Congestion Identification in a Radio Access Transport Network

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Degree project in
Communication Systems
Second level, 30.0 HEC
Stockholm, Sweden

Congestion Identification in a Radio Access Transport Network

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2014-03-03

Master's thesis

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Abstract

The convergence of mobile services and Internet has brought a radical change in mobile networks. An all-IP network architecture, an evolution of the radio access transport network, is required to support new high-bandwidth services. Unfortunately, existing control mechanisms are insufficient to guarantee end users a high quality of experience. However, coordinating radio and transport network resources is expected to yield a more efficient solution.

This thesis project investigates the interactions between the congestion avoidance protocols, explicit congestion notification, and the traffic engineering metrics for latency and bandwidth, when using Open Shortest Path First with traffic engineering (OSPF-TE) as a routing protocol. Using knowledge of these interactions, it is possible to identify the appearance of bottlenecks and to control the congestion in the transport links within a radio access transport network.

Augmenting a topology map with the network's current characteristics and reacting to evidence of potential congestion, further actions, such as handovers can be taken to ensure the users experience their expected quality of experience. The proposed method has been validated in a test bed. The results obtained from experiments and measurements in this test bed provide a clear picture of how the traffic flows in the network. Furthermore, the behavior of the network observed in these experiments, in terms of real-time performance and statistical analysis of metrics over a period of time, shows the efficiency of this proposed solution.

Keywords: *Radio Access Transport Network, Open Shortest Path First – Traffic Engineering, Explicit Congestion Notification, Interface to the Router System, Congestion Identification*

Sammanfattning

Tjänstekonvergensen av Internet- och mobila tjänster har medfört en radikal förändring i mobilnäten. En ”All IP” nätverksarkitektur, en utveckling av radios transportnät. Utvecklingen krävs för att stödja de nya bredbandiga tjänsterna. Tyvärr är befintliga kontrollmekanismer otillräckliga för att garantera användarens kvalitetsupplevelse. Med att samordna radio- och transportnätverkets resurser förväntar man sig en effektivare lösning.

Detta examensarbete undersöker samspelet mellan protokoll för att undvika överlast, direkt indikation av överlast och trafikal statistik för fördröjning och bandbredd med trafikstyrning baserat på fördröjning och bandbredd , vid användning av Open Shortest Path First (OSPF - TE) som routingprotokoll. Med hjälp av information om dessa interaktioner, är det möjligt att identifiera uppkomsten av flaskhalsar och för att styra trafikstockningar i transportförbindelser inom ett radioaccess transportnät.

En utökad topologikarta med nätverkets aktuella egenskaper kommer att reagera på en potentiell överbelastning. Ytterligare åtgärder, till exempel överlämningar, vidtas i mobilnätet för att säkerställa användarens upplevda kvalitet. Den föreslagna metoden har validerats i en testmiljö. Resultaten från experiment och mätningar i denna testmiljö ger en tydlig bild av hur trafikflödena framskrider i nätverket. Beteendet hos nätverket som observeras i dessa experiment, i termer av realtidsprestanda och statistisk analys av mätvärden över en tidsperiod, visar effektiviteten av denna föreslagna lösning.

Nyckelord: *Radio Access Transport Network, Open Shortest Path First – Traffic Engineering, Explicit Congestion Notification, Interface to the Router System, Congestion Identification*

Acknowledgements

We would like to express our most sincere gratitude to all the people that have helped us and supported us making possible this Master Thesis, namely:

Annikki Welin, our supervisor in Ericsson, for her constant support and project management, guiding us in a very didactic way, encouraging us always and proposing new ideas to open our minds, providing us the necessary equipment and a comfortable work space, resolving technical and organizational issues and introducing us people interested in the results of the projects and good contacts for our future. Summarizing, helping us in each aspect of the project.

Professor **Gerald Q. Maguire Jr**, our academic supervisor at KTH, for his indispensable feedback and continue brainstorm of ideas to improve our Thesis, guiding us when more we needed it and always with a teaching spirit in order to improve our apprenticeship.

Tomas Thyni, for helping us along the Thesis, providing interesting feedback and ideas, explaining deeply the most difficult concepts and motivating us to go further in our research.

To all our **families** and **friends** for their unconditional belief on us and encourage us to give our best for this project. Their advices and help, even in the distance, have been our main support during these months.

Table of contents

List of Figures

List of Tables

List of acronyms and abbreviations

2G	Second Generation
3G	Third Generation
3GPP	Third Generation Partnership Project
4G	Fourth Generation
ABR	Area Border Router
ACK	acknowledgement
API	Application Programming Interface
AQM	Active Queue Management
AS	Autonomous System
ASBR	Autonomous System Boundary Router
BGP	Border Gateway Protocol
CE	Congestion Evidence
CLI	Command Line Interface
CWR	Congestion Window Reduced
ECE	ECN-Echo
ECN	Explicit Congestion Notification
EIGRP	Enhanced Interior Gateway Routing Protocol
eNB	Evolved NodeB
eUTRAN	Evolved UTRAN
FIB	Forwarding Information Base
GPRS	General Packet Radio Service
GSM	Global System Mobile
GTP	GPRS Tunneling Protocol
HSS	Home Subscriber Server
I2RS	Interface To the Routing System
IANA	Internet Assigned Numbers Authority
iBGP	internal Border Gateway Protocol

IETF	Internet Engineering Task Force
IGP	Interior Gateway Protocol
IGRP	Interior Gateway Routing Protocol
IPv4/6	Internet Protocol version 4/6
IS-IS	Intermediate System to Intermediate System
ITU	International Telecommunication Union
LSA	Link-State Advertisements
LSDB	Link-State Database
LSU	Link-State Update
LTE	Long Term Evolution
Mbps	Megabits per second
MIB	Management Information Base
MIMO	Multiple Input Multiple Output
MME	Mobility Management Entity
NAS	Non-Access-Stratum
NAT	Network Address Translation
NTP	Network Time Protocol
NSSA	Not-So-Stubby-Area
OID	Object Identifier
OFDM	Orthogonal Frequency Division Multiplexing
OS	Operating System
OSPF	Open Shortest Path First
OSPF-TE	OSPF with TE extensions
OSR	Open Source Routing
PDN-GW	Packet Data Network Gateway
PTP	Precision Time Protocol
QoE	Quality of Experience
QoS	Quality of Server
RAN	Radio Access Network
RATN	Radio Access Transport Network

RED	Random Early Detection
RFC	Request For Comments
RIB	Routing Information Base
RIP	Routing Information Protocol
RRM	Radio Resource Manager
RTTA	Real-Time Traffic Aware
RTT	Round Trip Time
S-GW	Serving Gateway
S1-AP	S1- Application Protocol
SC-FDM	Single-Carrier Frequency Division Multiplexing
SCTP	Stream Control Transmission Protocol
SNMP	Simple Network Management Protocol
SON	Self-organizing Network
SPF	Shortest Path First
TCP	Transport Control Protocol
TE	Traffic Engineering
TED	Traffic Engineering Database
TLV	Type/Length/Value
UE	User Equipment
UMTS	Universal Mobile Telecommunicaitons System
UTRAN	UMTS Terrestrial Radio Access Network
WAN	Wide Area Network

1 Introduction

This chapter provides an introduction to the subject of this master's thesis project in order to help readers understand the scope of this project. Next, the problems addressed in this thesis project are described, followed by a statement of the project's aim and goals. The research methodology is described. The chapter concludes with an overview of the structure of this thesis.

1.1 Overview

The convergence of mobile services and Internet has brought about a radical change in mobile networks. The number of mobile broadband subscribers continues to grow at a tremendous rate. The number of active mobile subscriptions is over 2096 million according to International Telecommunication Union (ITU) statistics [1]. Cisco expects that the number of subscribers will be four times as large in four years. This is due to the radical increase in the number of devices and the number of mobile subscribers, with 3.4 billion Internet users expected in 2016 [2]. Along with this growth in the number of mobile subscribers there has been a substantial increase in mobile traffic. Some examples that illustrate this include Ericsson's prediction of a 10-fold increase in mobile traffic by 2016 as compared to 2011 [3] and studies by Cisco who expects an increase in the global mobile Internet data traffic of 10.8 Exabytes (i.e., 10.8 x 10^{18} bytes) per month by 2016 [2]. These expectations of growth do not stop and it is very likely that in the next decade the rate of growth will continue to increase. As the traffic demand due to mobile data applications continues to grow dramatically, mobile operators are investing heavily in infrastructure upgrades to support the network demands of their subscribers [4].

In order to organize and coordinate the evolution of mobile networks the 3rd Generation Partnership Project (3GPP) was created in 1998 to lead and produce technical specifications and technical reports for a third generation (3G) Mobile System based on evolved second generation (2G) Global System for Mobile Communication (GSM) core networks and the radio access technologies that they support[5]. A new generation of mobile networks, 3GPP's Long Term Evolution (LTE), is being rapidly deployed by mobile operators as they try to satisfy their subscriber's demands. By September 2013, nearly 100 cities around the globe had started commercial deployment of LTE systems [6].

LTE is a fourth generation (4G) wireless broadband technology[*]. LTE provides significantly increased peak data rates in comparison with 3G Universal Mobile Telecommunication Systems (UMTS), with the potential for 100 Mbps downstream and 30 Mbps upstream, reduced latency, scalable bandwidth capacity, and backwards compatibility with existing GSM and UMTS technology. Future developments are expected to yield peak throughput on the order of 300 Mbps [7].

The upper layers of the LTE protocol stack are based on many different protocols, but the IP protocol is fundamental to this protocol stack resulting in an all-IP network similar to the current state of wired communications. LTE supports mixed data, voice, video, and messaging traffic to and from user equipment (UE). Today, the radio base stations (called eNodeBs) are connected via IP technology to the core network, thorough a network of routers forming a so-called Radio Access Transport Network. This network is connected via Serving

[*] More correctly, LTE-**Advanced** is a 4G system according to the ITU-T criteria for 4G systems.

Gateways (S-GWs) to one or more packet data network gateways (PDN-GWs), which is in turn is connected with the Internet and other packet networks. This overall system architecture is shown Figure 1-1.

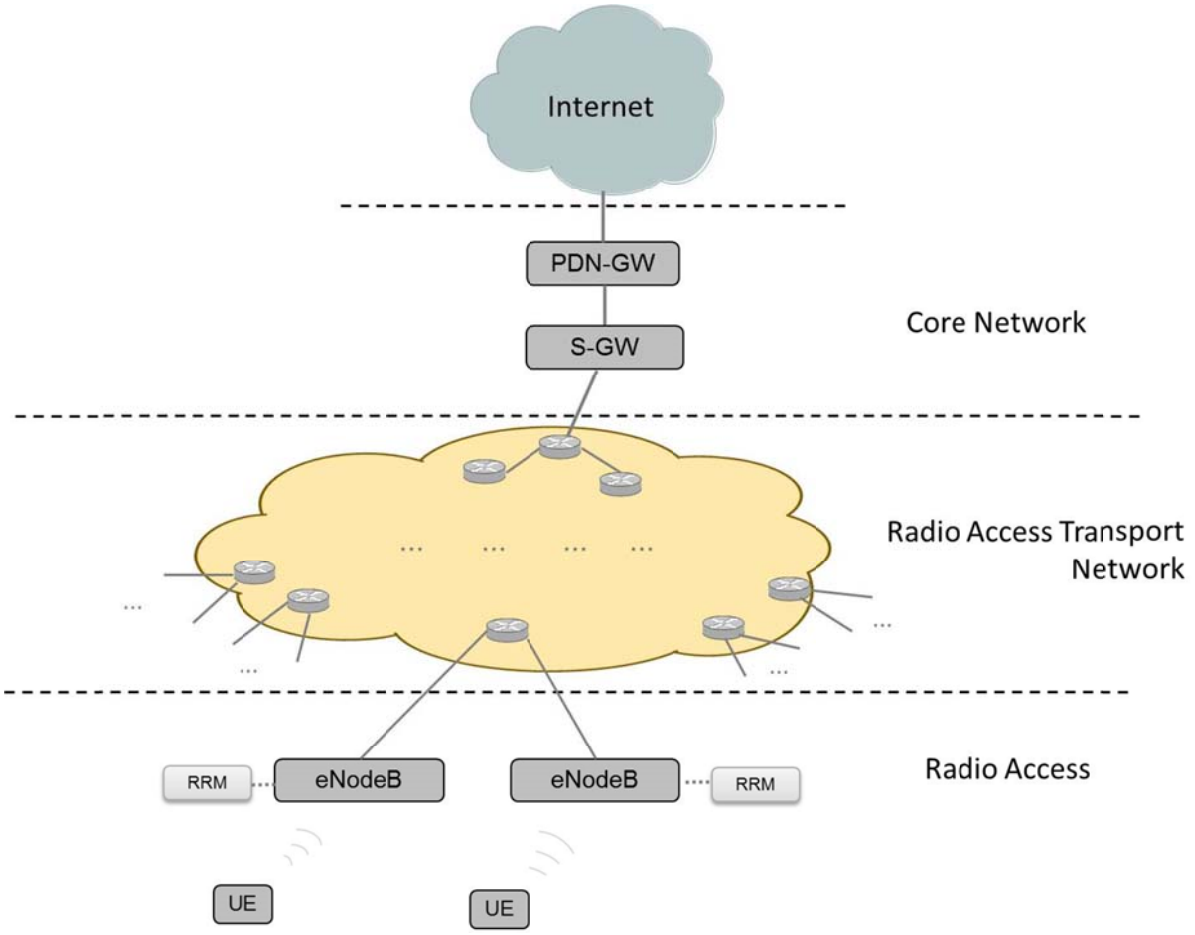

Figure 1-1: Relationship between the Radio Access Transport Network and the eNodeBs.

However, even the advances already offered by LTE are unable to handle the expected demands for capacity and increased data throughput. Due to the growth in number of users, the increasing number of both human users and devices, and the increasing aggregate data rates; the appropriate management and control of all this amount of traffic is becoming more and more relevant in order to ensure subscribers of a reliable Quality of Experience (QoE) [8].

Ideally the network should be accessible and fair to all the users. In order to ensure this fairness we should not allow some users to load the Radio Access Network (RAN), while other subscribers experience poor quality of service (QoS) or cannot even access the internet. Each user's traffic has to be prioritized such that the network allows all of the users to access services with the appropriate bandwidth and within the desired latency bounds, *even* when there is congestion. As in many other markets, mobile broadband and fixed line broadband service providers seek to provide their subscribers with exactly what they need and are willing to pay for - no more, no less - and that is why our efforts are directed to ensure the best service possible with the resources that have been deployed in order to satisfy the actual current needs of the subscribers.

The expectations described above explain why an all-IP network architecture, an evolution of the radio access transport network, is required to support the current and future high-bandwidth services. Unfortunately, the existing control mechanisms are insufficient to guarantee end users a high QoE.

Earlier studies focused on different solutions to address congestion in the transport and radio access networks. However, in this thesis project a more efficient solution is proposed based upon *coordinating* the radio and transport access networks.

1.2 Problem definition

The evolution of mobile networks towards new high-bandwidth services brings new challenges due to the mobile network's behavior. The number of IP applications is increasing and the applications are creating new requirements for mobile networks. Mobile network operators need to ensure QoE and equitable services for all of their subscriber [8]. In order to meet this goal the mobile network operators need to skillfully manage their network resources. In order to do this management the network operator needs to have as much information as possible about the network's current status and the expected demands upon their network.

In particular, the RATN equipment that was designed to provide UEs with access to the Internet and other packet data networks needs new mechanisms to meet the requirements of 4G networks. Providing adequate capacity is one of the main problems to solve for existing networks, especially providing adequate capacity in the aggregation network structure and the last hop (the wireless link between the UE and the base station) – as these are typically where bottlenecks appear[10]. Bandwidth bottlenecks can have a large negative impact on the user's perceived QoE, hence avoiding such bottlenecks forming becomes an important challenge for the mobile networks operators. For this reason, the existence and evolution of congestion avoidance and control mechanisms is crucial in such networks.

Unfortunately, the existing congestion mechanisms, such those employed in the transmission control protocol (TCP) or Explicit Congestion Notification (ECN), are only handled by the endpoints of the communication, leaving the networks in between with no part in this task (other than marking packets in order to provide ECN). Today nodes along the path between the UE and the PDN-GW generally deal with congestion imply by deciding *which* packets to drop. Moreover, the congestion information received at the endpoints is very limited, typically the endpoints only knowing the *presence* of congestion along a communication path, but do not have any details about which part of the network is congested nor what the network's state is along the path. As a result the endpoints cannot initiate actions to systematically deal with this congestion. Hence the network must collect and utilize information to deal with this congestion. Today reports of congestion in real-time are becoming necessary with a radio access transport network in order to avoid bottlenecks forming. For example, this information can be forwarded to a radio resource manager (RMM) or Mobility Management Entity (MME) for further action (e.g. dropping some UEs' connections or initiating a handover for one or more UEs).

For all of the reasons described above, new congestion identification mechanisms are needed and there needs to be a way to propagate information about congestion to the relevant entities within the networks and in some cases to the relevant endpoints. The relevant network entities are the RRMs, MMEs, PDN-GWs, eNodeBs, and the routers in the radio access transport network[*]. For this reason, this master's thesis project studies the *interaction* between congestion avoidance protocols (such as ECN) and traffic engineering (specifically Open Shortest Path First Traffic Engineering - OSPF-TE) in a radio access transport network in order to meet current and near future requirements.

1.3 Aim and goals

The aim of this master's thesis project is to provide Ericsson, the company in which this project is being conducted, with the basis for a mechanism to identify congestion in a radio access transport network. This is one of the challenges which the company is currently facing as part of their tremendous effort to offer solutions that can better adapted to the growth in the amount of mobile broadband traffic.

Given this aim, the goal of this thesis project can be split into two parts. The first part is to investigate protocols related to congestion control and the interactions between these protocols. The second part is to propose a mechanism that identifies congestion in a radio access transport network. The following specific sub-goals will guide us throughout the project to achieve the aim and goals proposed above:

- Study the ECN protocol and the OSPF-TE metrics for delay and bandwidth.
- Investigate ways to implement them in a radio access transport network of a LTE network.
- Investigate their interaction in order to gain enhanced knowledge of congestion, in order to be able to identify congestion in the radio access transport network.
- Propose a mechanism based upon the results of the investigation and integrate this mechanism in a test bed, based on Quagga (routing software) routers, that simulates the proposed solution for evaluation in a test scenario.

1.4 Methodology

To carry out and accomplish the objectives of this project, qualitative and quantitative methods will be used. The first parts of this master's thesis project are based on a qualitative research methodology and then, with the information collected from the test bed deployment of our scenario, initiate a quantitative study. The schedule and problems encountered in this thesis project can be described as the following three steps:

- **Background study and design proposal**: At the beginning of our project, our main task was to obtain the relevant knowledge required to understand and carry out this master's thesis project. In order to accomplish this part, we started with a literature and studied related work. The background study considered the three main prerequisites of our thesis. First of all we needed to have a wide overview of the new System Architecture Evolution (SAE), especially as LTE. While we reading the literature for this part, we mainly focused on the RAN. After this, we studied OSPF-TE and ECN, as these were the main protocols that we expected to use. The goals of this sub phase were: (1) to understand OSPF and OSPF-TE behavior along with

[*] For readers familiar with existing GSM and UMTS networks, the radio access transport network replaces the so-called "back-haul" network with an IP based network as part of the transformation to an all-IP network.

its LSA types and the architecture of Opaque LSAs, and all the functionality of ECN protocol; (2) identify relevant processing metrics and virtualization information that OSPF-TE Opaque LSAs would need to convey and how these can interact with ECN; and (3) investigate how OSPF-TE and ECN can be extended and implemented in our transport network scenario.
Finally with all the information gathered, we have proposed a design of a mechanism for achieving our goals.

- **Test-bed implementation**: This step involves the deployment of a test-bed and the use of this test bed with a test scenario to implement and evaluate a proposed solution for the congestion identification problem. This step took the largest amount of time during this thesis project. Part of the reason for this was the time needed for the development and study of a suitable scenario and for the implementation of the solution that we propose. Initially, we implemented all the extensions and developments that our test-bed would need to run OSPF-TE and ECN. After this, the experiments with our test bed started and we begin to obtain and collect the information necessary to identify the congestion.
- **Verification**: The final part of this project has been divided in two different steps. First a complete evaluation of the designed network and elements has been done. This step has appraised the functioning of the mechanism implemented; observing that the information about congestion identification collected is accessible for the nodes in the network. The second step formulates an analysis of different simulations of our test bed, in order to test the interaction between the evidence of congestion identified with ECN and the congestion information provided by the nodes through OSPF-TE. With visualization and monitoring of the information obtained, this interaction is verified.

1.5 Structure of the thesis

The chapter one is the introduction to the problem, aim, goals, and methodology. The motivation for the fulfillment of this project and the main objectives of it are explained.

The second chapter provides the background and the knowledge that a reader initiated in the mobile network subject needs to fully understands the rest of this thesis. This chapter gives and introduction to the evolution of the RAN and explains OSPF-TE and other mechanisms and protocols. It ends with an introduction to the routing software used

Chapter three describes the method that will be used to achieve the goals described above. The whole explain of the design of the network and its elements. Different options and methods have been taken into account and are discussed in this chapter

The fourth chapter explains in detail the implementation of the design explained in the previous chapter. This chapter takes into consideration the limitations of our test bed, explains them and its alternatives and finally discusses how these have influenced in the final solution.

The chapter five makes a complete analysis of the solution set out. A whole overview of the obtained information is given, followed with a study of certain simulated scenarios that corroborate this solution proposed.

The final chapter six states the conclusions reached during this project. Together with these conclusions a suggestion of the possible future works and a reflection of the consequences of this master thesis is considerate it.

2 Background

This chapter begins by introducing the reader to the evolution of radio access transport networks in order that the reader can understand the context of this project. After this, a brief introduction is given to the Internet Engineering Task Force (IETF) proposed Interface to the Routing System. The Explicit Congestion Protocol is explained, followed by an introduction to OSPF-TE. This chapter continues with an explanation of a wide set of protocols and techniques needed for a better understanding of the measurements, implementation, and design that will be presented in subsequent chapters. Section 2.6 presents the Quagga router and the OSPF API, as they will be used to implement the proposed solution that will subsequently be evaluated in a test bed. The final section of the chapter surveys related work.

2.1 Evolution of the Radio Access Transport Networks

Since GPRS first enabled user applications to easily send data packets over mobile networks, the use of packet based communication technology in wide area cellular mobile networks has experienced a rapid evolution to today's mobile broadband internet as experienced by increasing numbers of subscribers. 3GPP is responsible for evolving the GSM, UMTS, and LTE standards. This section summarizes the aspects of these networks that are needed to understand the context of the thesis.

Both, UMTS and LTE, introduced a redesign of the Terrestrial Radio Access Network (TRAN), referred to as UTRAN and Evolved-UTRAN (eUTRAN) respectively. UMTS combined properties of the earlier circuit-switched voice networks with properties of packet-switched data networks in order to support new services. Despite the improvements brought by UMTS, it has been limited by several of its design decisions in the same way as its predecessors were. LTE is an evolution of UMTS in which both the radio and core networks were redesigned. The main LTE improvements over UMTS are Orthogonal Frequency-Division Multiplexing (OFDM) with Multiple Input Multiple Output (MIMO) support for transmitting the data over the air interface and an all-IP approach to simplify the design and implementation of the air interface, radio network, and core network [11]. In this thesis we will focus on the later improvement, specifically we focus on how the network architecture has evolved and, in particular, how the radio and core networks are implemented in this new architecture.

The LTE network architecture is divided in to a radio network (referred to as eUTRAN) and a core network (referred to as the Evolved Packet Core). This architecture is shown in Figure 2-1. According with Olsson, et al. [7], two main principles have guide the design of this architecture: a flat architecture, as few nodes as possible handle the user data traffic, and a separation of control data from user data.

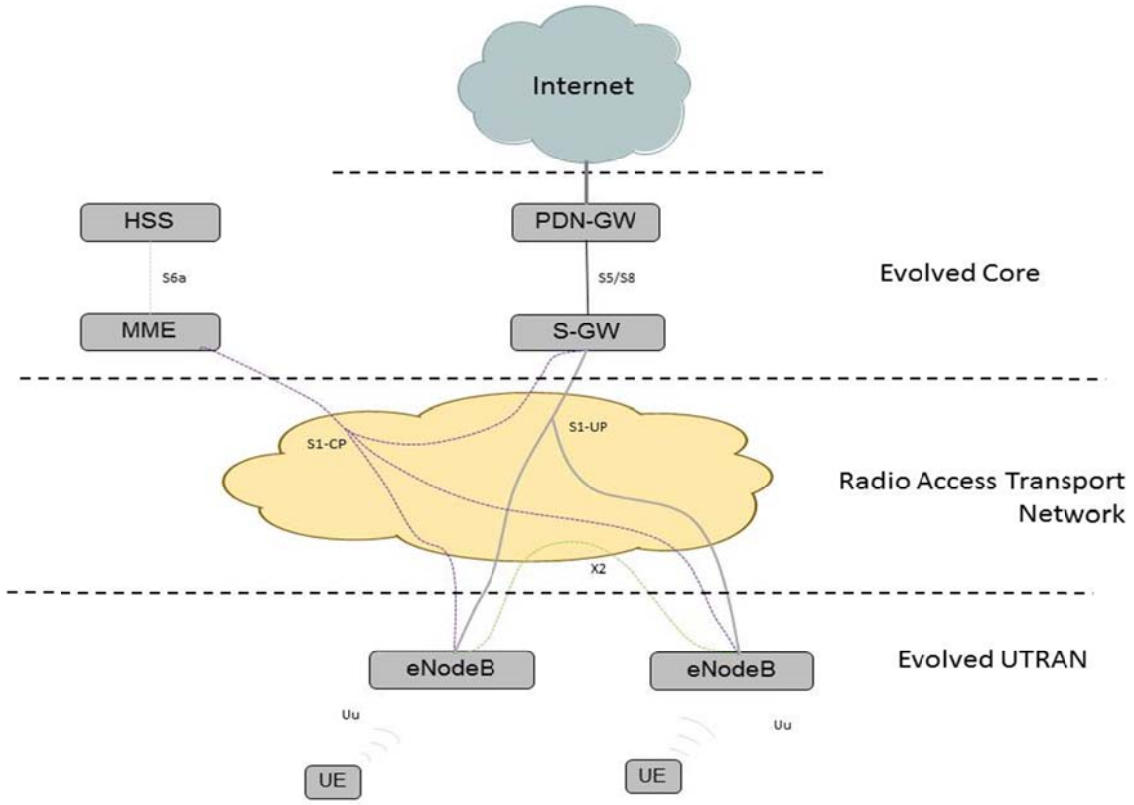

Figure 2-1: LTE network architecture.

2.1.1 Evolved UTRAN

The eUTRAN is composed of the UEs and eNodeBs that communicate over the air interface. This air interface uses OFDM for downlink and Single-Carrier Frequency Division Multiplexing (SC-FDM) for uplink transmissions. This air interface can take advantage of the MIMO technique to exploit multipath transmissions to increase the channel's peak data rate.

The eNodeB base station is one of the most complex entities in a LTE network due to the desire to realize a flat network where the number of entities handling user data is reduced. This node is responsible for the followings tasks:

- User management and scheduling of radio interface resources,
- Ensuring QoS depending upon the user profile, and
- Performing the operations necessary to support UE mobility.

In order to perform the complete set of tasks for which the eNodeB entity was designed, the eNodeB utilizes the following protocols:

- The eNodeB communicates with the core network via the S1 interface. This interface has two logical parts: user plane (UP) and control plane (CP). Both of these parts are IP tunnels. Figure 2-2 shows the protocol stack for the S1 interface.
- The S1-UP interface was designed to transport user data packets through the radio access transport network. As the endpoint of the S1-UP interface, the eNodeB is at one end of a tunnel (called a bearer) implementing a GPRS Tunneling Protocol (GTP) session. S1-UP manages the user traffic flowing through the radio access transport network. In order to achieve the goals of this thesis project we needed to understand the details of this interface and the traffic that flows across it.
- The S1-CP control plane interface is used for two purposes. First this interface is used for interacting with the core network. For example, configuring the eNodeB as a network node.

