Implementation of Traffic Engineering in MPLS Network by Creating TE Tunnels using Resource Reservation Protocol and Load Balancing the Traffic

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***Abstract*- Traffic engineering (TE) is most effective in networks where some links are heavily utilized and have little or no bandwidth available while others carry little or no traffic. It is of great importance to the recent development of mobile and wireless technologies. Without the process of TE, there is possibilities of having under-utilization and over utilization problems along the link. It is necessary to consider the implementation that would avoid the goal of network and unguaranteed bandwidth delivery. Therefore, the operations and service providers require seamless combination of network protocols with an improved quality of service (QOS). This paper will be focusing on Resource Reservation Protocol Tunneling Extension Multiprotocol Layer Switching (RSVP-TE MPLS) for sustainable mobile wireless networks. This will make provision of bandwidth allocation possible by implementing the configuration of the dynamic and static LSPs (Label Switching Paths). The network model designed will be used for this purpose by using simulation approach. The verification of the MPLS model will be presented. It will eventually maximize bandwidth utilization, minimize operation cost and improve QOS.**

Keywords: Traffic Engineering; Load Balancing; MPLS Networks

1. INTRODUCTION

Traffic engineering (TE) broadly defines the optimization of functional abilities of the network [1]. This optimization is done by diverting the traffic to the paths that are lightly loaded in order to balance the load amongst the paths as per the various metrics calculated. Methodologies for TE proposed all over the world can be divided in to state dependent and time dependent. Time dependent functionalities engineer the traffic on the basis of long time scale. On the other hand, state dependent methods alter the traffic in short time scale depending on the different metrics calculated online or offline of the present traffic. The aim of both these methods of course is to balance the traffic so as to avoid the congestion. Present day IP network relies on the best effort service but as there is considerable growth of the applications that rely on the services of the Internet for their operation, there is huge competition amongst the ISPs to provide Quality of Service (QoS). QoS refers to the transport of traffic in the network as per the agreement between the user and the ISP which is known as Service Level Agreement (SLA).

Internet Engineering Task Force (IETF) proposed Multiprotocol Label Switching (MPLS) for providing QoS in the Internet. MPLS is a very scalable, protocol independent, data-carrying mechanism. In MPLS the forwarding decision are made solely on the content of the assigned labels without a need to examine the networks layer header of the packet itself. In MPLS, one or more.

Virtual path similar to the Asynchronous Transfer Mode (ATM) or Frame Relay is set up which is known as Label Switched Path (LSP). Forwarding the packets on the basis of label facilitates source routing and QoS. In this paper, we survey the work done in load balancing in the network by various authors that had the following motivations:

1) Reducing Congestion in the network.
2) Reducing packet loss and packet delay.
3) Providing QoS parameters like fault tolerance.
4) Increasing overall efficiency of the network.

RELATED WORK

Researchers have proposed various bandwidth management techniques. In addition, research in the area of Multiprotocol Layer Switching (MPLS) technology had been in existence for decades. However, much work has not been employed 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.

The routine maintenance of network performance is an important challenge for the respective operators in in-built features of wireless networks. The main purpose of operators is to satisfy their subscribers by providing the QoS 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 [9-12]. These two aspects of the resource allocation are called “admission control” and “scheduling”.

Radio resource management (RRM) is the system level control of radio transmission characteristics in wireless communication systems [13, 14]. This system allows parameters such as transmit power, transmit rate and modulation scheme to be controlled in order to utilise the limited radio spectrum resources and radio network infrastructure efficiently. In order to achieve an improved and efficient utilisation of resources, adaptive RRM schemes that can adjust the radio communication parameters dynamically to the QoS and throughput requirements are considered.

These schemes are particularly considered in the design of wireless systems [15-17], in view of maximizing the system spectral efficiency without sacrificing the system performance.

I. MPLS TRAFFIC ENGINEERING (MPLS-TE)

MPLS is one of the tools that can be used to implement traffic engineering. An MPLS network is of the type that gives preferential treatment to certain types of traffic, which needs to have TE-configured differently from a network that does not. TE implementation that accommodates traffic of different priorities is said to be DiffServ-aware. MPLS. Core networks shown in Fig.1.2 and Fig.1.3 include two LSP scenarios: a dynamic (full mesh) LSP model and a static LSP model. The dynamic LSP can be configured with explicit or Constraint-based SPF (CSPF) routes. This will calculate an optimum explicit route (ER), based on specific constraints. CSPF relies on a Traffic Engineering Database (TED) to do those calculations. The resulting route is then used by RSVP- TE. At the beginning of the simulation, all dynamic LSPs are signaled using RSVP or Constraint-based LDP (CR-LDP). Static LSPs are not signaled.

