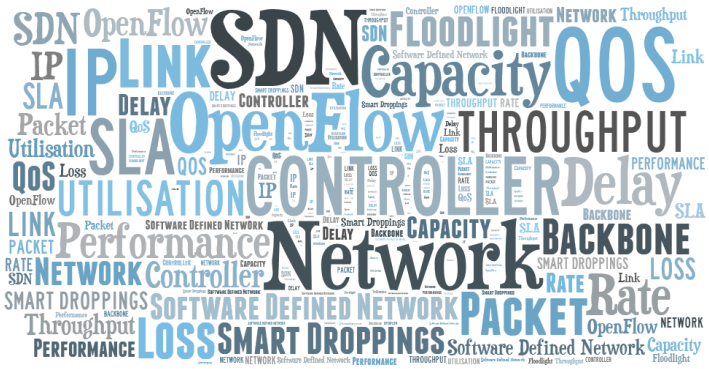
TÉCNICO LISBOA




# omniCluster - Virtual Network Allocation in Data Centers using Software-Defined Networking

**Daniel Gomes Fonseca Caixinha**

Thesis to obtain the Master of Science Degree in

## Telecommunications and Informatics Engineering

Supervisor: Prof. Luís Manuel Antunes Veiga

### Examination Committee

| | |
|---|---|
| Chairperson: | Prof. Paulo Jorge Pires Ferreira |
| Supervisor: | Prof. Luís Manuel Antunes Veiga |
| Member of the Committee: | Prof. Fernando Henrique Corte-Real Mira da Silva |

**October 2015**

# Acknowledgments

First and foremost, I would like to thank my advisor, Professor Luís Veiga, for his invaluable contributions to this work. From day one I have been motivated by his unceasing drive, tremendous sharpness and never-ending support. Besides improving this thesis, our prolific discussions have made me grow and learn a lot.

I would also like to thank the open-source community in general and the floodlight and mininet mailing lists in particular. Their prompt answers and guidance helped me lot to navigate in these giant projects.

Last but not least, I would like to leave here my gratitude to my girlfriend and life companion, Raquel. It is in periods of fear and stress we can see that the whole is greater than the sum of its parts. Thank you for everything.

# Resumo

A criação de centros de dados permitiu o acesso a pedido (e sem compromisso) a recursos computacionais. Ao alugar o poder computacional desejado, os clientes não precisam de se preocupar com grandes investimentos. Nos centros de dados atuais, um cliente pode requisitar uma instância computacional de variados tamanhos, e o fornecedor do serviço assegura níveis de *performance* (através de um SLA) a essa instância computacional.

No entanto, estas garantias não são estendidas até à camada de rede. Fornecedores de *cloud computing* não oferecem garantias de *performance* de rede aos seus clientes. Uma vez que a comunicação é feita através de uma rede partilhada por todos os clientes, a *performance* que a aplicação de um cliente irá ter é imprevisível e dependente de vários fatores - alguns deles fora do controlo do cliente.

Neste trabalho propomos *omniCluster*, que soluciona estes problemas usando a abstração de uma rede virtual. Redes virtuais são isoladas umas das outras, providenciando garantias de *performance*. Desenhámos um controlador *OpenFlow* escalável, que aloca redes virtuais (com garantias de largura de banda) num sistema com conservação de trabalho, e que consegue atingir uma alta consolidação na alocação de redes virtuais e um alto uso dos recursos do centro de dados.

A nossa avaliação mostra que as propriedades mencionadas em cima foram alcançadas, sendo esta realizada testando duas comuns topologias de centros de dados: Árvore e *Fat-tree*.

**Palavras-chave:** Centro de Dados, Redes Virtuais, Garantias de Largura de Banda, Conservação de Trabalho, Redes Definidas por Software

## Abstract

The creation of data centers allowed on demand (and without commitment) access to computational resources. By renting the desired computational power, clients do not need to worry about large investments. In current data center environments, a client can ask for a computational instance of various sizes, and the service provider assures levels of guaranteed performance (through an SLA) for that computational instance.

However, these guarantees are not extended to the networking layer. Cloud providers do not offer network performance guarantees to their tenants. Since communication is carried over a network shared by all tenants, the performance that a tenant's application can achieve is unpredictable and dependent on several factors - some outside the tenant's control.

In this work we propose *omniCluster*, which solves these problems by using the abstraction of virtual networks. Virtual networks are isolated from each other, providing performance guarantees. We designed a scalable OpenFlow controller, that is able to allocate virtual networks (with bandwidth guarantees) in a work-conservative system, and achieves both high consolidation on the allocation of virtual networks and high resource utilization of the Data Center's resources.

Our assessments show that the above mentioned properties are achieved, being carried out in two common data center network topologies: Tree and Fat-tree.



**Keywords:** Data Center, Virtual Networks, Bandwidth Guarantees, Work-Conservation, Software-Defined Networking.

# Contents

# List of Tables

# List of Figures

# Glossary

| | |
|---|---|
| **AIMD** | **A**dditive **I**ncrease **M**ultiplicative **D**ecrease |
| **API** | **A**pplication **P**rogramming **I**nterface |
| **CPU** | **C**entral **P**rocessing **U**nit |
| **FIFO** | **F**irst **I**n **F**irst **O**ut |
| **HDD** | **H**ard **D**isk **D**rive |
| **HTB** | **H**ierarchical **T**oken **B**ucket |
| **ICMP** | **I**nternet **C**ontrol **M**essage **P**rotocol |
| **IETF** | **I**nternet **E**ngineering **T**ask **F**orce |
| **ILP** | **I**nteger **L**inear **P**rogramming |
| **IP** | **I**nternet **P**rotocol |
| **LLDP** | **L**ink **L**ayer **D**iscovery **P**rotocol |
| **MAC** | **M**edia **A**ccess **C**ontrol |
| **MPLS** | **M**ulti**P**rotocol **L**abel **S**witching |
| **NIC** | **N**etwork **I**nterface **C**ard |
| **ONF** | **O**pen **N**etworking **F**oundation |
| **OS** | **O**perating **S**ystem |
| **OVS** | **O**pen **vS**witch |
| **OVSDB** | **O**pen **vS**witch **D**ata**b**ase |
| **QoS** | **Q**uality **o**f **S**ervice |
| **RAM** | **R**andom-**A**ccess **M**emory |
| **RPC** | **R**emote **P**rocedure **C**all |
| **SDN** | **S**oftware-**D**efined **N**etworking |
| **SFQ** | **S**tochastic **F**airness **Q**ueueing |
| **SLA** | **S**ervice-**L**evel **A**greement |
| **TBF** | **T**oken **B**ucket **F**ilter |

| | |
|---|---|
| **TCAM** | **T**ernary **C**ontent-**A**ddressable **M**emory |
| **TCP** | **T**ransmission **C**ontrol **P**rotocol |
| **ToS** | **T**ype **o**f **S**ervice |
| **UDP** | **U**ser **D**atagram **P**rotocol |
| **VLAN** | **V**irtual **L**ocal **A**rea **N**etwork |
| **VM** | **V**irtual **M**achine |
| **XML** | E**X**tensible **M**arkup **L**anguage |

# Chapter 1

# Introduction

The creation of data centers allowed global access to huge computational resources, previously only available to large companies or governments. By renting the desired computational power, small companies (or even an individual person) do not need to worry about large investments. In current data center environments, a client can ask for a computational instance of various sizes, and the service provider assures levels of guaranteed performance (through an SLA) for that computational instance. This guarantee is possible due to the huge evolution of virtualization technologies. Nowadays, hypervisors can control the behaviour of the virtual machines (VMs) it hosts, ensuring that a VM cannot use more CPU than what it was requested (except when the hypervisor allows). In this way, clients (or tenants) are not harmed by the misbehaviour of others.

This simplicity of computational resources on demand has generated a lot of interest around the world. However, there are still a lot of things to improve in this area. One of them is the lack of network accounting in this renting of resources. Cloud providers do not offer network performance guarantees to their tenants. In fact, a tenant's compute instances (i.e. VMs) communicate over the network, that is shared by all tenants.

Thus, the network performance that a certain VM can get is dependent on several factors, most of them outside the tenant's control, such as the network load on a given moment or the placement of that VM in the network. Moreover, this is aggravated by the oversubscribed nature of a Data Center network.

The lack of guarantees in a shared communication medium leads to unpredictable application performance (as well as tenant cost). This is well documented in [47]. This work shows the impact of machine virtualization on network performance, and conclude that virtualized machines often present abnormally large packet delay variations, which can be a hundred times larger than the propagation delay between the two hosts they used for measuring. Another interesting finding by this work is that TCP and UDP throughput can fluctuate rapidly (in the order of tens of milliseconds) between 1 Gb/s and zero, which

shows that applications will have a very unpredictable performance. This can be very important, since many applications running in the cloud are data intensive, such as video processing, scientific computing or distributed data analysis. This may severely degrade the performance achieved by an application. For instance, with intermittent network performance, MapReduce[16] applications will experience harsh issues when the data to be shuffled amongst mappers and reducers is quite large.

As a curious note, Amazon solves this problem by giving tenants (at high cost) a dedicated connection of either 1 or 10 Gb/s without oversubscription[1]. However, this solution is definitely not scalable. Furthermore, the problems presented in the last chapter severely reduce cloud adoption. There are many use cases that are not applicable to cloud scenarios because of the lack of guarantees in network performance. With current solutions, it is not possible to seamlessly migrate a network of an enterprise to the cloud, while maintaining the same reliable performance of running it locally.

Solving this problem is hard for many reasons. Nonetheless, the major one is the difficulty of obtaining the current state of the Data Center network. In turn, this is hard to do because the network state is distributed: both physically among network devices that are running distributed protocols; and logically, since each network device has its own convoluted way to configure and monitor. Software-Defined Networking (SDN) [34] is an emerging networking concept, that decouples the control and the data planes. In Figure 1.1 we can see the abstraction provided by SDN.

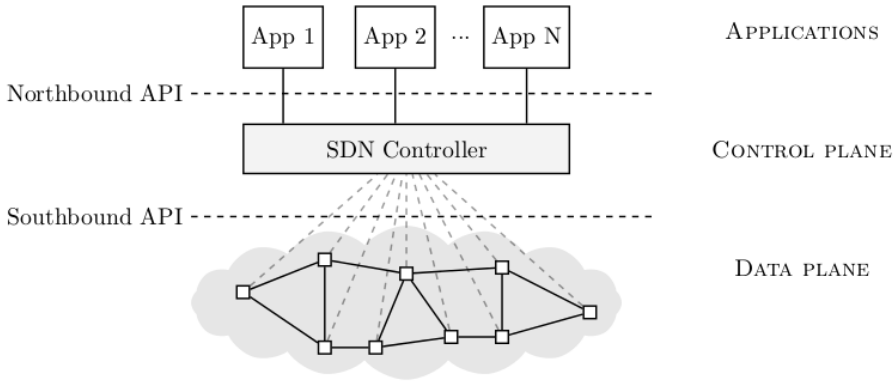



Figure 1.1: Abstractions provided by Software-Defined Networking: North- and Southbound interfaces.

This abstraction thereby decouples the forwarding hardware from the control software, which means that the control mechanism can be extracted from the network elements and logically centralized in the SDN controller. The controller creates an abstraction of the underlying network, and thereby provides an interface to the higher-layer applications. By introducing layers and using standardized interfaces, SDN brings a modular concept that enables the network to be developed independently on each one of the three layers. Most notably, it creates the opportunity, for others than networking hardware vendors, to innovate and develop control software and applications for the network.

[1] `https://aws.amazon.com/directconnect/` (Last Access: 12-10-2015)

In this project, we use SDN to have a global view of the state of the network. This way, we are able to give *virtual networks* to tenants. Such as a VM, a virtual network gives performance isolation, but at the network level. This allows tenants to express their requirements in terms of bandwidth, which is then enforced through virtual networks.

## 1.1 Shortcomings of Current Solutions

A great number of works identified the problems we have outlined above, and currently network virtualization is a research topic with a lot of attention in the networking research community. However, as later described in the document, those solutions focus either on providing exact guarantees to each tenant or on maximizing network utilization and bandwidth sharing among tenants (i.e. work-conservation).

This is not enough because these works either increase the satisfaction of tenants (providing them with guarantees), or improve the network management of data centers (favouring revenue gains of the service provider), but can not combine both goals.

## 1.2 Objectives

As expected, the objectives are focused on improving the state-of-the-art, providing properties that are lacking in current systems. Our objectives are characterized as properties of the system we want to build. Thus, we develop a network management solution that:

- Is scalable to Data Center environments;
- Achieves high consolidation (within the placement of virtual networks);
- Achieves high resource utilization of the Data Center's physical resources (namely servers and network);
- Provides bandwidth guarantees in a work-conservative system.

## 1.3 Document Roadmap

The rest of this work is organized as follows: In chapter 2 we present the related work, describing our enabling technologies as well as the research works that are similar to ours. Chapter 3 describes the architecture, algorithms and data structures of our solution, focusing on the main components and their interactions. In chapter 4 we present the implementation we have done by following the design depicted

in chapter 3. Then, in chapter 5 we evaluate our implementation, taking into account the objectives defined in the previous section. Lastly, in chapter 6 we make some concluding remarks about this work and point out some possible directions for future work.

# Chapter 2

# Related Work

In this section it is presented industry and research work considered relevant for the development of this thesis. In section 2.1, we start by describing our key enabling technology - OpenFlow. In the next section (2.2), we present the taxonomy we have created to classify the works related with this thesis. On section 2.3 we describe in greater detail the most important works from the categories presented in 2.2. Finally, in section 2.4 we make a comparative analysis of all the studied systems in order to wrap-up the related work.

## 2.1 Software-Defined Networking

Sofware-Defined Networking is an emerging network paradigm that separates the data plane from the control plane. With this separation, the control of the network is decoupled from forwarding and is directly programmable, as defined in [2]. SDN aims to move the intelligence from the network elements to the SDN controller, which greatly simplifies and cheapens the network devices, since they no longer need to process hundreds or thousands of protocol standards, but merely have to accept orders from a software-based SDN controller. The controller has a logically centralized global view of the network, which enables applications to view the network as a logical entity and thus have a more accurate reasoning about the network state. With this network abstraction, administrators can programatically configure the network and apply network-wide policies.

It is a common misunderstanding to view SDN and OpenFlow as the same. As explained above, SDN is an architecture where the control and data planes are separated, while OpenFlow is a protocol that interfaces between the network elements and the control plane (also known as the southbound interface). Besides OpenFlow, other systems such as SoftRouter [31] and IETF's Forwarding and Control Element Separation (ForCES) [52] follow this architecture. Obviously there are differences between

these systems, for instance ForCES allows multiple control and data elements within the same network and the logic can be spread through all the elements, while OpenFlow aims at centralizing the logic of the control plane. IETF documented in detail the differences between ForCES and OpenFlow in [46]. As OpenFlow became the SDN implementation that received the most research attention and industry support [33], this thesis's focus will reside in it.

### 2.1.1 OpenFlow Specification

McKeown et al. started OpenFlow [34] on Stanford University, but currently it is maintained by the Open Networking Foundation (ONF). Its initial goal was to enable researchers across the world to run experiments on their own campus networks, and is based on a previous work by some of the same authors called Ethane [14]. Ethane already had an architecture with a centralized controller in order to simplify the management and policy enforcement on enterprise networks. OpenFlow developed on this architecture, but not so focused on security and policy enforcement. Instead, the main idea was to deploy several OpenFlow-compliant switches across campus networks, which could logically separate production from research traffic. This idea gives researchers a real test bed to evaluate their work on (as opposed to evaluation based on simulations), which increases the credibility (and possibly deployment) of new systems/protocols.

In Figure 2.1, we can see the OpenFlow switch architecture, as defined in the OpenFlow white-paper [34].

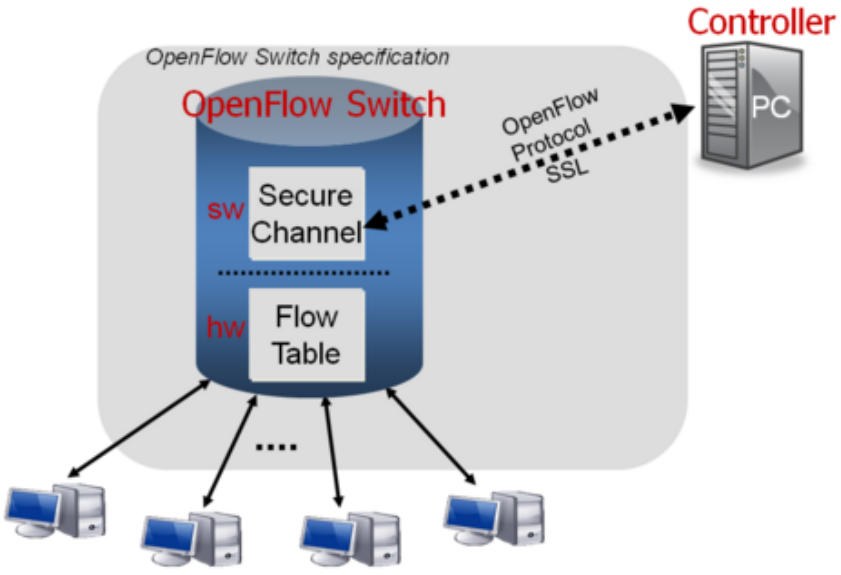



Figure 2.1: The OpenFlow switch architecture [34].

This architecture consists of three concepts: the network is made of OpenFlow-compliant switches, that form the data plane; the control plane is made by at least one OpenFlow controller; and there is a secure channel between each switch and the control plane.

Each switch is a ”dumb” forwarding device, that simply uses its flow table to determine a packet's next hop. A flow table is composed by a series of flow entries. Each flow entry has the **header fields** on which it will try to match incoming packets (e.g. Ethernet destination address), **actions** to perform when an incoming packet matches this table entry (e.g. forward to a specified port or flood to all ports), and **counters** that hold statistical information about each flow (e.g. number of packets and bytes transmitted in this flow). Each OpenFlow switch uses Ternary Content Addressable Memory (TCAM) to allow fast lookup of wild-card matches, and thus fast packet forwarding. When an incoming packet does not match any entry on the flow table, it is sent to the controller using the secure channel, which will decide what to do with this packet. In Figure 2.2 we can see an example of a switch's flow table.

OpenFlow-enabled Network Device

| MAC src | MAC dst | IP Src | IP Dst | TCP dport | ... | Action | Count |
|---|---|---|---|---|---|---|---|
| * | 10:20:. | * | * | * | * | port 1 | 250 |
| * | * | * | 5.6.7.8 | * | * | port 2 | 300 |
| * | * | * | * | 25 | * | drop | 892 |
| * | * | * | 192.* | * | * | local | 120 |
| * | * | * | * | * | * | controller | 11 |

Figure 2.2: Flow table of an OpenFlow switch (adapted from[2]).