Second, this interface is used for signaling messages that concern the users of the system, such as signaling for establishing a tunnel between the eNodeB and SG-GW, for maintaining a user service, or for perform handovers as directed by the MME.

- The X2 interface enables one eNodeB to communicate directly with other eNodeBs. This eNodeB enables a handover between two eNodeBs to be performed *without* involving the core network. Such a handover can only be performed if the target eNodeB is known and reachable through the identification of this eNodeB made by the UE.

The eNodeBs communicate through the transport network using the S1-UP, S1-CP, and X2 interfaces to entities connected to the evolved core network.

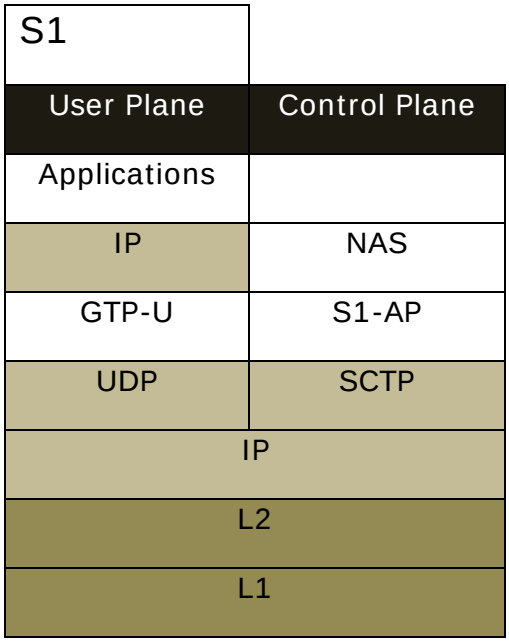

Figure 2-2: The protocol stack of the S1 interface in a LTE network.

In addition, self-optimizing and self-organizing network (SON) functions has been introduced in LTE networks that leverages network intelligence based on measurements performed by the eNodeB in order to automate the configuration and optimization of the Radio Access Network. Some of the use cases of SON functions are[12]:

- Automatic Neighbor Relation (ANR), enabling automatic configuration of neighbor cells.
- Load balancing: adjustment of cell handover parameters to balance the traffic between cells.
- Handover: optimizing the mobility functionalities as identification and avoidance of Ping-Pong behavior or decrease the number of handovers.
- Network monitoring.

2.1.2 Evolved core network

The evolved core network includes MMEs, S-GWs, PDN-GWs, and the Home Subscriber Server (HSS).

A MME is responsible for all the signaling messages between a UE and the core network and between the eNodeBs and the core network. The MME performs the following tasks [11]:

- Helping in the exchange of authentication information between a UE and the HSS.
- Establishing a tunnel (bearer) for user data packets between an eNodeB and a S-GW.
- Handover support when no X2 interface is available and modifying the tunnels after a successful handover.
- Manage idle mode UEs when the UE has released its air interface connection and released its resources in the radio network.

In addition to the use of the interfaces described above, the MME uses the interfaces S1-CP and S6a to communicate with the others entities of the core network, as shown in Figure 2-1.

The S-GW manages the user data packets by acting as a bridge between the radio network, where the S1-UP GTP tunnels terminate, and the core network, where an end of the S5-UP GTP tunnel is situated. The S1 and S5 tunnels, for a UE, are independent of each other and can be separately modified as required. If there is a handover where the UE changes from one eNodeB to other eNodeB managed by the same MME, then only the S1-UP GTP tunnel is modified.

The PDN-GW is a gateway to the Internet or other packet data networks. Here, the user data packets from an external network are encapsulated in a S5-GTP tunnel and forwarded to the S-GW which is currently responsible for this UE. The PDN-GW is responsible for assigning IP addresses after a UE is authenticated via the MME's communication with the HSS. The PDN-GW can utilize network address translation (NAT) to map many internal IP addresses to a smaller number of public IP; enabling many IP tunnels to connect to one UE.

The HSS stores all the subscriber related information. This includes credentials for authentication and access authorization. The MME requests subscription-related data via messages to the HSS over the S6a interface.

2.1.3 Radio Access Transport Network

As it is shown it the Figure 2-1, the radio access transport network is situated between the eUTRAN and the Evolved Core carrying out the communications between both parts of the network. It can be divided in an access part, from the eNodeBs to concentration points, and an aggregation part, to transport mobile flows from concentration points to the core network. Both parts are composed of different technologies: microwave, copper cables or optical fiber (GPON), depending of the available resources in the concerned area and bandwidth requirements. The routers, located in each node of the Radio Access Transport Network, are the responsible of forwarding packets through the network from the access to the aggregation part. Although, none difference is conceived in the routing and the forwarding between both parts. OSPF or IS-IS are used as routing protocols to define the network topology configuring the forwarding tables of the routers.

In this project we want to identify congestion occurring within this network and to provide information about this congestion to the relevant entities. The information will be used by these entities to determine *how* the user's IP tunnels between PDN-GW and the UE's current eNodeB (and potentially a future target eNodeB) *should be treated* to ensure QoE or any other SON use case as it is described before. However, how this determination is to be made is out of the scope of this thesis project, hence this is left as future work.

2.2 Explicit Congestion Notification

The ECN protocol is a congestion avoidance protocol that extends IP and transport protocols to reduce the impact of loss on latency-sensitive flows [13]. Usually at the transport layer TCP is the responsible for avoiding congestion. However, without ECN the only evidence of congestion to TCP is packet loss. ECN takes advantage of active queue management (AQM), which detects congestion in routers *before* the queue overflows. ECN uses the IP protocol to transport this notification and the transport protocol, usually TCP, is responsible for taking the appropriate action in order to reduce the congestion it is causing in the network.

In ECN, the endpoints of the communication are denoted as ECN-capable nodes and the routers in between as ECN-aware. ECN-aware routers uses AQM to detect congestion based on the average queue length exceeding a threshold, rather that reacting only when the queue overflows. In case of congestion, when the threshold is exceeded, the router marks the packets currently in its queue as Congestion Experienced (CE) in the IP header. However, ECN-aware routers may apply different policies to these packets, for example dropping or marking as many packets as a specific router desires.

As described above, an extension of the IP protocol carries the notification of congestion to the end nodes. For this purpose, bits 6 and 7 of the IPv4 Type of Service[14] octet were designated as the ECN field [13]. Table 2-1 shows all the possible configurations and their associated meanings. The two codepoints for ECT, ECT(0) and ECT(1), are set by the end nodes to announce that they are ECN-capable, details are discussed in RFC 3168[13]. The CE codepoint is set by an ECN-aware router to indicate congestion to the end nodes.

Table 2-1: ECN field in the IP header

ECN	CE	Meaning
0	0	Not-ECT
0	1	ECT (1)
1	0	ECT (0)
1	1	CE

Another point worth considering is how ECN works in the presence of IP tunneling. When an IP tunnel is implemented in a network between the two endpoints, then different actions may be taken to use ECN depending of the particulars of this network. There are two kinds of IP tunnels with respect to ECN: a tunneled that supports ECN or one that does.

In an IP tunnel, a new “outer” IP header, that encapsulates the original or “inner” header, is added. If the tunnel supports ECN, then the ECN IP bits are copied from the “inner” to the “outer” header at the entry to the tunnel and if congestion has occurred in the IP tunnel then the CE codepoint is copied from the ”outer” to the “inner” at the egress of the tunnel. In this way, if a CE codepoint is set before or after entering the tunnel, then the CE codepoint is conveyed to the receiver. On the other hand, if the tunnel does not support ECN, then the not-ECT codepoint may be set in the “outer” header and the “inner” header is not altered after the packet exits the tunnel, hence the only mechanism available for controlling congestion concerning this tunnel is dropping packets.

In addition, ECN requires support in the end nodes at the transport layer. With regard to ECN, the transport protocol is used for two main purposes: for establishing the ECN communication and for congestion control, i.e., making the appropriate decisions when there is evidence of congestion.

2.2.1 TCP support

The functionality of the TCP protocol has been extended to support the requirements of the ECN protocol. In order to do this, two new flags are defined in the Reserved Field of the TCP header[13], the ECN-echo (ECE) and the congestion window reduced (CWR) flags (as shown in Figure 2-3).

0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	Bit
Header length				Reserved				CWR	ECE	URG	ACK	PSH	RST	SYN	FIN	field

Figure 2-3: ECN field in the TCP header

The ECN initialization takes place during the TCP connection setup phase. In this process the end nodes use the CWR and ECN-echo bits to announce to the other its own ECN capability. After this, if both ends are capable, then the ECN communication is setup and subsequently the end nodes set either ECT(0) or ECT(1) in their transmissions to inform the routers in the middle that these packets can be marked. Once the ECN communication is established and a packet is received with the CE codepoint set (i.e., the CE value is set in the ECN field of the IP header), then the receiver reports this congestion to the sender by setting the ECN-echo (ECE) flag in its next acknowledgement (ACK). After receiving this flag the sender halves its congestion window for this TCP connection, in the same way that it would in the case of packet loss; and then it sets the CWR flag to report this action to the receiver. This procedure is repeated for all packets received with the CE codepoint set. In this way ECN reduces the flow in the event of congestion without relying simply on packet losses, where there is a need to wait for timeouts or triple ACKs – all of which increase the latency in reacting to congestion and increase the number of packets that were sent despite the congestion – thus further increasing congestion [15]. Note that if multiple IP packets in a TCP flow are currently in a queue and are marked by the router, then the result is a binary exponential decrease in the congestion window.

2.2.2 Support from higher layers in a RAN

The use of ECN in a radio network for identifying congestion in a data communications path needs to adapt to the radio requirements to ensure proper functioning, e.g. most radio technologies uses UDP rather than TCP, for their transport and tunneling mechanism. In the Ericsson´s patent, "Identify bottleneck in RAN transport and avoid congestion with optimized radio and network usages"[16], a mechanism for congestion control based on ECN in a backhaul network is addressed. In such an approach, ECN is enabled between the core network and base stations controlling traffic entry into these networks. The gateway and base stations are the end nodes of the ECN connection, but they are **not** the ingress or egress of an IP tunnel – rather they are end points of an ECN aware communication path. Additionally, this mechanism is compatible with the ECN tunneling feature (encapsulation of IP packet headers in tunnels [13]) of another ECN aware tunnel that already exists between end users outside of the mobile network.

The general principle is similar to the use of ECN with TCP, but some functionality is added in the *application* layer to support the functionality that would normally be provided by TCP when using ECN – but now supports UDP as a transport mechanism. This functionality comprises the ECN initialization, the setting of either ECT(0) or ECT(1), the addition of a congestion bit identifier, a notification module for echoing in case of evidence of congestion, and a decision module for controlling traffic entry.

2.3 OSPF

The Open Shortest Path First (OSPF) [17] protocol belongs to the general category of routing protocols called link-state protocols and it is classified as an Interior Gateway Protocol (IGP) due its area of use.

The main idea behind this protocol is that each router in the routing domain is responsible for describing its local piece of the routing topology using link-state advertisements (LSAs). These LSAs are then reliably distributed to all the other routers in the routing domain. Taken together, the collection of LSAs generated by all of the routers is called a link-state database (LSDB). As a result of the flooding of LSAs, all the OSPF routers within the routing domain will have the same contents in their LSDBs, except during brief periods of convergence.

Each OSPF router computes a Shortest Path First (SPF) tree using Dijkstra's Algorithm [18] and then with this information, the router updates its LSDB. The SPF tree is used to update the router's routing table.

One of the best features of OSPF is that it is a dynamic routing protocol and its response to a network topology change is rapid. In the case of a change in the network, the OSPF routers notify the other routers by flooding new LSAs, known as Link-State Updates (LSUs). Due to the LSUs the other OSPF routers within the routing domain will learn of the change in the network. When an LSU is received each of these routers will update its LSDB and recalculate its routing table. The OSPF protocol can support a large number of networks and hosts grouped together into an area or Autonomous System (AS). The routers in the same area are called intra-area routers. These intra-area routers exchange routing information with the same routing protocol, thus they have identical topological information. This topology is hidden from the other ASs' areas. These OSPF areas are interconnected via a backbone area which is referred to as area zero (0.0.0.0). A router located on the border of an OSPF area, is called an Area Border Router (ABR). An ABR interconnects the area(s) to the backbone area. ABRs leak IP addressing information from one area to another in OSPF summary-LSAs. This enables routers in the interior of an area to dynamically discover destinations in other areas (the so-called inter-area destinations) and to pick the best ABR when forwarding data packets to these destinations.

The OSPF protocol runs directly over IP and has five different message types to accomplish its operation, see Table 2-2.

Table 2-2: OSPF Message Types [17]

Type	Packet Name	Protocol Function	Description
1	Hello	Discover/Maintain Neighbors	These packets are sent periodically on all interfaces in order to establish and maintain neighbor relationships.
2	Data Description	Summarize Database Contents	These packets are exchanged when an adjacency is being initialized. They describe the contents of the link-state database.
3	Link State Request	Database Download	The Link State Request packet is used to request the pieces of the neighbor's database that are more up-to-date.
4	Link State Update	Database Update	These packets implement the flooding of LSAs. Each Link State Update packet carries a collection of LSAs one hop further from their origin.
5	Link State ACK	Flooding Acknowledgment	To make the flooding of LSAs reliable, flooded LSAs are explicitly acknowledged.

2.3.1 OSPF Functionality

When an OSPF router is initialized, it first checks its directly connected links and networks, and then detects which of these are going to participate in the OSPF routing process [19]. At this point the OSPF router creates a LSA based upon the information about its closest and directly connected neighbors. The LSA includes the interface's IP address, link cost(s), and network type.

Once the OSPF router has determined which links and networks belong to the OSPF routing process, the OSPF router starts to flood via these interfaces "Hello" messages in order to discover its neighbors. When these messages arrive at another OSPF router, this router will send a Hello message in response. As a result the initial router will receive a Hello message and then they will form an adjacency. This adjacency is a relationship with a neighboring router. When a network is in a steady state (no routers or links are going in or out of service) then the only OSPF routing traffic is periodic Hello packets between neighboring OSPF routers and the occasional refresh of parts of the LSDB. Each OSPF router constructs its LSAs containing link-state information (such as neighboring routers and links) and floods the OSPF domain with these LSAs. This flooding enables all of the other OSPF routers in the OSPF domain to receive this LSA. Finally, the OSPF routers construct their LSDB using the information from all the LSAs that they have received. This LSDB is used for calculating a SPF tree and this tree is used to update the router's routing table.

Every OSPF packet starts with a standard 24 byte header; as shown in Figure 2-4. This header contains all the information necessary to determine whether the packet should be accepted for further processing.

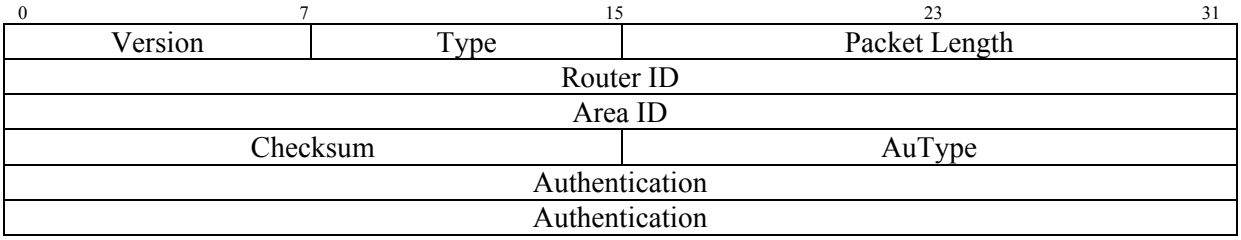

Figure 2-4: OSPF Packet Header [17]

If the link between two routers fails, then the physical or data-link protocols in the routers will probably detect this failure within a small number of seconds; as a last resort, the failure to receive OSPF Hello packets over the link will indicate the failure in less than a minute. As soon as the router detects this failure it will update its LSDB and propagate this information by re-originating its router-LSA. This new router-LSA will say that the link no longer exists. The involved routers will start to flood these new LSAs by sending it to their neighbors, and these neighbors will continue this flooding process and so on. Eventually all of the routers in the routing domain will know of the link failure and will update their own LSDB and as a result compute their new routing table entries.

2.3.2 Link State Advertisements (LSAs)

Each OSPF router originates one or more LSAs to describe its local part of the routing domain. Together all of these LSAs create a LSDB; this LSDB is used as input to the routing calculations. Each LSA provides some bookkeeping information, as well as topological information. All OSPF LSAs start with a common 20-byte header, see Figure 2-5.

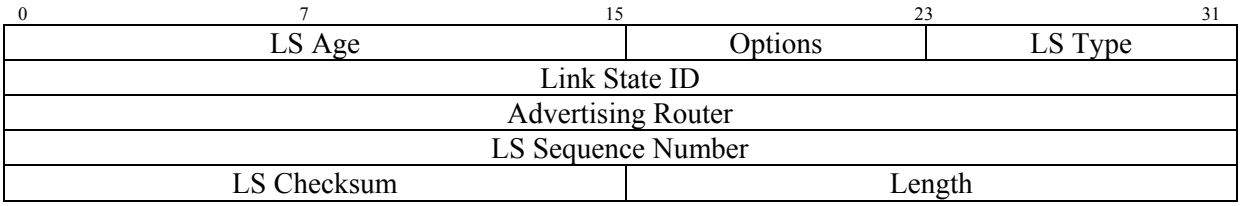

Figure 2-5: Common LSA Header[17]

The LSA header contains the LSA type, the link-state ID, and the Advertising Router fields. These fields are used to identify and verify the originality of a LSA message. These fields in the LSA header uniquely identify each LSA*. The OSPF router can originate one or more types of LSAs. Eleven distinct types of LSA are registered for the OSPF protocol[20]. Some of them have been already mentioned, but for this master thesis only worth focusing in the Opaque LSAs.

The Opaque LSAs (types 9, 10, and 11) are designated for upgrades to OSPF for application-specific purposes; for example, OSPF-Traffic Engineering (OSPF-TE). Opaque LSAs are used to flood metric information. Standard LSDB flooding mechanisms are used for distribution of Opaque LSAs. Each of these three types of LSA has a different flooding scope. In order to enable the use of these Opaque LSAs the routing domain has to be aware of its use. To do this the O-bit of the options field of the LSA header should be set. This bit of the options field is shown in Figure 2-6. In this thesis we will not discuss these other bits, the interested reader is referred to RFC 5250 [21]. By setting the O-bit routers communicate their willingness to receive and forward Opaque LSAs. Opaque LSAs are explored in detail in the next subsection.

* Modulo the effects of wrap-around of the LS Sequence Number.

DN	O	DC	EA	N/P	MC	E	MT

Figure 2-6: Options Field [21]

2.3.3 Opaque Link State Advertisement

Opaque LSAs are Type 9, 10, and 11 link state advertisements. These advertisements may be used directly by OSPF or indirectly by an application wishing to distribute information throughout the OSPF domain.

As for any LSA, Opaque LSAs use the link-state database distribution mechanism for flooding this information throughout the topology. However, an Opaque LSA has a flooding scope associated with it so that the scope of flooding may be link-local (type-9), area-local (type-10), or the entire OSPF routing domain (type-11).

As with all OSPF LSAs a standard LSA header is used, but with some differences in the link-state ID structure. The link-state ID field for an Opaque LSA consists of two parts: "Opaque type" field (the first 8 bits) and an "Opaque ID" (the remaining 24 bits). The packet format of an Opaque LSA is shown in Figure 2-7.

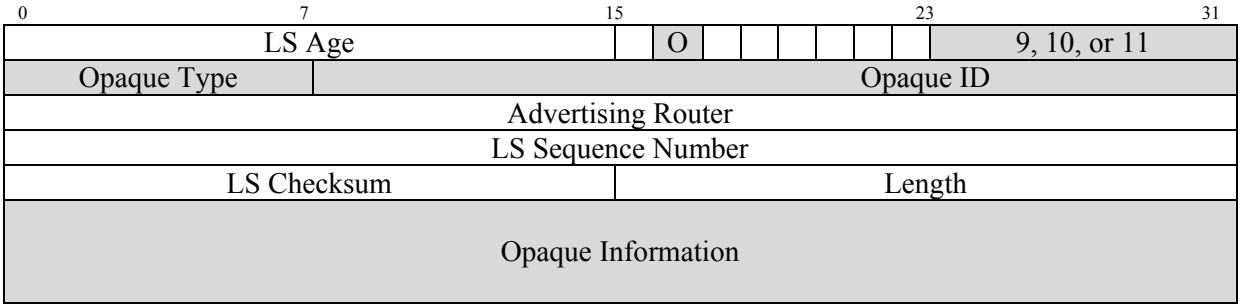

Figure 2-7: Opaque LSA Header Format [21]

The Opaque Type field defines the application for this Opaque LSA. At the moment, six type values have been defined by the Internet Assigned Numbers Authority (IANA) [22]. The values and their assignments are shown in Table 2-3.

Table 2-3: Opaque Link-State Advertisements (LSA) Option Types [22]

Value	Opaque Type	Reference
1	Traffic Engineering LSA	RFC 3630 [23]
2	Sycamore Optical Topology Descriptions	John Moy[22]
3	grace-LSA	RFC 3623[24]
4	Router Information (RI)	RFC 4970[25]
5	L1VPN LSA	RFC 5252[26]
6	Inter-AS-TE-v2 LSA	RFC 5392[27]
7-127	Unassigned	
128-255	Reserved for private use	RFC 5250[21]

In the next subsection the first Opaque Type is described, the traffic engineering LSA, as this is the only LSA type used in our project.

2.3.4 OSPF Traffic Engineering

The Traffic Engineering (TE) Extensions to OSPF [23], also known as OSPF-TE, is an upgrade of the OSPF protocol to add more information about the traffic and performance to OSPF messages. With this improvement, traffic engineering capabilities were added to OSPF. OSPF-TE uses Opaque LSAs with different flooding scopes to carry this TE information. These LSAs are similar to Router LSAs in that a TE LSA identifies the originating router and the router's neighbors, but adds additional TE parameters.

The information made available by these extensions can be used to build an extended LSDB, just as router LSAs are used to build a "regular" LSDB; the difference is that this Traffic Engineering Database (TED) has additional link attributes (e.g., bandwidth) compared to the "regular" LSDB. It is worth mentioning that if there are non-TE capable routers in an OSPF network, these routers can still forward the Opaque TE LSAs that they receive and flood them in the network, as just another Opaque LSA. Hence, an OSPF network can have both non-TE and TE capable routers, while still providing (to some extent) traffic engineering functionality.

Each TE LSA starts with a common LSA header with LSA Type 9, 10, or 11 and Opaque Type 1. The LSA payload consists of one or more nested Type/Length/Value (TLV) triplets for extensibility.

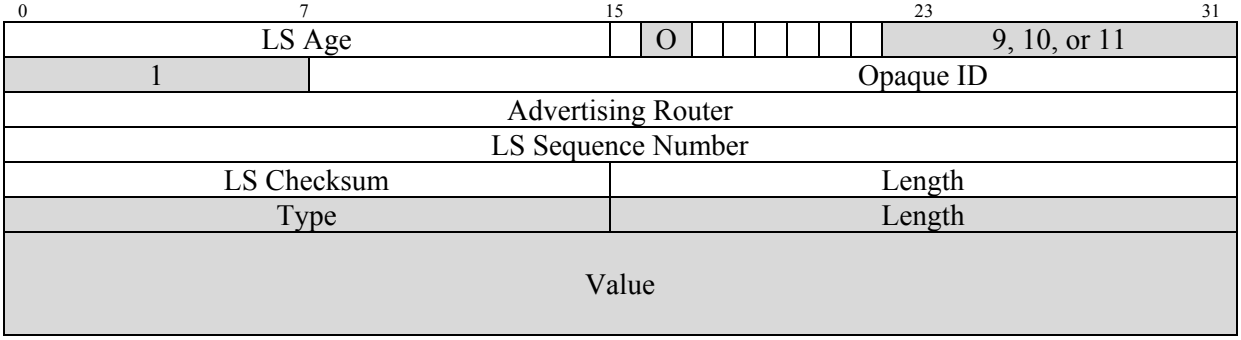

Figure 2-8: TE-LSA header format and TLV [23]

The "Type" field of a TLV triplet indicates the type of the TLV and the "Length" field contains the length of the "Value" in octets. Table 2-4 shows the IANA registered top level TLVs for OSPF-TE.

Table 2-4: Top Level Types in TE LSAs [28]

Value	Top Level Types	Reference
0	Reserved	RFC 3630 [23]
1	Router Address	RFC 3630 [23]
2	Link	RFC 3630 [23]
3	Router IPv6 Address	RFC 5329 [29]
4	Link Local	RFC 4203 [30]
5	Node Attribute	RFC 5786 [31]
6-32767	Unassigned	
32768-32777	Reserved for Experimental Use	RFC 3630 [23]
32778-65535	Reserved	RFC 3630 [23]

In this thesis the Node Attribute TLV will be an essential part of our study. This attribute describes the TE information from a single node. It is constructed of a set of sub-TLVs. There are no ordering requirements for the sub-TLVs. Only one Node Attribute TLV is carried in each LSA, allowing for fine granularity changes in topology. The Node Attribute TLV is type 5, and the length is variable. It is worth mentioning that some TLVs in a TE-LSA are constructed of a series of sub-TLVs and in this way, the Node Attribute TLV has a number of sub-TLVS defined by IANA [32] as can be seen in .

It is worth mentioning that these sub-TLVs allow a source to include relevant information about the sending node. As a result information such as delay, bandwidth, and other useful data about the router's performance can be flooded into the network and hence be known by all of the other nodes of the routing domain.

Table 2-5: Sub-TLVs types of a Node Attribute TLV[32]

Value	Sub-TLV	Reference
0	Reserved	RFC 5786 [31]
1	Node IPv4 Local Address	RFC 5786 [31]
2	Node IPv6 Local Address	RFC 5786 [31]
3-4	Unassigned	
5	Local TE Router ID sub-TLV	RFC 6827 [33]
6-11	Unassigned	
12	Inter-RA Export Upward sub-TLV	RFC 6827 [33]
13	Inter-RA Export Downward sub-TLV	RFC 6827 [33]
14-32767	Unassigned	
32768-32777	Reserved for Experimental Use	RFC 5786 [31]
32778-65535	Reserved	RFC 5786 [31]

2.4 Interface to the Routing System

Managing a network of routers running a variety of routing protocols involves interaction between multiple components within a router. A router has information that may be required for applications to understand the network, verify the forwarding plane, measure flows, evaluate routes, or to select forwarding entries, as well as to understand the configured and active states of the router. A new IETF draft "Interface to the Routing System Problem Statement"[34] proposes a new protocol to allow applications to manage or manipulate these components. This protocol is intended to incorporate and extend existing mechanisms in order to give applications appropriate access, support authentication & authorization, and to enable policies to manage these components.

The draft proposed Interface to the Routing System (I2RS) [35] architecture consists of an I2RS Client, controlled by one or more applications, and an I2RS Agent, associated with a router element. Both of these communicate over the I2RS protocol to carry asynchronous messages between the clients and the agent in order to transfer state into and out of the Internet's routing system. Figure 2-9 despites the I2RS model.

Agents gather information from different components of the router element, such as the topology database, policy database, routing and signaling protocols, Routing and Forwarding Information Base (RIB and FIB) manager, and data plane. In addition, other useful information such as measurements, events, or QoS metrics can be collected by the agent. Clients request their desired information from the agent and deliver this information to an

application. Furthermore, the clients can interact with the agent to modify and access the state of the routing element. However, some direct modifications are **not** allowed, such as updates to the link-state database, in order to preserve network state consistency. Additional architecture properties are discussed in these drafts including: simplicity, extensibility, data-models, authorization, and authentication. A deeper study of these additional properties is out of the scope of this thesis project.