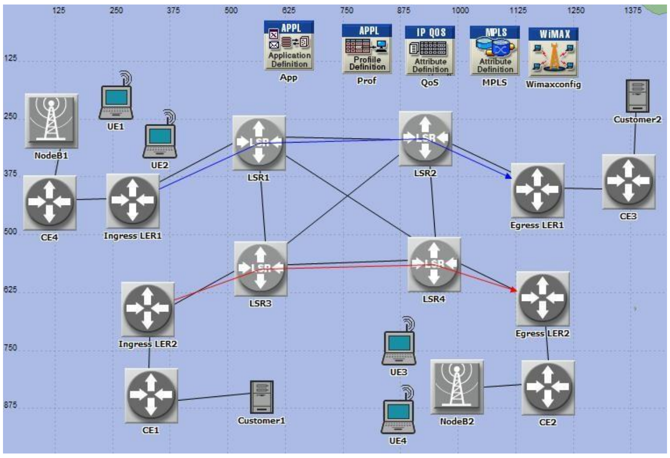


Fig.1.1. MPLS Network model design

A. Differentiated Services MPLS Traffic Engineering (DSTE)

The Differentiated Service MPLS Traffic Engineering (DSTE) is an aspect which combines the capabilities of QoS and DSTE capabilities of MPLS to allocate bandwidth and control QoS for various virtual networks (also known as the class of service in DSTE). The allocation of bandwidth to each class type and provision of bandwidth protection and QoS can be implemented using admission control. There are three "bandwidth constraint models" which have been experimental (Request for comments) to control bandwidth allocation/protection within the DSTE framework.

It is illustrated that with the implementation of the constraint models, 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.

B. Resource Reservation Protocol (RSVP)

In the past, packet switch networks have been supporting multimedia applications for those that integrate 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. RSVP is a state-establishment protocol that will enable the Internet to support real-time and multimedia applications, such as teleconferencing and videoconferencing applications. These applications require reservations to be in the Internet routers, and RSVP is the protocol to set up for these reservations. The key features of RSVP include flexibility, robustness, scalability through the receiver control reservations, sharing of reservations and use of IP multicast for data distribution. RSVP is not a routing protocol but works in conjunction with the routing protocol. It is usually referred to as signaling protocol used in MPLS Traffic Engineering.

C. MPLS Model Scenarios

The label forwarding in MPLS begins at the ingress edge router called Label Edge Router (LER router) in which the label is assigned and imposed by the IP routing process. This is followed by the swapping of labels on the contents of the label forwarding table in the core using Label Switch Router (LSR). At the egress edge router, the label is disposed and a routing lookup is used to forward the packet. Therefore, LSR form the basis for labeled packets forwarding (label swapping) while Edge LSR labels IP packets and forwards them into the MPLS domain, or removes labels and forwards IP packets out of the MPLS domain.

Dynamic LSPs with CSPF routes must use a routing protocol based on the SPF algorithm. Specifically, make use of OSPF or IS-IS as a routing protocol. 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 signaling protocol (RSVP- TE) to establish an LSP from source to destination.

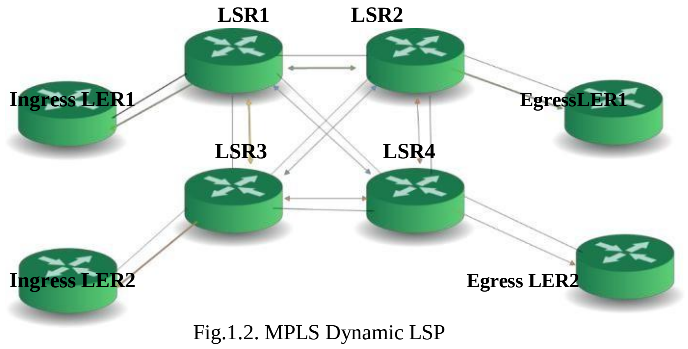


Fig.1.2. MPLS Dynamic LSP

Also, network model in Fig.1.3 shows 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. Each connection request has a unique LSP identity (ID) assigned by either the ingress LER1 or ingress LER2. All the signaling messages generated by a request will contain this ID: the reply to the signaling messages will also contain this ID.

