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Novel SDN architecture for smart MPLS Traffic Engineering-DiffServ Aware management

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HIGHLIGHTS

- A novel SDN architecture for smart MPLS DS-TE Management is proposed.
- The architecture aims to dynamically manage the bandwidth and to ensure the segment routing.
- The proposed architecture can manage Legacy equipment.

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ABSTRACT

Large scale networks are still a major challenge for management, guarantee a good level of Quality of Service (QoS) and especially optimization (rational use of network resources). Multi-Protocol Label Switching (MPLS), mainly used in the backbone of Internet service providers, must meet these three major challenges. Software-Defined Network (SDN), is a paradigm that allows, through the principle of orchestration and layers abstraction, to manage large scale networks through specific protocols. This paper presents a new SDN-based architecture for managing an MPLS Traffic Engineering Diffserv Aware (DS-TE) network. The architecture manages the QoS and routing with QoS constraints, following a new smart and dynamic model of allocation of the bandwidth (Smart Alloc). The proposed architecture is suitable for SDN equipment and especially the legacy equipment. We tested our architecture by simulation on a hybrid network made up of SDN equipment and another legacy. The results of the simulation showed that thanks to our architecture we can not only efficiently manage hybrid architectures but also achieve good QoS levels for convergent traffic. The performance evaluation was performed on VoIP, video, HTTP, and ICMP traffic increasing packet load.

1. Introduction

Multi-Protocol Label Switching (MPLS) [1] is a protocol deployed primarily by Internet service providers to improve the performance of their IP network. The main motivation for using the MPLS protocol was the speed of routing process. With the label switching principle, MPLS provides a quick and simple treatment over IP packet relay. Traditionally, in an IP network, each router consults the IP header of the packet to determine the destination, then searches the Routing Information Base (RIB) to determine the next hop address or the outgoing interface. Thereafter an Address Resolution Protocol (ARP) request is generated to determine the physical address of the next hop if the media is Ethernet. This processing is probably very expensive in terms of delay. To overcome this limit, manufacturers have included new methods of routing such as Forwarding Information Base (FIB) which consist of creating a routing and adjacency tables. The first table is used to store, in a sorted manner, the resolutions made by the RIB table, and the second table is used to insert the information related to the encapsulation linked to the data link layer. Cisco Express Forwarding (CEF) [2] is an example of these methods.

MPLS facilitates the routing of packets by predefined routes, called Label-Switched Path (LSP). The intermediate routers called Label-Switched Router (LSR) process the primary information contained in the labels quickly, since the LSP is determined in advance. The Label Distribution Protocol (LDP) is responsible for assigning labels along the path. The Resource Reservation Protocol-Traffic Engineering (RSVP-TE) protocol [3] is responsible for determining the path.

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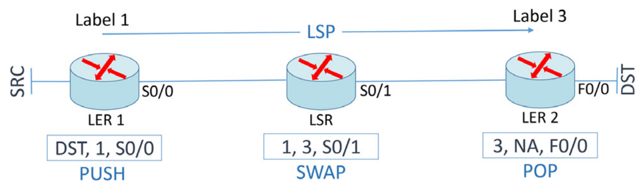


Fig. 1. MPLS network components.

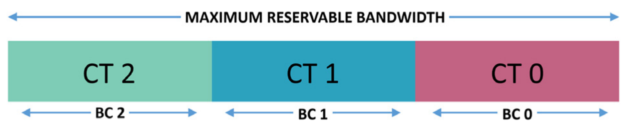


Fig. 2. MAM allocation model.

Fig. 1 illustrates an example of the MPLS network connecting two machines (the source SRC and the destination DST). The LDP protocol has already assigned labels along the LSP. The main routing operations of a packet in an MPLS network are as follows:

1. Label Edge Router 1 (LER) inserts the label 1 to reach the destination DST (PUSH operation).
2. LSR consults the FIB Label table to perform the switching and sends the packet through the Serial 0/1 interface (SWAP operation).
3. LER2 receives the packet and sends it to the destination by removing the label (POP operation).

1.1. MPLS DS-TE

Traffic Engineering (TE) is one of the strengths of MPLS technology. TE allows determining the LSP not only according to the shortest path (Interior gateway protocol metric) but also according to the constraints of the transported traffic, such as the bandwidth, the delay, and the resources available in the network. The problem with TE is that it does not take into account the different classes of traffic, which means, the bandwidth reservation is done for an LSP tunnel between a source and a destination and does not set up several tunnels per traffic class where each tunnel has its own resources.

Diffserv[1] Aware Traffic Engineering (DS-TE) [4] allows the reservation and the establishment of LSP tunnels according to a Class Type (CT). CT, as defined in RFC 4142, is "*the set of Traffic Trunks crossing a link that is governed by a specific set of bandwidth constraints. CT is used for the purposes of link bandwidth allocation, constraint-based routing and admission control. A given Traffic Trunk belongs to the same CT on all links*". DS-TE supports and allows up to eight CT. To synthesize, a gateway supporting DS-TE must first determine the class type for each stream, then establishes an LSP tunnel by CT. The LSP routing is done according to the CT priority. In case of links congestion, the LSP with the highest experimental (EXP) field will be deployed over the lower priority.

1.2. Bandwidth Allocation Model

Bandwidth Allocation Model (BAM) allows to reserve, distribute or limit the bandwidth for CT. There are several BAM models:

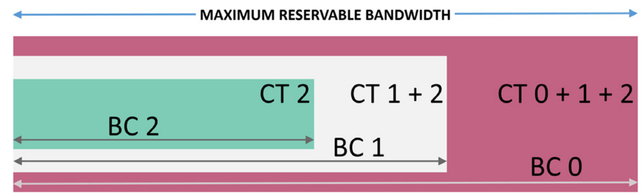


Fig. 3. RDM allocation model.

Fig. 4. AllocTC-Sharing allocation model.