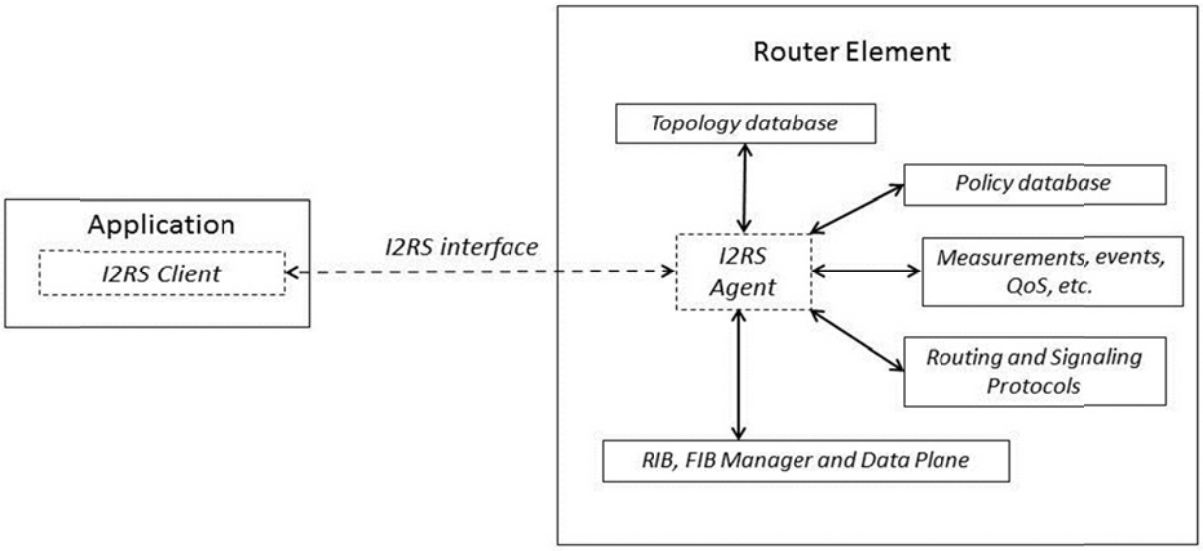

Figure 2-9: I2RS model

The IETF proposes a set of use cases of I2RS, in which the I2RS interface can be implemented. For example, Liu, Zhang, and Li describe the requirements of I2RS for network monitoring in their IETF draft "Use Case of I2RS in Mobile Backhaul Network" [36]. These requirements are basically: using centralized controllers, associated with the clients, to control and manage the entire network, while detecting traffic congestion or packet dropping in real-time. This makes state information dynamically available all of the time.

2.5 Potential sources of data for our measurements

This section describes a number of tools and measurement methods that could be used in conjunction with our experiments to collect data.

2.5.1 TCP Extensions for High performance

The TCP protocol [37] was designed to operate reliable over almost any transmission medium regardless of transmission rate, delay, corruption, or reordering of segments. TCP is utilized extensively as the transport protocol by a large amount of applications, such as web browsers and file sharing. In addition, due to the introduction of new technologies, e.g. fiber optics, which offer ever-higher transmissions speeds, some extensions for high performance have been introduced in the TCP domain [38]. These extensions solve three fundamental performance problems: window size limit, recovery from losses, and round-trip time measurement (RTTM). In this subsection we will study the extensions to support RTTM as an alternative for measuring round trip time and one way delay between the routers of the radio access transport network.

TCP implements reliable data delivery by retransmitting segments that are not acknowledged within some retransmission timeout (RTO) interval. Accurate dynamic determination of the RTO interval based on RTTM is essential to improving TCP's performance. The TCP timestamps option contains two four-byte timestamps fields: Timestamp Value field (TSval) that contains the timestamp of the TCP sender and Timestamp Echo Reply field (TSecr) that echoes a timestamp value that was sent by the remote TCP peer in the TSval field. The timestamp clock is a virtual clock proportional to the real time. This pair of timestamps is used in order to estimate the RTTM. Figure 2-10 shows the Timestamps option of the TCP header.

1	1	4	4	bytes
Kind = 8	Length= 10	TS Value (TSval)	TS Echo Reply (TSecr)	field

Figure 2-10: TCP Timestamp Header Field

In a TCP communication with peers A and B as the end points of the TCP connection, the RTTM mechanism works as follows: TCP packets sent by A carry timestamps from A's clock in TSval and echoes the last timestamp clock received from B in TSecr. In the same way, B sends its timestamp values and echoes values received from A. Both nodes dynamically calculate RTTM during the TCP connection and use it to accurately set their RTO value.

2.5.2 Clock requirements and the NTP protocol

As it is designed the RTTM mechanism no synchronization is needed between the TCP end points since each end point uses its own clock in RTTM. However, if measurements, such as one-way delay, are to be done using the TCP timestamp option, then some synchronization of clocks is required.

The Network Time Protocol (NTP) can be used to synchronize the clocks of the nodes of a network, using one or more reference clocks. This is achieved by exchanging NTP packets between each node (acting as a NTP clients) and the node with the reference clock (acting as a NTP server). The NTP packets carry information related to the reference clock in order to synchronize the client node's clock with the reference clock. These packets consist of a UDP header followed by the NTP data. For details see [39], [40].

2.5.3 Simple Network Management Protocol (SNMP)

The SNMP [41] protocol was developed in 1988 due to the need to manage ever growing networks, and the need to verify certain conditions being experienced in those networks. SNMP has three distinct versions SNMPv1, SNMPv2 [42], and SNMPv3 [43]. The different SNMP versions are very similar except for small syntax distinctions and their support or lack of support for security. All three versions follow the agent/manager model.

The SNMP architecture is composed of three major elements:

- SNMP managers are responsible for managing network devices that implement SNMP agents.
- SNMP agents reside in network nodes such as switches, routers, etc. and provide information to SNMP managers.
- Management Information Bases (MIBs) describe data objects to be managed by an SNMP agent within a device. The values in this MIB, called data objects, are exchanged in a conversation between SNMP managers and agents.

The SNMP protocol allows an SNMP manager to control an SNMP agent by exchanging SNMP messages. The main purpose of an SNMP message is to control (set) or monitor (get) parameters from a SNMP agent. In a SNMP agent, parameters are structured into a tree. SNMP uses an Object Identifier (OID) to specify the exact parameter to set or get within the tree. An OID is encoded as a list of numbers separated by dots.

Every SNMP agent has an address book of all its objects, called a Management Information Base (MIB) [44]. The MIB provides the name, OID, data type, read/write permissions, and a brief description for each object in an SNMP agent. When an SNMP manager requests information from an SNMP agent, the SNMP agent retrieves the current value of the requested information from the MIB.

The MIB defines the managed objects that a SNMP manager monitors or configures on a SNMP agent. Each system in a network (server, router, bridge, etc.) maintains a MIB that keeps track of the status of all of the managed resources on that system, such as the IP address assigned to each port or interface, the number of packets received by a specific interface, the number of packets sent by each interface, the number of packets in the queue to be sent, etc. The MIB is a dynamic database that provides a collection of managed object definitions that are continuously updated by system. The MIB defines the data type of each object and describes the object.

2.6 Open source routing software

Many open source projects have developed routing software and many of these projects are still under development and updates continue. In this project, the network routing suite Quagga was chosen. Quagga was chosen for many reasons. The main reason is that this open source routing (OSR) software supports many different routing protocols, including OSPF, RIP, BGP, and IS-IS. Additionally, Quagga provides an API to access to its OSPF daemon. Using this API it is possible to add additional functionality to routing suites without making any changes in the router's core source code. Finally another important factor that should not be underestimated is that Quagga is one of the most used open source router project and there is a large internet community which provides continued updates and support.

2.6.1 Quagga router

Quagga is a software based routing package that provides TCP/IP based routing functionalities with support for commonly used routing protocols, such as RIP, OSPF, IS-IS, and BGP. A machine where Quagga is installed acts as if it were a dedicated router. With Quagga, this computer exchanges routing information with other routers using routing protocols. Quagga uses this information to update the kernel's routing table so that the correct data goes to the correct place. Quagga allows dynamic changes to the configuration and the routing information can be displayed via Quagga's terminal interface.

A Quagga router consists of Zebra, OSPF, RIP, and BGP daemons, see Figure 2-11. The core daemon* Zebra is an abstraction layer of the underlying operating system (OS), while the OSPF, RIP, IS-IS, and BGP daemons are processes over this abstraction layer that provides routing functionality for these four different routing protocols.

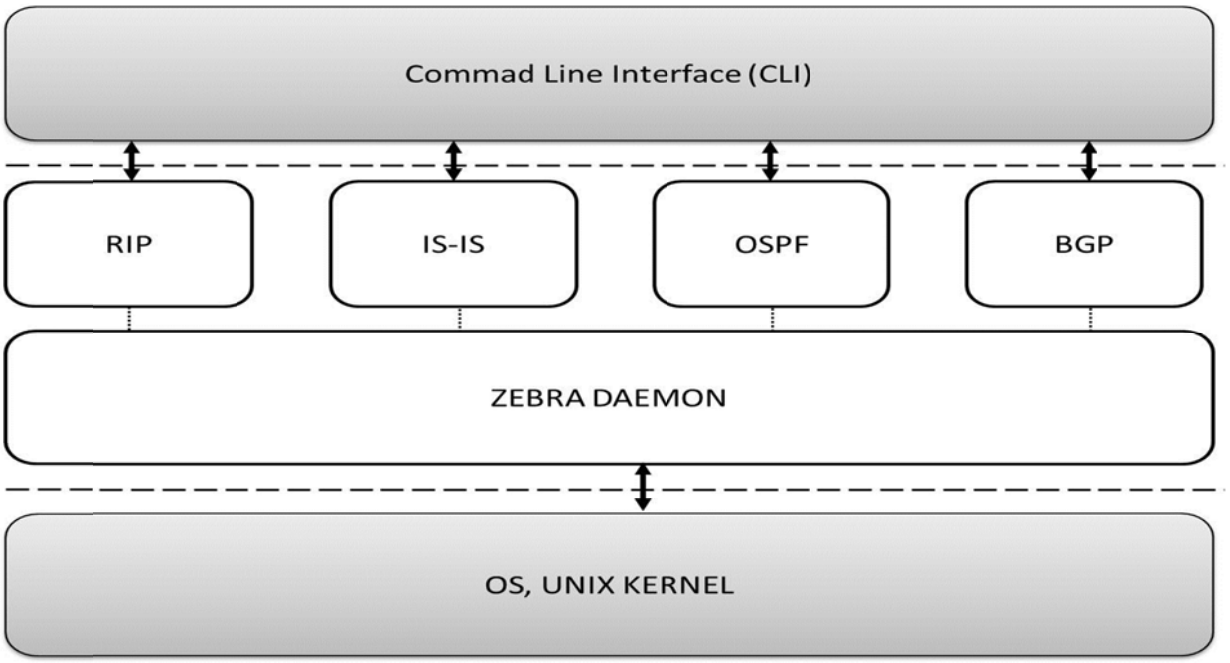

Figure 2-11: Quagga Router Architecture

The OSPF daemon is divided into different modules: an OSPF core, Opaque LSA, OSPF-TE, and OSPF API (as shown in Figure 2-12). The "OSPF core" is the module that implements the main tasks of the OSPF protocol and has responsibility for neighbor discovery and exchanging neighbor state.

The Opaque LSA module permits the OSPF daemon to exchange Opaque LSAs with other OSPF routers. The "OSPF-TE" and the OSPF API modules can generate Opaque LSAs that will be flooded by the Opaque LSA module into the OSPF domain.

Quagga send all this information to the other routers in our network in order that all these routers will converge to the same (extended) LSDB contents. Many studies have been performed to improve the switching and converge time and they confirm that with Quagga it is possible to have better routing performance than with commercial routers [45].

* In multitasking computer operating systems, a daemon is a computer program that runs as a background process.

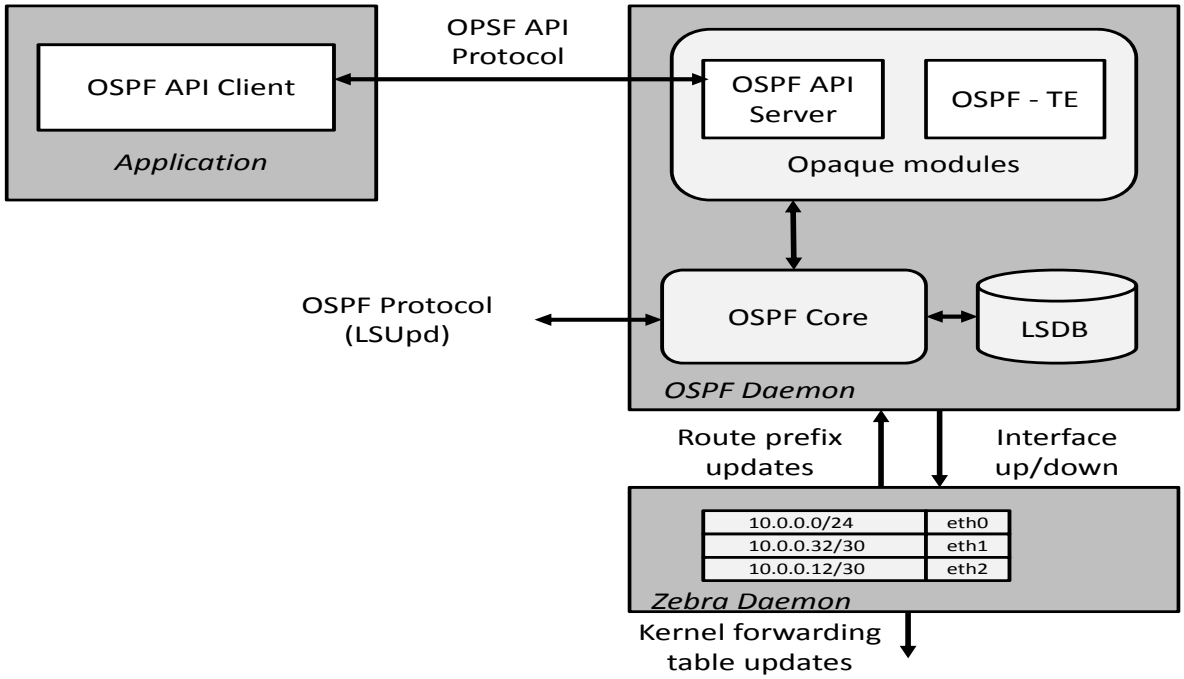

Figure 2-12: Quagga OSPF Daemon Architecture

2.6.2 OSPF API Extension

The OSPF API is divided into two parts: OSPF API-Server and OSPF API-Client. The OSPF API-Client can establish a TCP connection with the OSPF API-Server using the OSPF API Protocol [46]. Using this API, applications can get the complete or partial of the LSDB of the OSPF daemon. Additionally, they can originate application-specific LSAs. Opaque LSAs are then transparently distributed to other routers within the LSA's flooding scope and can be received by other applications through the OSPF API.

External applications connect via the OSPF API client library. The OSPF client library provides an API that directly accesses the LSDB and can transparently distribute customized Opaque LSAs. Given a request from an external application, the client library establishes a connection to the OSPF daemon to retrieve LSA updates and emit Opaque LSAs. The OSPF API server module can handle multiple clients concurrently.

The OSPF API protocol phases are explained below and illustrated in Figure 2-13:

Connection initiation — The connection between the OSPF API-Client and the OSPF API-Server is based on two channels: one for synchronous and the other for asynchronous messages. Requests operate synchronously using a two-way scheme, while notifications operate asynchronously by sending messages one-way from either the server or the client. These notifications are one-way messages and can be a result of new, updated, or deleted LSAs received via the network, changes of neighbor connectivity, or failures of interfaces.

Link-state database synchronization — After communication has been established between the OSPF API-Client and the OSPF API-Server, the client application can send a LSDB synchronization request to the server. The OSPF API-Server will send its entire internal LSDB by sending a sequence of LSAs update notifications to the OSPF API-Client.

Opaque type registration	Once LSDB synchronization has been achieved, the OSPF API-Client can register the Opaque type which it wants to originate as an Opaque LSA.
Origination of own Opaque LSAs	Once the OSPF daemon has learned that an Opaque-capable neighbor's state is complete, then the OSPF API-Server will notify the OSPF API-Client that it is ready to flood Opaque LSAs. There must be at least one Opaque-capable neighbor before a router can originate Opaque LSAs. The OSPF API-Client application can originate its own Opaque LSAs and then invokes the OSPF demon to flood them throughout the OSPF network.
Update own Opaque LSAs	The OSPF API-Client can update the content of self-originated LSA and asks the OSPF API-Server to re-flood it with new content.
LSA updates from neighbors	Whenever an LSA of any type is received over the network, the OSPF daemon immediately notifies all applications.
Deletion of own Opaque LSA	The OSPF API-Client application can delete its Opaque LSA from all OSPF routers by sending a deletion request to the OSPF API-Server.
Connection shutdown	When an application terminates, it should first delete all self-originated Opaque LSAs before it shuts down its connection to the OSPF daemon. If the application closes the connection without issuing delete requests, then the OSPF daemon takes care of cleaning up obsolete Opaque LSAs, making sure that no stalled Opaque LSAs remain within the network.

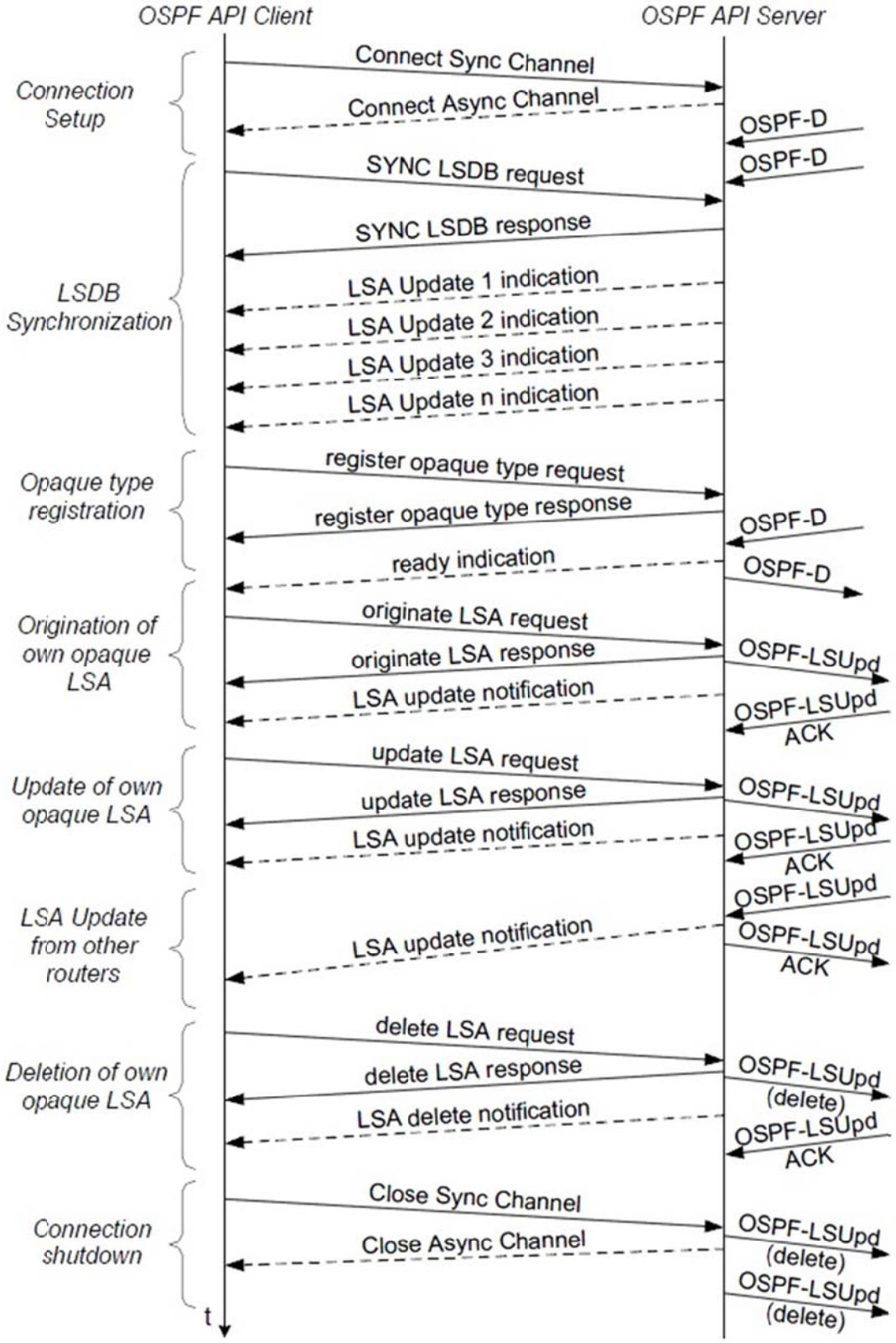

Figure 2-13: The OSPF API Protocol phases[46]

2.7 Related work

In this section, several previous studies conducted in areas relevant to this master's thesis are summarized. This previous research can be divided into the following topics: Congestion Flow Control in a RAN, Congestion Notification in a RAN, I2RS, and Monitoring & Flooding Methods.

In the field of the Congestion Flow Control, Rácz, Pályi, and Nádas researched mechanisms to control and identify congestion in the RAN, but their studies were limited to the performance of transport network layer and the associated data packets flows. Rácz, Pályi, and Nádas [47] describe two different methods that when combined improve the performance and fairness of the congestion identification of the transport flows.

Nádas, Nagy, and Rácz have also worked on congestion identification and avoidance for the air interface, specifically with HSUPA [48], [49]. In this research they propose a per-flow HSUPA transport network flow control algorithm. All of their flow control studies are related to the topic of this master's thesis, but they do not cover the same breath as our work.

Work has been done to identify and notify entities of congestion in the RAN. Kekki [50] has researched and patented a method, apparatus, and system to notify entities of the occurrence of congestion in a 3G radio access network. This work could be a good combination with the approach proposed in this master's thesis, because while Kekki focuses more on radio access network congestion notification, this master's thesis focuses more on the identification of radio access transport network congestion. Companies are also working and researching in transport TE. ARISTA has developed a network analyzer LANZ [51] that provides queue congestion notification in the network with a granularity of microseconds. Ericsson has continued research in this field and one of the last documents is a U.S. Patent application P10259PC00 [16]. The ideas described in this patent application were the starting point for this master's thesis. By combining the methods and procedures proposed in this master's thesis it is possible to identify congestion in the RAN and provide this information to the system as this patent application proposes. In combination with the two previous patents and the work presented in this master thesis, we introduce a complete congestion avoidance method and set of control management tools which provide different functionalities, for example improved handover control.

Active control and management of the RAN is becoming more and more relevant as the demands on this network increase. To provide network nodes with the ability to react to congestion and interact with the different control modules of the network is a major research challenge. The AQM tools have helped with this task as Pacifico et al. discuss in [15], but the next step in network management is to provide an I2RS to the network to enable management of the network. As explained in Section 0 the IETF is still developing I2RS and many drafts have been released [34]–[36]. With the functionalities of this new interface and the methods that this master's thesis proposes to provide congestion information, it should be possible to manage all this information, react, and interact with the network nodes in order to avoid and control the congestion in any module of the RAN.

Another topic related to this master thesis is monitoring and flooding methods. Keller [46], [52] has proposed a client-server method for use in the router node. This solution has been adopted in this master's thesis project in order to acquire the monitored information and then, forward and flood this information to the rest of the network. This method, as explained previously in the OSPF API Extension Section 2.6.2, was implemented by Keller in the Quagga routing software. This implementation provided us with an important and helpful resource. Amir Roozbeh has also supported and used, in his master's thesis [53] the work of

Keller and, in the same way as this master's thesis proposes, he has used OSPF-TE TLVs to flood a network with information about monitored resources. While the research field of Amir Roozbeh´s master's thesis is completely different from the RAN, his work and Amir himself have provided ideas to this master's thesis project with respect to monitor the congestion of resources and flooding this information into the network.

Finally, a lot of research has been done in order to select the most suitable method to monitor the different resources of the network. This discussion is extended in the next chapters of this master's thesis. Earlier studies have been made in order to measure the delay [54], [55] and bandwidth [56]. This research has been taken into account as we tried to select our method to monitor these and other properties of the resource of the network.

3 Design of a Congestion-Identification Mechanism

This chapter describes the architecture of a mechanism for identifying congestion in a radio access transport network. The mechanism is structured into different modules and elements which are explained and discussed, as well as the relationships between them. The chapter starts with an overview of the proposed solution and continues by focusing in each of the elements (Real-Time Traffic Aware Router and Receiver), describing modules within them and its relationships. The design ends a summary of the structure of the design.

3.1 Overview of the Design

Our Congestion-Identification solution is designed to be a mechanism for use in a mobile operator's network, such as an LTE network. The goal of this mechanism is to identify congestion within the radio access transport network. This mechanism is based upon the interaction of two different actions: real-time monitoring **for evidence of congestion and** collecting **measurements (such as bandwidth, round-trip time, etc.). To understand how this mechanism operates we will consider**

Figure 3-1 from a conceptual point of view as it shows the relevant elements that we will consider via a scenario.

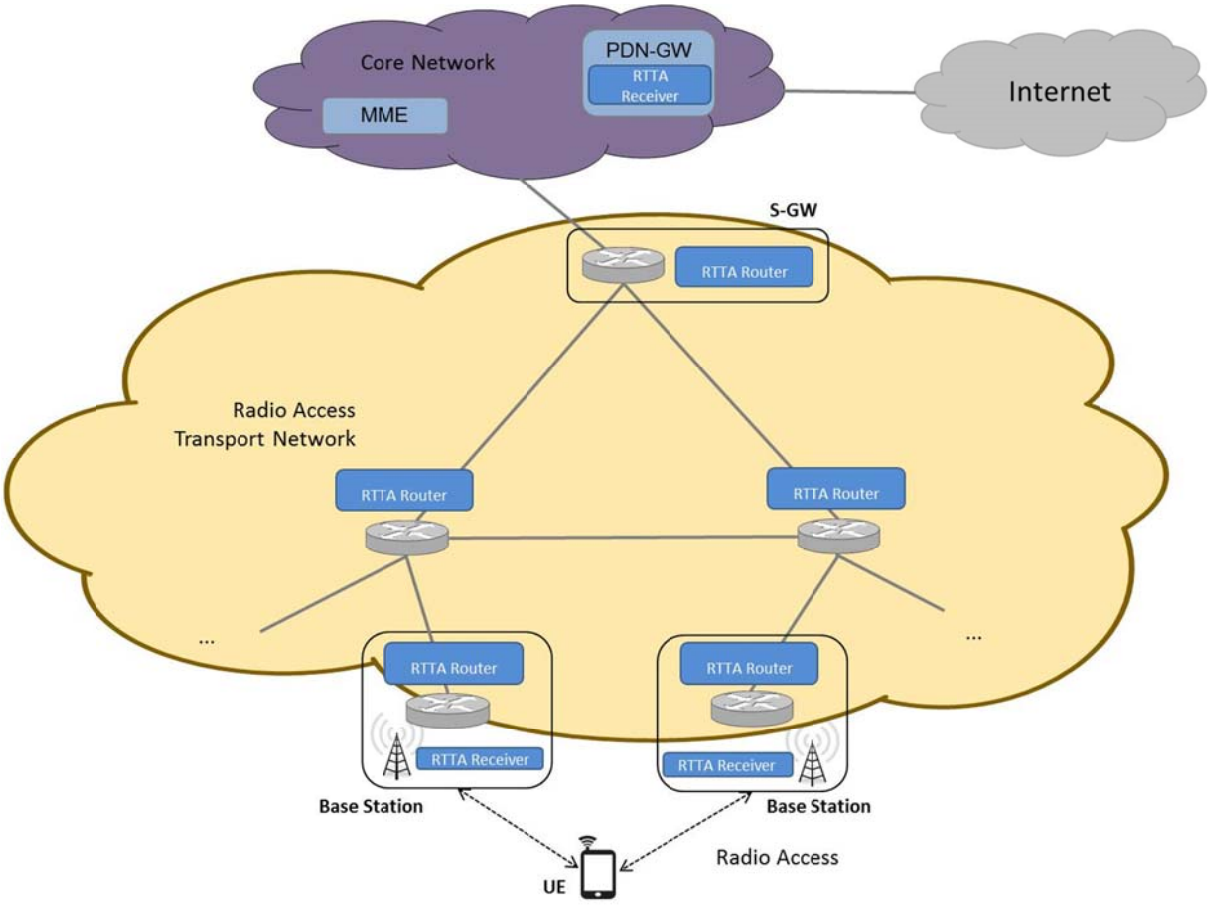

Figure 3-1: Congestion-Identification scenario

Two new elements are introduced in the proposed Congestion-Identification solution: the Real-Time Traffic Aware (RTTA) Router and the RTTA Receiver:

RTTA Router	The RTTA Router is an extended router which has the same functionalities as a "normal" router in such network, but has an extension for measuring and forwarding relevant information regarding its interfaces. This type of router is located at all the nodes of the radio access transport network, as well as in the S-GW and the base stations.
RTTA Receiver	This is a new element is located in the PDN-GW and in the base stations. This element is responsible for detecting evidence of congestion in real-time and collecting the information forwarded by the RTTA Routers. The RTTA Receiver also combines and presents the information in a suitable way in order to enable further actions (e.g. dropping some UEs' connections or initiating a handover for one or more UEs).