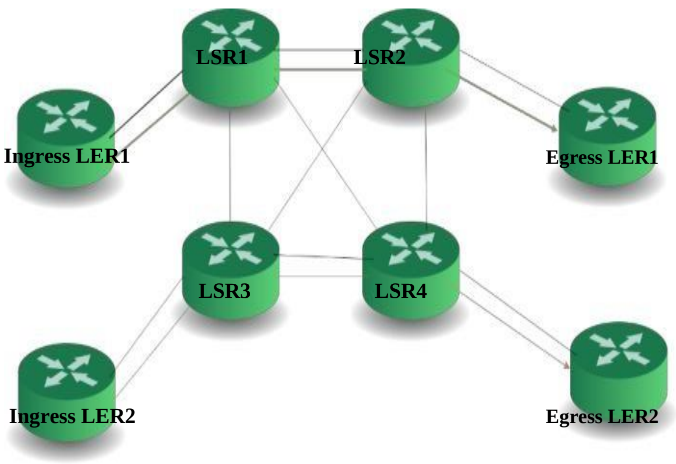
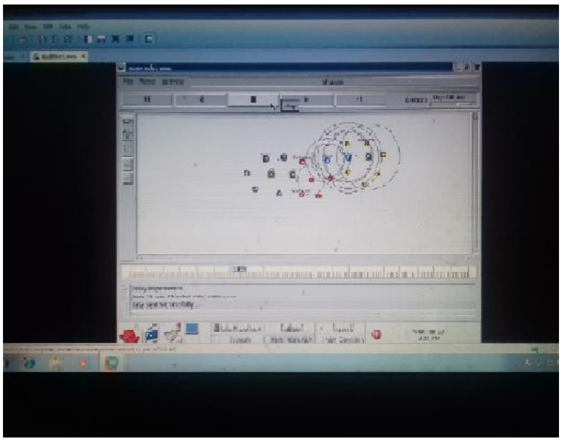


Fig.1.3 MPLS Static LSP

D. MPLS Model Verification and Analysis

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 behavior and to verify the configurations of the devices. Generally, probability theory and statistics will be used for the validation and further verification of the network model. As the simulation model has gained an improvement, the need to verify and validate the model is 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.



Let the Average rate and utilisation be A_r and ρ respectively

C = buffer service capacity (bits/sec)

n = number of sources

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

$$Mean\_delay(w) = \frac{1}{\mu C - \lambda} = \frac{1}{\frac{C}{1/\mu} - \lambda} \tag{4}$$

Furthermore, the Mean Opinion Score is presented for all simulations. It is defined as a measure of Quality of Experience (QoE) for voice user in the network. The E-Model defined as the statistical estimation of quality measures. 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. It is expressed as follows [22-29]

$$R = 94.2 - I*d - I*e \tag{5}$$

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 below are given in [22-27].

R < 0 $MOS = 1$

0<R<100

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

R>100 $MOS = 4.5$

Fig1.4

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, which implies that error is minimized when minimizing the value of c.

E. Results and Discussions

It can be seen that all the results are tentatively to change for further research work by way of validation and refinement. As for the results of the implementation, the dynamic and static configurations of voice and video conference are used which yielded results as shown in Fig.6 to Fig.12. Table I depicts rating value and users satisfaction in term of services.

Linear relationship between mean opinion score and rating factor using theoretical and simulation outputs is shown in Fig.4. 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. While the graphical representation of the throughput and delay is illustrated in Fig. 5. This implies that throughput is inversely proportional to the delay in the network with an approximate exponential decrement as shown in equation 4.

In Fig. 6, the plot demonstrates the overall throughput of the designed model. The output of the performance indicates that there is an absolute packet delivery from one access point to another. This is likely to be a better communication point-to-point link with an approximate value of 63% packets received for both voice and video.

As shown in Fig. 7, packet deliver fraction of the dynamic configuration has considerable amount of throughput than static configuration with application of video-conferencing. There is a tremendous increase in the transmission of packets from one end of the site to another end of the site. However, there exist sloppy decrease at the maximum for both.

As can be seen from Fig. 8, 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 a slight gap between dynamic and static configuration with indication that conversational and streaming classes are mainly intended to carry real-time traffic flows as stated in [28].