- Maximum Allocation Model (MAM) [5] is a strict allocation model of the bandwidth. Each CT has its proposed bandwidth, if the latter is not used, it cannot be allocated to another CT. Fig. 2 illustrates an example of MAM allocation.
- Russian Doll Model (RDM) [6] is a nested allocation model of the bandwidth. Each CT is defined by a weight. The highest CTs priority can reuse the free bandwidth of lower priority CTs. So, the reservation is made from top to bottom and not the reverse. Fig. 3 illustrates an example of RDM allocation.
- AllocTC-Sharing model allows an opportunistic sharing of the bandwidth between the different classes. It is considered as an enhancement of the RDM model because it not only allows a top-down but down-top reservation as well. Fig. 4 illustrates an example of AllocTC-Sharing allocation.

1.3. Software-Defined Networking for MPLS

Software-Defined Networking (SDN) [7] is a paradigm for orchestrating the network infrastructure through a centralized controller, in an agile and highly dynamic manner. SDN decouples the control plan of the data plan making the infrastructure more programmable and rather scalable. The SDN architecture is based on four logical layers:

1. The application layer: contains a list of applications and services used by network users.
2. The data layer: contains the network equipment responsible for information forwarding.
3. The control layer: typically, a Network Operating System (NOS) providing an abstraction suite (application programming interface) to access the resources (such as routing table, forwarding, and calculations) of a network node. It decides on routing, traffic engineering, and fault detection.
4. The management layer: this layer is responsible for the configuration and supervision of the network equipment (such as routers, servers, and switches) the management layer and the control layer are often executed on the same processor.

Fig. 5 illustrates an overview of the SDN architecture.

The interfaces are designated in order to communicate two layers. As we represent the hardware components at the bottom and the applications and services at the top, the interfaces are often called "North" and "South".

[1] Diffserv is a QoS model in which traffic is processed with relative priorities based on the Differentiated Services Code Point (DSCP) field for the IP or EXP protocol for the MPLS protocol.

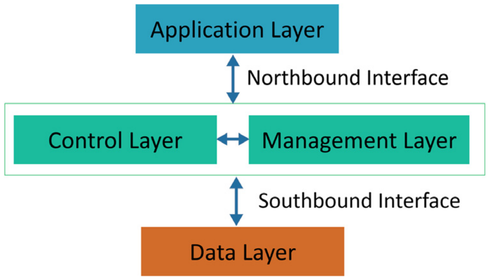


Fig. 5. SDN architecture.

- The North interface is intended to pass on information from applications so that the controller can establish an LSP that meets the Service Level Agreement (SLA) of the application.
- The south interface allows to deliver configuration commands to data layer, and raises the network statistics from the latter to the control layer. Several protocols can be used in this interface such as OpenFlow (OF) [8], Network Configuration protocol (NetConf) [9], Open vSwitch Database (OVSDB) [10], and Simple Network Management Protocol (SNMP) [11]. In the case of a hybrid network (containing recent SDN nodes and legacy nodes) two or more protocols can be used combined, such as OpenFlow and NetConf.

Several research projects have been carried out proposing either new bandwidth allocation models or new approaches for the QoS management in the communication networks by adopting SDN. Following works dealt with the dynamic bandwidth allocation [12–17], these contributions were based on RDM and AllocTC-sharing strategies. However, these allocation models do not take into account various SLA to adjust the bandwidth. Our solution Smart Allocation (Smart Alloc) aims, firstly, to classify flows based on their threshold severity (high, medium, and low). Whatever the priority of the flow belonging to the high threshold, the latter can benefit from the loans of the other categories. Secondly, to collect bandwidth from other categories and to calculate fairness index in order to allocate resources precisely to all flows taking into account their priorities. Smart Alloc was implemented on a controller to manage QoS and routing for MPLS DS-TE networks.

The rest of the paper is organized as follows: in Section 2, we will list the current limits in an MPLS DS-TE network and position our contribution. In Section 3, we will review and discuss the recent related works. Next, in Section 4, we will present and describe our SDN architecture for the intelligent management of the MPLS DS-TE network. A performance evaluation of our architecture will be carried out in Section 5. Last but not least, we will conclude in Section 6.

2. Issue and positioning of the contribution

As we mentioned earlier, the solution we offer is suitable for SDN and legacy equipment. The management of an MPLS DS-TE network in this hybrid environment faces a high number of challenges:

- **Deployment side**:
  Customizing QoS policies is a tedious task, whose command lines depend on the router's constructor and range. Also, The design of QoS policies suffers mainly from scalability and modularity. Which means that with the extension of the CEs number and the nature of their contract flows, PE equipment must undergo a new customization of QoS policies.
- **Routing side**:
  Setting up QoS policies in border routers is a difficult task. However, another constraint must be handled in the MPLS environment, it is the routing with QoS constraints. In other words, how to ensure that the used path meets the bandwidth requirements of users. RSVP-TE combined with CSPF (Constrained Shortest Path First) [18] strives to find the most suitable path to meet QoS requirements of users. Despite this, this combination represents several limits, mainly of scalability. The tunnels must be established a priori in a partially or completely meshed way. If a node changes its state, a topology calculation must be carried out, leading to an overload of the network by the signaling messages.
- **QoS routing side**:
  While the LSP path is identified, a differentiation of traffic classes and resource allocation must be performed. The DiffServ approach can be adopted, but the latter is passive, which means the congestion generated on the DownStream side is not reported to the UpStream side. DS-TE allows overcoming this limitation by combining MPLS-TE and DiffServ. However, MPLS DS-TE represents several limitations, the specific routing to a traffic class can only be guaranteed through features such as Policy-Based Routing (PBR). The thing that is not feasible in a large-scale network.

To answer the above problems, we propose a new architecture based on the SDN model allowing to:

1. Solve the problem of deploying QoS policies: converting QoS requirements into command lines compatible with the equipment manufacturer's range. By adopting a solution based on a controller, we will guarantee the Internet service providers scalability and especially modularity.
2. Solve the routing problem: routers are not required to maintain tunnels in order to decide which path meets the bandwidth requirement of the clients. The controller will have a topology table of the entire network and will decide on the best path to take. Even if a link state change occurs, only the controller will be notified.
3. Detect the reliability of QoS policies on the UpStream side: ensuring that they do not result in congestion on the DownStream side. In case of congestion, which is often due to an asymmetry of the performances of the communicating pairs, the controller will carry out a QoS policy control as far as possible (if the regulation will cause a deterioration in the quality of the VoIP, then the regulation will not be executed).
4. Ensure a new Smart Alloc of bandwidth resources: based on the SLA constraints of different CT (such as jitter, latency, loss rate, objects loading times, and RTT).