The mechanism presented exploits the interaction of two sub-mechanisms: one based on ECN and the other on OSPF-TE. The information provided by these two sub-mechanisms is integrated by the RTTA Receiver.

On one side, ECN communication occurs between the PDN-GW and the base station in order to detect congestion in real-time as was described in Section 2.2.2. The RTTA Routers mark packets depending of the state of its queues and the selected policy. The RTTA Receivers detect evidence of congestion as packets pass through the PDN-GW or base stations. At the same time, via the other sub-mechanism, each RTTA Router is measuring relevant information (e.g. available bandwidth, one-way delay, round-trip time, number of packets marked, etc.) for each one of its links. This information is flooded over the network using OSPF-TE Opaque packets depending upon the policies implemented in the RTTA Routers. Thus, each RTTA Router maintains updated information about all of the other nodes in its OSPF-TE database and the receiver simply accesses this information via one of the RTTA Routers, usually the closest one. Finally, the RTTA Receiver combines the information from the ECN connection with measurements from each node, in order to characterize congestion in the network. This is presented in two different ways: in real-time and periodically.

The next sections of this chapter describe this mechanism in greater detail by defining each element together with its new modules and its interactions with other modules.

3.2 Real-Time Traffic Aware Router

The Real-Time Traffic Aware Router (RTTA Router) is one of the main elements of the proposed Congestion-Identification solution. This element provides congestion information to the whole network. In order to supply this information, it must be situated in all the routing nodes of the radio access transport network, including the routers situated in the eNodeBs and the S-GWs. The RTTA Router element is divided into the following modules: measuring, monitoring and policies, RTTA sender, and OSPF router. Together these modules measure the congestion of resources associated with each interface of the routing node.

The design of the mechanism proposed has been done using the OSPF-TE protocol, but it is worth mentioning, that other protocols can take its place in a real scenario. The Interior Gateway Routing Protocol (IGRP) [57] and its optimized version Enhanced IGRP (EIGRP) [58] provides similar updating and neighboring detection features as OSPF, and despite they don't warranty the use of the best route they are easier to configure than OSPF. These protocols also spread the traffic over the network and discover different paths between source and destination, but two main factors provoke its denied: They are Cisco property what

implies that its use is more restricted [59] than OSPF and furthermore, the stable implementation of the OSPF-TE in the open source routing software convinced the work team to choose this protocol and the features that it provides as the more suitable one for this design.

By using the OSPF-TE protocol, the RTTA Router constructs a TLV; includes it in an Opaque LSA and injects this LSA into the network in order to flood the radio access transport network with information concerning the congestion observed by this router node. The result is that all routers have an updated database with the relevant congestion information (even performance information) for all of the links of every router in the radio access transport network. Figure 3-2 shows the modules of a RTTA Router. Each of these modules will be explained in detail in the following sections.

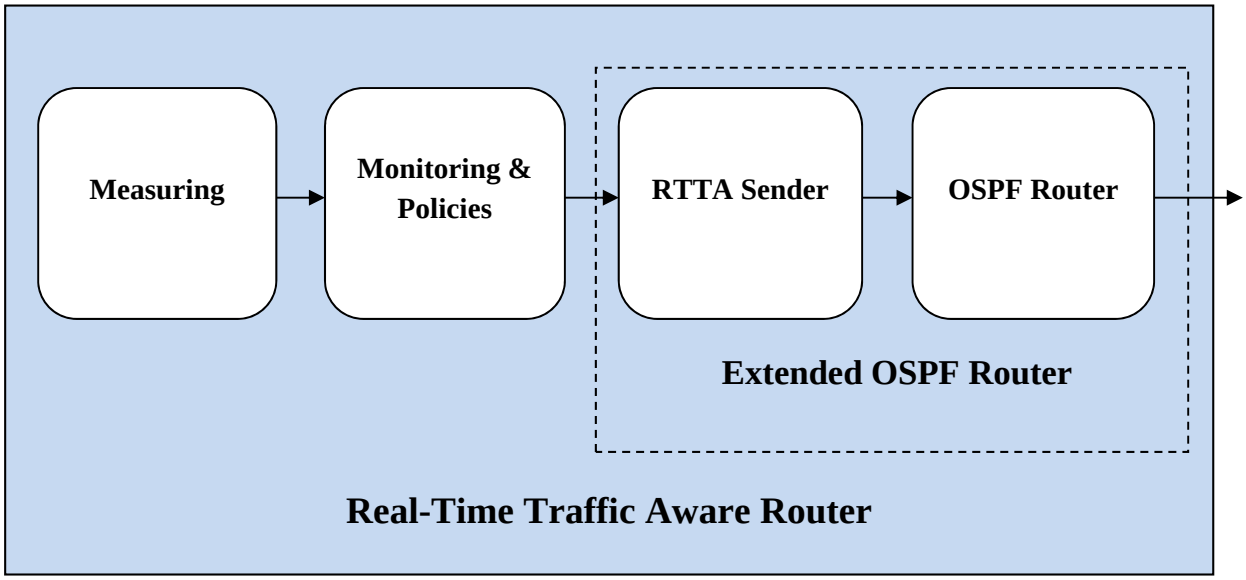

Figure 3-2: Architecture of the Real-Time Traffic Aware Router

3.2.1 Measuring

The first module has the task of measuring the traffic for all the interfaces of the RTTA Router. For each interface, it measures selected traffic parameters.

In order to obtain this data the module has to identify the interface. This is done utilizing SNMP. We will use the (first) IP address bound to this interface as the identifier for a set of measurements. This set comprises the available bandwidth, the one-way delay, the RTT of the link that is connected to this interface; and the number of packets marked by the Random Early Detection (RED) queue as Congestion Experienced (CE) packets. The design of these different measurements required more research. Details of this research will be explained in this section and section 4.1.

3.2.1.1 Available Bandwidth

Many ways to measure the available bandwidth have been studied. Some tools such as Pathchar [56] or BART [60] have been considered as possible ways to measure this parameter, but the main problem of these methods is that they load the link by sending a number of probes, thus adding more traffic to what may be an already congested link. Unfortunately, adding more traffic in this situation is counter-productive. These methods were mainly designed to be used to discover the available bandwidth along a whole path (for example, from the eNodeB to the S-GW). These methods are not very convenient for measuring the bandwidth of a link.

We decided to use the SNMP, as SNMP can directly provide the number of bytes sent and received by each interface. Given this data we can simply calculate the arithmetic mean, from which the total bandwidth used can be obtained. If the total bandwidth of the link is known, then the used bandwidth can be subtracted from this total bandwidth, to give the available bandwidth.

This choice was made in order to provide a simple (simply access a SNMP counter and perform the arithmetic operations) and lightweight (because the network is not loaded with any additional packets) method to obtain the available bandwidth. If the available bandwidth for a whole path is desired, then the tools referred to above are better options – or *within* the radio access transport network we can exploit the fact all of the routers know the available bandwidth for each link, hence they can simply compute the minimum link bandwidth along any path.

3.2.1.2 Round-Trip Time and One-Way Delay

In order to identify congestion of a link, the measurement of the link delay provides essential information. Two possible ways to learn this delay were considered: measure the one-way delay (the latency from a router to its neighbor) or measure the RTT (the round-trip time of a packet from a router to its neighbor and the return). Both measurements provide sufficient information to identify congestion, but are conceptually quite different. For instance, RTT also provides information about the time required for the forwarding processing in the neighbor router. Using one-way delays measures both uplink and downlink delay separately. These two different latencies can be quite different depending upon the current network conditions. For making one-way delay measurement it is mandatory to synchronize all the nodes' clocks in the network. Fortunately, we can use NTP at each node of the network to synchronize the local node's clocks with a single NTP server. Now that all of the clocks are synchronized we can measure the one-way delays to each neighbor.

The measurement method has two different alternatives: one based on communications between routers to share timestamps and the other using TCP timestamps.

In the first method, each router sends another router its current time in a timestamp. Figure 3-3 illustrated the sequence of messages of the proposed design.

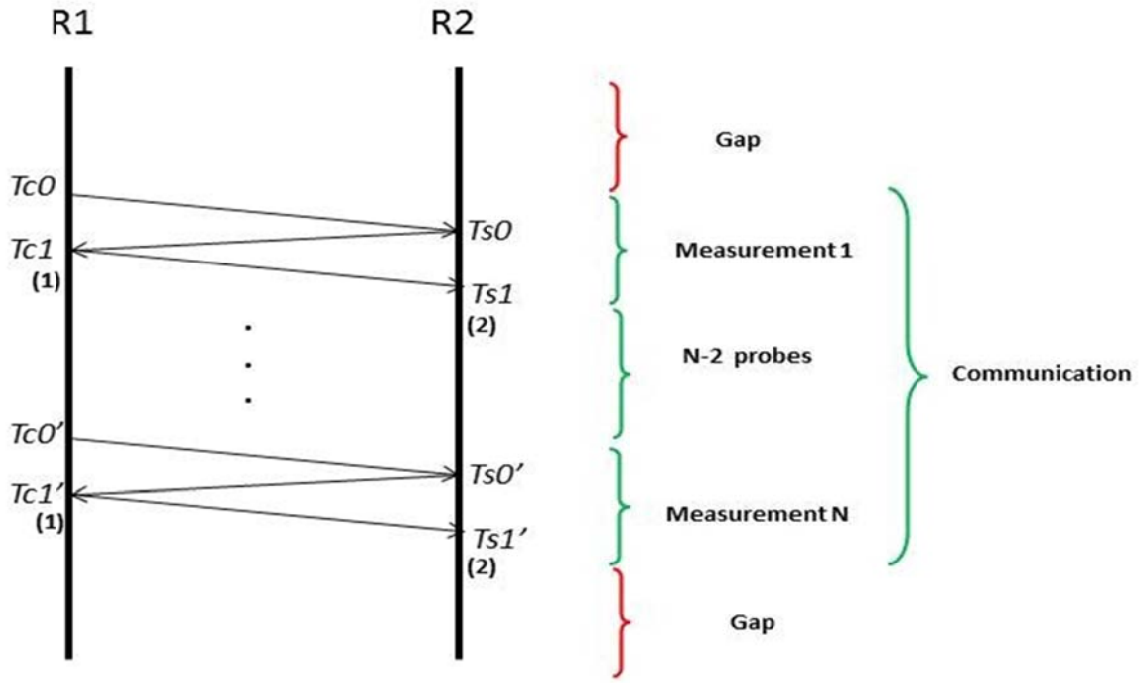

Figure 3-3: Sequence of messages between two routers, R1 and R2, for measuring RTT and one-way delay, based on sharing router timestamps

For each measurement N probes are sent. Using each probe the one-way delay and the RTT are measured by each end of the link. Each probe consists of three packets of same size, each containing the timestamp of the router who sent this packet. Using these three messages each router is able to compute the RTT and one-way delay as it is described with the equations given in Table 3-1. In addition, Table 3-1 and Figure 3-3 indicate the moment when each measure is obtained.

Table 3-1: Timestamp Equations

	Client	Sender
One-Way Delay	$T_{c1} - T_{c0}(1)$	$T_{s1} - T_{s0}(2)$
RTT	$T_{c1} - T_{s0}(1)$	$T_{s1} - T_{c1}(2)$

After the N probes, the averages of these two series of measurements are computed. These two averages provide the input values for the next module in the RTTA Router. Each router waits for a predefined gap time before it performs another set of measurements to compute the next two values.

The alternative method uses TCP timestamps, as described in section 2.5.1, by adding a Timestamp field to the TCP header. Using this procedure it is possible to provide a reliable and accurate measurement of both round-trip and one-way delays. This mechanism also requires sequences of packets and its operation is similar to the method described above. However, if this timestamp is not already in the TCP packet header, then this method requires that the timestamp needs to be inserted into the TCP packet header. Unfortunately, this will

increase the processing in the routers, but has the advantage that it can be piggybacked on top of existing traffic – hence only requiring the number of bytes for each timestamp to be added to existing packets. This approach discussed in detail in section 4.2.1, but due to software limitations of the nodes used in the test-bed, the first alternative was chosen for this the measurements used in this thesis.

Other active methods could have been chosen, as Pathchar or other ICMP based tools [61]. These tools have been researched and considered, but they are not suitable for our solution since they use are designed estimating the bandwidth of paths through network, rather than for individual links. Additionally, according to Strowes [54] these techniques are less accurate than the results provided using a passive method, as a TCP timestamp.

3.2.1.3 Marked packets

The main purpose for measuring this parameter is to identify which nodes have been involved in marking packets after receiving a packet with the ECN congestion experienced bit set. This parameter helps us to identify the congested link. At the beginning of this thesis project, this parameter was not going to be sent. Instead the first choice was to simply send the size of the interface queue. We thought that this parameter could easily be obtained from the MIBs via SNMP, but this method had to be rejected because of limitations of the software used in the test-bed. This alternative will be explained in detail in section 4.2.1as this method could be implemented when using other systems. Moreover, consideration of this method guided us to a similar and complementary solution.

A Random Early Detection (RED) output queue with ECN marking was implemented on each interface. With this design the total number of packets marked as Congestion Experienced can be obtained giving us direct evidence of congestion. This method is more intuitive than the queue size parameter that was studied early in this thesis project.

As it has already said, the set of measurements is made for each interface. When the module has finished measurements for each interface it passes this information on to the next module.

3.2.2 Monitoring & Policies

The next module implemented in the RTTA router has two tasks: monitoring and deploying policies. Throughout this explanation this module is also referred to as the monitoring or decision module, depending on which task is being considered at that moment, although both functionalities are found in a single module. After the measuring module, the router continues monitoring in real-time the measurements that have been made. This monitoring module, for each router interface, receives the latest set of measurements and then decides if the information is relevant to send to others. If the changes in measurement values are determined to be relevant, then the router will forward these measurements to the next module, otherwise the latest measurements will not be forward.

The importance of this decision module is worth mentioning as it prevents the unnecessary flowing of updates that might unnecessarily load the network. To avoid this, the decision module implements policies to determine for each of the different parameters measured when a change in the measurements is relevant to other nodes in the network. The implementation of these policies is explained in the next chapter. Here we simply state that if any of the measurements has a relevant change, then the whole set of measurements are forwarded regardless of whether they have changed or not.

3.2.3 Extended OSPF Router

This module is divided in two sub-modules: RTTA Sender and OSPF Router. The RTTA Sender constructs the Opaque packet while the OSPF Router floods the LSA to the network.

3.2.3.1 RTTA Sender

The RTTA Sender sub-module is one of the two sub-modules of the module that extends the standard OSPF router module. The main task of this sub-module is to build an Opaque LSA packet with the relevant traffic congestion information obtained from the previous modules. Section 2.3.4 gives the necessary background about the construction of this packet. The LSA type field is 10 because the Opaque LSA is to be flooded throughout the whole OSPF area. The Opaque Type chosen is type 1, as this corresponds to a Traffic Engineering LSA and this packet contains information relevant to traffic control. The next field is a TLV. The TLV type is 5 because the packet contains Node Attributes, and finally a sub-TLV with type 4, an unassigned sub-TLV type [32]. In this final field the set of measurements are included, as shown in Figure 3-4.

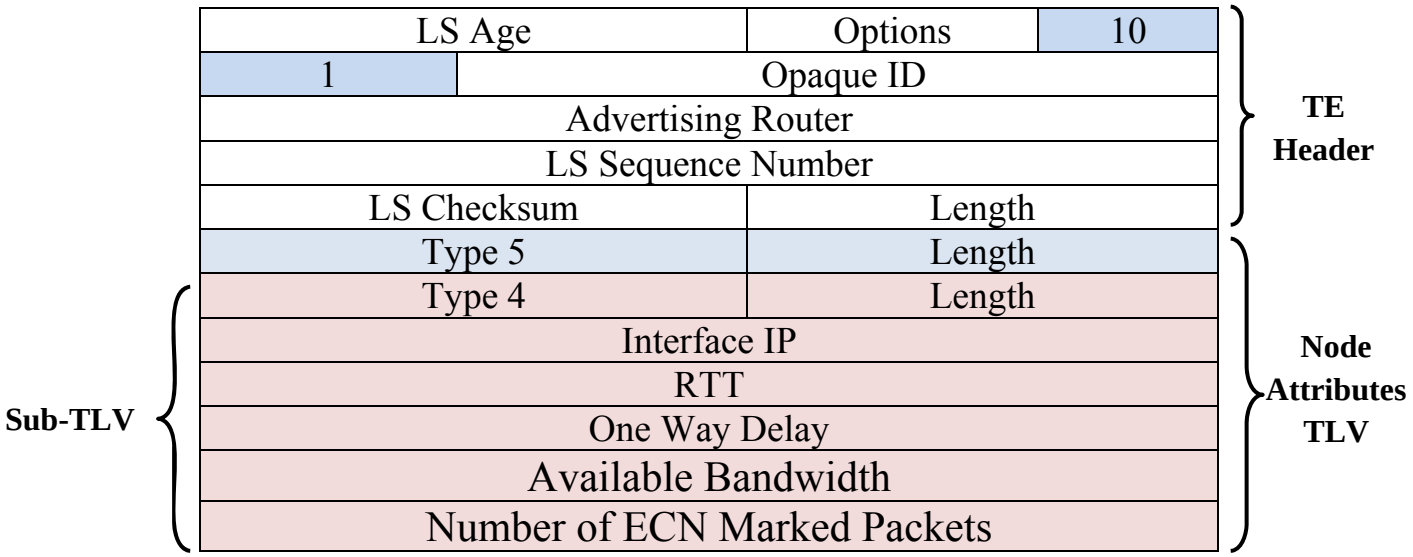

Figure 3-4: LSA-Opaque Packet

The details of the construction of this packet were a major point of discussion and research by the team. Many options were researched, three of which are shown in Figure 3-5. Instead of sending only one sub-TLV with the parameters as shown in Figure 3-5 (a), two other options could have been chosen: send the parameters in a different sub-TLV field each one but in the same packet Figure 3-5 (b) or simply send one parameter in each sub-TLV and packet Figure 3-5 (c). Each of these alternatives has its own benefits and weaknesses. Selecting the best alternative requires additional research and analysis which are out of scope of this thesis project. In this master's thesis project we have followed the advice and research of Amir Roozbeh [53] and placed the parameters in the same TLV field and in the same packet. This alternative us allows to distinguish the measurements of each interface based upon the IP address of interface, thus, a larger sub-TLV is a better solution that the other alternatives.

(a)

Type 5	Length
Type 4	Length
Interface IP	
RTT	
One Way Delay	
Available Bandwidth	
ECN Marked Packets	

(b)

Type 5	Length
Type 1	Length
Interface IP	
Type 2	Length
RTT	
Type 3	Length
One Way Delay	
Type 4	Length
Available Bandwidth	
Type 5	Length
ECN Marked Packets	

(c)

Type 5	Length
Type 4	Length
Interface IP	

...

Type 5	Length
Type 4	Length
Bandwidth	

Figure 3-5: Three alternative Sub-TLVs proposals

It is worth mentioning that this decision has a big influence on the decision task of the previous module. If we had chosen to send just one parameter in one TLV in one packet, then the policy would check if each parameter has had a relevant change and would forward just those parameters, instead of forwarding all the parameters when any one of the parameters has a relevant change, was done in this design. This decision was supported by the fact that the effort to send one parameter is nearly the same effort as required to send all of them.

When the Opaque LSA is created it is forwarded to the next sub-module of the Extended OSPF Router.

3.2.3.2 OSPF Router

The last sub-module of the Extended OSPF Router is also the last step performed by the RTTA Router. This module plays the role of a "normal" OSPF router. When it receives the LSA built in the previous sub-module, it will inject this LSA into the OSPF network in order to flood the area with this traffic information.

As an OSPF router, it has all the functions and features of an OSPF router. One of these functions is to create a forwarding table with the information necessary to route packets that arrive at this router. To do this it creates a database containing all of the sub-nets of the network, its neighbors, and the shortest path to reach each sub-net. Another feature, essential to understand the design of the elements of this network, is that an OSPF router has the functions necessary to create a TED. This TED contains information from all the Opaque LSAs with TE information that was flooded to the network. Thus, a RTTA Router that injects Opaque LSA packets in the network is at the same time receiving Opaque LSAs from the other RTTA Routers and updating its TED based upon the information that these packets contains.

3.2.4 RTTA Router Summary

After the above explanation of the RTTA Router, it is worth summarizing the main ideas and the features that this router realizes. This RTTA router collects relevant traffic information (bandwidth, latency, and ECN marked packets) from all of its interfaces and floods the network with this information, thus all the traffic information from all the interfaces of all the routers in the network is stored a TED at all of the routers in the network. Therefore, except for periods of convergence the same information should be stored in all of these TEDs.

3.3 Real-Time Traffic Aware Receiver

This section describes the design of the RTTA Receiver and its role in the proposed Congestion-Identification solution. The RTTA Receiver may be located in both PDN-GWs and the eNodeBs. In this way, the information collected by the RTTA Receiver can help these nodes to manage and handle the traffic associated with UEs making use of the PDN-GW and eNodeBs. This element is responsible for integrating the two sub-mechanisms described above (i.e., OSPF-TE information and the CE detection), by collecting, sorting, and analyzing the information that has been obtained.

The RTTA Receiver can be divided in two modules (as shown in Figure 3-6): one for collecting all the congestion information and another in which the integration and analysis are performed. Figure 3-6 illustrates the architecture of the RTTA Receiver and the relationship between the two sub-modules. In the "Collecting of Congestion Information" module, the RTTA Receiver accesses the TED of the router situated in the same node as it (i.e., eNodeB) or to the closest one in case of none router is not located in either of both entities; and at the same time, it acts as a detection module (as described in the case of an ECN connection in section 2.2.2). Then, this information is processed by the RTTA Receiver in the "Analysis and Visualization" module, in which information is sorted and analyzed from two different perspectives: in real-time (where the information is processed as it is collected) and periodically (i.e. information is captured for a period of time and stored for subsequent analysis).

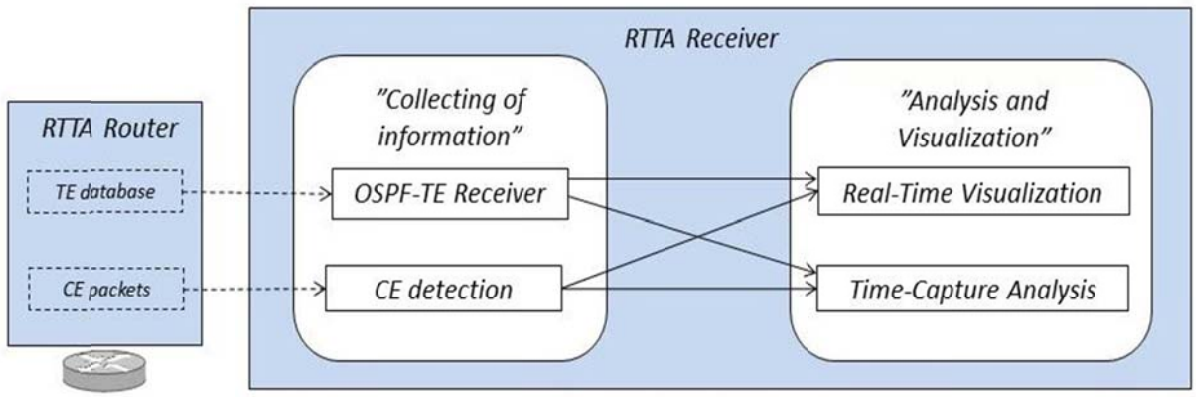

Figure 3-6: Architecture of the Real-Time Traffic Aware Receiver

3.3.1 Collecting of information module

As stated earlier, this module has two functionalities: OSPF-TE Receiver and CE detection. Both of them gather congestion information that will be the input for the next module.

OSPF-TE Receiver — The OSPF-TE Receiver functionality is responsible for collecting information located in the TED of a RTTA Router. Every RTTA Router maintains an updated TED, due to the use of OSPF-TE mechanism, with measurements for each link of the network. The OSPF-TE Receiver dynamically access and downloads this database. The communication between the OSPF-TE Receiver and the RTTA Router remains established from the moment that the RTTA Receiver starts to work. When a new Opaque LSA from any of the other routers arrives in the TED, the OSPF-TE Receiver downloads it. This design, in which the RTTA Receiver downloads the LSAs on the fly, is consistent with the inclusion of policies in the RTTA Routers for deciding when a new Opaque LSA should be sent, avoiding flooding the network with unnecessary Opaque LSAs and reducing the need to handle redundant information*.

CE detection module — The CE detection module is in charge of detecting CE packets by tracking packets marked (by sniffing CE packets) in the interface of the router where the receiver is located. It is worth to mentioning here that between logical pairs of PDN-GW and eNodeB an ECN connection is setup and the routers in between them mark packets depending of the state of their queues and the policies selected in those nodes.

The collected information is delivered to the next module for analysis, visualization and different actions or decision making.

3.3.2 Analysis and Visualization module

This is the module of the RTTA Receiver, the last step in processing the congestion information occurs. The information from the measurements carried out in the RTTA routers and the evidence of congestion detected either in the PDN-GW or the eNodeBs is processed. The purpose of this module is to help the eNodeBs or PDN-GW take further actions. The

* However, since the update containing all of the parameters is sent if any of the parameter changes are determined to be relevant there may be some amount of handling of redundant information.

analysis and visualization module has been separated into two sub-modules: one for real-time processing and the other for periodic processing. The reason is to exploit the latest update about network properties in real-time actions; while in parallel being able to study the behavior of the network for periods of time in order to consider other types of actions that concern longer term properties or to guarantee stability for a longer period of time.

- The real-time functionality monitors the state of the network by comparing the parameters (available bandwidth, RTT, one-way delay, number of packets marked) for each link of the network and computing from them different characterizations of the state of the network. At the same time, an ECN indication monitors the number of marked packets detected by the RTTA Receiver to complement the information about how many packets were marked by each of the routers.