The OpenFlow protocol defines three types of communication: Controller-to-Switch, asynchronous and symmetric communication. The **Controller-to-Switch** communication is responsible for feature detection, configuration, switch programming as well as statistical information retrieval. **Asynchronous** communication is initiated by the switch without any solicitation from the controller. It is used to inform the controller about unmatched packet arrivals (`PACKET_IN` messages), state changes at the switch and errors. Lastly, **symmetric** messages are sent without solicitation from either side. Examples for symmetric communication are Hello or Echo messages that can be used to identify whether the control channel is still alive.

The controller (also known as a network operating system - further detailed in section 2.1.2) is a remote software that uses the secure channel to manage the flow table of all switches. The controller can reprogram the forwarding logic of the network (through the OpenFlow protocol), as well as receive packets from a switch (`PACKET_IN` messages) and sending them back with instructions, either only for that packet or installing a new flow entry in the switch (for packets similar to that one). More commonly this is done by setting a new flow entry on a set of switches (and not only on the one that generated the

PACKET_IN message), which will define the flow for that kind of packets. The controller is also able to create a packet from scratch (i.e. without any event generated by a switch), and send it to one or more switches, through PACKET_OUT messages.

**OpenFlow versions:** The OpenFlow 1.0 specification [1] is currently the most widely used throughout the world [33]. As of the writing of this thesis, the most recent OpenFlow specification is version 1.4 [3]. In the OpenFlow 1.0 specification (the first non-draft version) the switch can process 12 header fields (for instance Ethernet, IP, TCP and UDP source and destination adresses), has only one flow table and can only connect to one controller. Along the years, with the evolution of the protocol, the OpenFlow 1.4 specification allows an OpenFlow switch to process, among others, IPv6, ICMPv6 and MPLS headers. It also introduces the OpenFlow Extensible Match, which drastically changes how packet matching is made and allows the definition of new match entries in an extensible way. Another improvement are: the support for multiple flow tables (and also groups of tables), which eases the implementation of complex forwarding tasks (such as multipath); and the switches' ability to connect to multiple controllers at the same time, which enhances scalability and fault tolerance.

OpenFlow began with a very basic support for Quality of Service (QoS). As the implementation of QoS mechanisms are very vendor specific, the protocol has to be general enough to cope with all these different mechanisms and hardware. In version 1.0, the protocol defines the creation of queues, with minimum rate as its only parameter. Instead of forwarding directly to a port, one can define the flow entries on a switch to forward to specific queues (with *enqueue* action), each with its own minimum rate. One switch port can contain one or several queues. On version 1.1 it was added the ability to process MPLS labels. On version 1.2, each queue can also have a maximum rate. The version 1.3 of the specification brought an important improvement for QoS enforcement, which is the meter table. This table contains per-flow meters that measure the rate of packets attached to each flow (as opposed to queues that are attached to ports). This new feature combined with queues allows the implementation of complex QoS frameworks [3], such as DiffServ (Differentiated Services).

It is important to refer that through the OpenFlow protocol the controller can only query information about the queues of a given switch, but not modify it, as it is out of scope of the OpenFlow specification [3]. To achieve this, the ONF created the OF-CONFIG protocol [4]. Using this protocol, one can configure the queues of a switch, but also other parameters not defined by the OpenFlow protocol, such as ports parameters or the administrative controller for a switch. The version 1.0 of OF-CONFIG requires switches with OpenFlow 1.2 or later. As of this writing, the latest version of OF-CONFIG protocol is 1.2 [4], which supports switches up to version 1.3 of OpenFlow.

### 2.1.2 OpenFlow Controllers

Gude et al. published the first OpenFlow controller - NOX [21]. The authors of this work identify the need for an abstraction to the management of networks, making an analogy to operating systems. In the early days of computing, programs were written in machine language and as a result they were hard to deploy, port and debug. Operating systems created an abstraction layer to the physical resources, which enhanced productivity and enabled programs to run independently of the underlying physical resources. The controller aims at providing this abstraction for networks, becoming itself a centralized control plane (and hence decoupling from the data plane), and being responsible for controlling the switches logic (using the OpenFlow protocol described in the previous section). This abstraction eases policy enforcement and increases optimal resource management, since applications running on top of a network operating system can get the ”big picture” of the network (network-wide view), as opposed to current network management done by distributed low-level algorithms.

**Industrial OpenFlow Controllers:** Since the release of NOX a great number of other OpenFlow controllers have been developed. Mainly they fall into two classes: closed-source commercial controllers; and open-source research and community oriented controllers. Examples of the former are Big Network Controller [1] from Big Switch Networks, Onix [30] from Nicira Networks (now owned by VMWare) and Contrail[2] from Juniper. Examples of the latter are: Trema and MUL (C based); Beacon [19] and Floodlight [3], a fork of Beacon (Java based); POX, Pyretic, and Ryu (Python based). OpenDaylight [35] is a Java based controller that can be seen as belonging to both classes, since it is an open-source project hosted by the Linux Foundation, but at the same time is supported by the networking industry with contributions (of both code and resources) from several companies. It is the first controller that supports communication with the switches (known as the southbound interface) not using the OpenFlow protocol (e.g. using NETCONF [17]).

The Beacon controller was one of the first ones to appear (after NOX) and its source-code was the basis of both Floodlight and OpenDaylight. However, natively Beacon does not support topologies with loops in it, making it more difficult to work with. Beacon is no longer in development. Both Floodlight and OpenDaylight support topologies with loops and both are in active development and with great acceptance by the community. They were both considered as an option to be used in our work, but the choice fell on Floodlight, with the justification presented in section 3.1.

There has been some research work on how to evaluate the performance of SDN controllers. In [45], Tootoonchian et al. propose Cbench, a benchmarking tool for SDN controllers that emulates a

[1] `http://bigswitch.com/products/SDN-Controller` (Last Access: 15-05-2015)
[2] `http://www.juniper.net/us/en/local/pdf/whitepapers/2000515-en.pdf` (Last Access: 15-05-2015)
[3] `http://www.projectfloodlight.org/floodlight/` (Last Access: 01-10-2015)

given number of OpenFlow switches that generate traffic to the controller to measure its performance. Performance metrics of Cbench are the throughput (events per second that the controller can handle) and the response latency (time to answer to a switch's request). Shalimov et al. created Hcprobe [39], a tool that is based on Cbench with some extra functionalities. The authors of Hcprobe state that previous evaluations of SDN controllers were only focused on performance. Thus, for reliability testing Hcprobe can measure the number of controller failures in a long-term run, and for security evaluation it tests if the controllers process correctly malformed OpenFlow packets and headers. Both of these works use their tools to make performance analysis on several OpenFlow controllers (presented in the previous paragraph), concluding that Beacon is the controller with the highest performance (confirming the results shown in [19]) The Python based controllers showed the worst performance, since they can only run in a single thread. Due to OpenDaylight recent publication date, as of this writing there was no evaluation work considering the OpenDaylight [35] controller.

**Research OpenFlow Controllers:** While having a centralized control plane brings many benefits to network management, it also raises several concerns regarding the scalability, robustness and reliability of the controller in large scale networks. Heller et al. worked on the controller placement problem [27], which studies the number of controllers needed and respective location for different topologies. The correct position is calculated by solving an optimization problem for a predefined metric (average or worst case latencies in their proposal) from all nodes in the network to the controller. They concluded that for most topologies, there is only need for one controller as long as it is correctly positioned. To address control plane failures, two distributed control plane approaches have been proposed: Tootoonchian et al. presented HyperFlow[44], based on a publish/subscribe system between the controllers, which propagate events between them to maintain a consistent state among different controllers. An OpenFlow switch connects to one controller only, but any controller in the network can control any switch, by intercepting messages that travel on the network and issuing new events in the control plane directed to an authoritative controller of a given switch; The second approach is the work by Botelho et al. called SMaRtLight [11]. In this case each switch connects to the same controller, and there are backup controllers that can take the primary role in cause of failure. The primary and backup controllers have a shared data store, implemented as a Replicated State Machine, in order to keep the state consistent among controllers. SMaRtLight shows only a preliminary evaluation, testing the throughput of the proposed controller as the works referred in the previous paragraph. HyperFlow also shows a rudimentary evaluation, describing only the tests procedure and stating that HyperFlow can withstand a level of dynamism of the network of up to 1000 changes per second in the network (links up/down or switches joining/leaving).

### 2.1.3 OpenFlow Network Simulators/Emulators

In this section it is explored the most relevant network simulators/emulators for this work. The widely known network simulator ns-3 [28] has support for an OpenFlow simulation model[4], however its current implementation does not model the OpenFlow controller as an external entity, therefore it does not support simulating multiple switches connected to an OpenFlow controller. Thus it does not allow to simulate a software-defined network with a controller dynamically changing the behaviour of the network elements.

**Mininet 1.0:** Lantz et al. developed Mininet [32], a network emulator system for prototyping large networks on the constrained resources of a single laptop. Mininet uses lightweight virtualization to scale to large topologies. The file system, kernel, device drivers, and other common code are shared between processes and managed by the operating system. It virtualizes the nodes by creating a different network namespace (i.e. a container for network state) for each host, which gives each one their own virtual Ethernet interface. The links are virtual Ethernet pairs (veth pairs), which makes virtual interfaces appear as a fully functional Ethernet port to all system. Mininet supports both user and kernel space virtual switches, such as the OpenFlow reference switch [3] (user space) and Open vSwitch [36] (user and kernel space). Open vSwitch consists in a virtualized switch, making it possible to deploy/remove or scale up/down according to the needs of the administrator. Mininet supports changing dynamically the network topology (i.e. links can be set to up/down and hosts can crash) and as this is an emulated environment, the authors state that the experiments made on Mininet can be ported to hardware seamlessly. The Mininet paper [32] indicates that one can create a network of 1024 hosts and 32 switches using 492 MB of RAM.

**Mininet 2.0 (Hi-Fi):** One of the main limitations the authors presented in Mininet is the lack of performance fidelity, especially at high loads. Since CPU resources are multiplexed in time by the default Linux scheduler, there is no guarantee that a host that is ready to send a packet will be scheduled promptly. So, the authors improved Mininet and presented Mininet Hi-Fi [25] (High-Fidelity). Mininet-HiFi extends the original Mininet architecture by adding mechanisms for performance isolation, resource provisioning, and monitoring for performance fidelity. This is done by using the following Linux features: control groups (*cgroups*), which allow a group of processes belonging to a virtual host to be treated the same way; CPU Bandwidth Limits are enforced for each virtual host, assigning a maximum time quota (configurable) for each *cgroup*; Each link (veth pair) is configured using the traffic control tool (*tc*), which sets properties such as bandwidth or delay. Mininet Hi-Fi then constantly monitors the performance of each

[4] `http://www.nsnam.org/docs/release/3.13/models/html/openflow-switch.html` (Last Access: 09-01-2015)

virtual host or virtual link, so that in the end of the experiment it is able to tell if fidelity was kept or the experiment should be reconfigured (or scaled down). Since in Mininet the only limitation to the size of the experiments is the underlying computational infrastructure, Wette et al. presented an extension to Mininet called MaxiNet [51]. MaxiNet is an abstraction layer connecting multiple, unmodified Mininet instances running on different computers. A centralized API (called MaxiNet API), which is very similar to Mininet API, provides access to this cluster of Mininet instances, which allows to control the experience. The connection between MaxiNet and Mininet instances happens through RPC calls. In this paper the authors show that using MaxiNet with 12 computers, they were able to emulate a data center with 3600 nodes (servers and switches), maintaining the CPU usage of each computer below 80% (to compensate for peaks in the CPU usage, which otherwise would bias the outcome of the experiment).

**EstiNet:** EstiNet [49], presented by Wang et al., is commercial tool that at the same time is a network emulator and simulator. It is an emulator since every node uses the real TCP/IP protocol stack and also the code developed in EstiNet can readily run on without any modification on a real scenario. To address the performance fidelity, EstiNet is at the same time a simulator, having its own simulation engine. Instead of scheduling the virtual nodes events using the underlying operating system (which may be unpredictable), it uses its simulation engine to schedule all events according to the simulation clock, which is slower than real time. This is done using a technique known as Kernel Re-entering, which uses tunnel network interfaces to intercept the packets exchanged by virtual nodes and redirect them into the EstiNet simulation engine.

Since EstiNet is based on a simulation clock, the authors claim that its results are accurate and repeatable. One of the authors published a study [48] comparing the performance fidelity and scalability of EstiNet versus Mininet. In this study the author evaluates both platforms creating a grid of NxN switches (where N goes from 5 up to 31) and then testing the ping time between two hosts at opposite edges of the grid. The author concludes that generally Mininet worked well but EstiNet gives results closer to the theoretical ones and that they are repeatable. However, it needs more time to simulate OpenFlow switches [48]. Nevertheless, according to [33], Mininet is the network emulator most used by the network research community.

## 2.2 Strategies for Network Flow Allocation in Data Centers

We now present the classification that emerged from the study of several works regarding network management in data centers. At the highest level, the studied works can be divided in two categories. Thus, our taxonomy for network allocation techniques in data centers is: dynamic allocation based

on network monitoring; and static allocation based on virtual network embedding. The former aims at maximizing resource utilization, fair bandwidth sharing of the physical network and sometimes give Quality of Service (QoS) guarantees for end hosts, while the latter provides support for virtual networks with QoS guarantees but with small (or even none) maximization of network usage. Each category is now explained in greater detail.

### 2.2.1 Dynamic Allocation based on Network Monitoring

This kind of approaches rely on a constant monitoring of the network performance to keep an up-to-date view of the data center network. For that purpose, it is periodically gathered statistical information from the network elements (and/or end hosts), in order to infer the bandwidth usage for each VM or physical server. With this updated view of the network, this type of systems can use traffic engineering techniques to maximize resource utilization, leading to an economical gain for the provider, since it can for instance aggregate requests and shut down unused resources. The maximization of resource usage is possible since these dynamic approaches can react to the changing demands from different tenants, and manage them to get the most of the providers' resources. The solutions we have surveyed can be categorized as based on: centralized traffic matrix estimation or bandwidth regulation in each end host.

**Centralized Traffic Matrix Estimation:** In current multi-tenant data centers there is a huge number of different applications sharing the infrastructure, which makes demand estimation very hard since the workload of each VM is very hard to predict. Traffic (or demand) matrix estimation approaches are based on a central node that collects information about the state of the network to detect large flows (in terms of bandwidth) between a certain VM/server pair. For traffic matrix estimation, all of the studied systems follow a very similar technique. They create an N by N matrix, where N is the number of hosts in the data center. Inside this matrix, in the row $i$ and column $j$ is the traffic demand from host $i$ to host $j$. Upon detecting the current demands for the data center, the central node is able to dynamically react and place large flows on links with more capacity and aggregate as much as possible the small flows.

Examples of works that follow this idea are Hedera [6], DevoFlow [15] and MicroTE [9]. These three works are very similar between them, thus we will only describe with detail MicroTE because it is the most related with this work. Heller et al. presented ElasticTree [26], which as these systems is based on traffic matrix estimation. However, the goal of the system is to know the matrix to save as much energy as possible in the data center, which makes the system different from these since ElasticTree does not provide neither bandwidth sharing nor bandwidth guarantees.

**Bandwidth Regulation at End Host:** Instead of monitoring the whole network and predict the current traffic demands on a centralized node, some authors defend the idea of managing the data center's bandwidth in a distributed manner. In this model, each entity (e.g. a process or a VM) gets assigned a weight (defined by the network administrator), which will be used to share the bandwidth across multiple tenants in a proportional way. There is a congestion controlled tunnel between each pair of communicating entities, that is used by the receiver to inform the sender of congestion when packets are lost. The sender upon receiving this congestion message adjusts its rate to spare the network resources. The mechanism proposed here is similar to Additive Increase Multiplicative Decrease (AIMD) used by TCP in congestion avoidance. This mechanism is implemented by a rate controller in each physical server (e.g. in the *hypervisor*), which throttles the sending rate according to: the weight of each entity as well as the congestion messages received. With this approach (and unlike the previous paragraph), malicious tenants cannot hog the network bandwidth since the resources are fairly shared according to the weights previously defined.

Works following this model are presented in [41, 38]. Although they are very akin to each other, they will both be detailed in section 2.3.1, because Gatekeeper [38] has some ideas inspired from Seawall [41], but it improves the state-of-the-art by introducing a new property (namely a work-conservative system with bandwidth guarantees at the same time).

### 2.2.2 Static Allocation based on Virtual Network Embedding

A different way to allocate network resources is to use virtual network embedding algorithms. Whereas the algorithms from the previous section aim at maximizing the resource usage and QoS guarantees for the tenants, in virtual network embedding the aim is to completely virtualize the network, providing performance isolation among tenants at the network level. So, virtual network embedding consists in the mapping of virtual networks (consisting of virtual nodes and links) onto the substrate network (consisting of physical nodes and links). Some works do not call this embedding, but are doing the same thing (mapping virtual networks in one or more physical networks). However, the most recent papers have used this nomenclature, which shows this name is generally accepted by the scientific community.

Virtual network embedding is the main challenge in the implementation of network virtualization [20]. In the data center context, network virtualization is advantageous because it allows clients to define the desired network topology, which will be allocated within the infrastructure exactly as defined by the client. This enables seamless migrations of current enterprise or scientific networks to a data center environment, since the client defines the topology and the wanted guarantees (either CPU or bandwidth between machines), which will be enforced by the embedding algorithm by placing the virtual network

only where there are sufficient resources left.

This means that virtual resources are first mapped to candidate substrate resources. Substrate resources are spent (and the entire network is embedded) only if *all* virtual resources can be mapped. Network embedding algorithms work similar to the Integrated Services architecture (IntServ [13]), making reservations in the substrate network according to virtual networks' requests. This gives strict bandwidth guarantees for each tenant since each link is never oversubscribed, but can lead to poor resource utilization as each reservation will not utilize the requested capacity all the time. This trade-off will be addressed again in section 2.4.

The virtual network embedding problem is proven to be $\mathcal{NP}$-hard (and related to the multi-way separator problem), as it deals with constraints on both physical machines and links. So, we classify virtual network embedding systems according to if they output embedding results according to exact optimization or if they employ some heuristic to lower the execution time (and thus give an approximate result). We first give a formal description of the virtual network embedding problem and then describe the two categories mentioned above.