3. Related works

The QoS management in an SDN-based MPLS DS-TE network is a broad subject, which consists of:

- Developing a new QoS algorithm for MPLS traffic (Smart Alloc).
- Defining a new method for MPLS routing that takes into account QoS constraints.
- Designing and implementing a new hybrid SDN architecture that takes into account QoS management and QoS constraint routing.

3.1. MPLS-TE SDN

Sharafat et al. [19] initiated the implementation of MPLS traffic engineering based on the SDN approach. They justified the adoption of this approach by the complexity of the control plan associated with the MPLS-TE technology. This technology is based on a set of OSPF, iBGP, mBGP, LDP, and RSVP-TE protocols, which because of their distributed natures can cause memory overflow or overuse of the routers' CPU resources. This leads to undesirable effects such as the rejection of certain HELLO messages responsible for the formation of adjacencies or the discovery of the best path with the QoS constraints. For all these reasons, the authors have shown the feasibility of using the OpenFlow protocol and a NOS to build a centralized network management, while avoiding the need for MPLS-TE protocols.

In fact, it is possible to have in the same network compatible SDN and legacy equipment. In this case, the controller relies on additional protocols such as SNMP, Netconf, LLDP [20] or even ICMP to detect the equipment manufacturer, its interfaces, its location in the topology, the capacity of its links, the already configured QoS mechanisms, etc. Several works showed the feasibility of adopting hybrid SDN architectures [21,22,32].

Recently, there is a strong tendency to manage and control the MPLS network based on SDN. Davoli et al. [24] propose an approach for traffic engineering with segmented routing (SRTE). This approach is very beneficial for the network because only edge routers must maintain the state of the LSP tunnels. This work has only processed the south bound interface (between the controller and the data plan), this implies that the approach does not take into account QoS requirements of applications and QoS constraint routing. Lee and Sheu [25] overcome this limit and include the bandwidth constraint for the selection of the shortest and optimal path.

3.2. Adaptive QoS for the convergent networks

Sadon et al. [12] propose a method of dynamic and hierarchical allocation of the bandwidth using the RDM strategy. This method is based on the classification and the prioritization of services. The authors applied the model in EPON networks, the algorithm provides the bandwidth required for the requests based on the fairness factor and the services priority.

The general problem of the algorithms based on RDM is that the resources reservation is carried out from the bottom to top; the low priority traffic shares its resources with the higher priority traffic and not the inverse. Several works have been carried out proposing new dynamic bandwidth sharing algorithms by adopting the RDM strategy [13,26,27].

To make a reservation from top to bottom and from bottom to top, the AllocTC-Sharing model [14] initiated a two-way approach of bandwidth dynamic sharing, the approach is to share the unusable bandwidth of high bandwidth applications priority with low priority applications. In [15,28] the authors evaluated the effectiveness of the AllocTC-Sharing model compared to the RDM model. The authors have shown by simulation that AllocTC-Sharing is more efficient in terms of maximizing the link use and that it is better suited for elastic traffic and high bandwidth usage. Adami et al. [16] present a new model called Generalized RDM (G-RDM), this model combines the characteristics of the MAM and RDM, defining for each class a private bandwidth and a public one. Unlike private bandwidth, public bandwidth can be shared with other classes. The model has been tested on the MPLS DS-TE network and has demonstrated its efficiency in terms of bandwidth utilization and scalability during load increase.

Reale, Bezerra, and Martins [17] initiate a new approach to switch autonomously between models (MAM, RDM, G-RDM, and AllocTC-Sharing) based on a controller. Switching from one model to another can be done in different metrics, for example link use, congestion probability, and packet number. The G-BAM solution [29] adopted the BAM approach to improve the efficiency and performance of the MPLS DS-TE network in an autonomous way.

3.3. Adaptive QoS allocation for multipath networks

Guaranteeing a good QoS level for multipath networks is one of the major challenges, whether for wired networks [30,31] or wireless networks more precisely, where energy consumption constraints arise [32,33]. The influence of multipath and QoS degradation can only really be felt when multimedia or real-time applications are routed through the network. More specifically and especially in wireless networks, whose bandwidth is one of the major concerns. Despite the many CMT (Concurrent Multipath Transfer) solutions, these remain limited when paths have asymmetric performances and especially when applications are sensitive to SLA constraints.

Wu et al. [34] present an improvement of the CMT solutions. They propose a Distortion Aware CMT solution (CMT-DA), taking into account the video distortion in the path selection process. In the first phase, this solution consists of estimating per path available bandwidth using the Round-Trip Time (RTT), the congestion window and the Retransmission Timeout (RTO). Then perform a flow rate allocation, ie, send the acknowledgment packets by the most reliable uplinks in order to adapt the congestion window. CMT-DA has been tested in a variety of heterogeneous wireless networks: WiFi, WiMax, and cellular, taking into account following criteria: Peak Signal to Noise Ratio (PSNR), goodput, and inter packet delay. The obtained results showed the efficiency of such solutions.

Maintaining the QoS of multimedia traffic in a heterogeneous network thanks to flow rate allocation algorithms, while minimizing the consumption of energy, is one of the major concerns of modern networks. In [35,36] the authors have contributed to this research field by developing a new algorithm EDAM (energy-distortion aware Multi Path TCP). As results, compared with the EMTCP [37] and MPTCP [38] solutions, respectively, the proposed solution allows: (i) reduce energy consumption by a factor of 26.3% and 40.6%. (ii) improve the PSNR by a factor of 25.5% and 39.3%. (iii) increasing the number of effective retransmissions by 46.3% and 58.2%.

4. Proposed architecture

Proposed architecture performs the following tasks:

1. Detect the applications and their SLA thresholds.
2. Translate the QoS requirements (priority, bandwidth, and EXP) of each user into an XML file.
3. Perform the admission control to verify if the bandwidth of the customer is sufficient for the QoS requirements.
4. If the network equipment is legacy, then, actively detect its constructor and translate the XML file into lines compatible with the equipment.
5. Verify the topology table to determine the end-to-end LSP meeting the client's QoS requirements.
6. Monitor the performance of the transported applications (according to the user SLA).
7. Adapt the bandwidth resources and define the new LSP that meets the traffic requirements.