  Many characterizations can be considered here, such as comparing the information by interface, router, link, or path. However, the current design of this module focus on presenting the information aggregated by paths, since this is a convenient characterization of the congestion information in a network, as will be discussed in later chapter. This visualization gives agile feedback about the traffic conditions of the radio access transport network that potentially allows elements of the mobile network (such as eNodes, PDN-GW, or MME) to make optimized decisions, such as initiating a handover or dropping one or more packets in order to guarantee end users a high QoE.

- Periodic Analysis consists of study the network properties over a period of time. Many characterizations can also be considered here. Following the same modus operandi as in real-time analysis, we focus on comparing the congestion information of different paths through the network. In this way, we hope to observe *trends* of the measured parameters in order to understand the behavior of the network over a longer period of time. The purpose of this functionality, besides monitor the traffic conditions over a period of time, is to study if actions motivated by the real-time analysis module are consistent with keeping the network stable over a longer period of time.

3.4 Summary of the design

Throughout this section, the design of the proposed Congestion-Identification solution has been described, giving the reader the necessary background to understand how this proposed solution works. Subsequent chapters will describe how this solution has been realized in a prototype and analyzed by using a test bed.

The proposed solution is based on the coexistence of two sub-mechanisms and their interactions. The RTTA Routers measure, monitor, decide upon relevance, and inject Opaque LSAs with information about RTT, one-way delay, bandwidth, and the number of ECN marked packet at each interface to its links. By this method, the relevant congestion information of the whole routing system is located in all the nodes of the network. In order to access to this information, this design performs a unique communication between the router and the receiver that allows transferring state out of the routing system, following the I2RS mechanism described in the background section 2.4.

With this interface designed between the RTTA Receiver (I2RS Client) and RTTA Router (I2RS Agent), the RTTA Receiver is able to access to the information of the whole routing system, due to this information obtained from all of the other nodes that is stored in the TED of each RTTA router of the network.

Additionally, ECN connections are enabled in the network which the RTTA routers mark in case of congestion and the RTTA Receiver evaluates this evidence. Finally, the RTTA Receiver integrates all this information by analyzing it and presenting it both in real-time and periodically.

In summary the proposed solution creates an updated map of the state of the network including information about the congestion state of each of its links. Subsequent chapter give some examples of show how this information can be utilized.

4 Congestion–Identification Implementation

In this chapter, after the explanation of the design, the implementation of the mechanism for identifying congestion in a Radio Access Network (RAN) is described. In order to make the reading easier, this chapter follows the same structure of the previous chapter. The chapter starts by introducing the implementation and operation of the test-bed, as this is the starting point for the rest of the deployment. The chapter next introduces the network and all the systems that are involved in the RAN. In the next sections, the thesis goes deeper into the implementation of each of the functionalities (explained previously in the Chapter 3). In this chapter all the problems and limitations that have been encountered and that have influenced the final proposed solution and the implementation are explained in detail. The chapter ends with a conclusion that sum up the main ideas presented in this chapter.

4.1 Set up of the Environment

In this section, the implementation and operation of the equipment used in the test-bed is explained. This test-bed, shown in Figure 4-1, is the starting point for the implementation of the design presented in the previous chapter. This test-bed allows us to explore different scenarios, for example, creating connections between PC1 and PC2 over different paths (such as PC1, VM-41, VM-32, VM-31, PC2 or PC1, VM-41, VM-42, VM-22, VM-31, PC-2)*. The deployment of this isolated scenario is also the main source of the limitations that have influenced the final decisions for the design. That is why it is important to go deeper in the devices and software used.

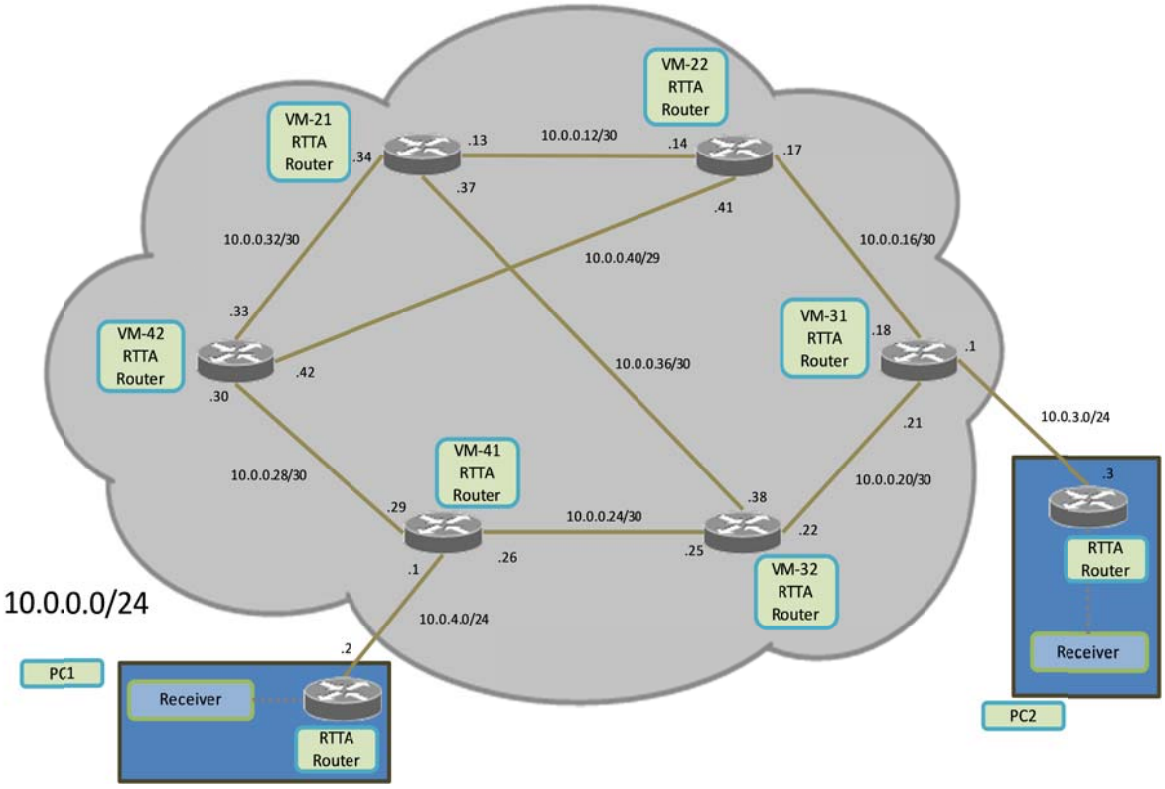

Figure 4-1: Test bed Deployment

* VM-xx represents a virtual machine numbered xx. Each of these virtual machines is running an instance of the Quagga routing software. The details of this naming are explained on the next page.

The realization of this scenario is done using five physical machines (see Table 4-1). Three of them host two virtual machines inside each of them. Each of these virtual machines plays the role of a router. The result is that we have emulated a total number of 6 routers in this test-bed. Two other machines play the role of end-nodes (in this scenario they emulate an eNodeB and PDN-GW) forming in this way a complete Radio Access Transport Network. The OS used in all the machines is Ubuntu 12.04 with Kernel 3.5.0-41-generic [62] and the virtualization software used to realize the virtual machines is Virtualbox-4.3 [63].

The routers are named with a number consisting of two digits, where the first digit indicates the host and the second digit indicates the number of the virtual machine (guest) inside that host. The end-nodes are called PC1 and PC2 playing the roles of the eNodeB and the PDN-GW (respectively). All the machines are physically interconnected using Cat 5-E [64] Ethernet cables connected to Ethernet interfaces configured to operates at 1000 Mbps, thus the maximum available bandwidth of each of these links is 1000Mbps.

Although the interfaces of the hosts are connected by these physical links, the guests inside them are linked by UDP tunnels. This is networking mode provided by VirtualBox to interconnect virtual machines running on different hosts[65]. This is done by encapsulating the Ethernet frames (sent or received) by the guest network card into UDO/IP datagrams, and sending them over the interface assigned to this guest interface. In order to configure this tunnel, the network adapters of the guest must be configured as generic drivers with UDP tunneling and the host's IPs interfaces must be the same to the corresponding one of its guest. Furthermore, it must be also configured the source and destination UDP port and the destination address, being this parameters swapped with the corresponding virtual machine in the other host. Thus there is one (virtual) network adapter for each of the guest's network interfaces. With this mode, each guest network adapter acts as if it is directly connected through a physical network interface to its (virtual machine) neighbor.

To interconnect the two guest VMs running on the same host, Virtualbox provides a different type of connection. An internal connection is created to pass the information from one guest to the other. VirtualBox also allows limiting the bandwidth of these internal links[65], providing a good mechanism to provoke different scenarios and simulated bottlenecks.

Once all the nodes are interconnected, each router has to build its forwarding database to route the traffic in this network. To provide this functionality to the elements of our network the routing software Quagga is installed in all virtual machine. With this software, as explained in the 2.6 section, an IP address is associated with each interface of the network via the Zebra daemon. With Quagga running in all the routers of the area, we can divide the area into different sub-networks and run the OSPF protocol over the network. The main reason to implement OSPF (explained in the 2.3 section), rather than another routing protocol in this scenario, is the traffic engineering features that OSPF provides. By running OSPF-TE in the area, the shortest paths will converge and the forwarding tables are built. Quagga provides also an API to interact with the OSPF router. This API serves as a bridge to inject the generated Opaque TE packets and to access the router's link state database.

The final step in the initial implementation is to enable ECN in the network. The end-nodes must be able to negotiate the use of ECN. In order to simulate this ECN feature in a Radio Access Transport Network, as it has been explained in section 0, TCP handshaking has been deployed to emulate it. On the other hand, the router nodes must be capable of marking ECN when they are congested. This last issue can solved enabling the router specifically to mark the packets, being this feature provided by the Ubuntu OS, setting the *tcp_ecn* [66]parameter correctly.

Following the above step, the test-bed is up and running. This means that all the initial elements of the network are connected and they can communicate to each other. In the following sections the implementation of the new elements introduced in the design chapter are explained. The RTTA Router is implemented in all of the virtual machines of the test-bed and in both end-nodes. The RTTA Receiver is only implemented in the end-nodes, i.e., the eNodeB and the Gateway.

Table 4-1: Configuration of each of the computers used to realize the test-bed

Computer	Role	Description
PC1	eNodeB	HP™ EliteBook™ 8560p Intel® Core™ i7-2620M CPU @ 2.70 GHz x 4, with 3.9 GiB memory, Integrated Intel 82579LM Gigabit Network Connection; 500 GB 5400 rpm SMART SATA II HDD[67]
PC2	PDN-GW	HP™ EliteBook™ 8560p Intel® Core™ i7-2620M CPU @ 2.70 GHz x 4, with 3.9 GiB memory, Integrated Intel 82579LM Gigabit Network Connection; 500 GB 5400 rpm SMART SATA II HDD[67]
RouterHost2	Hosts VM21&VM22	Dell™ OptiPlex™ 7010 Intel® Core™ i7-3770 CPU@ 3.40 GHz x 8, with 7.8 GiB memory, Integrated Intel® 82579LM Ethernet LAN 10/100/1000; 3.5" Hard Drives: 1TB 7200 RPM SATA[68]
RouterHost3	Hosts VM31&VM32	Dell™ OptiPlex™ 7010 Intel® Core™ i7-3770 CPU@ 3.40 GHz x 8, with 7.8 GiB memory, Integrated Intel® 82579LM Ethernet LAN 10/100/1000; 3.5" Hard Drives: 1TB 7200 RPM SATA[68]
RouterHost4	Hosts VM41&VM42	Dell™ OptiPlex™ 7010 Intel® Core™ i7-3770 CPU@ 3.40 GHz x 8, with 7.8 GiB memory, Integrated Intel® 82579LM Ethernet LAN 10/100/1000; 3.5" Hard Drives: 1TB 7200 RPM SATA[68]

4.2 Real-Time Traffic Aware Router

Following the general scheme of this chapter, this section explains the implementation of the RTTA Router element already introduced in the design chapter. All the modules and their design are revisited in order to go deeper into them. All of these modules, included the modules that Quagga provides, are implemented in the C programming language [69], except for the measurements of bandwidth and the number of marked packets as these two modules are written using Bash scripts [70].

The modules of the RTTA router are situated in different modules inside the Quagga software. As was shown in Figure 2-12: Quagga OSPF Daemon Architecture, when using the OSPF routing protocol the software creates a relationship between a server (included in the OSPF router) and a client. The client is part of an application that can introduce (or receive) information through Opaque TE packets. The implementation of the measuring, monitoring, and policies and the RTTA sender modules have been realized in the RTTA Router application. The OSPF router is actually just the routing part of the Quagga system with server functionality. These separate modules allow us to isolate the traffic engineering mechanism from the actual OSPF routing forwarding making the implementation modular, hence they are easier and safer to handle [52]. Figure 4-2 shows a flowchart which indicates the relationships between the different modules of the RTTA Router.

The three modules (measuring, monitoring, and policies) will each be explained in one of the following subsections. The implementation of the RTTA Sender will be described in section 4.3. The explanation of the deploying of the OSPF Router is unnecessary because it is already implemented in the Quagga system with the API Server functionality. This chapter ends, in section 4.4, with a summary of the implementation of the whole router system.

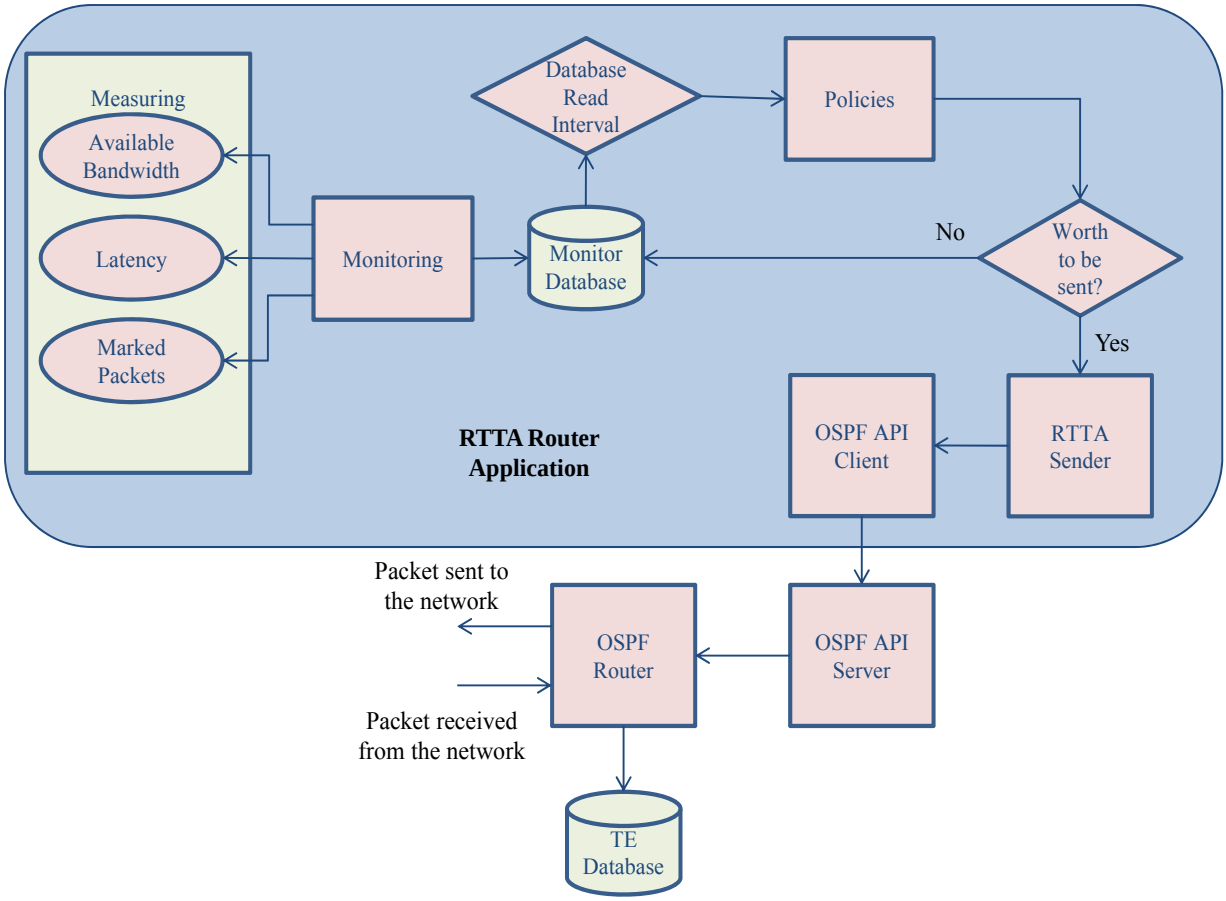

Figure 4-2: RTTA Router Flowchart

4.2.1 Measuring module

The first module has the task of measuring three traffic parameters for all of the interfaces of the RTTA Router. The idea of placing the implementation of these measurements in this module is to make them independent and isolate them from the rest of the modules. This implies that each measure is designed as a sub-module of the measuring module, as shown in Figure 4-2. This independency makes the module more robust and safer, avoiding overwriting and synchronization problems that could occur when implementing the measurements of the three parameters in one module. Additionally, separating the measurements into different sub-modules and handling them separately for each interface, makes the access easier and safer. These advantages lead to this method of realizing these measurements being the most suitable for this project. This approach should also facilitate future additions of additional measurements (if there is a need for them).

At the beginning of each set of measurements; this module has to identify which interface is actually being measuring, this is done by using the IP address of the interface being measured as the identifier. Thus the IP address of the interface is used as (1) the identifier of the set of measurements, (2) it is used by the RTTA Receiver to distinguish the interfaces of the network, and (3) it is used to identify the interface during visualization. This implementation has used SNMP to get the list of IP addresses for the active network interfaces. This approach decouples the implementation from worrying about the name of the interfaces and what their IP address(es) is(are). Given these identifiers the measurements are done as described in the following paragraphs.

4.2.1.1 Available Bandwidth

As explained in the design section, the use of active measurement methods to determine available bandwidth was rejected in order to provide a simple and light mechanism. The alternative method that has been selected exploits the information that SNMP can provide. Because SNMP keeps a number of different counters, we can easily learn the number of packets and bits received and sent by each interface. With this information we can simply calculate an arithmetic average of the total *used* bandwidth for the interface being measured. This data is collected using called ifstat [71], a tool that the Ubuntu OS provides. Given the configured maximum data rate of the interface and the average total bandwidth used we can simply subtract the used bandwidth from the maximum possible bandwidth 1000Mbps to calculate the available bandwidth.

At the same time, the interface on the other side of the link is also measuring the bandwidth. This implies that at least two measurements of the bandwidth for each link are available. Although these measurements are not precisely simultaneous, if they occur close enough in time we will have an up to date and consistent measurement of available bandwidth. This corroborated information is used in the RTTA Receiver to obtain a more accurate estimate of the link's available bandwidth. The module will report the bandwidth in units of kbps.

This straight-forward implementation demonstrates the simplicity and lightness of this mechanism. No probe or other packets need to be sent and the complexity of the mechanism is low compared with the other tools and active measurement methods referred to in the design.

4.2.1.2 Round-Trip Time and One-Way Delay

Obtaining the link's latency parameters via two similar but different methods was explained in section 3.2.1.2. Below the choice of method for the implementation is explained along with the limitations that influenced this decision.

The first method referred in the design chapter, is to make a delay measurement using the TCP Timestamp field included in the TCP header. As was explained in section 2.5.1, the value of this field is exchanged between the two machines at the endpoints of the TCP connection. The granularity of this parameter depends on the endpoint machine's clock. This implies that the minimum time granularity for a delay measurement using this method is limited by frequency of the clock that the OS uses. More specifically, for the Ubuntu OS used in this test-bed the granularity of the clock is 10 milliseconds [72], because the clock is defined to have a frequency of 100 ticks per second. This accuracy is insufficient for the purposes of this master thesis as a granularity of at least microseconds is necessary to provide a measurement of latency that reflects the link's congestion.

This limitation has been a factor in rejecting this first mechanism, but it is worth mentioning that in a real scenario this problem can be solved. Other commercial OSs provide clocks with finer granularity, such as Cisco's IOS [72]. Additionally, the Ubuntu kernel can be modified and recompiled to provide a higher frequency clock, although this option has been discarded to avoid introducing other unexpected TCP issues in our test-bed.

The rejection of the first mechanism leads to our use of the second mechanism to provide link latency measurements. This method can take advantage of the exchange of messages using almost any transport protocol. However, in this thesis only UDP and TCP have been considered due to our familiarity with these protocols. Both of them offer different benefits. UDP allows a light-weight exchange of messages, due to it connection-less behavior; while TCP utilizes reliable connection oriented communication. In this implementation, TCP has

been chosen to ensure that the measurements are made, even in case of congestion. Note that in the face of high packet loss rates it is not clear that the measurements will actually be accurate – but they will still have some meaning.

This TCP connection carries client/server communication between the endpoints of a link. The OS clock is read and placed as a timestamp that is send to the other party. This communication is configured to send a specified number of probes via each TCP connection with a specified time gap between each probe.

Since the granularity of the internal clock of the OS is in nanoseconds, then if the synchronization between the machines exchanging messages with such timestamp is tight it is possible to use this method to determine one-way delays. As described in section 3.2.1.2, this synchronization is done using NTP. An NTP daemon was configured in all the machines and all the nodes are synchronized with one of the machines acting as a NTP server. This allows the internal clocks of all of the nodes to converge. Even with this protocol, it was not possible to synchronize the clocks with a granularity of nanoseconds, obtaining Time Errors between nodes (function that calculates the difference in time values of two clocks at a certain time) of more than 1 millisecond. Hence the measurements provided for the One Way Delay do not offer any sensible values, due to the necessity of a precision of microseconds for measuring One-Way Delay in this network. Although, in other implementations using synchronization methods such as Precise Time Protocol (PTP) described in IEEE 1588 [73], which requires specific hardware and software, it is possible to obtain precise network synchronization to an accuracy for at least sub-microseconds, as it is discussed in the Mozhdeh Kamel's master's thesis [74]. This issue has led us to ignore the measurements of this parameter, thus only the RTT value is utilized.

Fortunately, the measurement of the RTT does not require synchronization of the nodes. This measurement uses the internal clock of a single node. While RTT values permit us to have a good characterization that there is congestion on the link it does not tell us which direction of the link is experiencing congestion. Additionally, the RTT value measured in the TCP connection implemented here is affected by the processing time in the nodes of the packets.

4.2.1.3 Marked packets

The implementation of the measurement of this parameter requires previous configuration of the queues of the interfaces in the router. These queues must include an AQM mechanism, more specifically a RED one. With RED the queue starts to drop packets with a certain probability, but if ECN is enabled for this queue, then instead of dropping packets it marks the packets with this probability. As a result each output queue is configured with three thresholds and one probability. The thresholds are: the minimum (when the queue starts to mark with the probability indicated), the maximum (when the packets are one hundred per cent marked), and the total limit of the queue. If this total limit is reached, then the packets that arrive to this queue are dropped. The system must also be configured to make it ECN-aware and to enable it to mark the packets while acting as a router or as an end-node for the communication.

As was explained in the design, one first mechanism, that procures the output queue length of each interface, was attempted in order to provide evidence that a queue in the network is marking packets. This value is included as a counter in a SNMP MIB with the name ifOutQLength [75]. With access to this counter we could spread this information to the network, as it provides a direct relationship between the number of Congestion Experienced (CE) packets sniffed in the end-nodes and this queue length parameter.

Unfortunately, the use of this SNMP counter is deprecated [75]. Although commercial routers from companies such as Cisco [76] implement this value, in other OSs this counter is not used, specifically the Ubuntu OS used in this thesis project does not implement this counter. In the Quagga routing software this counter is not included nor is it present in the SNMP upgrades [77]. As a result we could not use this parameter, but it is worth mentioning that this limitation is due to the OS used. In a network with routers that implement this counter, this method could be used to provide relevant additional information about the state of the queues.

Due to this setback, another tool, tc (traffic control)[78], was used as it allows us to explicitly manage the queues in Ubuntu. With this tool, not only is it possible to configure the queues of the interfaces as mentioned above, but the tool also provides statics about the traffic going through these queues, including the number of dropped and marked packets. With access to the actual number of packets marked by each interface, the information provided to interact with the sniffing mechanism of the RTTA Receiver is even more descriptive than would have been provided by the queue length parameter!

When the module has finished with the measurements of one interface, it stores the data in separate text files, ready to be read by the next module. These files keep the last measurement of each parameter for each interface being overwritten when a new specific measurement of that interface is done. As we have labeled the measurements with the IP address of the interface and the measurement results are stored independently for each measurement and interface, this module can continues with its task of making measurements of all the interfaces without any danger of overwrite the previously collect information about another interface.

4.2.2 Monitoring & Policies modules

This module was been divided in section 3.2.2 based upon the two tasks that it performs. Consistent with this design we implemented the two tasks as two separate processes that execute concurrently.

The first task monitors the text files with the measurements and writes them in a Sqlite3 [79] database. This sequential mechanism, even if it is not very efficient, allows assuring a problematic-less access and writing in the database fulfilling the requirements of the monitoring. When the information of an interface is read, the monitor fills a row which its columns are the four measures done: the interface IP, the available bandwidth, the RTT and number of packets marked. It does the same for the others interfaces, resulting in a database with a number of rows equal to the number of interfaces, as shown in the Table 4-2. Through this method, the set of measurement overwrites the previous corresponding to that interface, while at the same, the history of all the measurements done, are stored in a trace file. With all the information kept in a unique database, for the next process that it has to be done in this module, only one access to this table is required. This database usage allows to have the measurement data safely stored and at the same time does not need extra time access. This method also let to keep the data from as much interfaces as the node has, just adding more rows to the built database. The mechanism proposed fulfills the necessary requirements for the secure functioning of the module. The study of its efficiency is of the scope of this thesis and future works can be researched in order to improve it.

Table 4-2: Sqlite Database Contents

	IP	RTT(ns)	Available Bandwidth(kbps)	Marked Packets
Eth0	10.0.0.25	988623	925424.34	0
Eth1	10.0.0.38	889078	934242.23	30
Eth2	10.0.0.22	1002344	984542.27	70

After the information has been stored, the module can make a decision of where the information is relevant enough to be forwarded. This decision is made each time that this database is read. A "Database Read Interval" parameter can be configured to set this cycle. Each read cycle checks all the interfaces. To decide if the information is relevant enough to propagate, the new entry (Rn) is compared with the last relevant reading (Rp). This comparison is shown in equation (1).

$$\frac{|R_n - R_p|}{R_p} = threshold \quad (1)$$

This method is called a *relative threshold-based policy* [80] and as implemented in this thesis a percentage threshold is selected. This parameter can be separately selected for each type of measurement.

When the database is read the RTT and the bandwidth are checked in keeping with this *relative threshold-based* policy. If one of the measured values is considered to have a relevant, then the whole set of parameters for that interface are stored as the last relevant measure, and then these parameters are forwarded to the next module.

The number of packets marked does not use this policy because this number only increases and can reach large values for that the threshold chosen, making a relative threshold no longer suitable. In order to provide a constant flow of information, a *periodic update policy* is implemented when no relevant change has happened. A parameter can be configured as the number of "Database Read Intervals" after which a new update is made even if no single set of measurement has fulfilled the *relative-threshold*. In this implementation, the previous work of Amir Roozbeh [53] was used to provide a suitable solution for this problem.

4.2.3 Extended OSPF Router module

This module is divided into two sub-modules: RTTA Sender and OSPF Router. The implementation of the second sub-module is not part of our thesis project because is already is implemented in Quagga, hence this section primarily focuses on an explanation of the relationship of these sub-modules through the OSPF API and use of this API to exchange information with other nodes.

4.2.3.1 RTTA Sender

The RTTA Sender sub-module as explained section 3.2.3.1 is one of the two parts of the Extended OSPF Router module. This extension of the OSPF router has the task of building Opaque TE LSAs. The construction of these packets starts from the sub-TLV and continues with the addition of the required headers until the LSA is sent.