**Problem Formulation:** We now describe the virtual network embedding problem in a formal manner. Consider the following: $SN = (N, L)$ is the substrate network, in which $N$ is the set of nodes and $L$ the set of links of the network; $VNR^i = (N^i, L^i)$ is the set of $i = 1, ..., n$ virtual network requests, where $N^i$ and $L^i$ are the set of nodes and links of virtual networks request $i$, respectively; $\overrightarrow{R}$ is a vector space of resource vectors and $cap : N \times L \rightarrow \overrightarrow{R}$ is a function that assigns available resources to some elements of the substrate network; finally, $dem_i : N^i \times L^i \rightarrow \overrightarrow{R}$ is the function that assigns demands to elements of all virtual network requests, for each $VNR^i$. With this defined, the virtual network embedding problem consists on the results of the functions $f_i : N^i \rightarrow N$ and $g_i : L^i \rightarrow SN' \subseteq SN$. The functions $f_i$ and $g_i$ are called node mapping function and link mapping function, respectively. The two of them combined form the embedding for the virtual network request $i$ ($VNR^i$). The problem itself can then be described as for each $VNR^i$, these functions must comply with: $\forall n^i \in N^i : dem_i(n^i) \leq cap(f_i(n^i))$ and $\forall l^i \in L^i : \forall l \in g_i(l^i) : dem_i(l^i) \leq cap(l)$. The bottom line is that the sum of each node and link demand within the Virtual Network Requests ($N^i$ and $L^i$) must be less or equal than the physical resources available both for nodes ($cap(n)$) and links ($cap(l)$).

**Calculation of Optimal Solution:** The optimal embedding solutions can be calculated using optimization methods, such as Linear Programming. Namely, we can use Integer Linear Programming (ILP) to formulate the virtual network embedding problem, including both the node and link constraints in the same formulation. Although ILP is in most cases $\mathcal{NP}$-complete, there are algorithms (e.g. branch and

bound) that can solve the optimization problem in a legitimate amount of time. The work presented in [29] uses ILP to solve the virtual network embedding problem. It seeks the minimization of embedding cost and the maximization of the acceptance ratio of virtual network requests, including those criteria in the objective function of the ILP formulation. Other example of the usage of exact formulation of ILP is [12], which tries to make the virtual network embedding energy-aware. For that, it makes the embedding in the smallest set of resources of the substrate network (i.e. consolidate), with the objective to shut down unused resources (and thus save energy consumption).

In the fast-paced environment of a data center, the execution time of an algorithm is crucial, since it limits the amount and rate of virtual network requests a provider can process. So, algorithms that perform the calculation of the optimal solution are not suited for a data center, because it is a large environment, giving a huge search space for the network embedding algorithm. Moreover, as in a data center the tenants' requests arrive in real time, we do not know the whole set of virtual networks we want to map on the substrate network. This makes the calculation of the optimal solution possibly a waste of time, since it can spend time calculating a optimal solutions for conditions that will change very soon (upon the arrival of the next virtual network request). For these reasons, approaches based on optimal solutions are not further detailed, and thus are not presented in section 2.3.

**Heuristic-based Approaches:** As stated in the last paragraph, the calculation of an exact solution can be very time consuming. To deal with the $\mathcal{NP}$-hardness of the virtual network embedding problem, heuristic approaches "trade" a low execution time for a not optimal (but acceptable) solution.

Besides low execution time, these heuristics may have many different goals. Each heuristic-based algorithm is tailored to some specific objective as well as to some specific environment. For instance, in [37], the goal of the algorithm is to make the virtual network embedding with fast failure recovery times, which the authors call survivable networks. For this purpose, the heuristic used in this work is focused on creating a link of *backup paths* for each substrate link (e.g. using $k$-shortest path [18]), which will then give a fast re-routing of the current network embeddings when a link failure happens.

There are many other examples regarding heuristic-based approaches for virtual network embedding. However, we will now focus on works that are closely related to ours, i.e., virtual network embedding in data center environments. In the next section each work will be described in detail (along with the differences between each one). However, we were able to conclude that the core of all the studied algorithms is the same, since they all focus on delivering bandwidth guarantees in a data center network. The heuristic these works employ to reduce the search space for a solution (and hence reduce algorithm execution time) is to try to take into account server locality of the same virtual network embedding request. Therefore, upon a virtual network request, these systems first do the node mapping, by trying

to accommodate everything inside a single server (if the request is smaller than the VM capacity of a server), after that they try on servers of the same rack, then on adjacent racks, and so on.

The choice of the first server rack to analyse varies from work to work, but it is either random or in a round-robin fashion. The link mapping phase only starts if the virtual network request completes the node mapping phase. If it does not, it can be put into a queue to be processed later or the request discarded to alert the client there are not resources available for his request.

Thus, in the link mapping phase each node of the virtual network is already mapped to a substrate VM. The problem is now to connect these VMs according to the bandwidth requirements. To accomplish this, some works use the $k$-shortest path algorithm [18] (increasing $k$ until the bandwidth available complies with the request), while other works use the widest shortest path algorithm [50] to provide a path with the required bandwidth.

This kind of algorithm can be called *greedy*, since it picks the locally optimal choice at each branch in the road (i.e. it chooses the best solution that complies with the virtual network constraints). With this type of algorithms the virtual networks also benefit from the reduced number of hops in the substrate network, which will yield low latency (as much as possible, but with no guarantees) between the VMs communicating in the virtual network. Some works following this approach can be seen in [24, 8, 7, 53, 54]. Although these are all based on heuristics, the work from Yu et al. [53] does not reduce significantly the search space for a virtual network embedding solution, since it considers path splitting (i.e. bifurcated traffic) and migration of VMs at the same time. This results in a solution only appropriate for small data centers, since it produces results in a reasonable time only when there are tens to hundreds of nodes in the network. We will further detail in section 2.3.2 Oktopus [7] and SecondNet [24], as these have the most interesting properties and/or implementations to discuss.

## 2.3 Relevant Related Systems

We now describe the works most related with this thesis. Whereas in the previous section we have outlined the common characteristics of the surveyed works, we now detail the most relevant works, showing the particularities of each one. As in section 2.2, we will divide this section in: dynamic allocation based on network monitoring; and static allocation based on virtual network embedding.

### 2.3.1 Dynamic Allocation based on Network Monitoring

**MicroTE:** This system [9] uses OpenFlow as a framework to have centralized control over the data center and make routing decisions based on predictions of the traffic matrix. Based on measurements

made by the authors, they state that current traffic engineering techniques do not apply well to a data center, because they are too slow to react to micro-congestions. Therefore, the authors indicate that the reaction time must be under 2 seconds to be effective. For this purpose, they create a hierarchical structure to make traffic measurements that is scalable both with the data center size and with having a centralized controller.

This works as follows: on each rack, there is a server (called the designated server) that is responsible for collecting statistics from all servers in that rack every 0.1 s and also calculating the mean of the traffic demand for each pair of servers. This designated server then receives this demand and computes the new traffic matrix. If the average and this new value are within $\delta$ of each other, the designated server does not do anything (the authors empirically determined $\delta$ to be 20%). Otherwise, the designated server sends summarized information about the traffic demands (i.e. only the positions of the matrix that changed at least 20%) to the centralized controller.

With this structure, the controller only receives information about the traffic demands when they change, which decreases the amount of control traffic in the network. So, it is able to react to changes in a reasonable time, reconfiguring the network elements to place each demand in a link that best fits it. To determine this reconfiguration of the network, the centralized controller has a routing engine that can implement several routing algorithms, according to the preferences of the administrator. As an example, the authors describe a bin-packing heuristic they implemented in the routing engine. It begins by sorting the demand pairs in decreasing order of traffic volume. For each pair, it is computed the minimum cost path between them, where the cost of each link is the reciprocal of its available capacity. After assigning the traffic of a demand pair to a path, it is updated the cost of each link along the path by deducting the pair's traffic volume from the residual capacity of the corresponding links. In this way, a highly utilized link is unlikely to be assigned more traffic and the maximum link load will be minimized.

**Seawall:** Seawall [41] tackles the problem of fair bandwidth sharing and network performance isolation in data centers. Current data centers have strong performance isolation at the VM level (CPU, memory and so on), but they lack performance isolation at the network level, since an increase of a VM's bandwidth usage can lead to an decrease in performance of other VMs. The authors of Seawall propose to overcome this problem by assigning weights to each entity (can be a VM or a process inside a VM). Then, the share of bandwidth obtained by the entity in each network link is proportional to its weight.

To accomplish this, Seawall relies on congestion-controlled tunnels created between each pair of source and destination VM. This is implemented at each server in a shim layer, which forces each VM to use these tunnels. Sequence numbers are added to each packet sent trough the tunnel. These sequence numbers are stripped at the destination server and are used to detect packet losses due to

network congestion. Upon receiving congestion notification messages from receivers, senders use information from network topology to detect bottleneck links and adjust transmission rates at tunnels using that link. The adjustment is computed at each end host and is proportional to the weights associated with VMs sending traffic to that link. Rates are adjusted using weighted AIMD functions, as in TCP.

Since the main block of this system is the rate-limiter at each host, Seawall works irrespective to the data center topology and the communication protocols used by the VMs. It also achieves high scalability by maintaining the state distributed in each host. However, the hosts need updated information about the network topology so that Seawall can work, and the authors do not specify how this works. They relegate the updated topology dissemination problem to management software that is responsible for provisioning and monitoring the servers, and propose to use *traceroute* in case this software does not provide this.

**Gatekeeper:** Gatekeeper [38] is a system similar to Seawall [41], but providing minimum bandwidth guarantees. The authors recognize that Gatekeeper shares some design ideas with Seawall, but they extend them to achieve different goals.

Namely, Gatekeeper uses Open vSwitch [36] in each server to control all the VMs within a server. Each VM has a vNIC (virtual network interface card), that connects to the Open vSwitch. To each VM is assigned a minimum receive bandwidth guarantee as well as a minimum send bandwidth guarantee (it can also be assigned a maximum bandwidth for each VM). Minimum bandwidth guarantees are achieved using an admission control mechanism that limits the sum of guarantees to the available physical link bandwidth.

On the sender side, transmission bandwidth scheduling is done using a traditional weighted fair scheduler that provides these guarantees. On the receiver side, Gatekeeper monitors the receive traffic rate at each vNIC (and also the physical link) and determines the receive bandwidth allocation to each vNIC at periodic intervals (10 ms in the authors implementation), taking into account the link usage and the minimum rate for each vNIC.

If a vNIC receive bandwidth exceeds its computed allocation, Gatekeeper sends a feedback message to other remote Gatekeeper instances hosting VMs contributing to its traffic. The feedback message includes an explicit rate that is computed by distributing the desired vNIC receive rate among the senders. A feedback message is generated if the aggregate rate on the physical link exceeds a given threshold (95% of the link bandwidth in their implementation) or if a vNIC exceeds its predefined maximum rate. This message is the equivalent to the congestion message in Seawall, except in this case is to control the rates that are operating above predefined allocation. In this way, the system throttles the misbehaved VMs right away, which avoids congestions in the data center.

Also, each vNIC can exceed its guaranteed allocation when extra bandwidth is available, which is implemented by the weighted fair scheduler in the virtual switch (Open vSwitch). This is important in order to maximize resource utilization, since with a static allocation (for minimum bandwidth guarantee) the resources could be idle, resulting in economical losses to the resource provider. Although the Gatekeeper idea of mixing guarantees with a work-conservative system is very promising, it is yet to be determined its performance under a realistic scenario, since the authors only evaluated this work with two tenants and six VMs, postponing to future work an evaluation in a different data center environment (larger configuration and dynamic workloads).

### 2.3.2 Static Allocation based on Virtual Network Embedding

**Oktopus:** The authors of Oktopus [7] affirm that the unpredictability of network performance in data center environments is damaging both tenants and service providers, since the tenants' applications suffer from this unpredictability and the service provider can incur in avoidable revenue losses. So, their work is intended as a move towards predictable networks, giving network guarantees to tenants.

In Oktopus tenants require a virtual cluster, defining the number of VMs and the bandwidth desired between them (e.g. $<$3, 100$>$ means 3 VMs with a 100 Mbit/s connection between them). Tenants can also require a oversubscribed virtual cluster, where the links are oversubscribed and the guarantee is weaker, but with the advantage of lower cost. Upon receiving a request, Oktopus relies on a central node that does the embedding of this request onto the physical network. The algorithm running in the central node tries to do the embedding in the smallest sub-tree possible, starting in the same machine, then on the same rack and so on, checking at each stage if this sub-tree complies with both VMs and networking requests. This makes this system only applicable to data centers with tree-like topologies. The greedy algorithm to the oversubscribed virtual cluster is the same, but with the constraints on the links less severe (according to an oversubscription factor). Oktopus makes static allocations, which does not allow one tenant to use more bandwidth than what was defined in the request, even if it is free. To enforce this, it employs a rate-limiter in the hypervisor of each physical machine, throttling the bandwidth of each VM according to what was allocated.

**SecondNet:** This work [24] has the same structure as the one presented in the previous paragraph, i.e. a central unit that receives virtual network requests to process and allocate. SecondNet uses the central unit to run the embedding algorithm, so that it outputs the physical machine of each VM and the routes between VMs. But, unlike Oktopus, these routes calculated by the central node do not translate into routing rules to the switches because in SecondNet the physical machines keep information about

the routes of each VM it owns. The authors name this Port Switching Source Routing (PSSR) and it consists in injecting in each packet a list of output switch ports the packet will go through.

When a packet arrives to a switch, the switch pops the first element of this list, getting the output port to forward this packet. The next switch receives the list with one element less and applies the same procedure. With this approach the switches are stateless, passing this responsibility to the physical servers (making the system more scalable). Similar to Oktopus, SecondNet throttles VM traffic at the hypervisor to provide guarantees and performance isolation. But in SecondNet this throttling is not limited to the requested bandwidth, and the authors introduce three traffic levels (with decreasing priority): guaranteed bandwidth, guaranteed bandwidth on first and last hop, and best-effort. The best-effort traffic is only routed when the other two types of traffic are not using the entire bandwidth. This is not fair bandwidth sharing as presented in 2.2.1, but it is better than static reservations with no bandwidth sharing.

Regarding the allocation algorithm running on the central entity, it also tries to embed each request on the smallest subset possible (same machine and so on). But SecondNet does not use a sub-tree to check if the request can be mapped to these physical resources. Instead, it builds a bipartite graph, with VMs on one side and physical servers on the other, adding edges to the graph for matching pairs of $<$VM, Physical Server$>$. This makes the SecondNet system independent of the physical topology of the data center, making it deployable in more environments.

The authors also present (but do not detail) a second algorithm that runs periodically and is responsible to consolidate the VMs of the same virtual network as close as possible. The algorithm checks the virtual networks that are more scattered across the physical network and sees if it is possible to migrate some VMs to make the virtual network smaller (in terms of path lengths). The authors further state that this is a process that can be running in background when the load of the VMs to be migrated is low. However, this can lead to waiting too long for a certain VM load to get low, possibly causing the migration adverse (or inapplicable) at that time.

## 2.4 Summary and Final Remarks

After the classification and description of the related work throughout this section, we now present a table (Table 2.1) that shows a comparative analysis of the systems and solutions we have studied.

The criteria present in Table 1 is the criteria we have been discussing throughout this section and needs no explanation. Regarding the ”Partially”, each one was explained appropriately when the corresponding system was being described.

Other than that we can see that each system has its focus on a certain area, providing the set of properties ”expected” for that kind of system. The first four rows of this table are the system that employ

| System | End-to-end Bandwidth Guarantees | Work-Conservative | Support for Virtual Networks | Support for Consolidation | Optimization Technique | Scalable to Data Center Size |
|---|---|---|---|---|---|---|
| **DevoFlow** | No | Yes | No | No | Not Applicable | Yes |
| **Hedera** | No | Yes | No | No | Not Applicable | Yes |
| **MicroTE** | No | Yes | No | No | Not Applicable | Yes |
| **Seawall** | No | Yes | No | No | Not Applicable | Yes |
| **Gatekeeper** | Yes | Yes | No | No | Not Applicable | Yes |
| **Houidi et al.** | Yes | No | Yes | No | Exact | No |
| **Botero et al.** | Yes | No | Yes | Yes | Exact | No |
| **Yu et al.** | Yes | No | Yes | Yes | Heuristic-based | Partially |
| **VDC Planner** | Yes | No | Yes | Yes | Heuristic-based | Yes |
| **CloudNaaS** | Yes | No | Partially | No | Heuristic-based | Yes |
| **Oktopus** | Yes | No | Yes | No | Heuristic-based | Yes |
| **SecondNet** | Yes | Yes | Partially | Yes | Heuristic-based | Yes |

Table 2.1: Comparative analysis of the studied systems.

a dynamic allocation that is based on network monitoring. None of these systems provide bandwidth guarantees, but they are work-conservative systems, since they foster fair bandwidth sharing among tenants. The other eight rows of this table show systems that perform static allocations based on virtual network embedding. These systems provide bandwidth guarantees for their tenants and most of them support consolidation of the virtual networks they allocate.

The most ”all-around” system, and also the most similar to the solution we present in the next chapter, is SecondNet. We can see that it has almost all the properties presented in the table, with exception for supporting virtual networks, because they only employ classes of traffic (i.e. a tenant can not ask for an exact bandwidth guarantee). Another difference is the approach regarding consolidation. They employ an algorithm that runs periodically, while we do the consolidation incrementally, upon the reception of each request.

# Chapter 3

# *omniCluster*

We now present and describe our solution - *omniCluster*. We begin by providing a generic use case for our system, in the following paragraphs. Then, in section 3.1 we provide a detailed architecture description, focusing on the interactions between each component of our solution. In section 3.2 we go in-depth to the components presented in the previous section, and describe the internal working of each one. Then, in section 3.3, we give some concluding remarks regarding our solution and summarize what has been described in this chapter.

In Figure 3.1 we can see the high level architecture of the solution, in this case using a simple tree topology with depth equal to 3 and fanout equal to 2. Although many topologies are being proposed by the network research community, we will focus on the traditional tree-like topologies (Tree and Fat-tree) since they are still the most used data center topologies, as described in [10]. The OpenFlow controller (in the top-right corner) is the central component of the solution, as it is responsible for running the virtual network embedding algorithm in order to map the requests on the substrate network, and then program the switches to deploy the requested virtual networks.

As explained in section 2.1, each switch acts merely as a packet forwarder, according to the rules dictated by the controller. The controller has a connection to every switch in the network, represented by the dashed red lines that form the control plane. They are dashed to represent the logical (and not physical) separation between the data and the control planes. The control plane does not necessarily need dedicated connections, it can use the same physical links of the data plane.