Fig. 6 illustrates the proposed architecture.

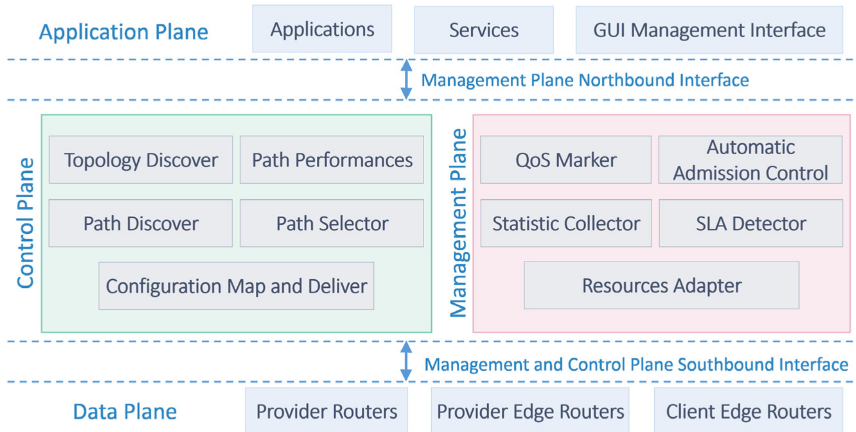


Fig. 6. SDN Architecture for MPLS DS-TE Network Management.

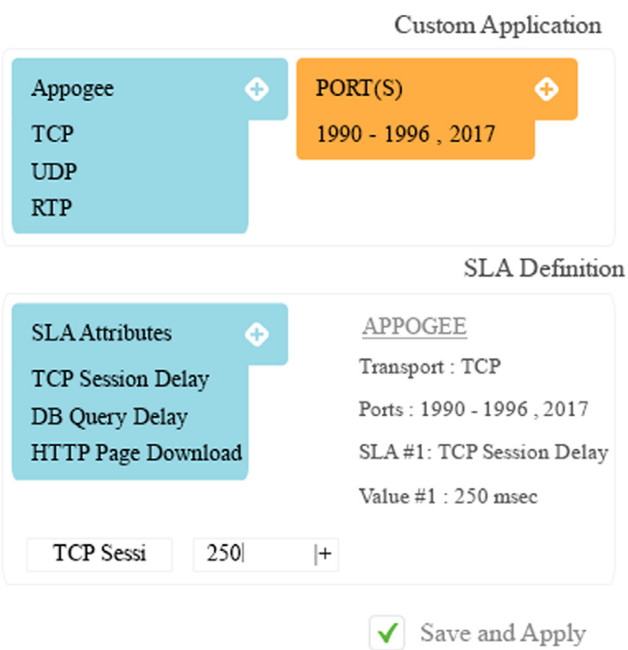


Fig. 7. Customizing a batch of applications.

4.1. Application plane

The layer in which all user applications and services are consolidated. These applications can be defined if they are not standardized. We have developed a new Human-Machine Interface to facilitate the customization of each user's applications. Fig. 7 illustrates an example of customization of a batch of applications.

The window is divided into two sections: the first for the application definition and the second for its SLA thresholds. The application created in this demonstration is called 'Apogee', the latter is based on the TCP transport protocol. The different SLA thresholds compatible with an application can be chosen from the drop-down list in the SLA definition section, in our case we choose as the SLA attribute the TCP session delay and 250 msec as an argument.

4.2. Management plane

This layer is responsible for creating QoS policies, admission control, monitoring client application performance, and adapting QoS policies to eventually meet SLA requirements. QoS Marker module allows assigning an application to a traffic class according to its priority and type. Table 1 shows the set of traffic classes allocated according to the application type. It is recalled that the EXP field is coded on 3 bits; 8 possible values ranging from 0 to 7, where 0 = Best Effort and 7 is the network traffic.

Table 1
Traffic classification.

Application Type	DSCP Code	AF	CT	RFC
Best Effort	0	0	0	2474
Scavenger	8	CS1	1	3662
Bulk-Data	10,12, 14	AF11, AF12, AF13	1	2597
Network Management	16	CS2	2	2474
Transactional Data	18, 20, 22	AF21, AF22, AF23	2	2597
Call Signaling	24	CS3	3	2474
Mission Critical	26, 28, 30	AF31, AF32, AF33	3	2597
Real-Time Interactive	32	CS4	4	2474
Interactive Video	34, 36, 38	AF41, AF42, AF43	4	2597
Voice	46	EF	5	3247
Routing	48	CS6	6	2474
Network	56	CS7	7	NA

After the classification phase, a QoS policy must be defined, i.e. priority and reserved bandwidth (absolute in kbps or relative in percent) for a specific application. Fig. 8 illustrates an example of implementing a QoS policy on previously created applications.

The window consists of two sections; the first concerning classification, the user can select one or more applications and assign them to a traffic class. The second section is dedicated to applying a QoS policy for previous classes. This policy may include:

- Shaping: or flows regulation, provides a means of controlling the trade volume. This control is carried out by delaying certain packets (without loss) that have exceeded the maximum throughput defined in the contract.
- Priority: the bandwidth not to exceed.
- Bandwidth: the minimum bandwidth to be reserved for an application. This bandwidth can be represented in Kbps or percent.

The auto-management of QoS policy is provided in the management layer. The auto-management is done when the following events occur:

- SLA statistics for applications exceed standard or tolerable thresholds (such as delays, loss rate, jitter, HTTP load time, and TCP session delay).
- Losses are detectable on the link.

Fig. 8. Example of implementing a QoS policy.

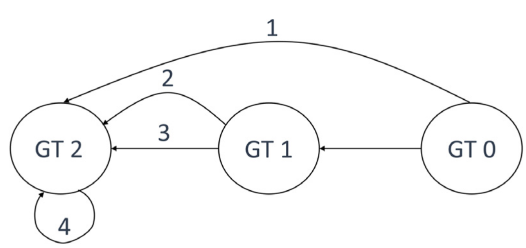


Fig. 9. Example of implementing a QoS policy.