When a set of parameters have been accepted by the previous module, the process of constructing an Opaque TE LSA begins. A sub-TLV with type 4 is created to carry this set of measurements as a field of 32 bits each. For a total of size of 5 x 32 bits (4 fields of 32 bits

for the IP address and the measurements and then another 32 bits for the Type and Length of the sub-TLV) for this sub-TLV. In the next step a type 5 TLV is added and then the OSPF Opaque header is added with a LSA type 10 (Area Opaque) and the Opaque type 1 (TE).

The Opaque ID field is filled-in during this process. A different value of the Opaque ID is used for each interface in order to provide an identifier that will make it possible in the TE database to distinguish between the Opaque TE LSAs of the different interfaces that come from the same router. Without this identifier the incoming Opaque TE LSAs would overwrite the previous LSAs of a given router even if they are for a different interface.

4.2.3.2 Quagga OSPF API

The Quagga routing software provides a means to communicate between the RTTA Router application and the routing system. As was introduced in section 2.6.2, by using this protocol the application can connect to the router and send the Opaque TE LSAs that it has built, just as if it was the router itself that created the LSA.

Using TCP the client/server endpoints establish a communication session. First the client opens a connection for synchronous request/replies to the server, and then, the server accepts this incoming connection and opens a reverse channel for asynchronous messages. It is worth mentioning that this communication could be done between any machines in the network, but for this scenario, the communication occurs with a machine.

When the Opaque TE LSA is created, it is handled by the API client, which forwards it through the TCP connection created to the API server side included within the OSPF router. When this LSA is accepted by the server, then the LSA is sent to the next and final sub-module, the OSPF router.

It is worth mentioning, that this implementation has simply used the API provided by the routing software provides to deploy this feature. This API provides the necessary modularity and isolation to implement an application. However, this communication could have been done by using a method other that this API, however an alternative implementation is outside the scope of this thesis project.

4.2.3.3 OSPF Router

The last sub-module of the RTTA router provides the features of a common OSPF router, included the task of building a TE database containing the traffic engineering (in this case congestion) information. When the OSPF router receives an OSPF packet from the API server, it stores it in its database after identifying this information as a new data structure, index by its router ID and Opaque ID. The OSPF router then injects the packet in the network to flood it (in this case distributing the congestion information).

When the OSPF router receives Opaque TE packets from other routers within the area, it stores them in the same way as described above. The TE database is updated whenever an Opaque TE LSA is received, from itself or other routers. The database also keeps the age of each packet as this provides information about *when* the last Opaque TE packet from a certain interface was received.

4.3 Real-Time Traffic Aware Receiver

The RTTA Receiver may be situated in the PDN-GW and in all the eNodeBs of an LTE network to provide information about congestion to these nodes. In our test-bed, two RTTA Receiver elements are located in PC1 and PC2, as shown in Figure 4-1, at the end-points of the network. In addition, other RTTA Receivers can be added to the test-bed; for instance, RTTA Receivers can be added to one or more eNodeBs in order to analyze different network

configurations and to describe network conditions of different paths. This will be further described in the next chapter.

The purpose of the deployment of RTTA Receivers is to collect all the congestion information in order to evaluate and comparing techniques (and combination of these techniques) for studying the interaction of the information obtained. Figure 4-3 illustrates how the information is processed within a RTTA Receiver.

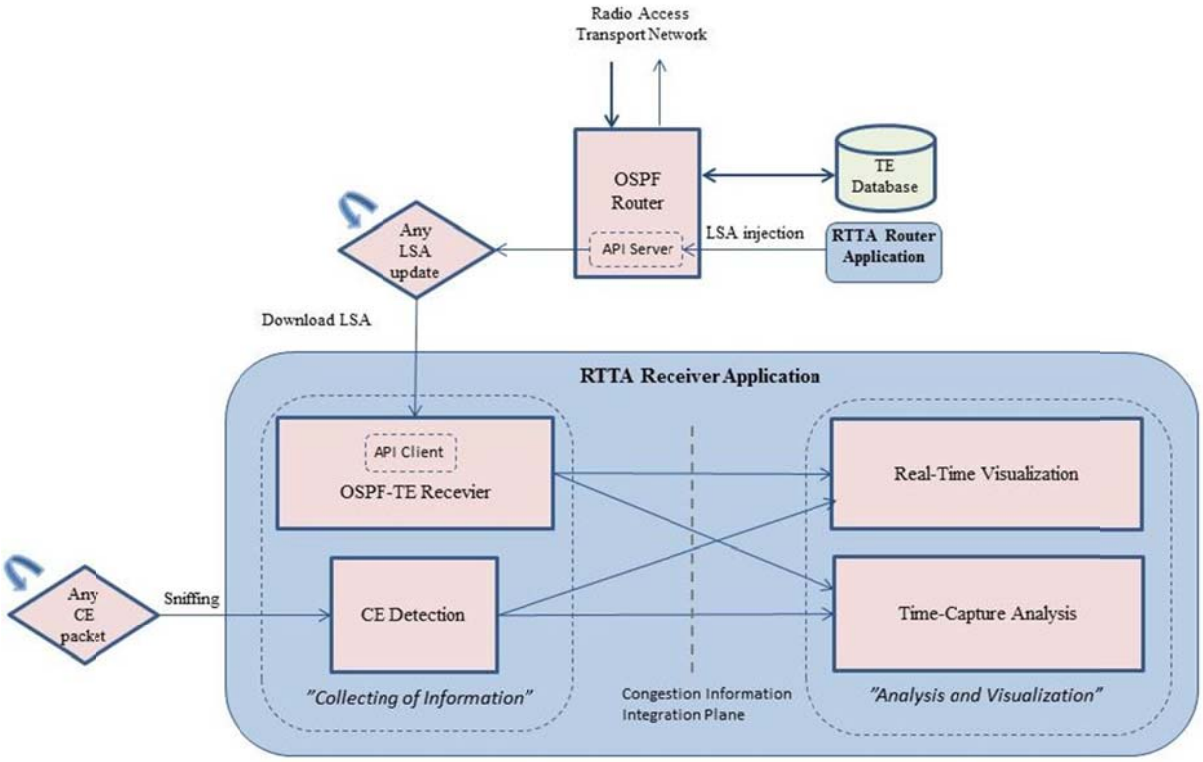

Figure 4-3: RTTA Receiver Flowchart

The RTTA Receiver Application has been divided in two modules: Collecting of Information and Analysis and Visualization. This application is responsible for handling the congestion information and evaluating its interactions. The next sections go into details about how the congestion information flows within the RTTA Receiver Application, explaining how the information is gathered, and how is it represented. All the code for this application has been written in Python, due to some of the facilities that Python offers to manage data and to display it.

4.3.1 Collecting of information module

This module collects information by implementing the collecting information tasks of the two sub-mechanisms: OSPF-TE and ECN for RAN. These tasks are indicated in Figure 4-3 as "OSPF-TE Receiver" and "CE Detection" (respectively). These tasks execute concurrently and provide updated information to the next module (Analysis and Visualization – described in section 0).

4.3.1.1 OSPF-TE Receiver

The OSPF-TE Receiver functionality is responsible for gathering information related to the OSPF-TE protocol. As it has been said earlier, using the API of Quagga it is possible to access the TE database stored in each router of the network. This API reflects the I2RS idea

described in the background section 2.4 where a communication between an I2RS client and an I2RS agent is performed in order to transfer out the state of the network.

To do so, TCP communication is established between the Quagga API client (located in the OSPF-TE Receiver) and the Quagga API server (located in the RTTA Router). It is worth mentioning that due to the use of a TCP connection, the RTTA Receiver can access the TE database of any of the RTTA Routers in the network. However, for studying the interaction with the ECN protocol, the RTTA Receiver in this implementation only accesses the RTTA Router located in the same node as it is located (in this case only at the two end nodes of the network). Initially the entire TE database is downloaded from the router, and then the connection remains established to download the TE information when an Opaque TE packet arrives, as shown in the Figure 4-3.

The OSPF-TE Receiver remains synchronized with latest version of the TE database and provides relevant data to the Analysis and Visualization module. Specifically it keeps track of the information forwarded from each interface of every RTTA Router of the RTT (in nanoseconds), Available Bandwidth (Kbps), and number of marked packets. In addition, the OSPF-TE Receiver computes for each data structure of the TE database, where all the TE metrics of each interface of every RTTA Router are stored and the average number of packets marked when an update arrives. This helps the receiver to understand the congestion status of the queues by comparing the number of CE packets marked by each interface. Equation (2) describes how this parameter is calculated, where "p.m." means "packets marked", new refers to a new Opaque packet received from a certain interface and stored refers to the latest Opaque packet information already stored in the TE database from the same interface.

$$\bar{N}\ of\ p.m. = \frac{N\ p.m._{new} - N\ p.m_{stored}}{Arrival\ timestamp_{\ new} - Arrival\ Timestamp_{stored}} \quad (2)$$

It is important to mention that the output values correspond to the characteristics of each of the links of the network. These values are sorted by interface. As a result two values from each link are forwarded to the next module, each one gathered from the correspondent interface of the router at one end of each link.

4.3.1.2 CE detection

The CE detection functionality detects CE packets coming from the network. This sub-module is constantly sniffing the data plane, using the Scapy software [81], and checking the ToS field in the IP header (see section 0) of each packet to identify CE packets coming from the network. The average number of packets marked is computed. The value is calculated using equation (2). This value, together with the collected OSPF-TE information, is input for the next module of the RTTA Receiver (i.e., the Analysis and Visualization module – described in section 0).

4.3.2 Analysis and Visualization module

Once the information from the Collecting of information module has been collected, it is time to integrate this information to provide information useful to the end nodes of the network – so that they can take further actions. When the information coming from the previous module crosses the "Congestion Information Integration Plane" all the congestion information is treated as a unique input in order to derive different interpretations of the state of the network. Different methods of combining this data have been attempted to determine what the most reliable means of identifying congestion is.

More related work has been done in order to provide this relevant data and combine it to identify congestion. The Arista company's project already mentioned [51] provides information about the queue size of the nodes in the network and implement an alarm system to make use of this. Another recent research studies the features that the Precision Time Protocol (PTP) provides and its deployment in a metropolitan area network[74]. These analyses provide relevant information about the state of the network, but the one proposed in this thesis, provide a whole combination of the measurements in order to identify the actual congested link in the network that can be used in coordination with Radio Networks.

Two analysis methods have been implemented: Real-Time Visualization and Time-Capture Analysis. The use of both methods depends of what the end-nodes want to do with this information; however, comparing them helps us to understand the behavior of the network. The Appendix A shows some examples of the visualization for both methods.

4.3.2.1 Real-Time Visualization sub-module

The motivation for real-time visualization is to monitor in real-time how the characteristics reported to the eNodeBs or the PDN-GW evolve. Ideally we want these nodes to take appropriate actions regarding how they handle and manage connections in an LTE network based in this congestion information.

The real-time visualization module monitors the transport network's characteristics by displaying them graphically. Each one of the characteristics (Round-Trip Time, Available Bandwidth, and Average of Packets Marked) can be compared in different ways. In this master's thesis they are compared based upon router interfaces and paths, although they could be compared by links or by specifics paths. Figure 4-4* is a snapshot of the output of the monitor at a specific moment (note that the names of the interfaces and paths in this figure correspond to those defined in Figure 4-1).

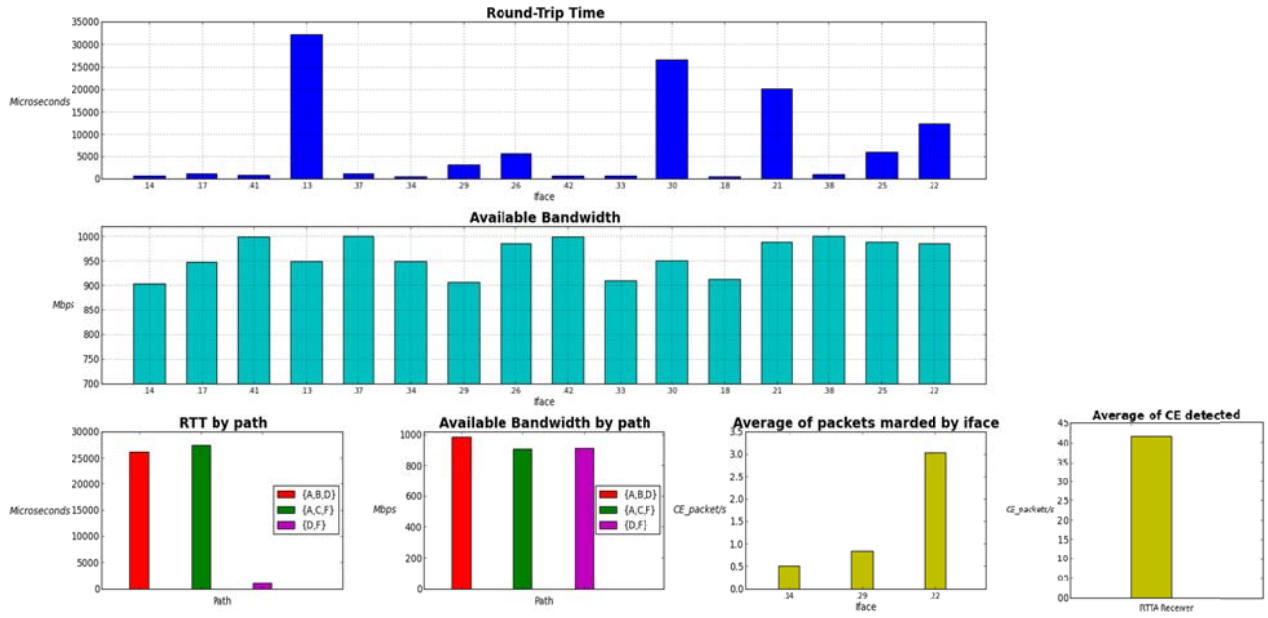

Figure 4-4: Real-Time Visualization Capture

* A larger version of this figure is shown in Appendix A.

In this implementation we have compared:

- **RTT and Available Bandwidth visualized by interface**: The actual values of both characteristics are dynamically displayed in order to comparing them against all the interfaces of the network. An example is shown in Figure 4-5. The names of the interfaces in this figure correspond to those defined in Figure 4-1.

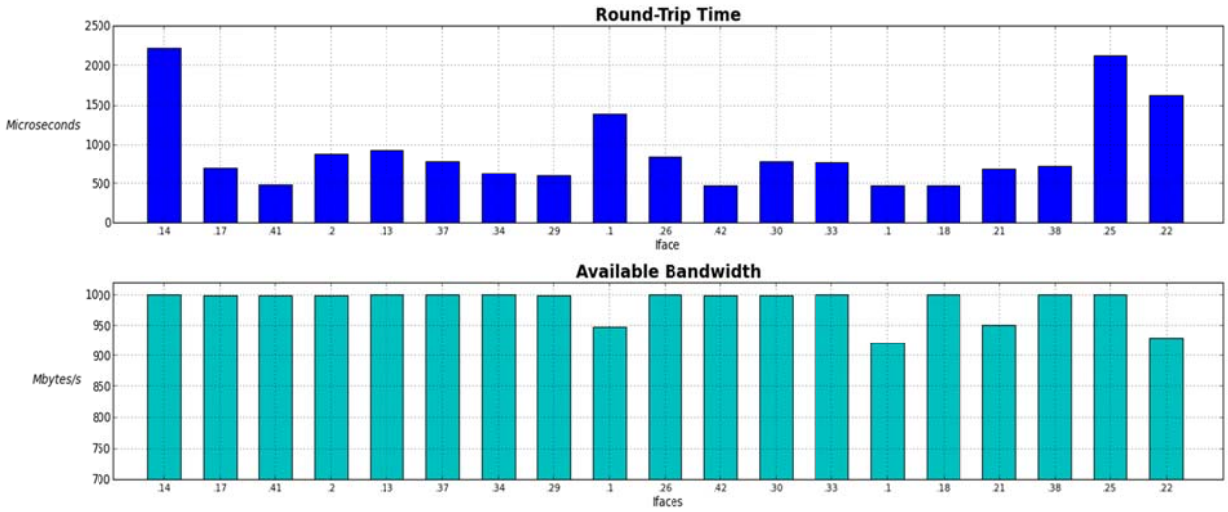

Figure 4-5: Example of Real-Time Visualization for RTT and Available Bandwidth by interface

- **RTT and Available Bandwidth visualized by path**: The RTT for a whole path in the network is shown in terms of the maximum of the actual RTT values of an each link, with the sum of the maximum RTT of the others links displayed for each path. For the Available Bandwidth the minimum Available Bandwidth of all the router interfaces along this path is computed and displayed. In both cases, for RTT and Available Bandwidth, the most restrictive parameter has been chosen. Figure 4-6 shows the output of the Real-Time Visualization of these two cases. The names of the paths in this figure correspond to those defined in Figure 4-1.

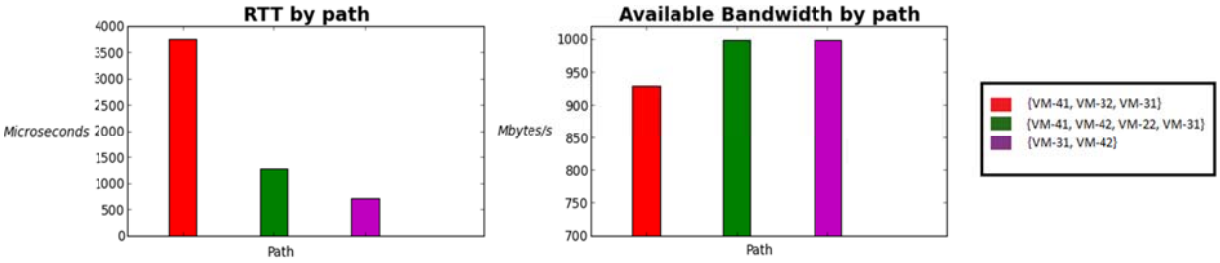

Figure 4-6: Example of Real-Time Visualization for RTT and Available Bandwidth by path

- **Average of packets marked**: In order to integrate the Congestion Evidence (CE) information for the visualization; the average number of marked packets at the interface of each router is compared with the average of CE packets detected by the CE. Detection module of the RTTA Receiver. Figure 4-7 shows an example of how this is visualized by the Real-Time Visualization module.

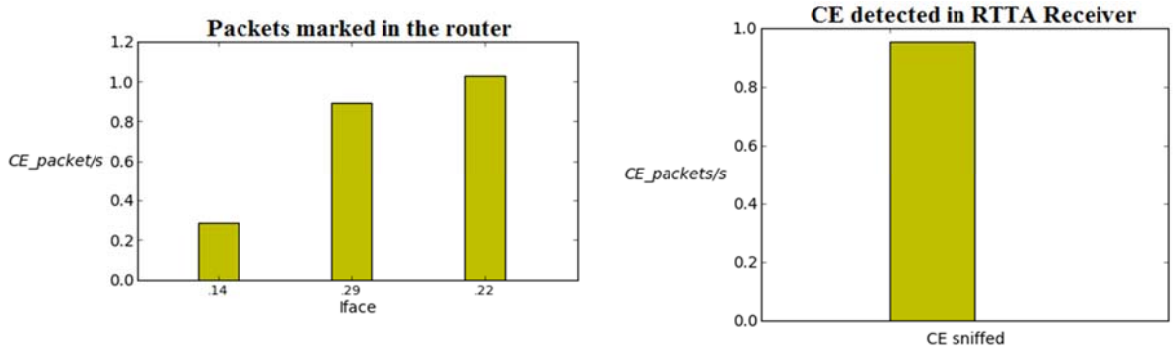

Figure 4-7: Example of Real-Time Visualization for CE packets

4.3.2.2 Time-Capture Analysis sub-module

The analysis done by this sub-module here is based recording the state of the network during a certain period of time and then visualizing the data after this period of time has concluded[*]. This sub-module records the values of the different characteristics gathered by the RTTA Receivers as samples for a pre-defined period of time. Each time that the TE database is updated, a new sample is stored by the Time-Capture Analysis sub-module. Later a comparison is made of a set of samples of the congestion information from each node of the network in order to understand the behavior of the network during this time period. The analysis performed here is in terms of:

- RTT, Available Bandwidth, and Average of Packets Marked between router interfaces are present as the samples recorded during a pre-defined period of time.
- RTT, Available Bandwidth, Average of Packets Marked along certain paths, and the Average of CE packet detected at the RTTA have been used to select the most restrictive parameters for each of these characteristics, then the samples for a period of time are compared.

This sub-module needs to manage different numbers of samples related to specific intervals of time within the pre-defined period. For example, we can re-samples the recorded values in order to time align different sets of samples. This can be done by linear interpolation of the recorded samples, using the Scipy library of Python [82], to generate a new set of virtual sample. Now the desired computations can be done to obtain the RTT and Available Bandwidth, while comparing the values of each characteristic for tuples of time aligned samples. In addition to the Real Time Visualization, this can also be done for the characteristic Average of packets marked for which we can compute the sum of all the values from every router interface along a path.

The result can be visualized graphically using Python's Matplotlib library [83]. The following Figure 4-8 shows examples of the results of this visualization.

[*] This means that this data can be viewed as being processed by a finite impulse filter whose duration is the specified period of time.

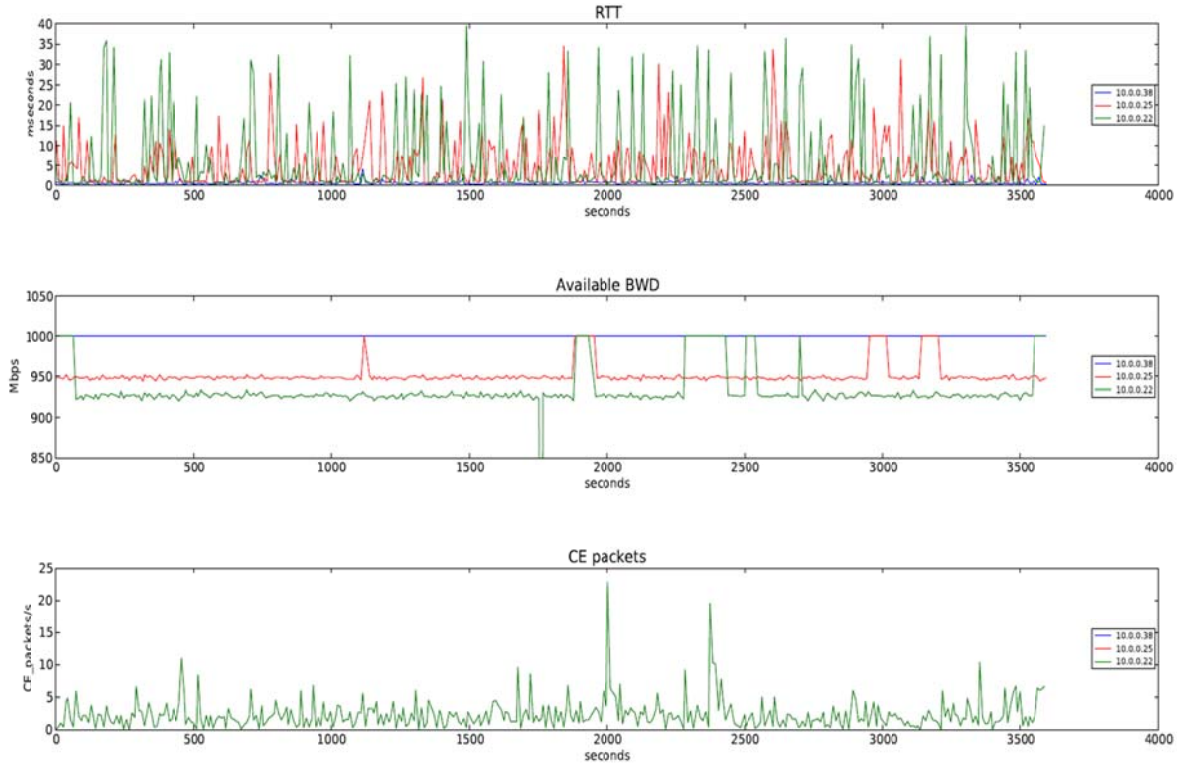

Figure 4-8: Time-Capture Visualization of the interfaces of the RTTA Router of the VM-32

This implementation allows uses to perform simulations with different network conditions in order to verify trends and watch the distribution of the congestion information as function of time. In this way, it is possible to contrast the consequences of actions taken based on the Real-Time Visualization functionality. The Time-Capture Analysis sub-module should help is to understand the behavior of the network over time.

4.4 Conclusion of this implementation chapter

In this chapter, an implementation in a test-bed of the Congestion-Identification scenario has been described along with details of how each of the modules has been implemented. Many issues have been discussed ranging from selecting the appropriate technical methods to acquire data to how to deploy the functionality in the network. In addition, the implementation is consistent with the proposed Congestion-Identification design; although some of the proposed solutions described in the design, such as TCP Timestamp for measuring RTT, have been discarded for this implementation.

Furthermore, this test-bed provides us with the expected information in a manner suitable for understanding the behavior of the network. This information should be useful to identify bottlenecks within the network (hence we have focused on the limiting characteristics of links and paths). The next chapter describes examples of characterizations of the network in the test-bed in order to illustrate the convenience of this method via some use cases.

5 Verification of the results

Following an explanation of the design and the implementation, this chapter verifies a mechanism for identifying congestion in a radio access transport network. The proposed solution is illustrated with two scenarios that confirm the functioning of the methods presented in this thesis. In these two examples, using the visualization tools implemented, the interaction between ECN and the OSPF-TE is monitored and access to the congestion information is confirmed.

Scenarios with different characteristics are compared to demonstrate the statements made in the previous paragraph. The details of these scenarios have been chosen to provide a clear visualization of the information provided and make the comparison of the state of the network easier for the reader. Different scenarios with different characteristics can be deployed to confirm that the features proposed in this thesis work as expected. In order to provide a verification of the results, the proposed scenarios are justified (to ensure that they are relevant and cover the most interesting and common cases of the network).

5.1 Congestion Emulation Tools

The simulation of the congestion characteristics has required implementing in the test bed different tools and devices. These tools are summarized in Table 5-1. These mechanisms provide the necessary features to characterize congestion in a network.

Table 5-1: Tools used for Congestion Emulation and Simulation

IPERF	Iperf [84] is a tool that can be used to perform network throughput measurements. To perform an Iperf test the user must establish both a server (to discard traffic) and a client (to generate traffic). The network performance measurement is done by loading the network in order to reach its limits (saturating the bottlenecks) without identifying the bottlenecks. For example, Iperf can inject TCP traffic in the network until the total available bandwidth is reached.
Portable Wide Area Network (WAN) Emulator	Linktropy Mini2 [85] is a device that can be configured to introduce congestion characteristics into a network. Connecting this equipment between two nodes, allows us to limit the bandwidth, increase the latency and jitter, and induce losses, thus emulating a congested link.
VLC Media Player	VLC [86] is used to introduce user data traffic into the test-bed, specifically downloading streaming video via the RAN. The VLC media player encodes and streams video using different CODECs and formats. The video CODEC used in these experiments is H.264 and the audio codec is MP3 [87]. This choice of CODECs was convenient for the scenario, but other CODECs could have been used to provide a streaming media source. However, consideration of others CODECs is outside the scope of this thesis project. The protocol used for the streaming is UDP, this enable us to distinguish which video stream is suffering congestion. UDP was used due to its connection-less characteristic, thus avoiding the head of queue blocking that TCP would introduce when frames are lost.
VirtualBox	The VirtualBox [65] virtualization software was not only used to deploy virtual machines in the test-bed equipment, it also allows characterizing of the internal guest links. With this option the bandwidth of the links can be limited and consequently, this allows modeling the link as if a specified portion of the bandwidth was reserved thus simulating that this amount of bandwidth is being used by other traffic.

The tools described in the table allow us to introduce different congestion circumstances that can be used to characterize and simulate different scenarios. Data from tests of these different scenarios enable us to explain and verify our results using visualization (see section 5.3.1).