The Fig. 9 and Fig. 10 illustrate the packet delay variation (jitter) while Fig. 11 and Fig. 12 show packet end-to-end delay for both video and voice traffics. As for 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 / 1.0 s) for video in static and dynamic configurations, 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

TABLE I. RELATION BETWEEN R-VALUE AND USER SATISFACTION [27, 28]

R-Value	***MOS***	***User satisfaction***
90	4.34	Very satisfied
80	4.03	Satisfied
70	3.60	Some users dissatisfied
60	3.10	Many user dissatisfied
50	2.58	Nearly all users dissatisfied

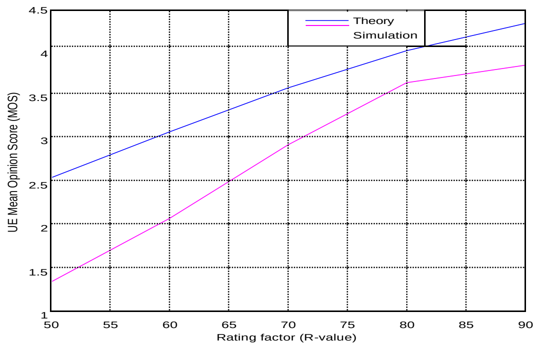


Fig. 4. MOS values against Rating factor

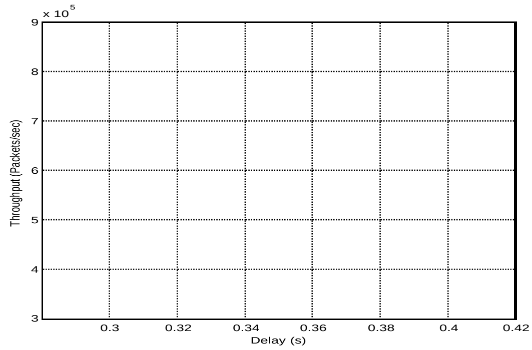


Fig. 5. Power of the Network

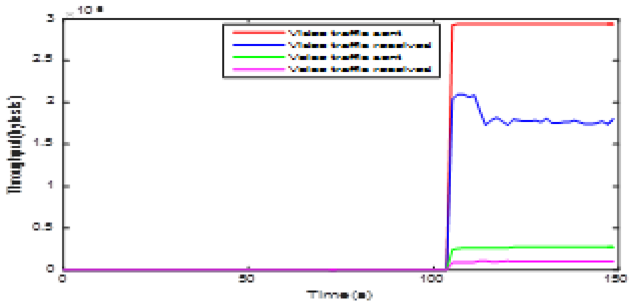


Fig. 6. Traffic sent and Traffic received (video conferencing and voice)

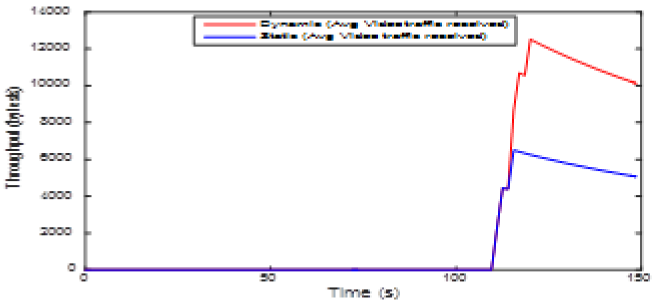


Fig. 7. Average Videoconferencing Traffic received using Static and Dynamic LSP

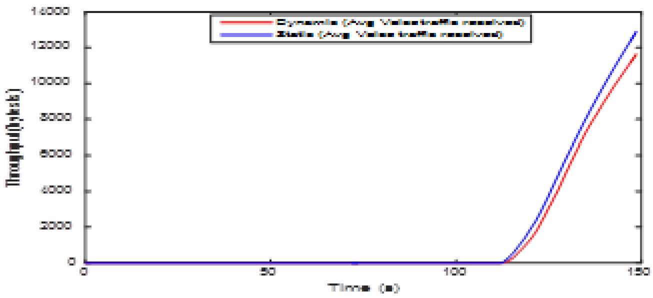


Fig. 8. Average Voice Traffic received using Static and Dynamic LSP

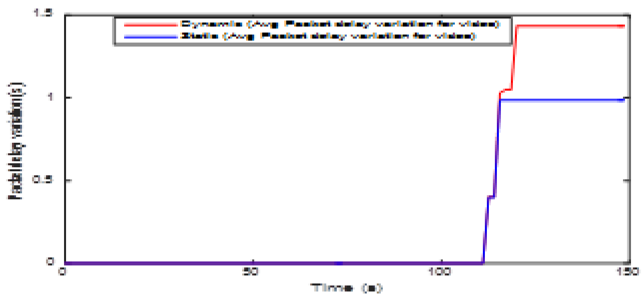


Fig. 9. Average Packet delay variation using Static and Dynamic LSP (video conferencing)