To perform an auto-management of QoS policy, the Statistic Collector module detects any trigger event, then performs an additional allocation or release the bandwidth resources. The allocation model that we propose for dynamic bandwidth management is defined in the following section.

4.3. A new model of bandwidth dynamic management (Smart Alloc)

The finite-state machine describing our bandwidth allocation model is as follows (Fig. 9):

Unlike other models that allocate bandwidth according to the video, voice and data classes. Our model is based on the state (performances) of the flows regardless of their traffic classes or priority. Our model is based on three Groups of Traffic (**GT**). **GT2** contains all CTs whose SLA thresholds exceed the thresholds defined in the contract, standardized thresholds or reserved bandwidth. **GT1** contains CTs that meet SLA thresholds without excess or deficiency. **GT0** contains the CTs with lower SLA thresholds of the contract, thus the best performers.

- **GTi**; represents a category of a group of Traffic Class C

$$GTi = \sum_{n=1}^{N} C_n^i \, ; \; C_n^i \textit{ is a Class } n \; \in \; GTi$$

- **BaGTi**; represents the bandwidth available in each **GTi**.

$$BaGTi = \sum_{n=1}^{N} BtotalC_n^i - BuC_n^i \, , \; \forall i \in \{0, 1\}$$

$BuC_n^i$is the bandwidth used by a class n of GTi

- **BdGT2**; represents the bandwidth demanded by GT2.

$$BdGT2 = \sum_{n=1}^{N} (BrC_n^2 - BtotalC_n^2)$$

- **BsGTi**; represents the additional bandwidth to be allocated to GT2 category

$$BsGTi = \frac{BaGTi}{BaGTi - |BaGT(i-1)|}$$

The BsGTi allocation is made if :

$$\sum_{i=0}^{1} BaGTi \leq BdGT2$$

The collected bandwidth BaGTi is distributed over the traffic classes of GT2 category according to their priorities.
$[B_a C_{p,n}^2]$ represents the bandwidth requested by a class n with priority p of GT2 category.

$$B_a C_{p,n}^2 = BaGTi * \frac{Pc_n}{\sum_{n=1}^{1} Pc_n}$$

The traffic classes in the GT2 category shall be treated in accordance with the RDM model; the traffic classes are divided into three sub-categories BC2, BC1, and BC0. The assignment of a traffic class to a sub-category is carried out according to its priority. In order to perform this division, our model consists in calculating the most optimal correlation coefficient, i.e. the closest one to 1. Let the pair xi: yj, x be the priority and y the subcategory, $j \to 0, 1, 2$. The correlation factor formula is calculated as follows:

$$r = \frac{n(\sum xy) - (\sum x)(\sum y)}{\sqrt{[n \sum x^2 - (\sum x)^2][n \sum y^2 - (\sum y)^2]}}$$

For classes with CT $= 7, 4, 3, 2$, and 1, the factor r reaches the score 0.9604 for assignments BC2, BC1, BC1, BC0, BC0. After finding the combination xi and yj with the factor r close to 1, our model consists in calculating a regressive linear function for a subsequent classification. This function is defined by the formula: $a + bx = y$ where x is the priority.

$$b = \frac{\sum_{i=1}^{n} xiyj - n\overline{xy}}{\sum_{i=1}^{n} xi^2 - n\overline{x}^2}$$

And $a = \overline{y} - b\overline{x}$

4.4. Control plane

This logical layer first serves, before any processing, to detect the topology of the controller domain. The SNMP protocol is used as a protocol for exchanging relevant topology information between legacy equipment and the controller, this information is generated by the Link Management Protocol (LMP) [39]. The information required to detect the topology is:

- Domain routers: all routers must belong to a community that can be read and written. The MIB object 'LmpNbrTable.LmpNbrNodeId' is consulted to determine the IP addresses of the routers. To detect the router constructor, the controller consults the MIB object 'SysObjectId'.

- Link bandwidth: the MIB object 'IfTable.IfSpeed' does not accurately indicate the bandwidth of a link. The calculation of the bandwidth was carried out according to the following formula:
$$\frac{max(\triangle IfInOctets + \triangle IfOutOctets) * 8 * 100}{(\triangle Value\ in\ second) * IfSpeed}$$
- Link between routers: the links detection between the various routers is essential to trace the network topology, and possibly to deduce the most optimal path. This task is done based on:
   1. The IP address of each interface of the routers: **Ai**.
   2. The network address of each interface: **SNi**.
   3. The interfaces name: **Ni**.
   4. The physical and logical state of the interfaces: **Si**.
   5. The Router name: **Rn**.

Algorithm 1 summarizes the procedure for topology detection:

Algorithm 1: Topology identification

Topology identification procedure;
Sort Ai;
Load Rn;
Load Ni;
for *each* $Ai \in SNi$ **do**
  if *Si = Active* **then**
    Create the link between the Ni and update the topology Rn;
  else
    Delete the Ni link and update the Rn topology;
  end
end
for *each* $Ai \notin SNi$ **do**
  if *Si = Active* **then**
    Display an Endpoint interface on Rn and update the topology;
  else
    Delete an EndPoint interface on Rn and update the topology;
  end
end

4.5. Path selection

After mapping the topology of the MPLS network, we must detect the available bandwidth of a link. This phase is paramount to selecting enough bandwidth path for a reliable routing of customer traffic. Several methods or approaches can be used for this task. The active approach consists in injecting traffic into the network to carry out measurements, this can cause disturbances on the network. The passive approach, for its part, makes it possible to collect statistics and analyze them later. The feedback to the controller can be made via the SNMP protocol. To calculate available bandwidth:

$$Ai = Ci - \overline{U}i$$

Where Ci is the capacity of a link and $\overline{U}i$ represents the used bandwidth.

We must determine the mean of a link use in a time interval $(t-\tau, t)$:

$$\overline{U}i(t-\tau, t) = \frac{1}{\tau}\int_{t-\tau}^{t} Ui(x)\,dx,\ \overline{U}i = \frac{Ti(t-\tau)}{Ci}$$

$Ti(t-\tau, t)$: quantity of traffic in a time interval.

