6 respectively. Subsequently by the discussions of the results obtained from the node and network models. This is likely to be a better communication link between two points or more. As shown in Fig. 4 (left), there is a tremendous increase in the transmission of packets from one end of the site to another end of the site. It is an exponential increment showing a considerable amount of packets transmitted from the source to the destination.

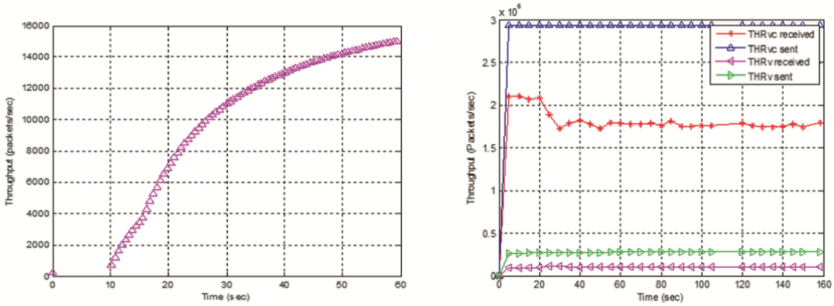

Fig. 4. Throughput of the link for the nodes model *(left)* Throughput sent and received for voice and video *(right)*.

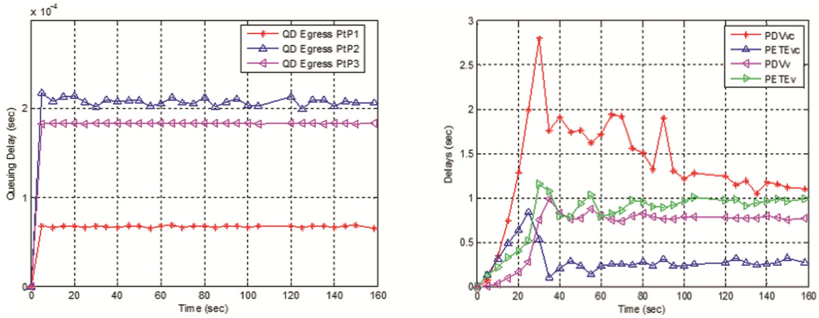

Fig. 5. Queue delay from point-to-point at the egress (*left),* Packet delay variation and end-to-end delay for voice and video *(right)*.

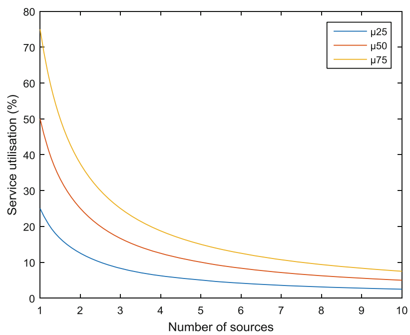

Fig. 6. Service utilization versus number of sources

As can be seen from Fig. 4 (right), the overall throughput is able to stabilize for both voice and video traffics. There appears a wide gap between video traffic (THRvc) sent and video traffic received with the effect of more packets transmitted than packets received. In the case of voice traffic (THRv), a small gap occurs between a number of packets transmitted and packets received which indicates that both have low throughput within the range of 0 and 500000 packets/s. In general, from Fig. 4 (right), it provides an important opportunity to advance the understanding of the throughput of video conferencing traffic is considerably higher than that of voice traffic due to the level of bandwidth consumption. However, voice transmission is quite easily manageable as compared to video transmission.

Figure 5 (left), the queue delay of point-to-point (PtP) of the different locations of nodes is as shown. The queue delay of PtP1 of 0.08 ms is far lower than that of PtP2 and PtP3 with 2.2 ms and 1.8 ms respectively. In other words, PtP1 has low latency compared to others which imply that the packets will likely move faster in the link with respect to PtP1 whereby enhancing the performance of the network.

The Fig. 5 (right) illustrates the differences between the packet delay variation (jitter) and packet end-to-end delay for both video and voice traffics. As for the packet delay variation, there is an uprising to the peak of about 2.8 s at the initial stage of the video traffic which later drop down by fluctuation to an approximate value of 1.1 s while a remarkable constant delay exists for voice with a minimum of 0 and maximum of about 0.75 s. This is because data rate of the voice is lower than that of the video. In addition, end-to-end delay appears to follow a different pattern in which that of the voice has to reach up to 1.2 s and video has the peak of 0.8 s with little variation.

4 Conclusion and Future Work

In summary, the approach used in this piece of research is similar to that reported in [2], but with different technique and implementation. Therefore, there is need for a thorough study of the performance of the MPLS technology and its implementation that would sustain the future exponential increment in user demand. This observational study of the node model designed are likely to have moderate performance due to low values of end-to-end delay, low queue delay, and high throughput. An implication of the packet delivery fraction is the possibility, which yields a positive performance of the voice packets delivered from the source to the destination due to its low bandwidth consumption as compared with a video packets delivery fraction.

The use of MPLS technology to implement bandwidth management in the future mobile wireless network is reliable and profitable due to its valuable cost to the both operators and service providers. Then it is possible that critical problem of delays such as end-to-end delay, queue delays, and packet delay variation are less likely to occur in the design. However, it will be an additional cost to deploy MPLS technology to the existing network, instead of eliminating existing IP technology. This will enhance the compatibility of the existing facilities.

Performance evaluation of the QoS schemes such FIFO, PQ, and WFQ will be further implementation to assess the services provision to the users. Comparative study of theoretical queue and OPNET queue designs are necessary. More analysis of the MPLS traffic engineering (MPLS-TE) will put into consideration for the next generation of mobile and wireless networks. Further 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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