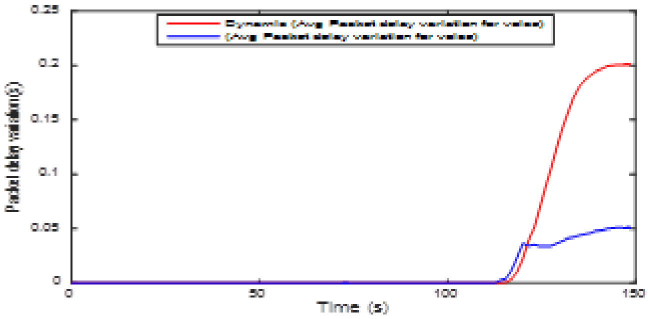


Fig. 10. Average Packet delay variation using Static and Dynamic LSP (voice)

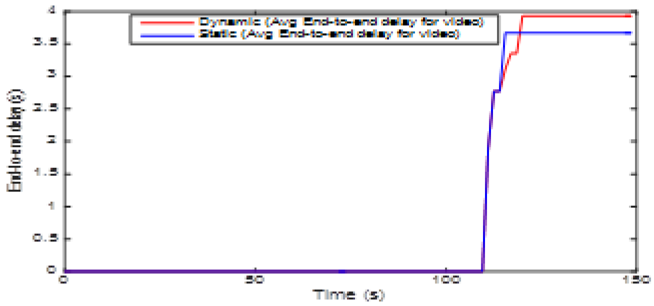


Fig. 11. Average End-to-End delay using Static and Dynamic LSP (video conferencing)

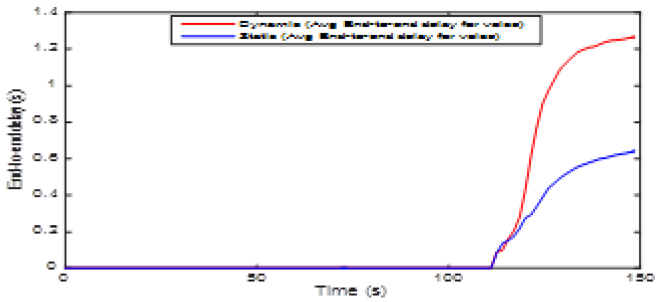


Fig. 12. Average End-to-End delay using Static and Dynamic LSP (voice)

IV. CONCLUSION AND FUUTURE PLAN

In summary, some of the proposed bandwidth management techniques had been reported in the literature review. The approach we used in this piece of research is similar to that reported in [1]. Therefore, a thorough study of the performance of the MPLS technology using static and dynamic configurations are implemented.

This would sustain the future exponential increment in user demand with adequate allocation of bandwidth. This is verified using the theoretical and simulation results of mean opinion score, which have moderate performance due to low values of end-to-end delay, low queue delay, and high throughput.

The use of MPLS technology to implement bandwidth management in future mobile wireless network is reliable and profitable due to its valuable cost to the both operators and service providers. As a result of this, it will yield sustainable quality of services to the users. Then the critical problem of delays such as end-to-end delay, queue delays, and packet delay variation would be drastically reduced.

However, it will be an additional cost to deploy MPLS technology to the existing network, instead of eliminating existing IP technology together with the facilities completely.

Performance evaluation of the QoS schemes such FIFO, PQ, and WFQ will be employed to assess the services provision to the users. Further analysis of the MPLS traffic engineering (MPLS-TE) will put into consideration for the adequate allocation of bandwidth to the next generation of mobile and wireless networks. 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.

REFERENCES

[1] E. D. Osborne and A. Simha, Traffic engineering with MPLS: Cisco Press, 2002.G. Ou, "Managing Broadband Networks: A Policymaker's Guide," *ITIF, December,* 2008.

[2] Optimisation Network Tools (OPNET) . Available from: http://www.opnet.com, [Accessed on 1st of March, 2016].

[3] J. Zander, "Radio resource management in future wireless networks: requirements and limitations," Communications Magazine, IEEE, vol. 35, pp. 30-36, 1997.

[4] K. S. Loh, B. E. LaVigne, V. V. Cavanna, and K. Y. Thoon, "Adaptive Bandwidth Management Systems and Methods," ed: Google Patents, 2007.

[5] J. K. Jones Jr, S. M. McLean, and C. E. Foley, "System and method for managing bandwidth utilization," ed: Google Patents, 2007.