So

$$Ai = Ci(1 - Ui)$$

$$Ai = \frac{1}{\tau}\int_{t-\tau}^{t} Ci(1 - Ui(x))\,dx$$

$$Ai = \frac{1}{\tau}\int_{t-\tau}^{t} Ci(1 - Ui(x))\,dx$$

$$Ai = Ci(1 - Ui(t-\tau, t))$$

$$Ai = Ci - \frac{Ti(t-\tau t, t)}{\tau}$$

Then, the link bandwidth is the min value of $Ai : A = \mathrm{Min}(Ai)$.

Until this phase, the controller obtains a global view on the network topology, its routers, the links capacity, and the available bandwidth. The next step is to select the best path taking into account the constraint of the available bandwidth of a link and the path cost.

To address the shortest path with QoS constraints, we assume that the graph $G = [X, U]$ represents a network of N routers and M links. Au represents the available bandwidth of a link for each $u \in U$. For each request to establish an $LSP(K)$, defined by a source $S(k)$, a destination $R(k)$, the bandwidth d^k, and P_n^k the nth path between $S(k)$ and $R(k)$ for all $n \in [1, P(k)]$. The links meeting the constraints of the bandwidth are defined by:

$$Min\left\{\sum_{K=1}^{K}\sum_{j=1}^{P(k))} P_j^k(u)d^k \le Au\right\},\ u \in [1, M]$$

Algorithm 2 summarizes the procedure for highest available bandwidth path selection:

Algorithm 2: Path selector algorithm

INPUT : A graph G[X,U] of N routers and M links.
OUTPUT : An LSP connecting a sender S(t) to a receiver R(t) with highest available bandwidth path (ρ).
PROCESS
for *each* *link* $U \in$ *path P* **do**
  Calculate available bandwidth Ai per interface;
  $$Ci - \frac{Ti(t-\tau t, t)}{\tau}$$
  Determine path available bandwidth; A$u \Leftarrow$ Min(Ai);
end
for *each* *(P ∈ LSP & Au > 0)* **do**
  Compute shortest path ρ based on available bandwidth;
  $$Min\left\{\sum_{K=1}^{K}\sum_{j=1}^{P(k))} P_j^k(u)d^k \le Au\right\},\ u \in [1, M]$$
end

After having defined the shortest path with QoS constraints, the controller loads the identifiers of all the routers of this path in the label stack, which is delivered to the source $S(k)$. The principle of routing is therefore based on Segment Routing technology [40].

Until this phase, the configurations must be delivered to the data layer equipment. The Map and Deliver Configuration module allows thanks to the topology discover module to determine the router's constructor and load a compatible configuration file. The configuration file is in XML format created and delivered by the NETCONF protocol.

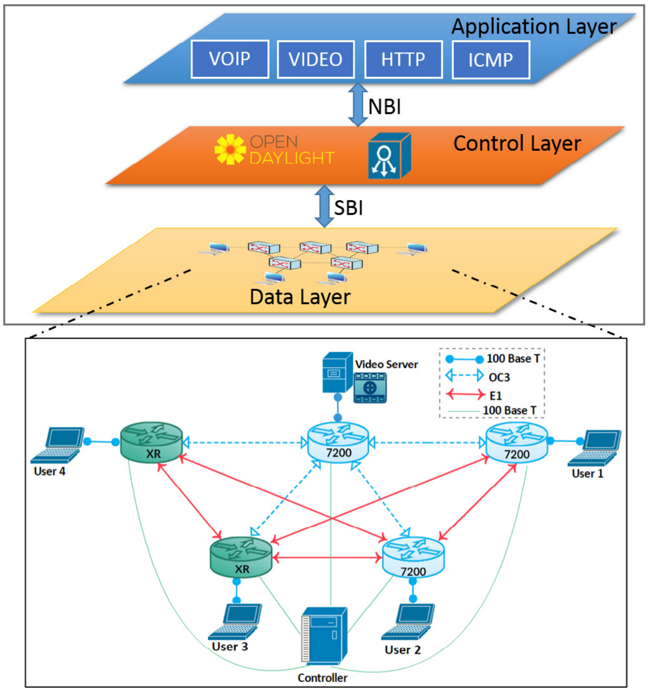


Fig. 10. Hybrid Test Network.

5. Performance evaluation

In this section, we firstly describe the evaluation methodology that includes a description of network testbed and performances metrics. Then, we present and discuss obtained results.

5.1. Evaluation methodology

In order to evaluate our solution, we realized a network architecture under Graphical Network Simulator 3 (GNS3) [44] according to Fig. 10. The routers used in the simulation are Cisco XR [45] and 7200 IOS [46] to represent legacy routers.

We used the OpenDayLight controller [41] in the simulations, the choice of this controller is justified by the wide tendency to its use and also by its rich documentation. The objective of the experiment is to evaluate the architecture on various traffic: real-time (VoIP with codec G.711 alaw), multimedia (video), best effort (HTTP version 1.0), and network management (ICMP).

The performance evaluation was carried out according to different criteria: VoIP jitter, VoIP latency, video Peak Signal to Noise Ratio (PSNR), goodput, retransmission, HTTP Response page, and the ICMP RTT. To deduce the efficiency of our bandwidth dynamic management model we compared it with the RDM and MAM models.