As in most of the works described throughout the previous chapter, our network embedding algorithm tries to allocate the requests on the smallest available subset of the substrate network (i.e. on the same physical server, then on the same rack, and so on). Thus, we aim to maximize the proximity of VMs belonging to the same tenant, which results in minimizing the number of hops between those VMs. This is advantageous for two reasons: first, with less hops the delay is normally reduced; and second,

keeping the VMs close (e.g. in the same rack) relieves the bandwidth usage in the upper links of the tree, which in the data center is where the bandwidth is scarcer[10]. With this approach, we will be able to accept more virtual network requests, since the core links will not be so likely to become the bottleneck of the data center.

In Figure 3.1, we can see an example of the placement of three virtual networks according to this algorithm. The virtual network of tenant A represents the best case possible where all the VMs of the virtual network can be mapped on the same physical server. In this case, there is no usage of the network (which saves bandwidth for future requests), and the bottleneck of the virtual network is only the speed within the server. In the virtual network of tenant B the request could not be mapped to a single server, and so it uses another server belonging to the same rack to accommodate the entire request. The virtual network of tenant C shows a case where the virtual network could not be mapped in the same rack, and has to use a server on the adjacent rack. As we can see, the bandwidth of the links on the top of the tree is only used in the worst cases (i.e. when the request is large or the data center is operating near saturation).

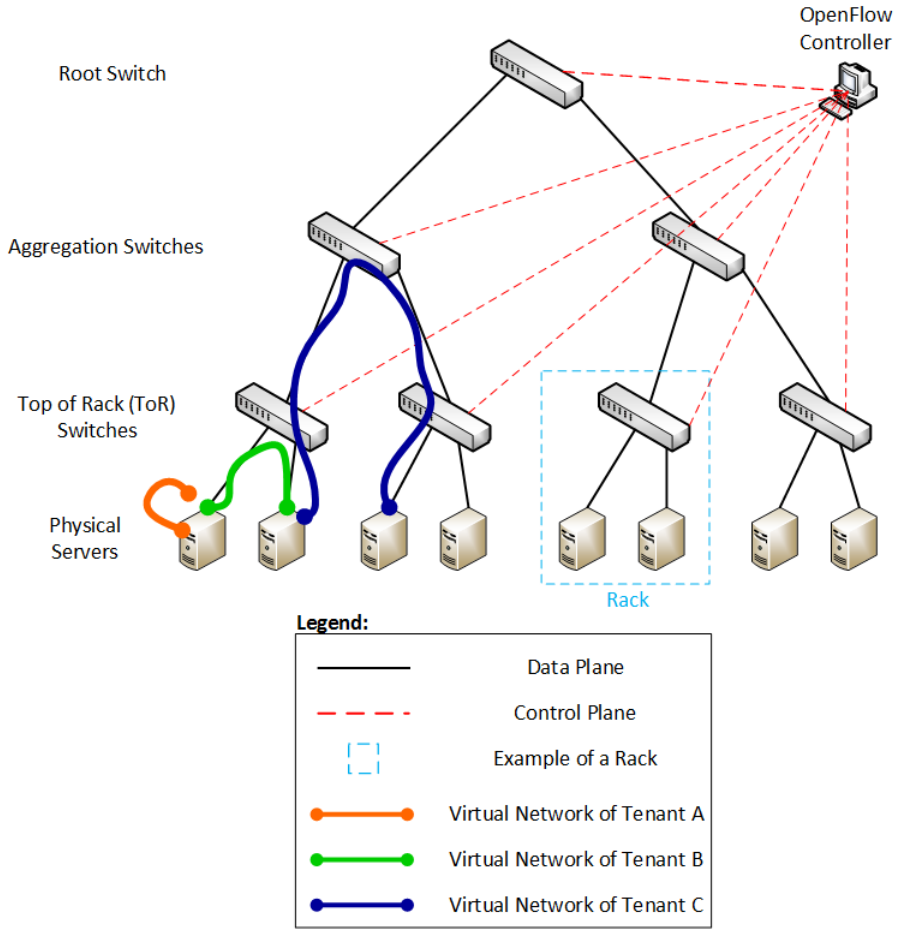



Figure 3.1: High level architecture of the solution with a tree topology.

Besides the network embedding algorithm, that ensures the network can provide the bandwidth

guarantees requested by each tenant, we make use of the centralized information in the controller to provide two properties not seen in the surveyed network embedding systems: fair bandwidth sharing (i.e. work-conservation) among tenants (non-existent in network embedding systems) and incremental consolidation of virtual network requests. The first is achieved by instructing every switch used by a virtual network (which is determined by the embedding algorithm) to create a new queue for that virtual network. As described in 2.1, the queues in OpenFlow are used to provide QoS guarantees (in this case bandwidth). This will be discussed in detail below in section 3.2.2. The second property is enforced in the network embedding algorithm itself (described in detail in the section 3.3), choosing the location of a virtual network according to a best-fit heuristic on the VM placement. To do this, the algorithm needs the current state of the network and the physical servers (which is kept by the OpenFlow controller - detailed in section 3.2.1). By managing all this information centrally, the controller could become itself the bottleneck of the network. However, as discussed in section 2.1.2, the controller does not need to be a single machine. For instance, it could be a cluster of powerful machines, or we could have several controllers controlling only a subset of the network, as proposed by [40].

## 3.1 Detailed Architecture

We will now describe in greater detail the architecture shown in the previous section. Figure 3.2 is the result of ”zooming in” in each component of Figure 3.1. Thus, Figure 3.2 contains the software that will run in each component as well as the most important interactions between those components.

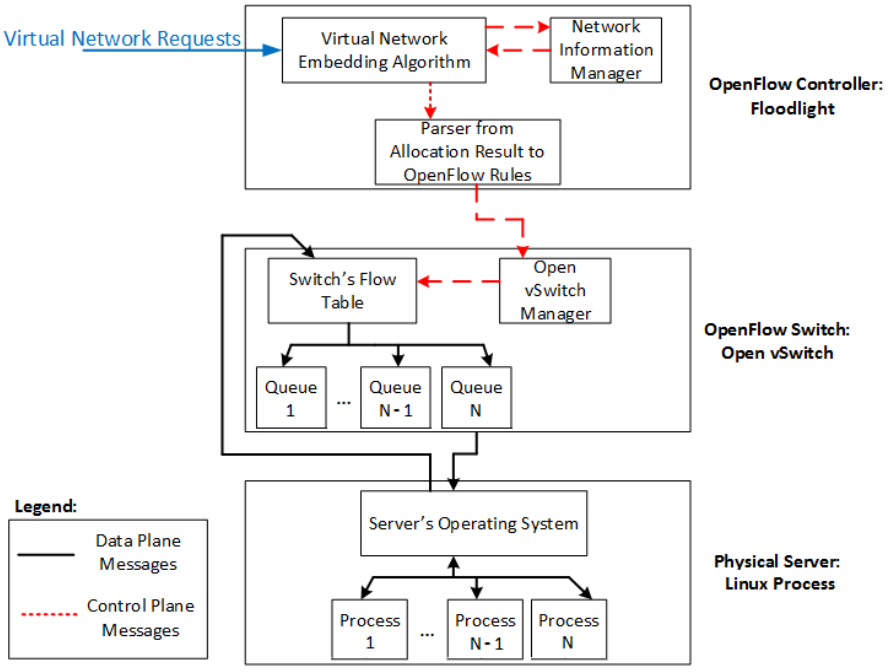



Figure 3.2: Software architecture of the components present in the solution.

First we justify our choices regarding the components shown in Figure 3.2. For the OpenFlow controller, we will use Floodlight for two reasons: first because it is based in the `Java` programming language, which as shown in chapter 2 gives a good enough peformance (i.e. a little worse than controllers that are written in the `C` programming language, but much better than the ones written in `python`, `ruby` and `javascript`); and second because it is the most robust and stable Java-based controller. We thought about using OpenDaylight but we felt that it is still immature, since we could not use it even in the simplest scenarios. Regarding the other components, Switch and Physical Server, the choices are Open vSwitch and Linux Process (respectively) because this solution will be implemented within the Mininet emulator, that uses those components. We will use Mininet both because it is open-source (which is very useful in research) and also because it is the *de facto* standard of OpenFlow emulators, as stated in section 2.1.3. It is also important to refer that all the source code that will be developed could be ported to a physical data center with few (or no) changes. The only exception is in the Linux Process, which in a real scenario would be running an hypervisor software to manage the VMs inside that host. In this work this is simplified to an operating system managing processes, where each process will simulate a VM (as detailed in section 3.2.3). Another important detail is that we assume all the physical servers have equal CPU, so that in a virtual network request a tenant asks for a percentage of a CPU (instead of a CPU with a certain frequency, as in real data centers).

We now describe the typical flow of information when the system is in operation. First of all, the tenant expresses its demands in a virtual network request (which consists in a XML file). This consists in defining two things: the number of VMs required (and also the percentage of CPU of each one) as well as the bandwidth required between the VMs that will be connected (expressed in MBit/s). This request is fed into the virtual network embedding algorithm (detailed in the next section), that is running in the OpenFlow controller. Upon receiving the request, the embedding algorithm contacts the network information manager to get the current state of the network. Based on this state, the algorithm determines (if the request is accepted) where this virtual network will be allocated. It is important to note that as all the requests are processed by the controller, this updated view of the network involves zero control messages over the network (both to switches and hosts), since the controller just has to update this information when it processes a new virtual network request. This update, and all the data structures that are involved in it, are detailed in section 3.2.1.

The controller then translates the resulting decision of the algorithm (i.e. the affected switches and hosts) to OpenFlow rule(s) to reprogram the switch(es). Upon receiving this message, the Open vSwitch manager takes two actions: creates a new queue for this virtual network, with the assured bandwidth present in the received message; and installs a new rule in the switch's flow table to forward packets

from a certain VM to the newly created queue. This means that there will be one queue for each pair of linked VMs in a virtual network. In this way, a VM can use more bandwidth than its minimum when the link is not throttled.

Moreover, the sharing of bandwidth between queues is made fairly according to the minimum bandwidth a queue has (a queue with a higher minimum bandwidth will use more spare bandwidth). Thus, the resource usage is maximized, since the tenants share unused bandwidth fairly, but at the same time get their minimum bandwidth guarantee when the network is saturated. If the network is saturated, packets may be dropped at the switch if a virtual network is generating more traffic than what it asked for. This ensures performance isolation at the network level. As the Open vSwitch is based on the Linux kernel, we will use the traffic control utility HTB (Hierarchical Token Bucket) to make the configuration of the above mentioned queues. This will be explained thoroughly in section 3.2.2.

As already mentioned, the VMs will represented by and implemented as Linux processes. To simulate tenants' workloads, each process will be running a traffic generator. As depicted in Figure 3.2, upon the necessary configurations, each process (i.e. VM) can communicate with other processes on the same virtual network, using either the operating system (in case the processes are on the same server), or contacting its adjacent switch, which will use the flow table to check to which queue it should forward this solicitation (in case the processes are on different servers). In this work we will only focus on communication inter-server (as the intra-server communication would be a responsability of the hypervisor in a real deployment).

## 3.2 System Components

Given the detailed architecture described in the last section, we now focus on each component of our solution. As we can see in Figure 3.2, our system has three main components: the OpenFlow controller; the OpenFlow switch; and the physical server. In the following subsections, the funcitioning of each component is defined in greater detail.

### 3.2.1 OpenFlow Controller - Floodlight

As stated above, the OpenFlow controller is the most important component of our solution. It is in the controller that resides the all the ”intelligence”, since it will run the algorithm that allocates virtual networks onto the physical network. Thus, we begin by showing the virtual network embedding algorithm, describing its rationale along the way. Then, we make a descripition of the important data structures on which the algorithm is based on, and we also make an analysis of some trade-offs inherent to some data

structures. We then finalize this sub-section by specifying an important design decision of our solution: how a virtual flow (i.e. between two VMs) is defined and how it is characterized at the network level.

#### 3.2.1.1 Virtual Network Embedding Algorithm

We now show the pseudo-code of the virtual network embedding algorithm that will be running on the OpenFlow controller. The main guidelines of the algorithm were already explained, but now we explain it in greater detail and formalism.

**Algorithm 1** Virtual Network Embedding Algorithm

```
Input: VNR (Virtual Network Request)
Output: True if request is accepted, False otherwise.
 1: totalVMLoad ← getVMLoadFromVNR(VNR)
 2: highestCPUAvailable ← getMostFreeCPU()
 3: if totalVMLoad < highestCPUAvailable then
 4:     appropriateList ← findAppropriateList(totalVMLoad)
 5:     for each server in appropriateList do
 6:         serverCPUAvailable ← getCPUAvailable(server)
 7:         if totalVMLoad < serverCPUAvailable then
 8:             allocVirtualNetwork(VNR, server)
 9:             return True
10: else
11:     server ← getMostFreeServer()
12:     firstServer ← server
13:     sortedVNR ← sortVNRByBWDemands(VNR)
14:     [preAllocatedVMs, remainingVMs] ← splitRequest(sortedVNR, highestCPUAvailable)
15:     VMsAlreadyAllocated ← (preAllocatedVMs, server)
16:     preAlloc(preAllocatedVMs, server)
17:     while True do
18:         server ← getNextServer(server)
19:         if server is firstServer then
20:             cancelAllPreAllocs()
21:             return False
22:         CPUAvailable ← getCPUAvailable(server)
23:         [preAllocatedVMs, remainingVMs] ← splitRequest(sortedVNR, CPUAvailable)
24:         BWDemands ← calcSumOfDemands(sortedVNR, VMsAlreadyAllocated, preAllocatedVMs, server)
25:         linksResidualBW ← calcResidualBW(VMsAlreadyAllocated, server)
26:         if BWDemands > linksResidualBW then
27:             clearLastSplitRequest()
28:             continue
29:         else
30:             preAlloc(preAllocatedVMs, server, BWDemands)
31:             VMsAlreadyAllocated ← VMsAlreadyAllocated + (preAllocatedVMs, server)
32:             if remainingVMs is empty then
33:                 allocAllPreAllocs()
34:                 return True
```

As stated earlier, the virtual network embedding has two objectives to fulfil: to guarantee that every VM pair in the virtual network request gets (at least) the requested bandwidth; and also to consolidate the virtual networks on the least amount of physical resources possible. Instead of making a consolidation algorithm that runs periodically, we have opted to take a new approach and do the consolidation incrementally (at each request), merging the network embedding and the consolidation algorithms into one. In this way, we avoid heavy migrations of VMs between servers, which would stop temporarily the work of those VMs and also possibly generate a lot of control and data traffic. Another advantage is that we relieve the controller from running this algorithm periodically, which would diminish its performance when processing new virtual network requests.

To explain the rationale behind the algorithm, we will divide it in two cases: when the virtual network request fits in on physical server and when it needs to be spread across multiple servers. This is the

first check made, comparing the total CPU load requested with the highest CPU available at the moment (line 3).

If the request fits in one server, we want to find the server with the least free CPU that fits the request (i.e. best-fit). After finding that server, we allocate this request on it (which includes updating the network state with this new request). To find best-fit server, we first find in which set we will search in (line 4). We will maintain 10 sets: the first keeps the servers with 0 to 10 % of CPU free, the second the servers with 10 to 20 % of CPU free, and so on. This is just to reduce the search space for the appropriate physical server, which in a large data center environment can reduce the run time of the algorithm considerably.

If the request does not fit in one server, we sort the request by decreasing bandwidth (line 13), and allocate as much VMs as possible in the most free server on the entire data center. In this way, the most consuming demands are on the same physical server, which significantly relieves the load on the network (and thus we can accept more virtual network requests). After pre-allocating (since the request can be rejected) what is possible on the most free server, we try to allocate the rest on adjacent servers. In line 18, the getNextServer(*server*) function returns the servers on the same rack of *server*, then the servers on an adjacent rack, and so on. Next we check if we already tried on every server of the data center (line 19), and reject the request if so (cancelling all the pre-allocations).

In the next lines (22-25), we see if the server we are checking has connection(s) with sufficient bandwidth (as defined in the request) to the other server(s) already pre-allocated in previous iteration(s). If it does not have enough bandwidth, we try on the next server (lines 26-28). If it does, we pre-allocate the VMs mapped onto this server. Finally, we check if the set remaining VMs to be allocated is empty (lines 32-34): if it is we allocate all the pre-allocations (i.e. commit) and return True; if it is not, we move on to the next server to allocate the remaining ones.

As we can see by the description of the algorithm, we will have high consolidation (and consequently low fragmentation) of VMs within the servers, since the algorithm either allocates a whole server (if the request does not fit one server), or finds the best-fit server (if the request fits in one server).

#### 3.2.1.2 Relevant Data Structures

Being the algorithm thoroughly described in the preceding paragraphs, it is now necessary to describe and analyze the data structures that support it. We will divide the data structures in two groups: the ones that are global and are always present; and the ones that are local and only exist when a virtual network request is being processed. We think that is very important to do this disction, because the data structures from the first group are global and thus will grow both with the size of the data center and also with number of requests, whereas the data structures from the second group have a limited scope,

which means that once a virtual network request is processed, they will be garbage collected and will not grow indefinitely. Of course it is important to treat all the data structures in the best way possible and to implement them according to all best pratices, however we think that the global ones should be addressed right from the design phase, since this way some bottlenecks can be avoided.

**Global Data Structures:**

- *freeCPUOnServer* - This data structure is a `Map`, where the key is a physical server and the value is the percentage of free CPU on that server. This `Map` is a core data structure of our algorithm, since it will define if a certain VM (or group of VMs) can be allocated in a certain server.

- *N$^{th}$CPUInterval* - This item is actually a group of data structures, rather than just one. As previously described (and shown in line 4 of Algorithm 1), we will keep 10 sets, where the first set contains the servers that have between 0 and 10 percent of CPU available, the second set contains the ones that have between 10 and 20 percent of available CPU, and so on. So, in this case, *N* goes from 1 to 10. These data structures are `Sets`, since duplicated values can not exist (both *intra* and *inter* `Sets`).

  These `Sets` are important because the `Map` described in the previous bullet point will have as many keys as physical servers in the Data Center. As in some real-world Data Centers there are up to 80000 servers[1], the queries on this `Map` have to be well thought. We do not want to sweep all keys of the `Map`, since it can seriously affect the performance of the OpenFlow controller.

  Thus, these `Sets` are important and can improve performance in the two scenarios of the algorithm: when a virtual network request fits within one server; and when a virtual network request has to be splitted among different servers. In the first case, we want to find the best-fit server to our request, i.e. we want the server which has the lowest $\delta$ to the CPU sum of the request. So, instead of searching all the servers for this ideal server, we will only look for the hosts that are in the `Set` that corresponds to the CPU sum of the request. In the second case, we want to find the server that has the most free CPU, so that we allocate as much as we can on it, and then try to allocate the rest on its neighbors. In this case, the `Sets` help again because they refrain us from searching across all the physical servers present on that `Map`. Instead, we search for servers in each `Set` in descendant order. We first try on the 90% to 100% `Set`, if it is empty on the next `Set`, and so on.