[6] F. J. Canova Jr and A. H. Ting, "Videoconferencing bandwidth management for a handheld computer system and method," ed: Google Patents, 2006.

[7] P. Bender, P. Black, M. Grob, R. Padovani, N. Sindhushyana, and S. Viterbi, "CDMA/HDR: Bandwidth efficient high speed wireless data service for nomadic users," *IEEE Communications magazine,* vol. 38, pp. 70-77, 2000.

[8] S. Salah Al-Majeed, Chih-Lin_Hu, Dhinaharan "Advances in Wireless, Mobile Networks and Applications" Internationational Conferences WiMOA 2011 and ICCSEA 2011.

[9] Y. Bandung, A. Z. R. Langi, and A. Narendra, "Bandwidth Management Technique for Improving Virtual Class in Rural Area Network."

[10] M. P. McGarry, M. Maier, and M. Reisslein, "Ethernet PONs: a survey of dynamic bandwidth allocation (DBA) algorithms," *Communications Magazine, IEEE,* vol. 42, pp. S8-15, 2004.

[11] C. Gallon and O. Schelén, "Bandwidth management in next generation packet networks," *MSF, August,* 2005.

[12] C. Palansuriya, M. Buchli, K. Kavoussanakis, A. Patil, C. Tziouvaras, A. Trew, *et al.*, "End-to-end bandwidth allocation and reservation for grid applications," pp. 1-9.

[13] S. Mehrotra, H. Chen, S. Jain, J. Li, B. Li, and M. Chen, "Bandwidth management for mobile media delivery," pp. 1901-1907.

[14] B. Li, Y. Qin, C. P. Low, and C. L. Gwee, "A survey on mobile WiMAX [wireless broadband access]," *Communications Magazine, IEEE,* vol. 45, pp. 70-75, 2007.

[15] P. H. P. de Carvalho, M. A. de Deus, and P. S. Barreto, "IP and 3G Bandwidth Management Strategies Applied to Capacity Planning," *Edited by Jesús Hamilton Ortiz,* p. 29, 2012.

[16] T. Takine, K. Okazaki, and H. Masuyama, "IP traffic modeling: most relevant time-scale and local Poisson property," pp. 221-228.

[17] S. Machiraju, M. Seshadri, and I. Stoica, "A scalable and robust solution for bandwidth allocation," pp. 148-157.

[18] T. Tjelta, M. Annoni, L. Tokarchuk, J. Bigham, C. Adams, and K. Craig, "Future broadband radio access systems for integrated services with flexible resource management," Communications Magazine, IEEE, vol. 39, pp. 56-63, 2001.

[19] A. Hills and B. Friday, "Radio resource management in wireless LANs," *Communications Magazine, IEEE,* vol. 42, pp. S9-14, 2004.

[20] P. K. Lakshmanan and Q.-A. Zeng, "Dynamic adaptive resource allocation scheme for multimedia services in wireless and mobile networks," pp. 1-7.

[21] J. P. Romero, O. Sallent, R. Agusti, and M. A. Diaz-Guerra, *Radio resource management strategies in UMTS*: John Wiley & Sons, 2005.

[22] G. R. Ash, Traffic Engineering and QoS Optimization of Integrated Voice & Data Networks: Morgan Kaufmann, 2006.

[23] M. Stasiak, M. Glabowski, A. Wisniewski, and P. Zwierzykowski, Modelling and dimensioning of mobile wireless networks: from GSM to LTE: John Wiley & Sons, 2010.

[24] M. Khan, "MPLS Traffic Engineering in ISP Network," International Journal of Computer Applications, vol. 59, 2012.

[25] A. Dua and N. Bambos, "On the fairness delay trade-off in wireless packet scheduling," pp. 5.

[26] Q. Zhao, "Intelligent Radio Resource Management for Mobile Broadband Networks," 2013.

[27] R. G. ITU-T and G. Recommendation, "107: The E-model, a computational model for use in transmission planning," Recommendation ITU-T, 2005.

[28] J. D. Chimeh, "Quality of Service in UMTS Mobile Systems," Quality of Service Architectures for Wireless Networks: Performance Metrics and Management: Performance Metrics and Management, p. 14, 2010.

[29] Z. Mammeri, Wireless and Mobile Networking: IFIP Joint Conference on Mobile Wireless Communications Networks (MWCN'2008) and Personal Wireless Communications (PWC'2008), Toulouse, France, September 30-October 2, 2008 vol. 284: Springer, 2010.