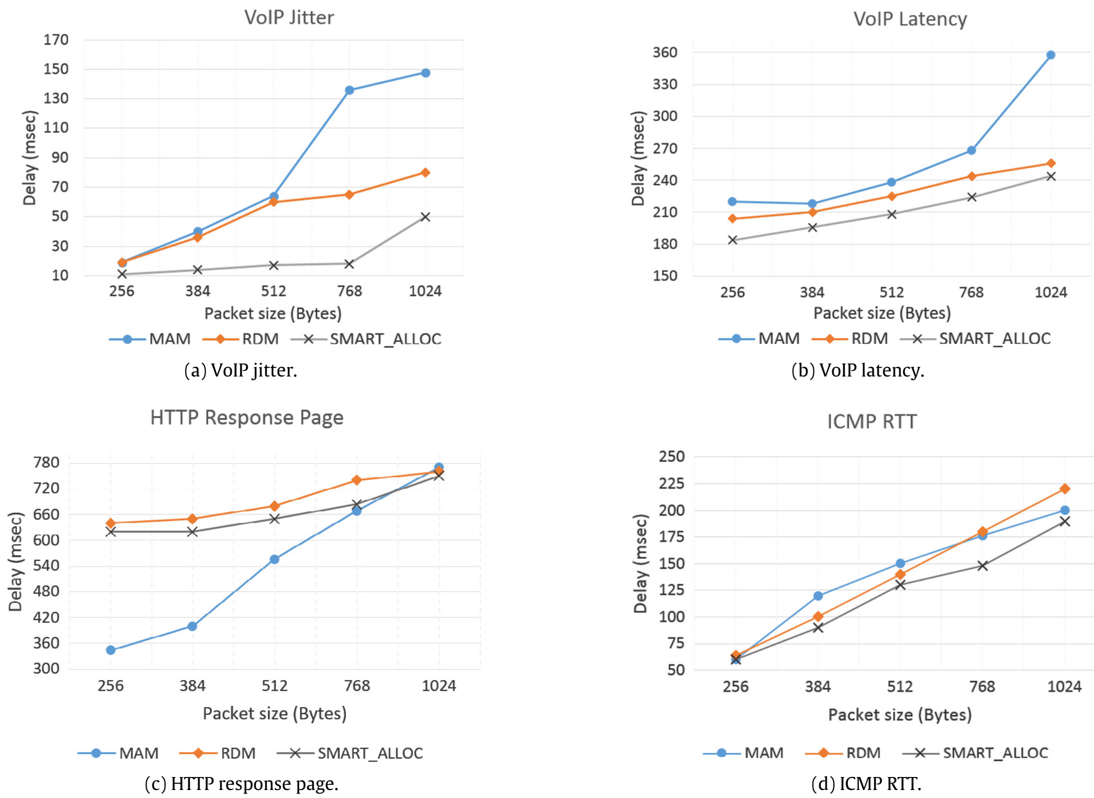
- Peak Signal to Noise Ratio (PSNR) [47]: the objective criterion for evaluating improvements in new perceptual video quality performance metrics. It consists of quantifying the performance of the encoders by measuring the reconstruction quality of the compressed image compared to the original image. PSNR is expressed in terms of the logarithmic decibel scale (dB).
- Goodput: the application-level throughput (size of payload bits delivered per unit of time). The amount of data considered excludes protocols headers bits as well as retransmitted data packets.
- Retransmission: the resending of packets which have been either corrupted or lost. A higher amount of retransmission can decrease goodput.
- Jitter: statistical variance of the transmission delay. It measures the temporal variation between the moment when two packets should have arrived and the moment of their actual arrival. This packet arrival irregularity is due to many reasons including packet encapsulation, network load, and paths performances.
- Latency: or end to end delay = network delay + encoding delay + decoding delay + compression delay + decompression delay. Given that network delay is the time at which the sender gave the packet to Real Time Protocol (RTP) to the time the receiver got it from RTP.
- HTTP Response page: required time to retrieve the entire page with all the contained inline objects.
- ICMP RTT: duration it takes for an "ICMP echo" message to be sent plus the duration it takes for and "ICMP echo-reply" to be received.

5.2. Obtained results

Obtained results are shown in Fig. 11.

Fig. 11(a) shows the VoIP jitter in the three scenarios: MAM, RDM, and Smart Alloc. Taking into account the VoIP tolerable jitter defined in ITU-T G.114 (50 ms), we find that our model offers the lowest and tolerable jitter in the various loads, unlike the RDM and MAM models which have deteriorated the quality of the VoIP from the 512-bytes scenario. Taking into account the increase in load, our model has increased by 39 ms from the 256-bytes to the 1024-bytes scenarios, the MAM and RDM models have respectively increased by 61 and 129 ms. So, as jitter, our model offers a lower delay by a factor of 36% compared to the RDM model and 69.76% compared to the MAM model. The reason the MAM model offers the highest jitter is that it does not perform additional bandwidth resource reservations even in the case of load increase. This explains its weakness in various scalability scenarios. The MAM, RDM, and Smart Alloc models, respectively, follow these exponential growth functions: $y = 12e^{0.53x}$, $y = 16e^{0.34x}$, and $y = 7e^{0.32x}$. This advantage is also noticeable in the results of the latency or the end-to-end delay (Fig. 11(b)), our model offers a reduced delay compared to the MAM model with a factor of 18.831% and a factor of 7.28% to the RDM model. The RDM model is slightly undergoing the same exponential growth of the Smart Alloc model but offers higher jitter and latency values. This is justified by the fact that RDM reacts only when the bandwidth becomes insufficient, in contrast, our model reacts according to the bandwidth consumed and the SLA thresholds. Our model, therefore, offers high-quality telephony.

Fig. 11(c) illustrates the HTTP response page delay. As expected, the MAM model offers the most optimal delays in most scenarios. This can be justified by the nature of this latter, which, despite the low priority of the stream, reserves a fixed portion of the bandwidth. Up to the 768 bytes scenario, MAM remains the best, but beyond this load, Smart Alloc shows its effectiveness. As noticed also, MAM model has undergone an exponential evolution, for the same reason in the case of VoIP, this is due to the static bandwidth reservation despite the increase in packet load. From the 512-byte scenario, RDM has evolved in terms of delay since Low-to-High reservation is not supported in this model. Our model offers the smallest delay compared to RDM. However, this delay has increased in the 1024 bytes scenario. This is justified by the fact that the VoIP in the same scenario was in a critical state (latency is close to 300 ms), made the High-To-Low allocation not beneficial for the HTTP traffic. Quantitatively, if we take the critical scenarios



(a) VoIP jitter.

(b) VoIP latency.

(c) HTTP response page.

(d) ICMP RTT.

Fig. 11. Results of the performance evaluation of our architecture.