  It is important to refer that there is a trade-off in the utilization of these `Sets`. One can diminish the size of each `Set`, making the search in each one faster since it would have fewer elements. However, this would come at the cost of extra memory usage, since the controller would be running

[1] `http://www.bloomberg.com/news/2014-11-14/5-numbers-that-illustrate-the-mind-bending-size-of-amazon-s-cloud.html` (Last Access: 01-08-2015)

more data structures. It may not be easy to find the sweet spot, but this is something that would be worth investigating in a real-world deployment.

- *highestCPUAvailable* - This data structure is the simplest one of our solution, since it is only a `Double`. It has to be a `Double` to allow the tenants to require fractional percentages of CPU. Although simple, this `Double` is important because it is what allows us to decide if we will allocate a certain request in one or more servers. As described in the previous bullet point, this is calculated using the `Sets` and the textitfreeCPUOnServer `Map`. As one would expect, this value has to be computed before each virtual network request is processed, so that the *highestCPUAvailable* is always accurate.

- *linkBandwidth* - As the first data structure, this is also a `Map`. It maps Links (keys) to the available bandwidth in each link (values). As in the textitfreeCPUOnServer `Map`, this data structure could become very big, since it will have as many keys as physical links in the Data Center. However, we will never need to loop through all the keys in this `Map`. The problem of virtual network embedding has a two-dimensional search space: servers (CPU) and links (bandwidth). Once we have fixed the first server (which was the one with the most available CPU), we try to allocated the remaining VMs in its neighbors. Thus, the routing engine of the OpenFlow controller will give us the path(s) between every server affected by a certain request, and we will only query this `Map` directly with a key (i.e. a link), so that we can check the available bandwidth of each link.

**Local Data Structures:**

- *virtualNetworkCPUs* - This data structure is a `Map`, and it is filled with the data present on the XML file that the tenant supplies. It has the name of each VM as key (which is used to identify each VM in the virtual network) and the required CPU as value. It is using this data structure that we know if a virtual network request fits in one server or not, by summing the CPU demands of that request. This data structure (as well as the two below) is recycled on each virtual network request.

- *virtualNetworkLinks* - As the previous data structure, this one is filled with the data present on the XML file that the tenant supplies. It is also a `Map`, where the key is a tuple formed by two VM names, being the first one the origin of the link (*from*) and the second one the destination (*to*). The value is the required bandwidth for this link, expressed in Mbit/s. It is not required that every VM pair has a connection between them (and the ones that do not have will not be able to communicate).

- *virtualMachineToServer* - Opposed to the last two data structures, this one isn't filled with information from the XML file of the tenant. Instead, this `Map` is populated while the virtual network

request is being processed. It has the VM name as key and a physical server name as value. This translation (from virtual host to physical host) is necessary to be able to check the bandwidth of the links that connect servers that allocate VMs of a certain virtual network. In the line 25 of Algorithm 1, we calculate the residual bandwidth between the ”current server” (the one we are trying to use at the moment) and the servers of all the VMs that were allocated in previous iterations. Thus, this data structures makes that computation possible.

#### 3.2.1.3 Definition of a Virtual Flow

With the virtual network embedding algorithm and its data structures already defined, the OpenFlow controller is almost completely described. However, there is still one important piece missing: given that the algorithm allocates VMs in the appropriate servers and guarantees the required bandwidth between those servers, everything is in place so that the VMs start to generate traffic. But, how is the traffic generated from a group of VMs routed to each other and enforced to behave as the virtual network request described?

The routing answer is simple: the OpenFlow rules that the controller installed in each switch will route the traffic accordingly. So, the problem is: in the network elements (the OpenFlow switches) onto what the traffic generated by the VMs will match against?

In this project, we use version 1.3 of the OpenFlow protocol, and as such we can match on a great variety of fields. It can go from the ingress port (of the current packet), to its either source or destination MAC address, VLAN identifier, MPLS label, IP address, TOS byte, ICMP type, TCP port and others. After analyzing all options, we identified some limitations that are common to some options:

- **Not scalable to Data Center size:** There are certain options which would limit considerably the number of communicating VM pairs we could have. For instance, if we used the TOS byte to identify each pair of communicating VMs, we could only have 256 pairs of VMs (maybe even fewer since two bits of the TOS byte are reserved). The same happens (but with a larger limit), if we used VLAN identifier as our solution. We would only be able to have 4096 pairs, which is definitely not acceptable to a real data center.

- **Not easy to deploy:** There are some options that are not easy to deploy, since we would need to force each VM to generate traffic with certain characteristics. For instance, if we used a MPLS label as our identifier, we would need to force each VM (or the hypervisor on that machine), to inject the appropriate MPLS label for each VM, so that the traffic carries it and is routed as expected by the adjacent OpenFlow switch. This is doable, but would insert a lot of overhead into our solution, for little or no gain.

With these limitations in mind, we have decided to use IP addresses as the matcher of traffic in the OpenFlow switch. This works as follows: when the OpenFlow controller starts, it initializes a variable *id* with the value 1. Then, when the controller is processing virtual network requests, this *id* is assigned to the first VM that needs to communicate with other VM that is not in the same server. Then, *id* is incremented by one and the process repeats for every VM that has inter-server communication. This happens in line 30 of Algorithm 1. Then, given that a request was successfully processed, in the line 33 of Algorithm 1, the *id* assigned to each VM is transformed into an IP address (e.g. the *id* `1` would turn into `10.0.0.1`) and OpenFlow rules are created for the appropriate switches, and the traffic is matched according to the tuple `<source IP address, destination IP address>`. The IP assigned to each VM will be unique, and thus we can match the traffic from a certain VM and forward it to the appropriate queue (with the required bandwidth).

In case one would argue the $2^{32}$ IP address space is a concern, we can always change this to use IPv6, which will scale well beyond our needs. We do not want to use the tuple `<IP, port>` because that way we would be controlling the port on which the tenant's applications run, which can be infeasible (e.g. the tenant could be running an application that has to run in a pre-determined port).

Regarding the configuration itself of the IP addresses the controller generates onto the NICs of the VMs, it is out of the scope of the current project to do it automatically from the controller. One interesting solution would be to integrate the OpenFlow controller (in this case Floodlight) with the OpenStack or similar software, so that the controller can automatically configure the IP address that it generates for each VM. Regarding the current project, this is done after the processing of a virtual network, using Mininet `python` scripts to do it semi-automatically.

### 3.2.2 OpenFlow Switch - Open vSwitch

As described in section 3.1, we use Open vSwitch as our OpenFlow switch. Open vSwitch is an important part of our solution, since it is responsible for providing the bandwidth guarantees as well as acting as a work-conserving system, enabling the share of spare bandwidth between tenants. There is certanly much research work to be done on switch and QoS schedulers, but given the needs of our solution regarding this, we think that it is not necessary to ”reinvent the wheel”, and as such we rely on Open vSwitch for our bandwidth guarantees and sharing.

As outlined in section 2.1, the OpenFlow protocol only specifies that the way OpenFlow switches provide QoS is through the *enqueue* action, and then delegate the responsibility of how it is done to the OpenFlow switch manufacturers. As this is such an important piece to our system, we think it is necessary to provide some context and detail on how Open vSwitch implements QoS, so that we know

what we are building on top of.

Since Open vSwitch is based on the Linux Kernel, it uses the *tc* (*traffic control*) utility as the switch's packet scheduler. *tc* operates between the IP layer and the network driver that transmits data onto the network. It is permanently active, even if no QoS options are selected. In this case, the switch uses *tc* with a basic FIFO queue. At its core, *tc* is composed by *queueing disciplines*, or *qdiscs*. The *qdiscs* are the scheduling policies applied to a certain queue. There are several types of *queueing disciplines*, such as FIFO (First In First Out), SFQ (Stochastic Fairness Queueing), TBF (Token Bucket Filter) and others. In fact, these are different scheduling algorithms that can make the queues act as we want.

Particularly, TBF was extended to an hierarchical mode called HTB - Hierarchical Token Bucket. HTB is what Open vSwitch actually uses to configure the OpenFlow queues, so we will focus on this *queueing discipline*.

As the name entails, it is based on the Token Bucket algorithm. When controlling bandwidth, it is not easy to find an efficient accounting method, since counting in memory is extremely difficulty and costly to do in real-time. Thus, instead of counting the packets (or the bits) transmitted, in this approach each queue has a bucket and each bucket receives a bucket at a regular interval. When a packet is going to be transmitted on a certain queue, it will check if that queue's bucket has enough tokens to send the current packet. If it has, the packet is sent. Otherwise, the packet waits for more tokens to be delivered on that bucket. The frequency at which the tokens are delivered to a bucket determines the rate (or transmission speed) of that bucket.

In HTB classes are linked together as a tree, with a root and leaves. Each class can have an Assured Rate (which is then translated to a bucket, which was described in the previous paragraph), as well as a Ceil Rate (along with other details). HTB also allows a class to borrow Rate from its parent, given that its parent has some spare tokens. This is what allows our system to be work-conserving.

Going back to Open vSwitch and the OpenFlow queues, given this description of how HTB queues work, here is how we intend to use them in our solution: first of all, we will create an HTB tree with a depth of 1, so that it only has the root and then all the other queues at the same level. This will allow all the switch's queues to borrow bandwidth from each other, with the queues that requested an higher bandwidth borrowing more (this make sense, assuming that the customers that requested more bandwidth are the ones that are paying more, and thus can borrow more and the Data Center is not saturated). Then, each queue is configured with an Assured Rate (that comes from the OpenFlow controller), and not with a Ceil Rate. This way, if a tenant for a certain period in time has no other tenants generating traffic, can use the full link capacity for himself. This is definitely arguable, and in a real-world deployment this would certanly follow some business rules and this traffic would end up

throttled. However, for our Proof of Concept we do not see any advantage in doing so.

As a final note on the OpenFlow switch, it is important to refer that during the development of this project, we found out that there is a limit on the number of queues per port on Open vSwitch. The limit is currently 65536 queues per port. This is not outrageous, since it is per port (and not per switch), but in a real deployment, depending on the Data Center size and number of tenants, this could become an issue. However, upon further discussion with some developers of Open vSwitch, we were informed that this limitation is due to the way Open vSwitch's implementation is using *qdiscs*, since they have a 32-bit identifier but Open vSwitch currently is only able to use 16-bit (which yields 65536). So, we think that this is not a real limitation, since it is not tied to a real bottleneck, and is something that is ought to change in the future.

### 3.2.3 Physical Server - Linux Process

Since we use Mininet to run our experiments, the physical servers in our Data Center are emulated as Linux processes. In a real-world scenario, each server would be running an hypervisor software, that manages the physical resources to the VMs that were allocated on that server.

Since in our work each server is emulated as a Linux process, each VM of the Data Center is also emulated as a child process of the server's process. This is done using the `&` operator in the Mininet's hosts, which creates a new child process that runs in the background. As long as each "VM" (or child process) is running in a different port, we can emulate several VMs running at the same time on a physical server, by running all the child processes at the same time as well.

It is important to note that this way we can mimic the functionality of physical servers and VMs, but this does not provide any guaranteee regarding performance isolation between child processes, as the hypervisor would do between real VMs. However, as described in one of Mininet's papers [25], the fidelity of an experiment can be maintained by monitoring the usage of resources (such as CPU and memory), and so this lack of isolation between child processes can be compensated by monitoring the resource usage of the base system.

## 3.3 Summary and Final Remarks

In this chapter we have presented our solution. We began by showing a common use case for our system, then described its architecture in detail, and we finally described each system component and how they function internally.

We have developed an algorithm that achieves high consolidation in the placement of virtual net-

works (incrementally), high resource utilization, as well as providing bandwidth guarantees in a work-conserving system. This algorithm is also tailored to be scalable to current Data Center networks, as depicted in section 3.2.1.

We intend to prove the claims made throughout this chapter in the Evaluation section, showing that we achieve the desired properties through emulation runs.

# Chapter 4

# Implementation

In this chapter we specify how we implemented the solution that was depicted in the previous chapter. This means that we need to translate the generic algorithm and data structures presented in the previous chapter into an actual implemenetation. As it was briefly pointed out in the previous chapter, we used Floodlight as our OpenFlow controller. Thus, in section 4.1 we describe the main part of our implementation, which consists in the modifications and extensions that we have made to the open-source Floodlight controller. Then, in section 4.2, we described a not so critical but also important part of our implementation, which is the creation network topologies using Mininet. This is what enables us to use and test the controller, since it is Mininet that creates a network of links and switches to connect to the controller. We then conclude this chapter by highlighting some key aspects of our implementation and also giving some concluding remarks about it.

## 4.1 OpenFlow controller - Floodlight

Since the OpenFlow controller is the centerpiece of our solution, naturally it is also the most important part of our implementation. As stated in section 2.1.2, the Floodlight controller is written in the Java[1] programming language. The modular architecture of the Floodlight controller is shown in Figure 4.1. The original architecture[2] is composed by the modules in blue and orange. The other modules are described later in this section.

The Floodlight controller is based on an event driven architecture, meaning that modules must subscribe to the messages they want to receive, i.e. in order for a module to receive an OpenFlow PacketIn message, the module must subscribe for this type of messages. When the controller receives an Open-

[1] `https://www.java.com/en/` (Last Access: 01-10-2015)
[2] `https://floodlight.atlassian.net/wiki/display/floodlightcontroller/Architecture` (Last Access: 10-10-2015)

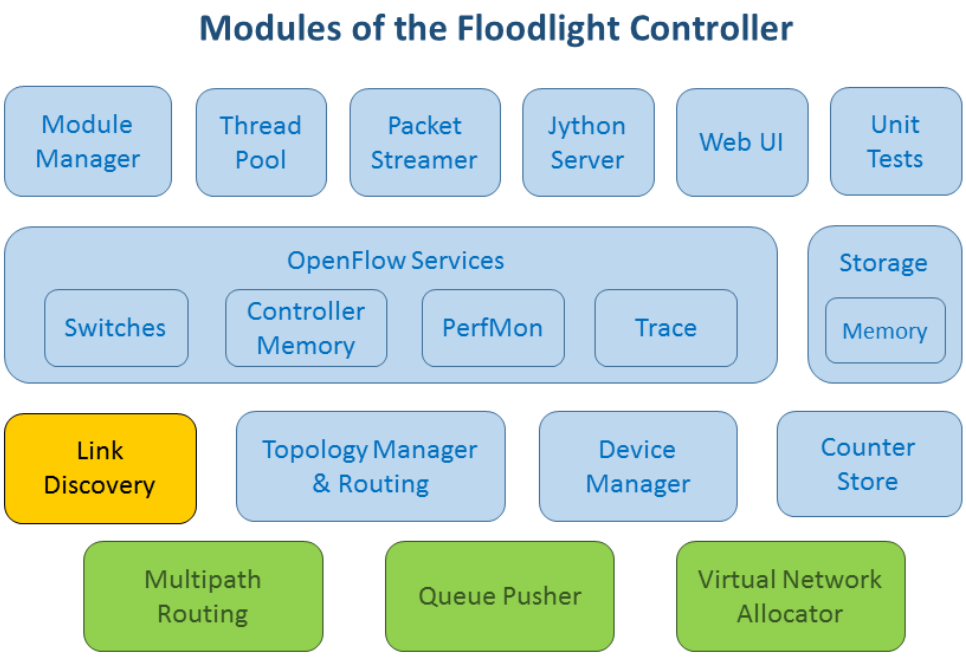



Figure 4.1: Modular software architecture of the Floodlight controller, and our extensions to it.

Flow message from a switch it will dispatch that message to all modules that have subscribed for that specific message type.

If the controller receives multiple messages from one or more switches, these messages are enqueued for dispatching because the controller only supports dispatching one message at a time. This means that the performance of the controller is dependent on the time it takes to process a message in each individual module, as the processing time of a single message is equal to the sum of all the processing time done in each individual module.

In order to add a new module to the Floodlight controller, one must create directly in the code base of the controller a new Java package. In order for a module to start up it must be added to a configuration file that defines which modules are launched when the controller is started. This is done by adding the module's path to the `floodlightdefault.properties` file. When the controller starts, it will start the different modules, and set references between them according to the inter-module dependencies. These dependencies are defined by implementing the appropriate methods (according to which dependencies one wants to set) of the `IFloodlightProviderService` interface.

As already said above, Figure 4.1 shows Floodlight's modular architecture. In this Figure, we can see modules that are displayed with three different colors. The modules in blue are the original Floodlight's modules that we simply use and did not modify in our implementation. The module in orange (Link Discovery) is also one of the original modules of the Floodlight controller, but this one was modified, since we needed a different behavior than what was originally implemented in the controller. Finally,

we have the green modules, which are the modules that we developed from scratch and weren't part of the original Floodlight architecture (which are the Multipath Routing, Queue Pusher and Virtual Network Allocator modules). We will now dive into the details of each of these four modules (the orange and the three green ones), specifying how they were built and what is their relevance in our solution.

### 4.1.1 Link Discovery Module

As already pointed out (and visible in line 25 of Algorithm 1), the controller needs to place the VMs of a request in physical servers that have links with enough bandwidth to accomodate what is requested in the XML file. Thus, each link has to know how much free bandwidth it possesses. However, in Floodlight's original implementation, the Link object did not have such information.

Hence, it was necessary to extend the Link Discovery Module, so that each link knows how much spare bandwidth it has. This consisted in adding a new field to the Link class, and defining the appropriate methods (*getters* and *setters*) to configure this new field.

It is important to note that the link's bandwidth isn't set automatically as one would expect (i.e. through the LLDP packets this Module sends), because Mininet emulates the all the virtual links with a bandwidth of 10 Gb/s, even if we configure it to have a lower bandwidth. We had to work around this, and we did it by initializing all links bandwidth when the controller starts, according to what is defined in a start-up XML file. In order to ease the process of running our controller, this XML file is automatically generated when the Mininet topology is created, according to the parameters in the Mininet script. This will be explained in greater detail later, in section 4.2.

### 4.1.2 Multipath Routing Module

This module was not mandatory for our solution to work, since Floodlight already has a module, Topology Manager and Routing, that computes the route between two endpoints. However, this is something that we have discovered during the implementation phase that we think was worth doing right, since this module could generate a higher number of accepted virtual network requests.