5.2 Description of the different scenarios

To illustrate the proposed mechanism; two different scenarios have been selected. These scenarios are described in this section. Each of them utilizes specific traffic conditions in the test-bed in order to characterize the state of the links (in terms of latencies and bandwidths), and to influence marking actions in the routers. The first scenario, called the Congestion-less scenario, characterizes a network without bottlenecks or congestion. The second, the Congestion scenario, induces congestion in some parts of the network.

Some parts of the configuration of the network are the same in both scenarios. The evaluation of the mechanism with different configurations of these parameters remains for future work. The configuration used in these parameters, using the notation used in Chapter 4, follows:

- **RED configuration of the router interface queues**: The values chosen for the three different thresholds and the indicated probability allow the router interfaces to mark packets in case of congestion. This configuration is the same for each output queue of every router interface in the test-bed. Table 5-2 shows the values set.

Table 5-2: RED queues configuration

Minimum	1 Kbit
Maximum	2 Kbits
Limit of the queue	1.46 Mbits
Probability	60 %

- **RTTA Router policies**: The values set for deciding when to report congestion information from the RTTA Routers were set to obtain information from the routers in both scenarios. It is important to mention here, that the policies in the RTTA Routers were selected to allow us to save resources and minimize the volume of congestion information traffic - thus avoiding redundancy across the network [88]. Therefore, additional implementations that study how to configure the RTTA Router policies are needed, as will be described in the section 6.2. This router policies were chosen as described in Table 5-3.

Table 5-3: RTTA Router policies configuration

Database Read Interval	5 seconds
Threshold	10 %
Periodic update policy	100 seconds

- **User traffic**: User traffic is crossing the network from the Internet to UEs in order to simulate user traffic passing through the network and to study evidence of congestion as well as to assess the achieved QoE. For this purpose, we assume that two UEs belonging to eNode1 and eNode2 respectively (notice the video icons in the eNodesB of Figure 5-1) are downloading a video from behind the PDN-GW, using an application such as VLC. Table 5-4 shows the traffic configuration of these two VLC streaming connections.

Table 5-4: VLC streaming configuration

Video Bitrate	1000 Mbps
Audio Bitrate	448 Kbps

In the characterization of the test-bed presented here, the Radio Access Transport Network comprises the links and nodes where the RTTA Routers (A, B, C, D, E, and F) are located; as shown in Figure 5-1. There are three RTTA Receivers in the network, as it is shown in Figure 5-1, in order to obtain results in a realistic scenario. These RTTA Receivers are situated in the PDN-GW, eNode1 (eN1), and in an eNode2 (eN2) that can be added to the test-bed. Each RTTA Receiver collects OSPF-TE information from the RTTA Router closest to it. Each RTTA Receiver also detects CE marked packets coming across the link where it is situated. In both scenarios, the results obtained are from the RTTA Receiver situated in eN1. Results could be obtained in a similar fashion from the other two RTTA Receivers.

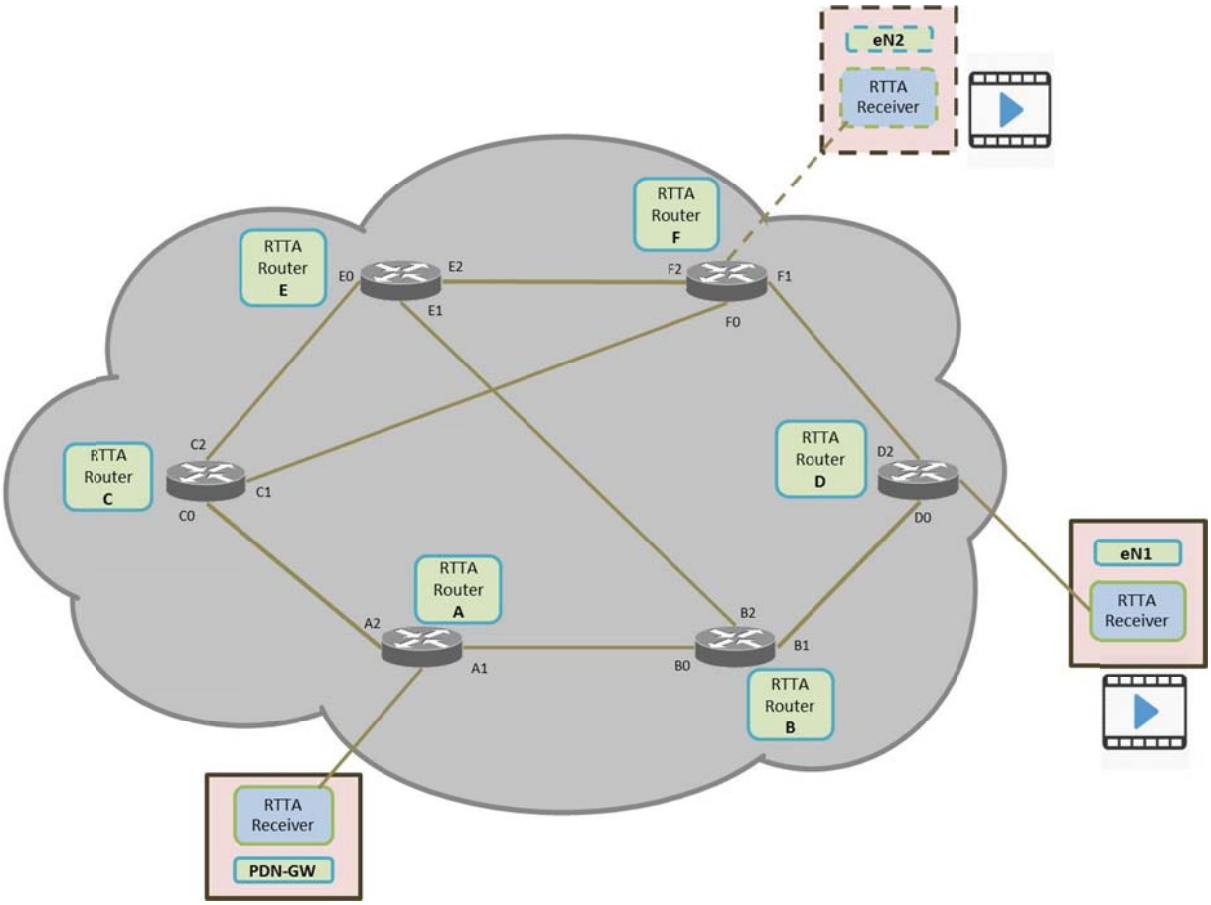

Figure 5-1: Scenario A

5.2.1 Scenario A: Congestion-less network

The results shown in this scenario are intended to show how the implemented mechanism works in a RATN free of congestion. The information that has been gathered is presented in order to understand the state of the network and to compare this state with the results obtained in the Congestion scenario. In the Congestion–less scenario no congestion emulation tool was used and only the desired user traffic is added. The properties observed here describe the state of the network defined by the hardware and software characteristics described in section 4.1. This scenario formed the basis for the Scenario B, which includes additional equipment that will be used to characterize the state of the network when congestion is induced.

5.2.2 Scenario B: Congested path in the network

This scenario simulates a RATN with congestion problems in one of its paths. Here, the links of a specific path, {A, B, D}, cause bottlenecks resulting in traffic problems due to the introduction of congestion (via the emulation tools). The sources of congestion introduced are the following:

- The Portable WAN Emulator is situated in the link AB, introduces delays of 2 seconds for all of the packets crossing the link in either the uplink or downlink direction.
- The Virtual Box Link Characterization is located in the link BD, emulating an existing bandwidth utilization of 950 Mbps for this link.
- An Iperf connection is established to load the whole {A, B, D} path in order to force the marking of packets.

These three modifications of the traffic conditions imitate circumstances that can actually occur in an LTE network, due to different factors as described in the introduction. Figure 5-2 illustrates Scenario B. This figure shows where the emulation tools have been placed, along with the congested paths and links.

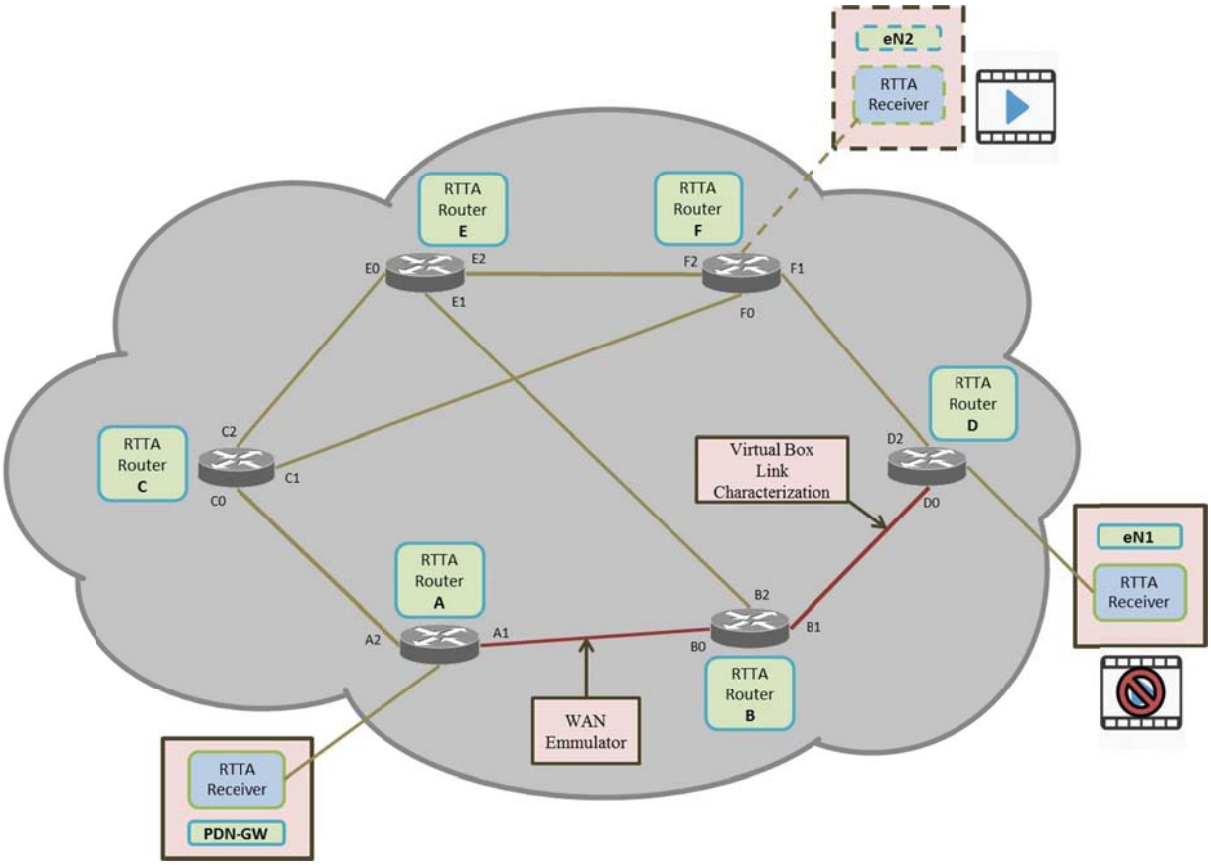

Figure 5-2: Scenario B

In addition, as described before, the same configuration of the network as used in Scenario A is deployed here, including the user traffic – so that we can compare the difference in QoE provided to a UE connected via different paths, i.e. {A, B, D} or {A, C, F}.

5.3 Analysis of the results

In this section, the results obtained from both scenarios are analyzed. We compare the network conditions of both scenarios by performing the analysis proposed in Chapter 4 using both Real-Time Visualization and Time-Capture.

The comparisons could have been done by selecting several router interfaces, links, or paths. However, for the purpose of this thesis project we are concerned about the *verification* of the proposed mechanism. For this reason a relevant selection of congestion information is gathered and analyzed. For each of the two visualization methods we explain which congestion information is being analyzed, along with which paths have been chosen and why. The paths studied are {A, B, D}, {A, C, F}, and {D, F}; as these paths allowing us to simulate realistic RATN within a LTE network. The results of this analysis should be conclusions regarding what further actions should be taken. These actions will be discussed at the end of this chapter, where we study potential alternatives that could be utilized in the event of congestion.

5.3.1 Real Time Visualization

In this section a real-time comparison between the scenarios is done. In this thesis the graphical presentation of the results visualized in real time are shown based upon the monitoring data at a certain moment; however, the tool actually shows the data in real-time.

The two scenarios are presented using bar charts at different moments of time, but when the characteristics are the same. This graphical limitation of a printed thesis is addressed in section 5.3.2. This visualization described in this section provides information to be analyzed or used in real-time in order for a network operator to react to the congestion occurring in the network.

From an UE's point of view, the congested state of the network is reflected in the QoE it experiences, for instance, in the visualization of a streaming video. In the following snapshots, the video experiences of the two UEs in the Scenario B are compared. Figure 5-3 presents the visualization of the video from a user connected to an eNode (eN2) when its transport path {A, C, F} is **not** congested, allowing the user to watch the video with a standard quality. In contrast, Figure 5-4 shows how the congestion introduced in the path between the eN1 and the PDN-GW {A, B, D} negatively affects the video experience of the user. As one can easily see the video in Figure 5-4 is degraded by the effects of many lost link frames.

Figure 5-3: Video seen by a UE connected to eN2

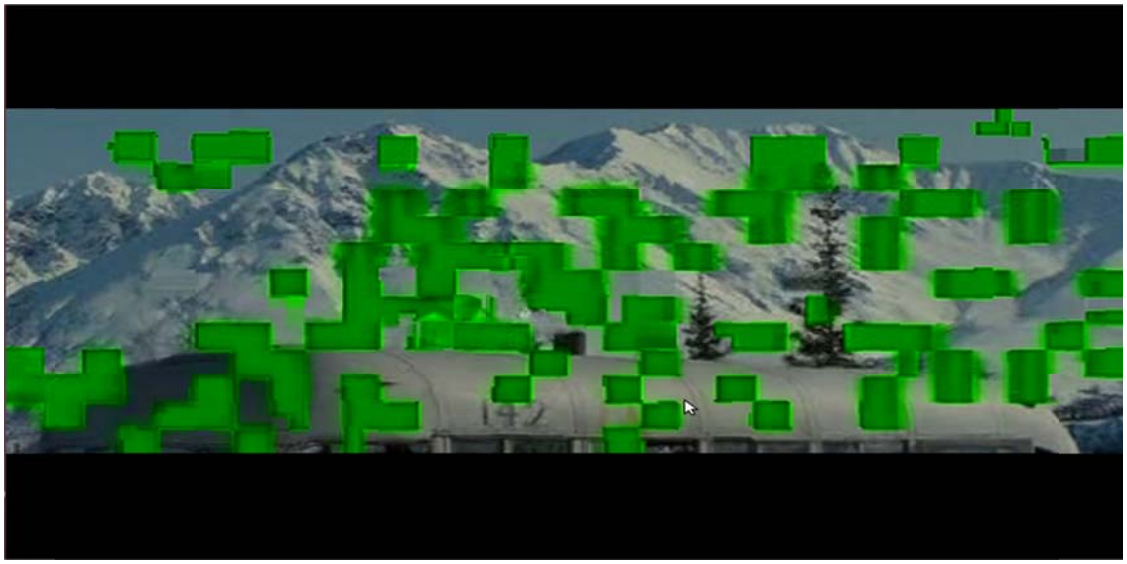

Figure 5-4: Video seen by a UE connected to eN1

In the following subsection, the reasons for the poor QoE experienced by the UE connected to eN1 are displayed with comparisons of the different congestion parameters organized by interface and by path. These comparisons verify that the information collected and visualized provides information about the state of the network in real time.

5.3.1.1 Per Router Interface Analysis

The congestion information shown here corresponds to information collected from all the routers of the network. The bar chart presents the interfaces of the routers, each identified by a letter and the number of the interface.

Round Trip Time: In this study the RTT and the available bandwidth are studied. In the Figure 5-5, congestion-less scenario, and the Figure 5-6 congested scenario, the RTT comparison between both scenarios is presented. In these graphs the delay introduced by the WAN emulator in the link AB (A1-B0) is visualized and the limited congested link BD (B1-D0) too. The interfaces included in the path {A, B, D} present higher RTT measurements than the others. Notice that the vertical scale is different Figure 5-5 than in Figure 5-6.

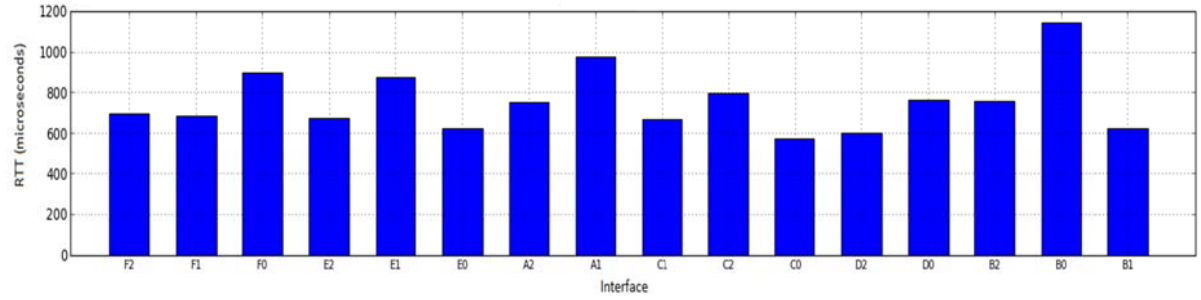

Figure 5-5: Real Time Visualization of the RTT by Interfaces in Scenario A

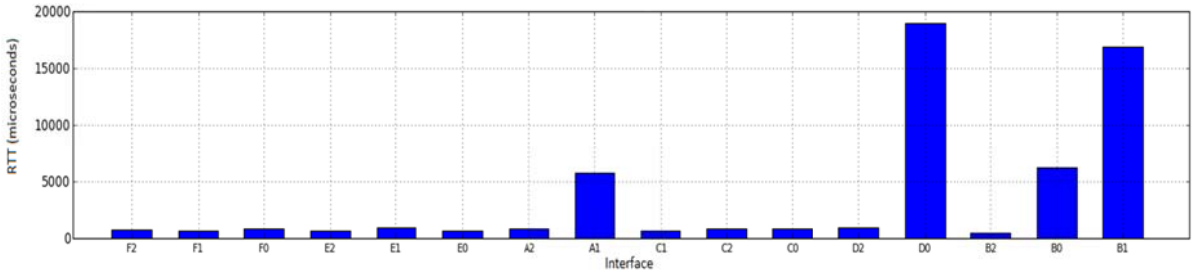

Figure 5-6: Real Time Visualization of the RTT by Interfaces in Scenario B

Available Bandwidth: Figure 5-7 and Figure 5-8 show the available bandwidth information for each of the different interfaces in the two scenarios. In Figure 5-7 we can see that no bandwidth is consumed by any interface, while in Figure 5-8, the information provided by interfaces B1 and D0 reflect the fact that the bandwidth limitation of a link has been reached (hence there is no available bandwidth via these interfaces). The reduction in the available link bandwidth of the other interfaces in the network reflects the amount of traffic injected by the tools.

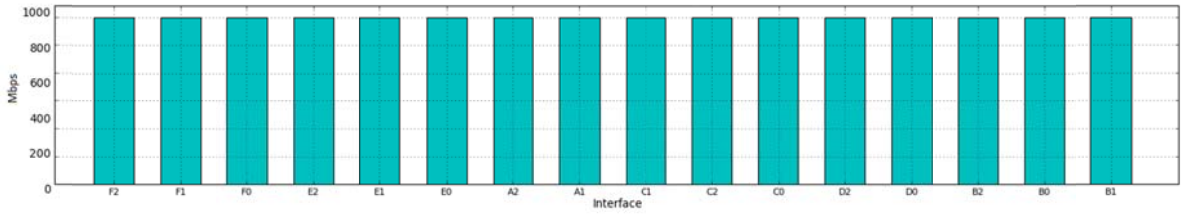

Figure 5-7: Real Time Visualization of the Available Bandwidth by Interfaces in Scenario A

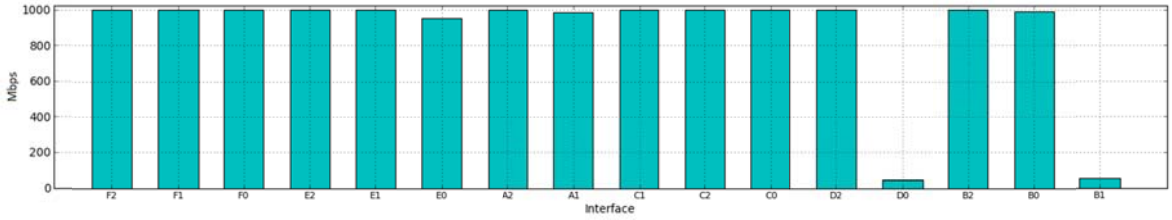

Figure 5-8: Real Time Visualization of the Available Bandwidth by Interfaces in Scenario B

Average of packets marked: In order to integrate the Congestion Evidence information into the visualization, we present in real-time the interaction between the proposed two mechanisms (ECN and OSPF-TE) and verify the results of this interaction.

No bar charts of Scenario A are shown, since due to its congestion-less characteristics there are no marked packets detected at the eNodes. Both of the following figures concern Scenario B. Figure 5-9 shows the average number of packets being marked by each interface. In this case, as was expected, the only interface that is marking packets corresponds to the output downlink queue of interface B1 of the simulated congested link. At the same time, Figure 5-10 shows the average number of CE packets detected (as provoked by the Iperf traffic) at the end point of the ECN communication, in this case, the eNode1. These CE packets are the same packets as marked by the B1 interface. The real-time visualization of the number of CE packets acts as a "congestion experienced" alarm for the network. Given the real-time visualization of the congestion parameters (bandwidth, latency, and number of packets marked), not only by path, but by interface, the network operator is now able to react and take further actions to solve the congestion problems in the network.

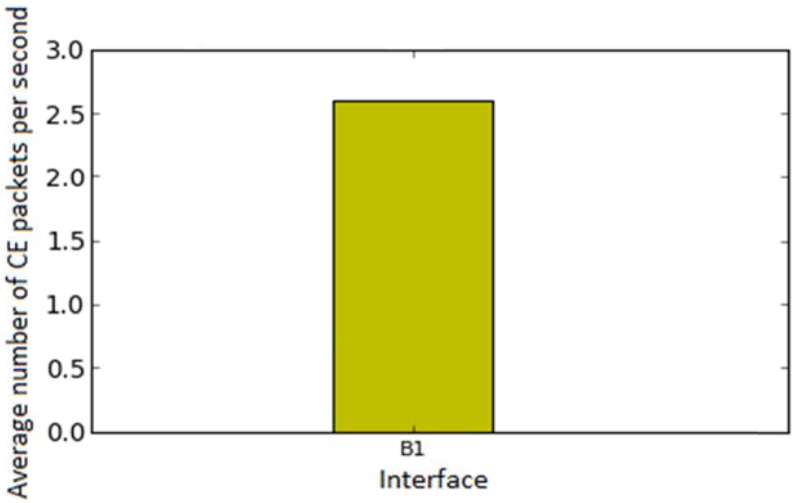

Figure 5-9: Real Time Visualization of the average number of packets marked per Interface in Scenario B

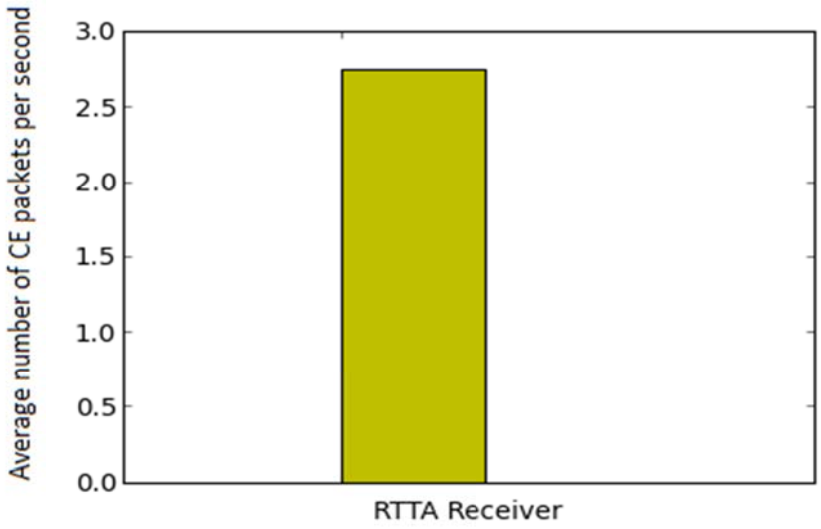

Figure 5-10: Real Time Visualization of the average number of CE marked packets detected at eN1

5.3.1.2 Per Path Analysis

An analysis of different paths was performed for two paths of Figure 5-2 {A, B, D} and {A, C, F}, in order to compare the traffic conditions of both paths (as was described in the introduction of this section). The comparisons are done of the RTT and the available bandwidth characteristics as presented by the Real Time Visualization functionality.

Round Trip Time: The study of the total RTT along the paths considers the difference between the sums of the most restrictive measurements of the RTTs along the paths considered. As shown in Figure 5-12 (corresponding to Scenario B), the RTT of the path {A, B, D} is higher than for the path {A, C, F}, due to the congestion implemented in this path. Figure 5-11 shows the corresponding RTTs for the congestion-less scenario (i.e., Scenario A). In this case the sums along the two paths are similar – which is as expected since both paths are equally loaded. Notice that the vertical scale is different in Figure 5-11 than in Figure 5-12.

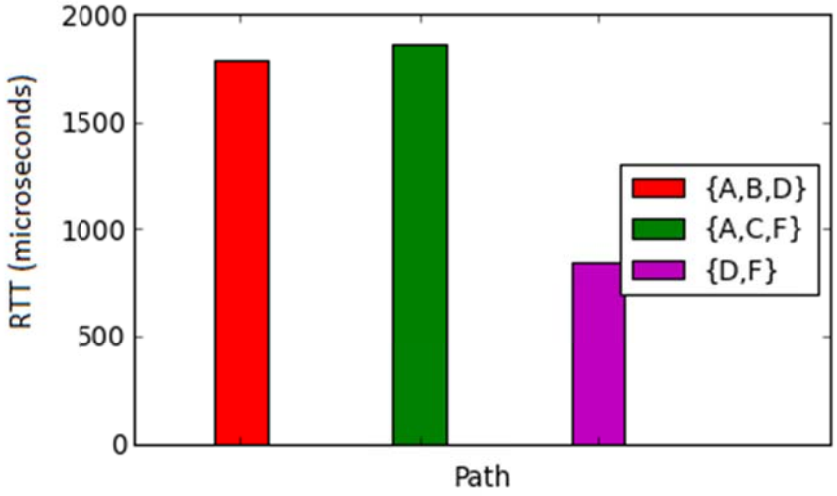

Figure 5-11: Real Time Visualization of the RTT by Paths in the Scenario A

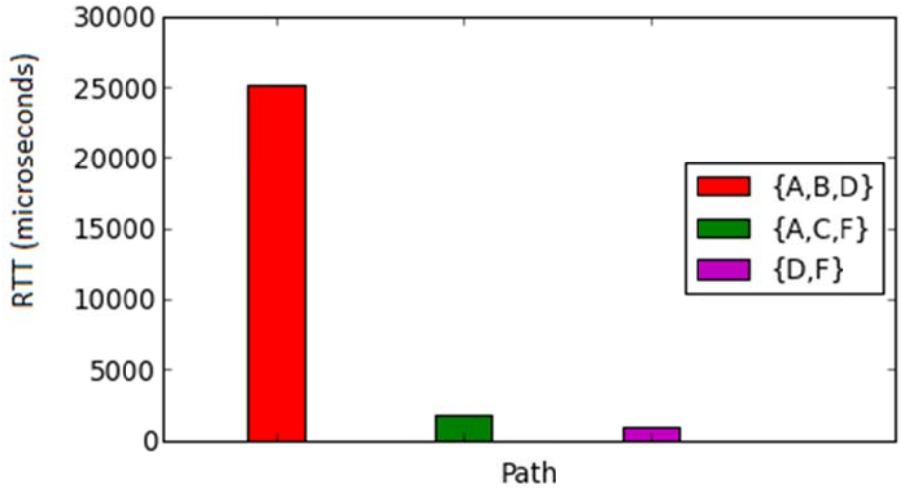

Figure 5-12: Real Time Visualization of the RTT by Paths in the Scenario B

Available Bandwidth: The comparison of both scenarios with regard to the available bandwidth reflects the differences between these two paths. Here the visualization shows the limited available bandwidth computed from the lowest available bandwidth measurement along that path. Figure 5-13 corresponds to Scenario A, where all the paths show the same available bandwidth as expected due the congestion-less characteristics of this scenario, while Figure 5-14 easily allows us to identify that the path {A, B, D} contains the simulated bottleneck (due to the BD link) and verifies that it is this bottleneck link that limits the total available bandwidth of the path.