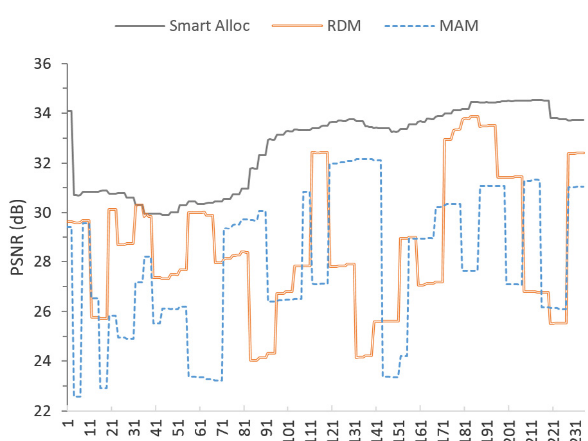


Fig. 12. Comparison of different PSNR.

starting at 512 bytes, the loading time of a page increases by 112 ms for the MAM model, 60 ms for the RDM model, and only 34 ms for our model. Therefore, our model despite the high-to-low and low-to-high allocation reduces the page response delay by a factor of 43.33% compared to the RDM model and 69.64% to the MAM model. As ICMP and HTTP traffic have the lowest priorities in the study, the same interpretation applies to the results of the RTT delay in Fig. 11(d).

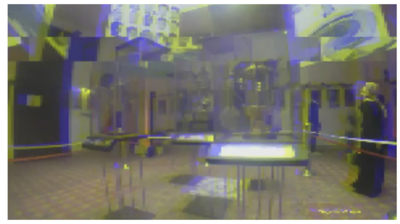
Fig. 12 illustrates a comparison of PSNR per video frame from 1 to 231 measured from exhibition room in different scenarios (Smart Alloc, RDM, and MAM). Obtained results were extracted from MSU Video Quality Measurement Tool [42]. We note that Smart Alloc model takes into account the SLA constraints of the different multimedia streams, while the other models are unaware of video traffic constraints. We can see that Smart Alloc achieves the highest PSNR values with marginal variations than RDM and MAM.

In order to have a perceptible view of the results, we plot a comparison of the 151th received frame in Fig. 13. It is clearly noticeable that Smart Alloc 13(a) achieve better quality while the received frame of the RDM 13(b) and MAM 13(c) are extremely damaged. We also notice that RDM and MAM models undergo frequent glitches and periodic distortions, while much smoother streaming is obtained with Smart Alloc.

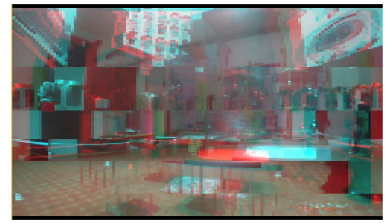
Fig. 14 illustrates a comparison of retransmission results and average goodput. Remind that the goodput is the application-level throughput, it is a good criterion in order to estimate the video quality [43], given that it measures the total rate of traffic from uncorrupted video frames arriving at a destination. Fig. 14(b) illustrates obtained average goodput for three scenarios. Clearly, the Smart Alloc model produce better video goodput than RDM and MAM, by a factor of 29.41% and 69.23% respectively, due to its ability to allocate dynamically bandwidth resources based on network congestion and application QoS constraints. However, among the factors that influence the goodput, we mention the size of the headers, used flow control mechanisms, and especially the retransmission of lost or damaged packets. Fig. 14(a) illustrates the average number of retransmissions and effective retransmissions. Thanks to adaptive resources allocation, the Smart Alloc model offers low rate retransmission and the highest ratio of effective re-transmissions, this contributes also to achieve the highest goodput. In summary :

- Smart Alloc reduces the retransmission by up to 74.19% and 45.45% compared to MAM and RDM respectively.
- Smart Alloc increases the effective retransmission by up to 166.66% and 60% compared to MAM and RDM respectively.

(a) SMART ALLOC.

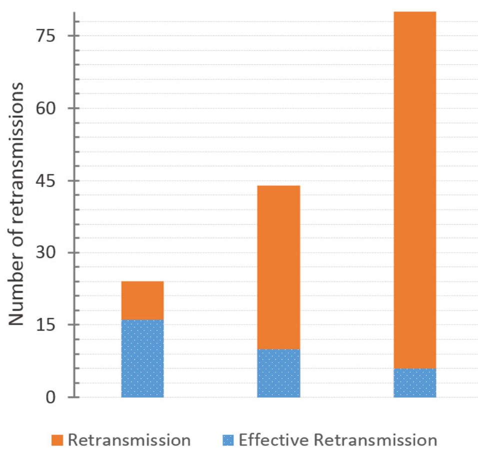


(b) RDM.

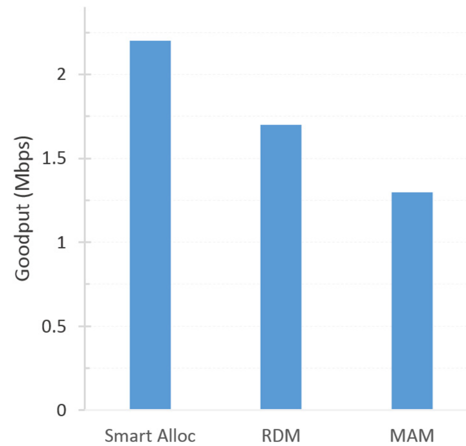


(c) MAM.

Fig. 13. Comparison of subjective video quality measured from exhibition room.



(a) Retransmissions.



(b) Average goodput.

Fig. 14. Comparison of subjective video quality measured from exhibition room.

6. Conclusion

Nowadays, the SDN approach is considered the future of traditional networks. This approach is promising in terms of flexibility, agility, and especially the ability to support new technologies. In order to make a network programmable and to control it, a controller entity must be used. However, the high investment cost in SDN equipment represents a major barrier, which the hybrid SDN approach can overcome. The latter can orchestrate both SDN equipment and those of traditional networks called legacy. Among the active issues raised up to the present day, we mention the management of QoS, especially in MPLS-based networks (DS-TE). In this paper, we have presented a new architecture for the management of MPLS DS-TE network by adopting a hybrid SDN approach. The aim of the architecture is to dynamically manage and allocate the bandwidth resources and to ensure the routing by segment taking into account end-to-end QoS constraints. The proposed architecture has been tested on different traffic (VoIP, Video, HTTP, and ICMP), and compared related to MAM and RDM bandwidth allocation models. The performance evaluation was carried out according to different criteria, such as VoIP jitter, VoIP latency, video PSNR, goodput, retransmission, HTTP response page, and the ICMP RTT. The results obtained showed the reliability of our model by providing the best VoIP quality, a smooth video quality, an acceptable HTTP response page delay, and the lowest ICMP RTT.

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