Floodlight's original routing module provides a Java API to use, from which one of the methods is `getRoute`. It calculates the route between two endpoints by applying the Dijkstra's algorithm [42] to the graph that contains the network topology. This means that it always gives the shortest path between two nodes in the graph, which for most applications is what makes sense, but not for ours. Since we want to maximize the number of allocated virtual networks by the controller, we want to test every possible route between two endpoints. The shortest path between two endpoins may not have enough bandwidth to accomodate a certain request, and a longer path may have that required bandwidth. Not using only the

shortest path can turn many otherwise rejected requests into accepted ones.

Now that the rationale behind this module is expressed, we will now explain how this module was built, and also its inner workings.

We began by studying how the original Floodlight's Routing Module works. This proved to be a more complicated task than initially expected, this module's code-base is quite large, and getting the logic from undocumented code is sometimes very difficult. After this study phase, we have implemented this module as follows: the module registers itself as a receiver for events of the type `topologyChanged`. By receiving these events, the module builds a graph, adding and removing links or hosts as the events dictate. This graph represents the network topology.

Having this graph, one must only apply a search algorithm on top of it to find the paths between two nodes. This is where our module and Floodlight's one differ, since instead of using the Dijkstra's algorithm, we use a Depth-First Search algorithm [43] to compute all possible paths between a pair of nodes.

This way, our module also provides a Java API to other modules, which only has the `getRoutes` method, which returns a `Map` of Routes between the two endpoints passed as parameters.

### 4.1.3 Queue Pusher Module

This module is responsible for providing an API to create queues in Open vSwitches. Queues in Open vSwitch are created using the OVSDB[3] protocol. So, we started out by studying how the OVSDB protocol works and how we can use it to create queues in Open vSwitchs right from the controller. In Figure 4.2 we can see the OVSDB schema, which is a set of tables that represent the Open vSwitch instances in a system.

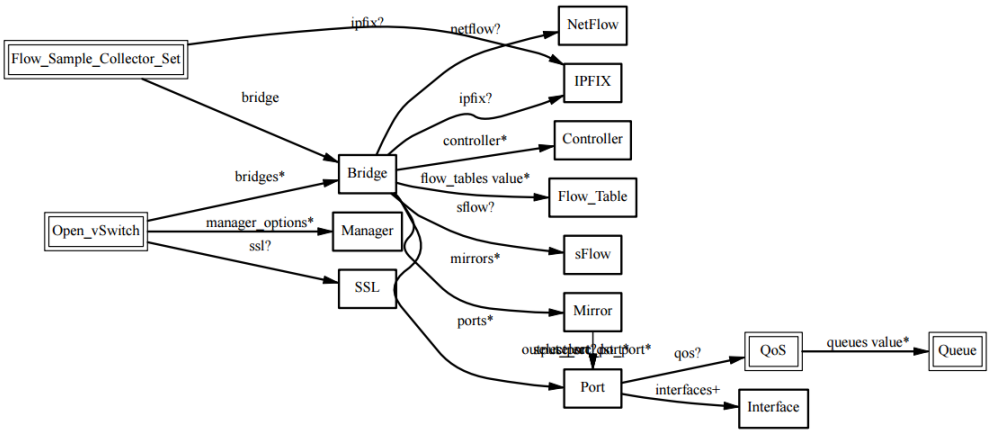



Figure 4.2: Table schema of the OVSDB protocol.

[3] `https://tools.ietf.org/html/rfc7047` (Last Access: 01-10-2015)

Looking at Figure 4.2, we can see that the Queue table is a child of the QoS table, and the QoS is in turn child of the Port table. Looking at this, we understand that for creating a Queue, one must first create a new entry in the QoS table (to which the Queue will be linked to), and this entry in the QoS table has to be linked to some entry in the Port table. The port creation is not a problem, since it is done automatically when creating a new network topology. Knowing this, we have implemented this module to create Queues as described in section 3.2.2, with only the Assured Rate configured (called `min-rate` in the OVSDB command) and no Ceil Rate (or `max-rate`) configured.

Given this information, this module is pretty straightforward. It only has to use the `ovs-vsctl` utility that comes pre-installed with the Open vSwitch, to create a new QoS entry and a new Queue below that QoS entry for each Queue the controller wants to create. There is only one noteworthy aspect left about this module: when creating Queues, each one gets assigned an identifier, which has to be unique per switch. This is important because the traffic will be directed to the Queue by matching this identifier, since the *enqueue* action receives it as argument. Thus, if the uniqueness of these identifiers is not kept (again, per switch), one may experience unexpected behavior.

### 4.1.4 Virtual Network Allocator Module

This is the main module of our solution. Not particularly because of its complexity, but rather because this is the module that actually fulfills our goal to process and allocate virtual networks onto the substrate network. At the top-level, this module runs an infinite loop, waiting for virtual network requests to be submitted. It processes them serially, which means that each request only gets processed if this module is in idle. Since this is merely a Proof of Concept and not a real-time system, the queueing mechanism to put requests on hold while the module is processing another request was not implemented. In our Proof of Concept, the requests to process are already present in a certain folder, and the module processes them one by one, until there are not more requests (or the module starts to return `False` upon processing virtual network requests, meaning the physical resources are full and it can not accomodate more requests).

While on its infinite loop, this module is waiting for XML files. Figure 4.3 shows an example of a virtual network request submitted to this module.

In this case, a tenant is requesting two VMs, one with 3.4% of CPU and the other with 4.2% of CPU. Then, he also requests a connection between these two VMs with a guaranteed bandwidth of 7 Mbit/s. Upon receiving this file, this module parses the XML file to fill the *local* data structures described in section 3.2.1.2. Then, as already pointed out, the algorithm is splitted into two cases:

– **When the request fits in one server,** i.e. the sum of CPUs requested can be accommodated in

```xml
<?xml version="1.0" encoding="UTF-8"?>
<virtualNetwork>
    <virtualHosts>
        <virtualHost>
            <name>vh1</name>
            <CPU>3.4</CPU>
        </virtualHost>
        <virtualHost>
            <name>vh2</name>
            <CPU>4.2</CPU>
        </virtualHost>
    </virtualHosts>

    <virtualLinks>
        <virtualLink>
            <from>vh1</from>
            <to>vh2</to>
            <bandwidth>7</bandwidth>
        </virtualLink>
    </virtualLinks>
</virtualNetwork>
```

Figure 4.3: Example of Virtual Network Request submitted to this module.

the same physical server, the processing of the request is fairly simple. In this case, this module does not need to use any of the modules described earlier in this chapter. It merely has to update the *global* data structures, defined in section 3.2.1.2, with the information from the *local* data structures, that was gathered from the XML file.

– **When the request does not fit in one server,** i.e. when this request has to be divided among two or more physical servers, this module will start by sorting the links of the virtual network request in descending order (since the tenant can provide the XML file in any order). Then, it will allocate as much VMs as possible in the server that has the most CPU available. After that, it will try to allocate the remaining VMs in the neighbors of this server. It will start from the server adjacent to this one, and it will continue this logic until there are no remaining VMs. In each iteration of going to the neighbor of the server with the most free CPU, this module is using two modules described above: the Multipath Routing Module and the Link Discovery Module. Every time it advances to an adjacent server, it uses the Multipath Routing Module to get all paths between this server and the ones that already have allocated VMs. Upon getting these paths, this module will then use the Link Discovery Module to check each link of each path, to make sure that those links have enough bandwidth to provide the guarantees required by this tenant's request. Once all the VMs have a physical server assigned (assuming a request where this happens), the module knows this request is going to be accepted. So, it is necessary to translate this request's results into real network rules. This, this module uses the Queue Pusher Module to create the required Queues on

each OpenFlow switch (creating the necessary OpenFlow rules at the same time), returning `True` after that.

As we can see by the previous paragraphs, this module is where the allocation algorithm runs, and (in some cases) uses the other modules that we have characterized above. It is also important to point out another detail: this module (as the Link Discovery Module) needs to load some information from a start-up XML file. In the case of this module, we need to get the information about the free CPU percentage on each server from that file, since this was not possible to do natively in Floodlight (because it does not connect to end hosts). This means that the *freeCPUOnServer* data structure is initialized with information from that start-up XML file, which will be described in the next section. We could simply initialize this data structure with 100% for each server (since in the initialization every server is empty), but this way our solution allows to simulate scenarios such as where some of the Data Center's servers are more powerful than others (i.e. one could normalize the CPU percentages, and start a physical server with 50% if it has half the clock of the other servers - assuming it has the same number of cores).

## 4.2 Network Topologies - Mininet

The core of our implementation was already described in the previous section, however, we think that it is important to give some details on the work we have done with Mininet. Mininet does not have a steep learning curve, and is fairly easy to start experimenting with it. However, in order to create real-world Data Center topologies, there are many options to choose from and it requires some investigation.

We have developed two `python` scripts that use Mininet's API and create two different network topologies: the first creates a Tree topology, as the one shown in Figure 3.1; whereas the second creates a Fat-Tree topology, as the one described by Al-Fares et al. [5]. The first script receives two parameters: depth and fanout. The depth specifies how many levels the tree will have (counting from the root), and the fanout details how many children each node in the Tree structure has. The Fat-Tree script receives only one parameter, named $k$, which defines how many ports each switch has. Then, in a Fat-Tree network, the number of connections to a node's father is the same that are going to that node's children (e.g. with $k = 4$, an Top of Rack switch would connect to two hosts and two aggregation switches). It would be interesting to develop new scripts that create different topologies, such as BCube[23] or DCell[22]. This would allow us test our controller (and algorithm) in different scenarios, which would be great for research purposes.

It only remains to state that both scripts generate a XML file with information about the created network topology. This is done by adding a new entry to the XML start-up file on each call to Mininet's

API, either `addLink` or `addHost`. As already described, this file is important since it provides information to our Modules that we could not get othetwise.

## 4.3 Summary and Final Remarks

Throughout this chapter we discussed *omniCluster*'s implementation. We described in detail the how we implemented the Open Flow controller, Floodlight, as well as the work we have done with Mininet.

While describing the modules we have developed, we also state which part of the architecture (portrayed in the previous chapter) defines the module being implement, and, when appropriate, we show how we translated the generic solution presented in chapter 3 to the actual implementation.

# Chapter 5

# Evaluation

In this chapter we describe and depict the evaluation we have made to the implementation described in the last chapter. This evaluation aims to prove the fulfillment of the goals that were set in chapter 1. This chapter is organized as follows: in section 5.1 we portray the environment on which we ran our experiments, indicating the hardware and software that were used as well as how the experiments were conducted. Then, in the next four sections (i.e. from section 5.2 to section 5.5), we show the results we were able to get from this evaluation, each one framed in the appropriate goal. Then, in section 5.6, we summarize the results achieved and provide some concluding remarks.

## 5.1 Evaluation Environment

### 5.1.1 Test Bed

Our experiments were run in a machine with the following hardware specifications:

- **CPU:** Intel® Quad-Core i7 870 @ 2.93 GHz
- **RAM:** 12 GB DDR3 @ 1333 MHz
- **HDD:** 450 GB Serial ATA @ 7200 rpm

The relevant software installed in this machine was:

- **OS:** Ubuntu 14.04.3 LTS (Linux Kernel 3.13.0)
- **Network Emulator:** Mininet - version 2.2.1
- **OpenFlow Switch:** Open vSwitch - version 2.3.1
- **OpenFlow Controller:** Floodlight - version 1.1

### 5.1.2 Setup

We now describe the setup that is common to all experiments, except the one made as part of Goal IV, which is described in its own section (5.5). The controller processes virtual network requests, and we stop an experiment when the controller returns `False` to an allocation, meaning it can not allocate more virtual networks. Then, each experiment is run a thousand times, in order to get meaningful results. In order to test our solution, we had to generate our own dataset. Unfortunately, since this is an emerging field, datasets of virtual network requests do not exist (at least publicly). This renders comparisons very difficult to do (and when possible not very reliable).

To generate our dataset, we have produced XML files similar to the one presented in Figure 4.3. We have produced requests with the following characteristics: in each request a VM asks for a CPU that is generated randomly (using an uniform distribution) between 0.1 and 5 %. The connections between VMs are also randomly generated (with an uniform distribution as well) between 0 and 10 Mbit/s (a connection of 0 Mbit/s is treated as a non-existent one). These parameters are similar (and in the same magnitude) to several works in this field [8, 24, 7, 54]. However, the evaluation scenario of these works differs a lot from ours (and between each other), so comparisons can only be done in certain graphs. For each size of the virtual network requests (i.e. number of VMs in it), which we defined as going from 2 to 40, we generated 10000 virtual network requests. This totalized 390000 XML files, which together make up around 10 GB in data. The experiments we show in the following sections run until the servers and/or network is satured, for each size of virtual network requests (i.e. use only virtual network requests with 2 VMs per request until we get the first rejected request, the same with 3 VMs per request, and so on).

Regarding the network topologies used, with the machine described above we were able to generate two topologies in Mininet with the following characteristics:

- A Tree topology with 125 servers, which entails 31 switches and 155 links. This consists in a Tree topology with a depth equal to 3 and a fanout equal to 5. All the links were supposed to be 1 Gb/s, however we were not able to set the bandwidth of links due to a limitation between Mininet and Open vSwitch. This only interferes with Goal IV, and as such will be detailed in section 5.5.

- A Fat-tree topology with 128 servers, which entails 160 switches and 384 links in the network. This consists in a Fat-tree with a factor $k$ equal to 32, which means switches with 32 ports. As with the previous topology, the bandwidth of links was not set.

We intended to use the MaxiNet[51] project in order to scale our experiments horizontally, creating larger and more realistic Data Center networks. However, for the configuration it requires, this was not possible to do in the time frame available for this project.

## 5.2 Goal I - Scalable to Data Center Environments

In this goal we want to assess if our solution is scalable to environment (and size) particularly found in Data Centers. To this end, we have measured the time it takes to process each virtual network request. As pointed out in section 4.1.4, our system processes the virtual network requests serially. Thus, the amount of time it takes to process each request is crucial to the operation of the entire system.

This is calculated using the method `currentTimeMillis` from Java API.

### 5.2.1 Tree Topology

In Figure 5.1, we can see the results of obtained with a Tree topology:

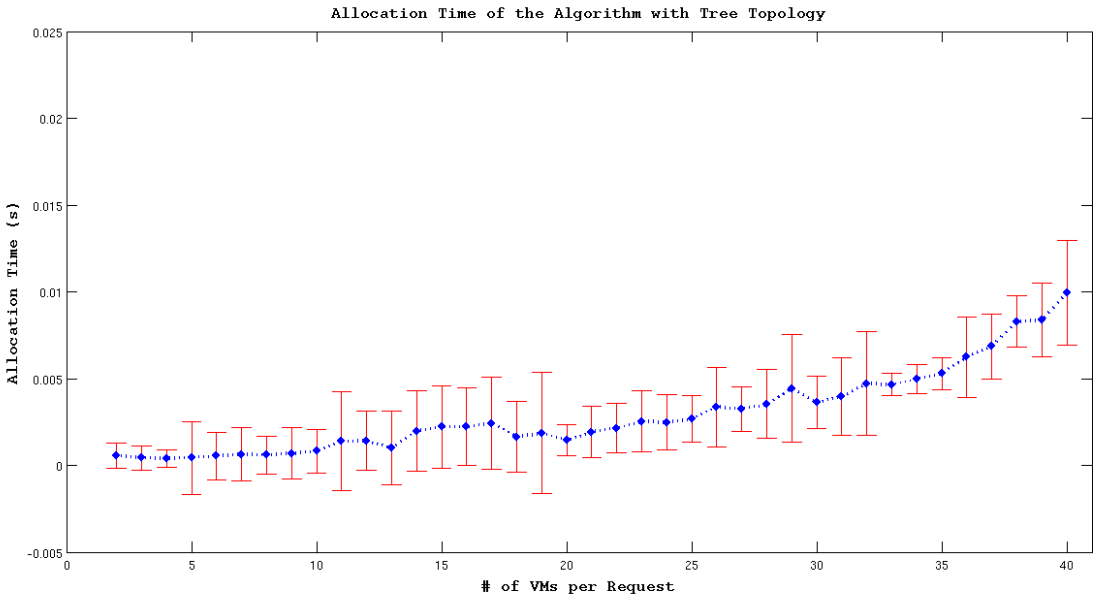



Figure 5.1: Allocation time for a virtual network request using a Tree topology.

By analyzing Figure 5.1, we can see that up to about 25 VMs in a request, we have a processing time under 5 ms, which we think are great results. As a comparison, SecondNet[24] achieves 10 ms in requests with 10 VMs, which is twice the processing time in requests with less than half the VMs. However, their setup is not completely equal to ours and there are a lot of details missing to make this comparison meaningful. We can see that our system takes about 10 ms to process requests with 40 VMs. Although the emulated network is not close to the size of a Data Center, we want to point out that a request with 40 VMs is almost one third of the number of physical servers in the network. Even in these conditions, our processing time did not grow abruptly, which we think is a good indicator of the scalability of our algorithm.

### 5.2.2 Fat-tree Topology

Regarding the results for allocation time using a Fat-tree topology, these are presented in Figure 5.2:

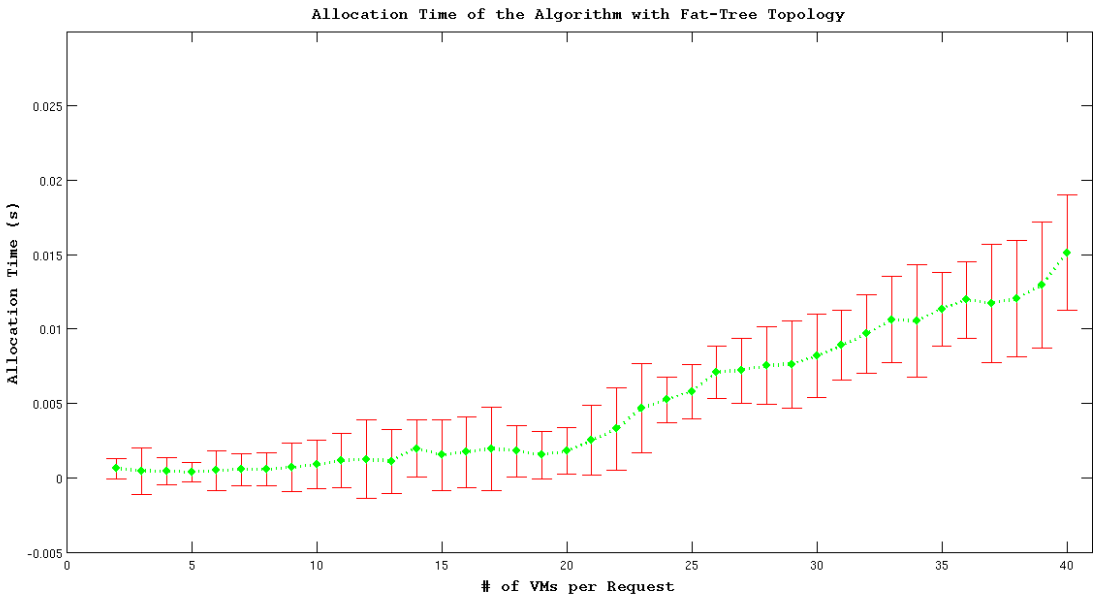



Figure 5.2: Allocation time for a virtual network request using a Fat-tree topology.

Looking at Figure 5.2, we can see that the processing time using this topology is higher than the one using a Tree topology, peaking at around 15 ms, which is 5 ms more than what we got using the Tree topology. Nevertheless, we think that the results we have are pretty exciting, since we can see that the care around the data structures described in chapter 3 paid off. As described in section 5.1, this topology has a lot more links (and switches) than the Tree topology. This means that there are more paths between two endpoints. Thus, this higher processing time can be explained by the extra work the controller had by checking more routes. This means that the extra processing time was not wasted, since with the Fat-tree topology we got a total of 15688 accepted requests, *versus* 15055 with the Tree topology. We have tried to compare these values with similar works, but the only work that computes this metric has a network topology (and size) very different from ours.

## 5.3 Goal II - High Consolidation

The evaluation around this goal aims to measure how consolidated virtual networks are, since our algorithm strives to take server locality into account, allocating the VMs of a virtual network as close as possible. This is important since a low number of hops leads to a low latency in the communication between the VMs of a virtual network. This metric is calculated by counting the numbers of physical servers there are in a virtual network allocation.

### 5.3.1 Tree Topology

In Figure 5.3 the results of using a Tree topology are depicted.

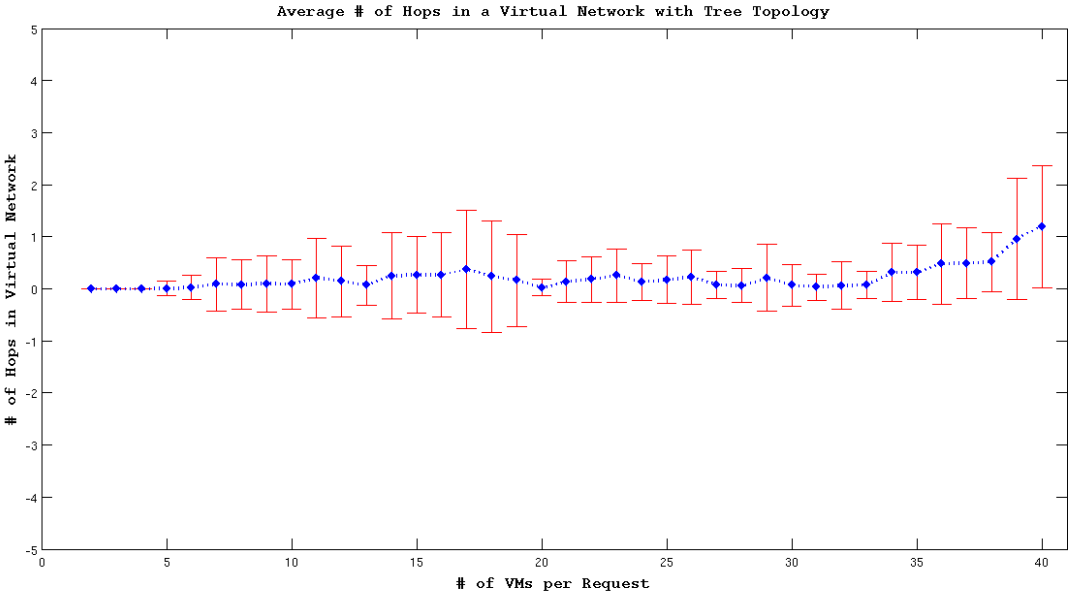



Figure 5.3: Average number of hops in a virtual network using a Tree topology.

Analyzing Figure 5.3 we can see that we achieve high consolidation of the virtual networks, since the average is almost always near zero. It starts out equal to zero up to 5 VMs per request, then the average is almost the same but the variance increases, meaning most of the virtual networks do not have any hop, but some do. It keeps this behavior along the line, with the average increasing less than linearly. On 40 VMs per request we get an average number of hops close to 1. Again, we want to point out that 40 VMs in a request is big since our data center network has 125 servers, and even in that case the virtual networks only need, on average, one hop.

### 5.3.2 Fat-tree Topology

The average number of hops using a Fat-tree topology as our network is shown in Figure 5.4.

Considering what is displayed in Figure 5.4, we can state that it exhibits a similar pattern to the one presented in the Tree topology case, and also that we achieve an high consolidation in this case as well. However, we can see that average number of hops is higher, as well as the standard deviation, meaning we have a higher number of hops using this topology. This can be explained by what has already been stated in the previous section, since we have more hops because we are processing more requests with this topology. It is worth noting that the extra requests we get with this topology are processed when the data center is near saturation (since we are stopping on the first rejected request, and with

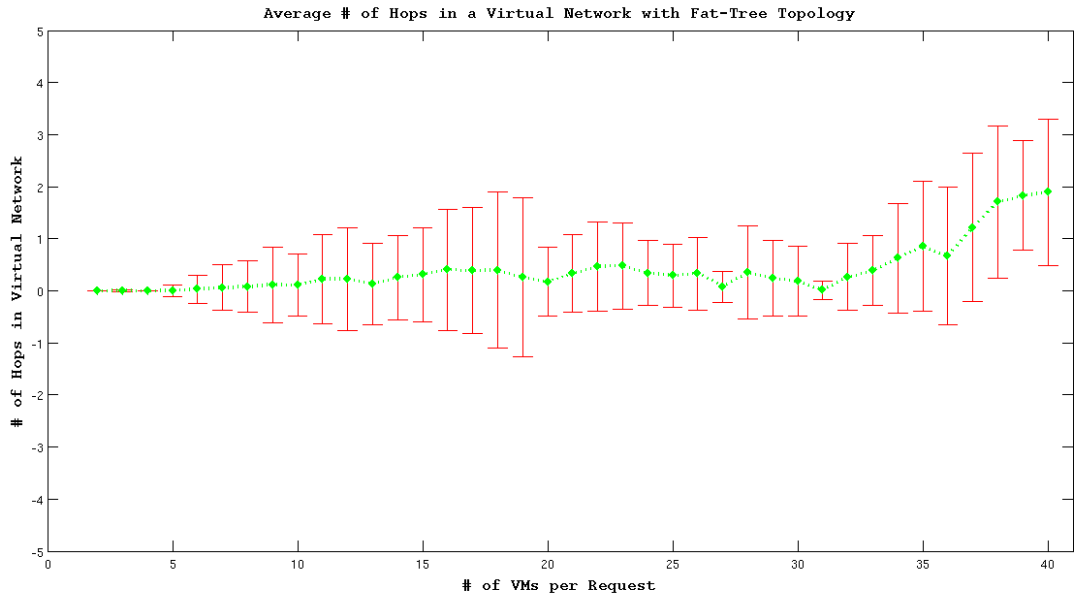



Figure 5.4: Average number of hops in a virtual network using a Fat-tree topology.

this topology we go further). Since the data center is near saturation, each request will more likely need a high number of hops, which explains the increase we see from this Figure to the Tree topology one.

## 5.4 Goal III - High Resource Utilization

By evaluating this goal we want to measure the resource utilization within the Data Center: this means calculating the server and link utilization. This metric is computed when we have the first rejected request by the controller. We want to point out that, since our algorithm is deterministic, it will get the same values for server and network utilization if the given input is the same. The graphs that we present in this section do not have the standard deviation on each datapoint because we have only generated our dataset once. As already stated in section 5.1, our dataset is composed by several hundreds of thousands of files, which makes its generation a time (and storage) consuming task. We could do it a few times, but not enough for it to have statistical relevance. The server utilization is calculated by the following formula:

$$server\_utilization = \frac{\%\ of\ Used\ CPU}{100}$$

This is calculated for every server in the Data Center and then averaged. Following the same logic, the network utilization is given by:

$$network\_utilization = \frac{Bandwidth\ Used\ in\ Link}{Initial\ Bandwidth\ of\ Link}$$

This is also calculated for each link and then averaged to produce the final result.

### 5.4.1 Tree Topology

The resource utilization results using a Tree topology are portrayed in Figure 5.5.

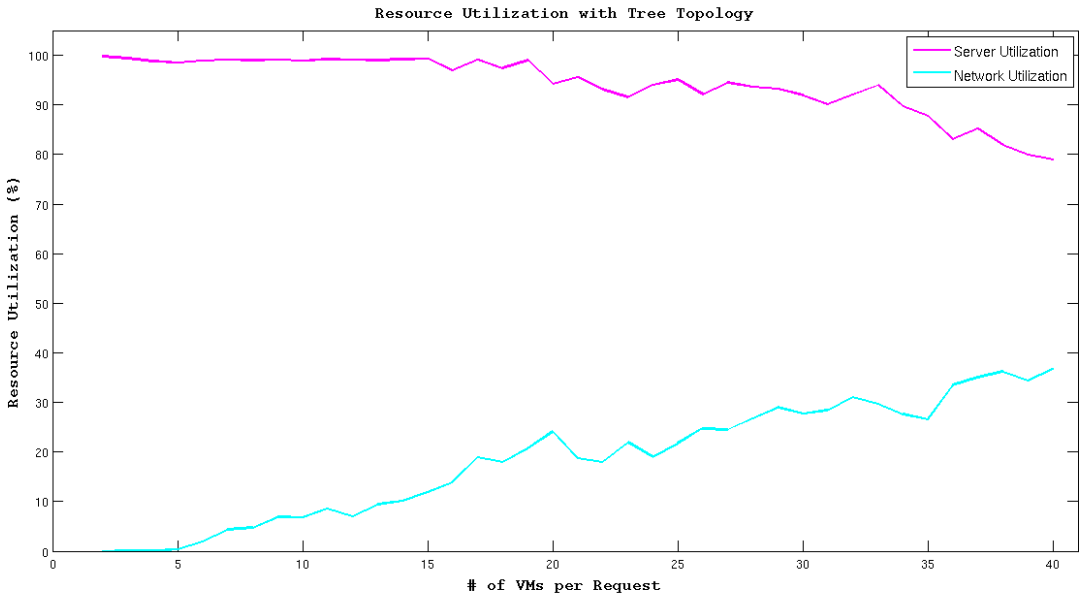



Figure 5.5: Resource utilization using a Tree topology.

Analyzing Figure 5.5 we can see that most of the time our system achieves high server utilization. This makes sense since one of the main concerns of our algorithm is doing a best-fit placement of the VMs within the servers. This also shows that our decision of not having a consolidation algorithm running periodically in the controller is correct, since our algorithm already does an incremental consolidation. The server utilization starts to drop when the number of VMs in a virtual network is around 20. This happens because with a request of this size (and larger), some of the VMs have to be placed on different servers, which causes fragmentation of the CPU utilization by a server, which results in a lower server utilization. Obviously, when this happens the network utilization starts to grow, since we have more and more utilized links across the network.

We want to point out that our low network utilization is a result of getting all the servers full before we get some virtual networks that require link usage, since this is just a matter of which resource is exhausted first. Since this is doing what we would expect by looking at the algorithm, we do not think this is relevant. However, in a real world deployment this would be something to adjust, e.g. by not allowing virtual network requests with few VMs, since the small requests give a greater chance for the algorithm to consolidate them - which fills servers and leaves the network unused.

### 5.4.2 Fat-tree Topology

Figure 5.6 depicts the resource utilization results when the network has a Fat-tree topology.

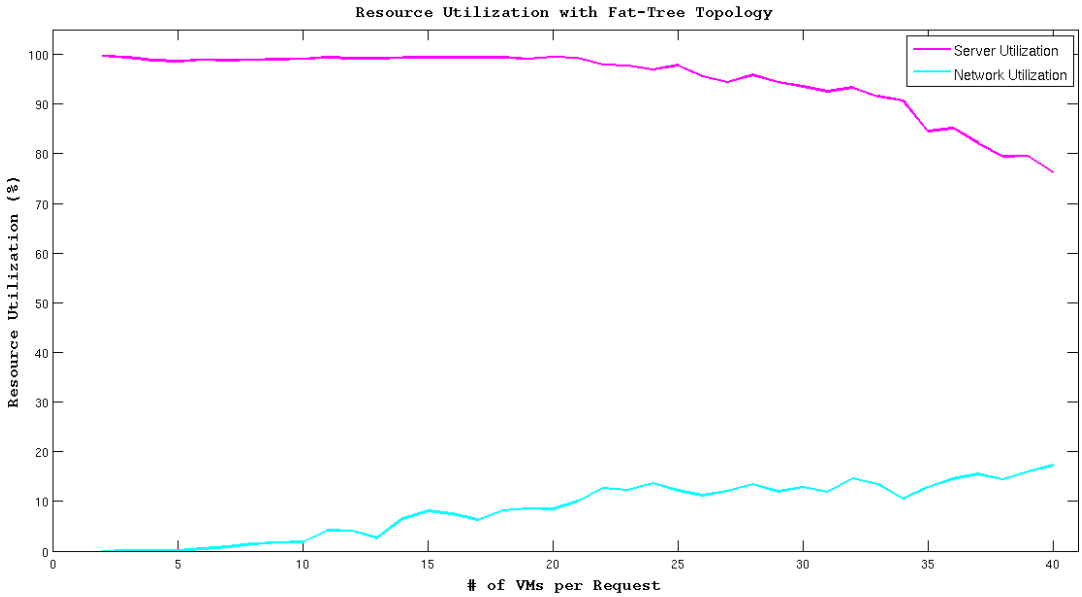



Figure 5.6: Resource utilization using a Fat-tree topology.

In Figure 5.6 we can see that these lines are similar to the ones shown in Figure 5.5. However, using Fat-Tree as a topology allows the server utilization to remain high for longer (up to 25 VMs per request), as we can see by the smoother magenta line (which in the Tree topology starts to drop at 15 VMs per request). This can be explained by higher number of requests served by this topology, since more requests allow to decrease the fragmentation in CPU usage across servers, which in turn causes the server utilization to increase.

However, in this case the network utilization is significantly lower (again comparing to the Tree topology case). This is due to the much higher number of links that this topology has (more than double), which all add up to the denominator of this metric and causes it to decrease significantly.

## 5.5 Goal IV - Bandwidth Guarantees in a Work-conservative System

We now intend to demonstrate that a server can get more bandwidth than what was requested when the link is free, and gets at least what was requested when the link is saturated. However, we were not able to do a full-fledged evaluation of this goal, beucase of a software bug between Mininet's links and Open vSwitch's queues. Basically, it is not possible to use bandwidth-limited links in Mininet and queues

in Open vSwitch at the same time, because they both use the *tc* utility at the same time, and this causes a conflict. This is a known bug and it is described in Mininet's repository[1], but a fix has not yet been released. This limits our evaluation possibilities, because this way we can not emulate a saturated link, which means a server would always get the bandwidth it requires (as long as the machine that is running the emulation has CPU available for that).

To work around this limitation and to prove the concept of our idea, we have devised a different way to test this. In Figure 5.7 is the network topology that we have created in Mininet, specifically for this test.

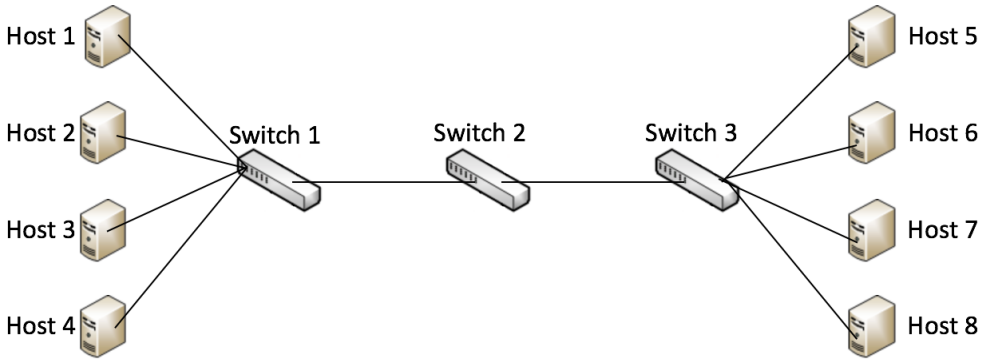



Figure 5.7: Network topology for the bandwidth test.

In this topology we can see 8 hosts and 3 Open vSwitches. The network was configured in the following way: *Switch 1* was configured with 4 queues, one for each host it connects to, all with the same configuration, a `min-rate` of 25 Mbit/s. The appropriate OpenFlow rules were also created to forward the traffic from each host to its own queue. Then, *Switch 2* is emulating the link throttling. This switch has one queue configured with a `max-rate` of 100 Mbit/s, and just one OpenFlow rule - to forward packets that come from its connection with *Switch 1* to the other connection (with *Switch 3*). Finally, *Switch 3* is configured with 4 queues, one for each host, and they also have the `min-rate` parameter set to 25 Mbit/s. Then, there are four OpenFlow rules, dictating that trafffic that comes from *Host 1* goes to the queue in the port connecting to *Host 5*. The same logic is applied to the pairs *Host2-Host6*, *Host3-Host7* and *Host4-Host8*.

Given this configuration, we used the `iperf` tool to make the hosts on the left generate traffic with a constant bit-rate, each one generating traffic with a rate of 50 Mbit/s. We used a constant bit-rate to make sure that changes we see in the rate on the receiver side is due to the network changes and not from changes in the sending side. In Figure 5.8 we can see the graph generated according to these configurations.

[1] `https://github.com/mininet/mininet/issues/243` (Last Access: 10-10-2015)

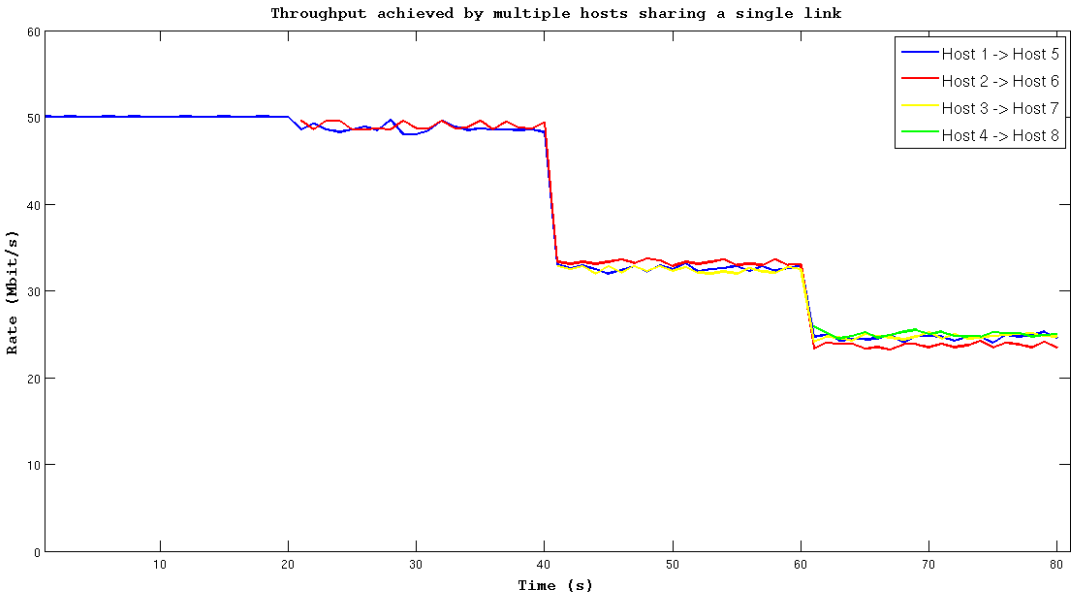



Figure 5.8: Rate achieved by each host when all of them are sharing a single link.

In the beginning ($t = 0s$), only *Host 1* is generating traffic with the bit-rate stated above, and with *Host 5* as destination. As he is the only one doing so, he gets the full bandwidth that he is requesting - 100 Mbit/s. This goes on until $t = 20s$, when *Host 2* starts to generate traffic towards *Host 6*, and there are two hosts generating traffic at 50 Mbit/s, which is the ”link capacity” (i.e. the `max-rate` allowed by *Switch 2*). Each host on the right gets pratically the bandwidth that its correspondent is generating, but we can already see the action of *Switch 2*, portrayed by the irregularities in both lines. When we reach $t = 40s$, *Host 3* starts to generate traffic to *Host 7*. Now, the sum of the traffic generated by the hosts exceeds the ”link capacity”, which will cause dropped packets. However, each host gets more than its assured bandwidth (25 Mbit/s), as the three of them divide the link, each one getting about 33 Mbit/s. Finally, at $t = 60s$, *Host 4* begins to generate packets towards *Host 8*. Now, the 100 Mbit/s link is divided by the four hosts, and each one gets about its assured bandwidth. The scheduler in *Switch 1* forces the packets coming from the four hosts to be schedule equally, since they all have the same configuration. The same happens in *Switch 3*, while on *Switch 2* the ”extra” packets are getting dropped, and each host gets real close to the required bandwidth.

As we can see by this example, our solution allows hosts to get guarantees about bandwidth, while using more when there are spare resources. This remains to be tested at a large scale, where several hosts generate traffic at the same time, varying the level of saturation of the data center. This type of proof would allow us to claim that our system fully achieves all goals in the large scale, and that would be very interesting to publish as innovative research work.

## 5.6 Summary and Final Remarks

In this chapter we have assessed the quality and goal fulfillment of our solution, by performing an extensive set of evaluation experiments and showing here its results.

We began by describing the evaluation environment, which comprised describing the test bed and the evaluation setup. Then, goal by goal, we have evaluated our solution according to different perspectives. Throughout this evaluation we made our analysis of the results presented by the several graphs, drawing parallels between the different test cases when appropriate.

We think that we can say that the goals we have set out in the first chapter were achieved, although the last one in a simpler scenario.

# Chapter 6

# Conclusions and Future Work

## 6.1 Conclusions

In this dissertation we have addressed the topic of network management and configuration in Data Center networks. As described in chapter 1, current Data Centers lack network performance guarantees, since all tenants interchangeably share the network. This makes the performance of a tenant's application unpredictable, since it is depedent on factors outside of its control. This unpredictability severely prevents a wider cloud adoption, since there are many use cases that require network performance guarantees.

These problems are solved using the abstraction of virtual networks. Virtual networks are isolated from each other, providing performance guarantees. We have undertook this challenge using Software-Defined Netwoking.

We designed an OpenFlow controller that is able to allocate virtual networks (with bandwidth guarantees) in a work-conservative system, achieving high consolidation on the allocation of virtual networks and high resource utilization of the Data Center's resources. We have also designed our allocation algorithm taking into account the environment where it will run in, a Data Center, which means that it has to scale well. We did this by appraising the data structures that our algorithm uses, and designed them to lower the execution time of our algorithm.

With our solution well defined, we have implemented our controller on top of the Floodlight open-source project. We began by studying how this controller works, in order to define which modules could be used as is, which needed modification and which had to be created from scratch. With this knowledge we have implemented our solution within this controller.

In order to assess our goal fullfilment, we have evaluated our implementation. With the objective of doing a comprehensive evaluation, we have developed scripts to create two Data Center network

topologies: Tree and Fat-tree. Our evaluation shows that we have pretty much accomplished the goals we have set out in the beginning. Throughout the evaluation, we can see that our system has: low execution time, high consolidation of virtual networks and high resource utilization.

Regarding the last goal (providing bandwidth guarantees in a work-conservative system), due to a conflict in our software stack, we were not able to make the full-fledged evaluation we would like. With this conflict solved, we could undoubtedly achieve all goals, and with it, make some more interesting research work (towards publishing this complete work). Notwithstanding, we have devised a work around in order to make a proof of concept on what we intend to build.

## 6.2 Future Work

Although we consider that we have developed a good solution, there is certainly room for improvement. Now, we leave some possible directions for future work on this project:

- **Solve the software conflict** - We think that the most important work to be done with this project is to solve the software conflict - or use different software. This is what is preventing a better evaluation to take place, and should be one of the priorities. The conflict is between Mininet and Open vSwitch. Since they are both open-soure, it is possible to try to read through the source code of both of them and try to fix the problem. If this renders undoable, another possible solution would be to use different software for running the emulations.

- **Make the controller fault tolerant** - An interesting follow-up work would be to make the controller fault tolerant, both to failures in links and in nodes (switches and servers). Since the OpenFlow controller receives events when a link or a node is down, one could develop new logic to deal with this information, and then take this into account when making allocations of virtual networks. Implementing this task correctly and perform an appropriate evaluation on this would increase very much the research value of this work.

- **Create new network topologies** - By creating new network topologies, one could assess the solution in different environments. As we can see by our evaluation with two topologies, the results vary considerably amongst them. Creating and running new tests on new topologies woud allow to draw new conclusions about this work.

- **Parallelize the processing of virtual networks** - If the controller could parallelize several requests at the same time, the number of processed requests would increase significantly. This is a great challenge since it introduces the need of concurrency control in the controller. Solving

this challenge also allows the creation of some different evaluations of the controller (such as the number of processed requests with and without the parallelization).

# Bibliography


[1] Openflow consortium: Openflow switch specification version 1.0.0, 2009. Available online at: https://www.opennetworking.org/images/stories/downloads/sdn-resources/onf-specifications/openflow/openflow-spec-v1.0.0.pdf (Last Access: 09-01-2015).

[2] Open networking foundation: Software-defined networking: The new norm for networks. 2012. Available online at: https://www.opennetworking.org/images/stories/downloads/sdn-resources/white-papers/wp-sdn-newnorm.pdf (Last Access: 09-01-2015).

[3] Open networking foundation: Openflow switch specification version 1.4.0, 2013. Available online at: https://www.opennetworking.org/images/stories/downloads/sdn-resources/onf-specifications/openflow/openflow-spec-v1.4.0.pdf (Last Access: 09-01-2015).

[4] Open networking foundation: Of-config specification 1.2.0, 2014. Available online at: https://www.opennetworking.org/images/stories/downloads/sdn-resources/onf-specifications/openflow-config/of-config-1.2.pdf (Last Access: 09-01-2015).

[5] M. Al-Fares, A. Loukissas, and A. Vahdat. A scalable, commodity data center network architecture. *ACM SIGCOMM Computer Communication Review*, 38(4):63–74, 2008.

[6] M. Al-Fares, S. Radhakrishnan, B. Raghavan, N. Huang, and A. Vahdat. Hedera: Dynamic flow scheduling for data center networks. In *NSDI*, volume 10, pages 19–19, 2010.

[7] H. Ballani, P. Costa, T. Karagiannis, and A. Rowstron. Towards predictable datacenter networks. In *ACM SIGCOMM Computer Communication Review*, volume 41, pages 242–253. ACM, 2011.

[8] T. Benson, A. Akella, A. Shaikh, and S. Sahu. Cloudnaas: a cloud networking platform for enterprise applications. In *Proceedings of the 2nd ACM Symposium on Cloud Computing*, page 8. ACM, 2011.

[9] T. Benson, A. Anand, A. Akella, and M. Zhang. Microte: fine grained traffic engineering for data centers. In *Proceedings of the Seventh COnference on emerging Networking EXperiments and Technologies*, page 8. ACM, 2011.

[10] K. Bilal, S. U. Khan, J. Kolodziej, L. Zhang, K. Hayat, S. A. Madani, N. Min-Allah, L. Wang, and D. Chen. A comparative study of data center network architectures. In *ECMS*, pages 526–532, 2012.

[11] F. Botelho, A. Bessani, F. Ramos, and P. Ferreira. Smartlight: A practical fault-tolerant sdn controller. *arXiv preprint arXiv:1407.6062*, 2014.

[12] J. F. Botero, X. Hesselbach, M. Duelli, D. Schlosser, A. Fischer, and H. De Meer. Energy efficient virtual network embedding. *Communications Letters, IEEE*, 16(5):756–759, 2012.

[13] R. Braden, D. Clark, S. Shenker, et al. Integrated services in the internet architecture: an overview, 1994.

[14] M. Casado, M. J. Freedman, J. Pettit, J. Luo, N. McKeown, and S. Shenker. Ethane: Taking control of the enterprise. *ACM SIGCOMM Computer Communication Review*, 37(4):1–12, 2007.

[15] A. R. Curtis, J. C. Mogul, J. Tourrilhes, P. Yalagandula, P. Sharma, and S. Banerjee. Devoflow: scaling flow management for high-performance networks. In *ACM SIGCOMM Computer Communication Review*, volume 41, pages 254–265. ACM, 2011.

[16] J. Dean and S. Ghemawat. Mapreduce: simplified data processing on large clusters. *Communications of the ACM*, 51(1):107–113, 2008.

[17] R. Enns, M. Bjorklund, and J. Schoenwaelder. Netconf configuration protocol. *Network*, 2011.

[18] D. Eppstein. Finding the k shortest paths. *SIAM Journal on computing*, 28(2):652–673, 1998.

[19] D. Erickson. The beacon openflow controller. In *Proceedings of the second ACM SIGCOMM workshop on Hot topics in software defined networking*, pages 13–18. ACM, 2013.

[20] A. Fischer, J. F. Botero, M. Till Beck, H. De Meer, and X. Hesselbach. Virtual network embedding: A survey. *Communications Surveys & Tutorials, IEEE*, 15(4):1888–1906, 2013.

[21] N. Gude, T. Koponen, J. Pettit, B. Pfaff, M. Casado, N. McKeown, and S. Shenker. Nox: towards an operating system for networks. *ACM SIGCOMM Computer Communication Review*, 38(3):105–110, 2008.

[22] C. Guo, H. Wu, K. Tan, L. Shi, Y. Zhang, and S. Lu. Dcell: a scalable and fault-tolerant network structure for data centers. *ACM SIGCOMM Computer Communication Review*, 38(4):75–86, 2008.

[23] C. Guo, G. Lu, D. Li, H. Wu, X. Zhang, Y. Shi, C. Tian, Y. Zhang, and S. Lu. Bcube: a high performance, server-centric network architecture for modular data centers. *ACM SIGCOMM Computer Communication Review*, 39(4):63–74, 2009.

[24] C. Guo, G. Lu, H. J. Wang, S. Yang, C. Kong, P. Sun, W. Wu, and Y. Zhang. Secondnet: a data center network virtualization architecture with bandwidth guarantees. In *Proceedings of the 6th International COnference*, page 15. ACM, 2010.

[25] N. Handigol, B. Heller, V. Jeyakumar, B. Lantz, and N. McKeown. Reproducible network experiments using container-based emulation. In *Proceedings of the 8th international conference on Emerging networking experiments and technologies*, pages 253–264. ACM, 2012.

[26] B. Heller, S. Seetharaman, P. Mahadevan, Y. Yiakoumis, P. Sharma, S. Banerjee, and N. McKeown. Elastictree: Saving energy in data center networks. In *NSDI*, volume 10, pages 249–264, 2010.

[27] B. Heller, R. Sherwood, and N. McKeown. The controller placement problem. In *Proceedings of the first workshop on Hot topics in software defined networks*, pages 7–12. ACM, 2012.

[28] T. R. Henderson, M. Lacage, G. F. Riley, C. Dowell, and J. Kopena. Network simulations with the ns-3 simulator. *SIGCOMM demonstration*, 2008.

[29] I. Houidi, W. Louati, W. Ben Ameur, and D. Zeghlache. Virtual network provisioning across multiple substrate networks. *Computer Networks*, 55(4):1011–1023, 2011.

[30] T. Koponen, M. Casado, N. Gude, J. Stribling, L. Poutievski, M. Zhu, R. Ramanathan, Y. Iwata, H. Inoue, T. Hama, et al. Onix: A distributed control platform for large-scale production networks. In *OSDI*, volume 10, pages 1–6, 2010.

[31] T. Lakshman, T. Nandagopal, R. Ramjee, K. Sabnani, and T. Woo. The softrouter architecture. In *Proc. ACM SIGCOMM Workshop on Hot Topics in Networking*, volume 2004, 2004.

[32] B. Lantz, B. Heller, and N. McKeown. A network in a laptop: rapid prototyping for software-defined networks. In *Proceedings of the 9th ACM SIGCOMM Workshop on Hot Topics in Networks*, page 19. ACM, 2010.

[33] A. Lara, A. Kolasani, and B. Ramamurthy. Network innovation using openflow: A survey. 2013.

[34] N. McKeown, T. Anderson, H. Balakrishnan, G. Parulkar, L. Peterson, J. Rexford, S. Shenker, and J. Turner. Openflow: enabling innovation in campus networks. *ACM SIGCOMM Computer Communication Review*, 38(2):69–74, 2008.

[35] J. Medved, R. Varga, A. Tkacik, and K. Gray. Opendaylight: Towards a model-driven sdn controller architecture. In *2014 IEEE 15th International Symposium on*, pages 1–6. IEEE, 2014.

[36] B. Pfaff, J. Pettit, K. Amidon, M. Casado, T. Koponen, and S. Shenker. Extending networking into the virtualization layer. In *Hotnets*, 2009.

[37] M. R. Rahman, I. Aib, and R. Boutaba. Survivable virtual network embedding. In *NETWORKING 2010*, pages 40–52. Springer, 2010.

[38] H. Rodrigues, J. R. Santos, Y. Turner, P. Soares, and D. Guedes. Gatekeeper: Supporting bandwidth guarantees for multi-tenant datacenter networks. In *WIOV*, 2011.

[39] A. Shalimov, D. Zuikov, D. Zimarina, V. Pashkov, and R. Smeliansky. Advanced study of sdn/openflow controllers. In *Proceedings of the 9th Central & Eastern European Software Engineering Conference in Russia*, page 1. ACM, 2013.

[40] R. Sherwood, G. Gibb, K.-K. Yap, G. Appenzeller, M. Casado, N. McKeown, and G. Parulkar. Flowvisor: A network virtualization layer. *OpenFlow Switch Consortium, Tech. Rep*, 2009.

[41] A. Shieh, S. Kandula, A. Greenberg, and C. Kim. Seawall: performance isolation for cloud datacenter networks. In *Proceedings of the 2nd USENIX conference on Hot topics in cloud computing*, pages 1–1. USENIX Association, 2010.

[42] S. Skiena. Dijkstra's algorithm. *Implementing Discrete Mathematics: Combinatorics and Graph Theory with Mathematica, Reading, MA: Addison-Wesley*, pages 225–227, 1990.

[43] R. Tarjan. Depth-first search and linear graph algorithms. *SIAM journal on computing*, 1(2):146–160, 1972.

[44] A. Tootoonchian and Y. Ganjali. Hyperflow: A distributed control plane for openflow. In *Proceedings of the 2010 internet network management conference on Research on enterprise networking*, pages 3–3. USENIX Association, 2010.

[45] A. Tootoonchian, S. Gorbunov, Y. Ganjali, M. Casado, and R. Sherwood. On controller performance in software-defined networks. In *USENIX Workshop on Hot Topics in Management of Internet, Cloud, and Enterprise Networks and Services (Hot-ICE)*, volume 54, 2012.

[46] T. Tsou, X. Shi, J. Huang, Z. Wang, and X. Yin. Analysis of comparisons between openflow and forces. *Analysis*, 2012.

[47] G. Wang and T. E. Ng. The impact of virtualization on network performance of amazon ec2 data center. In *INFOCOM, 2010 Proceedings IEEE*, pages 1–9. IEEE, 2010.

[48] S.-Y. Wang. Comparison of sdn openflow network simulator and emulators: Estinet vs. mininet. In *Computers and Communication (ISCC), 2014 IEEE Symposium on*, pages 1–6. IEEE, 2014.

[49] S.-Y. Wang, C.-L. Chou, and C.-M. Yang. Estinet open flow network simulator and emulator. *IEEE Communications Magazine*, 51(9):110–117, 2013.

[50] Z. Wang and J. Crowcroft. Bandwidth-delay based routing algorithms. In *Global Telecommunications Conference, 1995. GLOBECOM'95., IEEE*, volume 3, pages 2129–2133. IEEE, 1995.

[51] P. Wette, M. Draxler, and A. Schwabe. Maxinet: distributed emulation of software-defined networks. In *Networking Conference, 2014 IFIP*, pages 1–9. IEEE, 2014.

[52] L. Yang, R. Dantu, T. Anderson, and R. Gopal. Forwarding and control element separation (forces) framework. Technical report, RFC 3746, April, 2004.

[53] M. Yu, Y. Yi, J. Rexford, and M. Chiang. Rethinking virtual network embedding: substrate support for path splitting and migration. *ACM SIGCOMM Computer Communication Review*, 38(2):17–29, 2008.

[54] M. F. Zhani, Q. Zhang, G. Simona, and R. Boutaba. Vdc planner: Dynamic migration-aware virtual data center embedding for clouds. In *Integrated Network Management (IM 2013), 2013 IFIP/IEEE International Symposium on*, pages 18–25. IEEE, 2013.