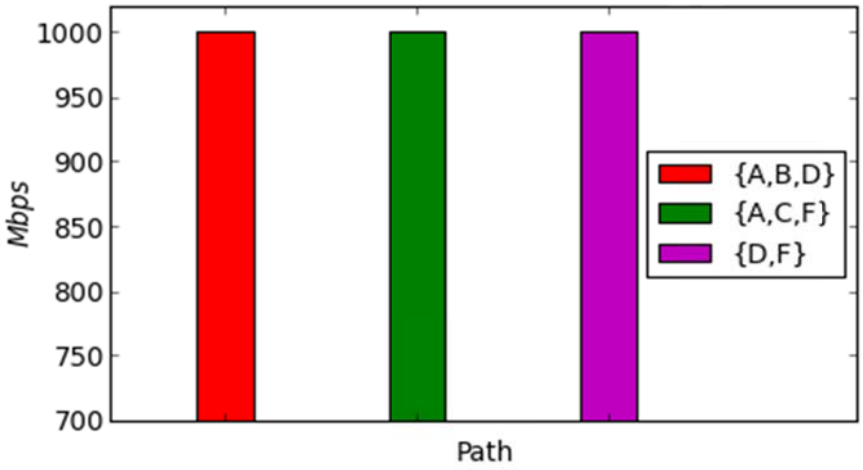

Figure 5-13: Real Time Visualization of the Available Bandwidth by Path in Scenario A

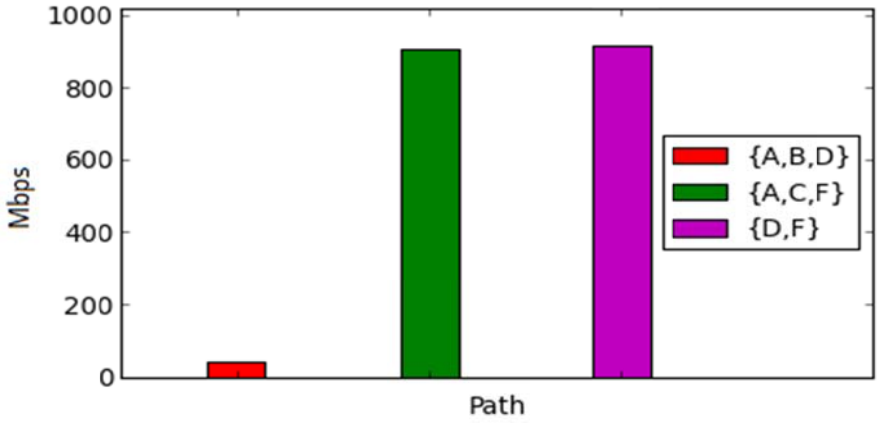

Figure 5-14: Real Time Visualization of the Available Bandwidth by Path in Scenario B

5.3.2 Time-Capture Analysis

As same way as in the previous section on Real-Time Visualization, in this section we make a comparison between the Scenario A and B to illustrate the results obtained by the proposed mechanism to analyze the state of the network *over a period of time*.

Once again we use two video streams. During the period observed, a UE connected to the eN1 experiences the same video quality as was observed for a UE connected to eN2 in the Scenario A, and much worse in the Scenario B, as several time frames in the real-time visualizations showed.

The analysis is done in terms of router interfaces and paths over 60 minutes. This pre-definite time period was selected to guarantee a steady state of the network in both scenarios over time. However, another pre-definite period that satisfies this steady state requirement could be chosen. The following paragraphs describe the results obtained in the analysis by router interface and path (respectively).

5.3.2.1 Per Router Interface Analysis

The congestion information shown in this analysis corresponds to the router located in the node B, randomly chosen for illustrate the desired example for the verification. As shown in Figure 5-2, node B has three interfaces B0, B1, and B2 which belong to the links AB, BD, and BE (respectively).

Round-Trip Time: The following graphs compare the distribution of the RTT values of the three interfaces over the pre-defined period. As we can see in Figure 5-15, the RTT distribution for Scenario A falls between 0.2 and 2.5 milliseconds during the entire period. This result is as expected due to the congestion-less characterization of this network. However, in Scenario B as shown in Figure 5-16 the values of RTT increase for interfaces B0 and B1, as they are part of the two congested links, AB and BD respectively. Note that the RTT values for interface B2 are of the same order of magnitude in both scenarios. In addition, we can observe that in Scenario B the RTT values are higher for the interface B0 than for the B1, this is due to the WAN emulator tool introducing a fixed delay in the AB link, while in the BD link the congestion conditions have less affect - as can be seen in the low RTT values. Notice that the vertical scale is different in Figure 5-15 than in Figure 5-16.

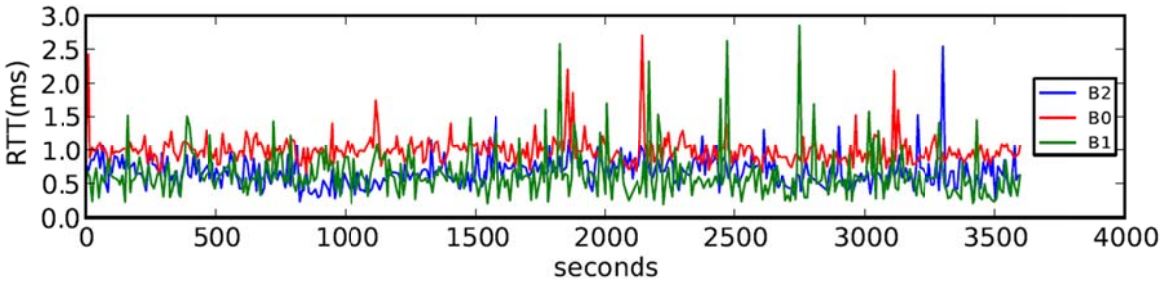

Figure 5-15: Time Capture Analysis of the RTT by Interfaces in Scenario A

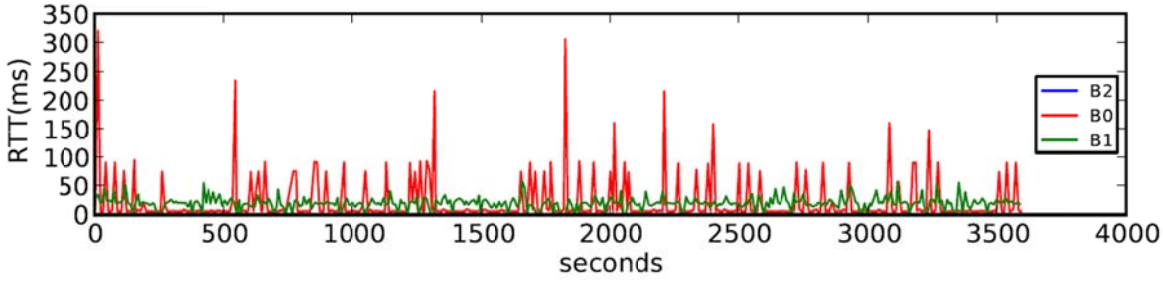

Figure 5-16: Time Capture Analysis of the RTT by Interfaces in Scenario B

Available Bandwidth: Figure 5-17 and Figure 5-18 show the Available Bandwidth at the interfaces of the router B for each scenario. The three interfaces have the same Available Bandwidth in Scenario A (Notice that the values coincide for the three interfaces in the Figure 5-17). Interface B2 has the same Available Bandwidth in both scenarios. However, the Available Bandwidth of interfaces B0 and B1 decrease in Scenario B, with the Available Bandwidth of interface B1 being the most limited due to the Virtual Box Link Characterization tools that simulates 950 Mbps of existing load on the link BD and sue to the

Iperf traffic crossing the path {A,B,D}. In Scenario B the Available Bandwidth for the interface B0 decreases by roughly 50 Mbps, due to the Iperf traffic on the link AB.

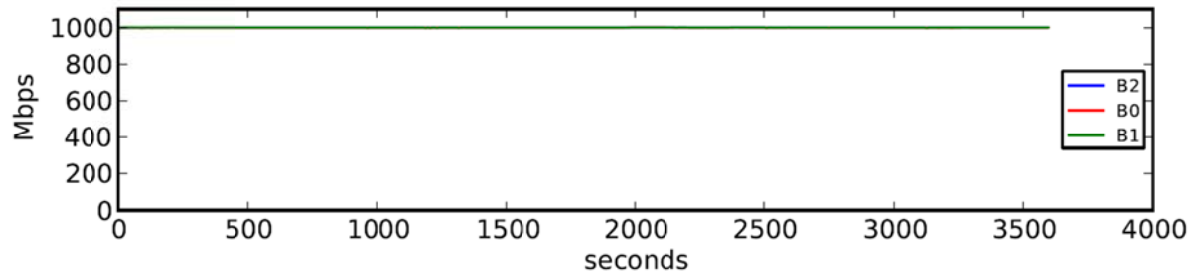

Figure 5-17: Time Capture Analysis of the Available Bandwidth by Interfaces in Scenario A

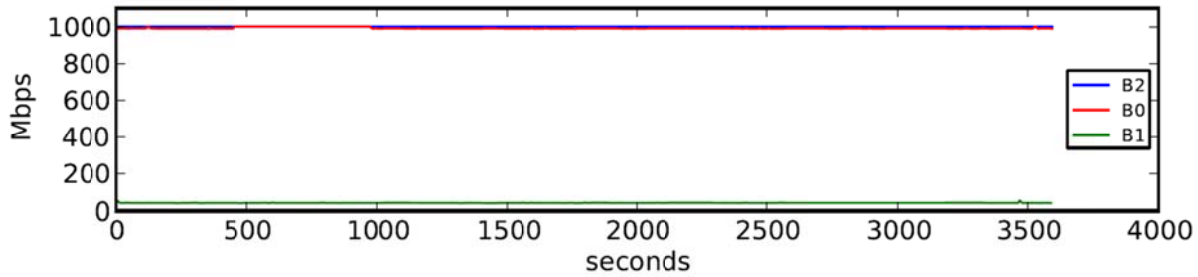

Figure 5-18: Time Capture Analysis of the Available Bandwidth by Interfaces in Scenario B

Average number of Marked Packets: The final parameter analyzed with our Time-Capture Analysis functionality is the average number of marked packets sorted by router interface. The Figure 5-19 shows these values for the three interfaces during the entire period in the Scenario B. In the Scenario A, the Average Number of Marked Packets for all three interfaces is 0 during the entire period, as the traffic in the network is really low and the queues of the router do not experience any significant load. In scenario B, the only interface that experiences high load of its output queue is the interface B1. This occurs because of the simulation of 950 Mbps of existing bandwidth being used for the link BD together with the Iperf and user traffic crossing the path {A, B, D}. However, even with the WAN emulator in the link AB introducing delay, no packets are marked at the interface B0, since the packets are not accumulating in the output queue of the interface, as the communication is downlink and there is still sufficient available outgoing bandwidth; however, these packers do suffer delays waiting for transmission. Therefore, interface B2 does **not** mark packets since there is no traffic problem for the link BE.

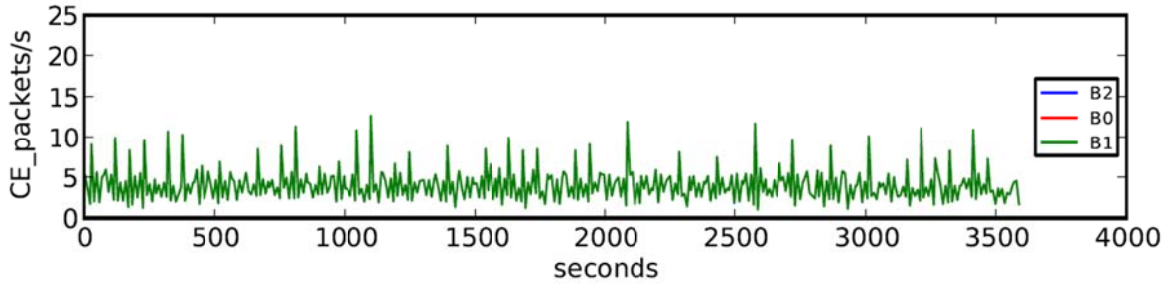

Figure 5-19: Time Capture Analysis of the Average Number of Marked Packets by Interface in Scenario B

5.3.2.2 Per Path Analysis

This set of paragraphs analyzes paths {A, B, D} and {A, C, F} of Figure 5-2, in order to compare the traffic conditions of both paths. The comparisons are done for the three characteristics recorded by the Time-Capture Analysis functionality.

Round-Trip Time: Figure 5-20 and Figure 5-21 show the RTT for the two in scenarios A and B (respectively). It is important to mention that the RTT values calculated here, as it is explained in section 4.3.2 corresponds to the sum of the most restrictive RTT values of each link along the path. As can observed in the Figure 5-20, the RTT values during the entire time period are similar for both paths and range between 0.5 and 4 milliseconds, as in Scenario A no traffic is being added to the network. However, as can be observed in Figure 5-21, in the congested scenario due to the congestion emulation tools, the RTT values for the path {A, B, D} increases to values in the range of 20-30 milliseconds, but with peaks of 100-250 milliseconds (due to timeout events in the TCP connections for these measurements). Note that the RTT values correspondent to the path {A, C, F} are the same as in Scenario A, and for this reason, its representation in the graph can be confused with the X-axis. Notice that the vertical scale is different in Figure 5-20 than in Figure 5-21.

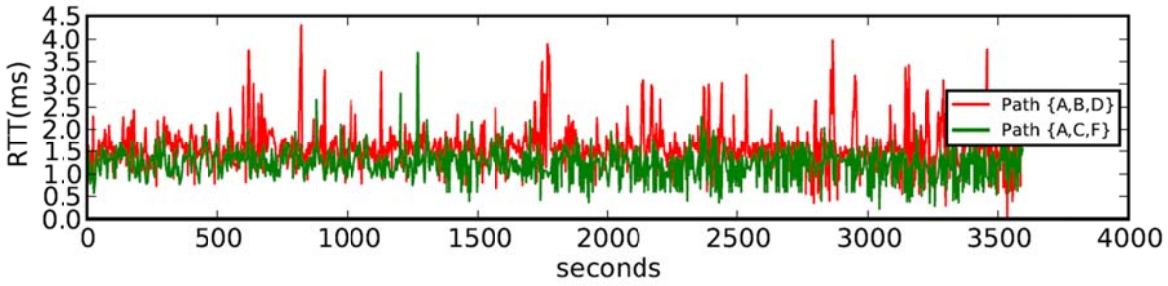

Figure 5-20: Time Capture Analysis of the RTT by Path in Scenario A

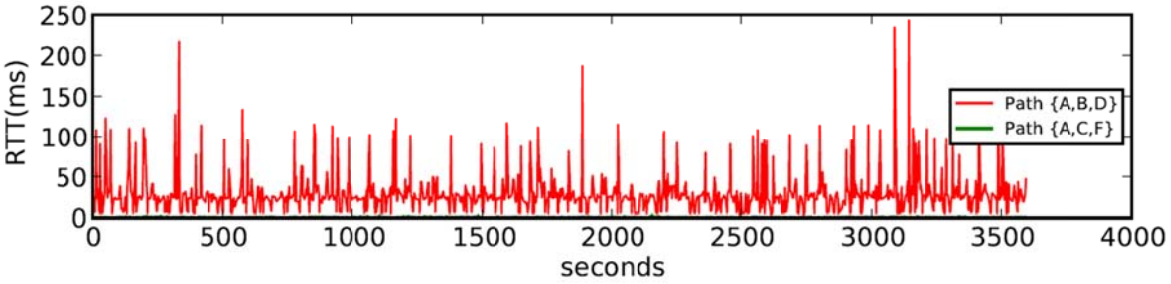

Figure 5-21: Time Capture Analysis of the RTT by Path in Scenario B

Available Bandwidth: The available bandwidth corresponds, as has been explained in section 4.3.2.2, to the value of the least available bandwidth of a link along a path. In other words, the available bandwidth is that of the bottleneck link along a path. The Figure 5-22 shows this information for the scenario B. The results obtained for Scenario A are the same available bandwidth for both paths, almost 1Gbps, as was. However, in the Scenario B, low values of Available Bandwidth for the path {A, B, D} can be observed due to the introduction of the congestions tools in the links AB and BD. Furthermore, if we compare Figure 5-22 with Figure 5-18, we can understand that the bottleneck in terms of bandwidth is the link BD along the congested path due to the Virtual Box Link Characterization tool which simulates 950 Mbps of bandwidth already being used. On the other hand, as expected, the values of this parameter for the path {A, C, F} in Scenario B are the same as in Scenario A.

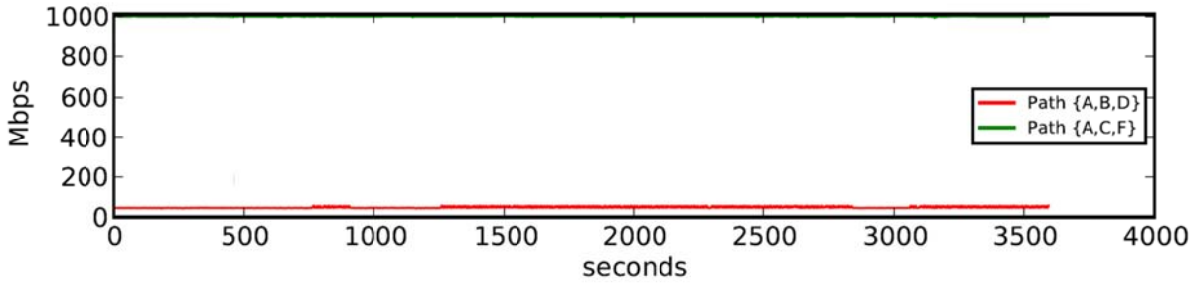

Figure 5-22: Time Capture Analysis of the Available Bandwidth by Paths in the Scenario B

Average number of marked packets: The Figure 5-23 shows the Average number of marked packets for the two paths and the Average Number of Congestion Evidence packets detected in the RTTA Receiver (located in the eN1). In the Scenario A, no packets have been marked on any paths and no CE packets has been detected at eN1 (note that the three lines of the figure coincide during the entire period of time), as no congestion traffic is added. In Scenario B, it can be observed that the Average Number of Marked Packets increases in value ranging in value between 0-8 packets marked per seconds for the {A, B, D} path; and the average number of marked packets detected at eN1 is similar for this path. In addition, comparing the Figure 5-23 with the Figure 5-19, it can be observed that the Average Number of Packets Marked in the path {A, B, D} almost coincides with the same parameter for interface B1, since the link BD is loaded with 950 Mbps of traffic and both Iperf and user traffic are downlink through this link. On the other hand, the Average Number of Marked Packets for the path {A, C, F} has the same behavior in both scenarios.

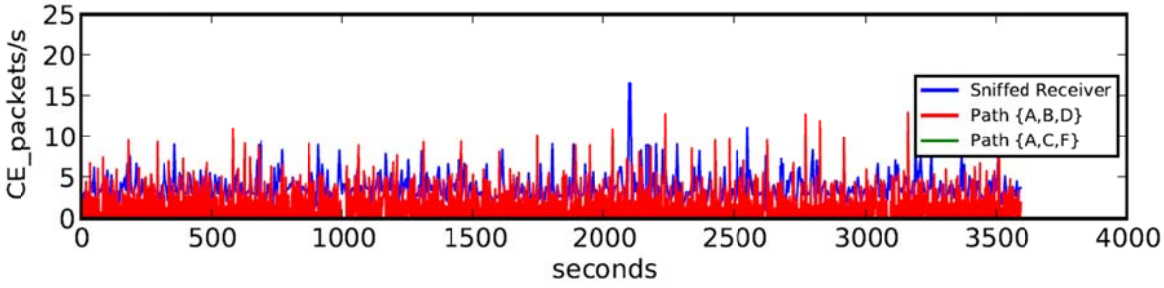

Figure 5-23: Time Capture Analysis of the Average of Number Marked Packets and CE Detected by Path in Scenario B

5.4 Conclusion of the Verification of the Results

The study of the two scenarios has confirmed that we have achieved the main goals of this thesis project. Scenario A considers the initial network *without* congestion and traffic, while Scenario B purposely introduces congestion, specifically introducing congestion along one of the paths between eN1 and the gateway; while the path between the eN2 and the gateway does not suffer congestion.

A real-time study has been conducted to monitor the state of the network in two scenarios. In Scenario B, the consequences from the UE's point of view are that it suffers due to the congestion happening on the path through the RATN. Moreover, these real-time results have been corroborated with an analysis of the congestion information done for a long period of time. In the Appendix B is given a complete visualization for the scenarios proposed.

In addition, the Scenario B illustrates the difference between the QoE of a video that two UEs, connected to two different eNodeBs, will experience. This difference is supported by the study of the links' congestion parameters (available bandwidth, RTT, and number of ECN marked packets) and their interaction with the CE evidence observed at eN1. This analysis demonstrates that the RTTA Router in the links of the congested path, and specifically the one which is the objective of the most restrictive congestion emulation features, provide more critical measurements of these parameters compared with those supplied by an RTTA Router of the other path. At the same time, the observation of these parameters in the eN1 corroborates the information concerning the evidence of congestion in the network, as provided by the detection of CE packets done by the RTTA Receiver in the eN1.

As it has been shown in this case study, the implementation of the proposed mechanism allows us to obtain information about the state of the links of the network. With the information gathered and processed by the RTTA Receiver, further actions can be proposed in order to improve the QoE of the UE connected to the eNodeB whose path to the gateway is congested. These actions could imply a handover of the UE to another eNodeB (if this eNodeB is close enough) or balancing the load between these two base stations. The combination of these visualization methods that have been implemented, confirm that: The interaction of the two proposed mechanisms provides relevant traffic information and consequently helps to identify congestion in a RAN.

6 Conclusions and Future work

This chapter describes a general conclusion based upon the work done during this thesis project and provides suggestions for next steps in order to continue the work. The intention of this chapter is to point out the most relevant ideas of the thesis and then consider how to build upon them in future research.

6.1 Conclusions

The mechanism proposed in this thesis project gives a solution that can be used to identify congestion in a Radio Access Transport Network. This thesis has described, discussed, and verified in a test-bed the proposed mechanism, giving a complete framework that supports it for continuing working on it or even deploying it into real networks. Taking into account the sub-goals stated in section 1.3, they can be divided into two: (1) "investigate protocols related to congestion control and the interaction between these protocols" and (2) "propose a mechanism that identifies congestion in a radio access transport network". Having divided them in this way several reflections may be made:

- The investigation of previous work allowed us to comprehend the scenario central to this thesis project, enabling us to detect and understand the most important relevant concepts. The study of mobile networks, in particular, understanding the RATN and its interaction with other parts of the network, guided us to focus on the RATN it the next steps of the project. This lead to research concerning the ECN and OSPF-TE protocols and how these might help us locate the congestion and then exploit the congestion control protocols operating in the RATN. The study of the interaction of these protocols and the consideration of new concepts, such as I2RS, leaded us to the design the proposed new mechanism.
- The design, implementation, and verification of the proposed mechanism address the second sub-goal of the thesis. Although, many issues can be discussed as part of an effort to optimize the mechanism proposed here. Some of these issues are related to the reliability of the mechanism when measuring the congestion information (as was discussed in section 3.2.1.1 concerning the design and in section 4.2.1 concerning the implementation). In addition, if we could travel back in the time, we would have focused on optimizing the policies regarding forwarding or not forwarding certain congestion information, and from an implementation point of view, we would have placed more attention on the efficiency of the code that we have developed.

As a result of the above reflections, the next section proposes new actions that could be undertaken in order to improve the proposed solution.

We conclude this thesis with a summary of some of the personal achievements of the authors during the project. This thesis project has given us a greater understanding of mobile networks, particularly of the interaction of the Radio Access Transport Network with the Radio Access part and the protocols used therein, learning new programming languages (Bash, Python), and improving our command of C programming language and Linux environment. We are especially thankful for the knowledge that we have gained about network configuration and traffic control. Moreover, carrying out this thesis project has improved the authors skills related to solving problems, carrying out a large project, and working as part of a team.

6.2 Future work

Following the description of the work done during this Master's thesis project, we have a number of different suggestions about work that could follow and build upon our research. This future can be mainly divided into two areas: (1) research and discussion focused on improving the design and implementation and (2) the exploitation of the possibilities and ideas that have been introduced throughout the thesis. In the paragraphs below we summarize the proposed future work.

The first direction for proposed additional research is the study and use of protocols other than those described here. The research would focus on the features that these other protocols provide that could be used to obtain the required information. In this thesis the ECN protocol has only considered the case of IPv4, hence an interesting related effort would be an implementation of the proposed mechanism for IPv6, following RFC 3168 [13] and the work done by the KAME project [89]. Some of the other possible protocol options that could be considered and studied are MPLS-TE [27], IGRP-DUAL [57], and IS-IS [90]. These protocols provided similar traffic engineering features to those provided by OSPF-TE (OSPF-TE is even part of MPLS-TE), hence these protocols could be used to distribute routing and congestion information throughout the network. Together with this research, a very interesting extension of the work here presented would be the distribution of the traffic engineering information between different areas or ASs, for example by utilizing inter-domain routing protocols such as BGP [91].

The next area for potential future research is implementing and performing the traffic measurements via the RTTA Router. During this thesis, the focus was finding a suitable way to extract the desired parameters via a simple implementation in our test-bed. We have not focus on their accuracy. Therefore an obvious area of future research is to improve the *quality* of these measurements. For example, in the case of the available bandwidth parameter, deeper research should be undertaken to estimate the total available bandwidth. As was explained in the section 4.2.1.2 the measurement of the one-way delay had to be excluded from this thesis project due to the low synchronization accuracy provided by NTP alone. However, by deploying PTP [73] in the network, this parameter could be accurately and reliably measured, providing more precise information about the state of the links. Finally, the configuration of the output queue of the routers interface that was deployed did not follow any standard or recommended model, rather the aim was simply to provide evidence of packets being marked within the network. Once this evidence has been verified, a following step may be an investigation comparing different AQM configurations together with a complete study in order to provide the most suitable characterizations of these queues. Moreover, the software developed during the thesis project sometimes lacks efficiency and some sections and modules could be improved to achieve better performance.

In conjunction with section 3.2.2, relevant future research could be done about what policies should be followed in order to control the flooding of the traffic, for example building upon the work presented by Roozbeh in his thesis [53]. However for the case considered in our study, a complete investigation of more suitable updating policies could provide information relevant to improving the efficiency of the distribution of the congestion information. At the same time, this study could be combined with a deeper analysis of different simulated scenarios. In section 5.2, two different characteristic scenarios were proposed and monitored in order to verify the proposed solution. In addition to these two scenarios, more complex simulations can be performed, for example by changing the configuration of the traffic parameters and adding more tools to characterize the network in

different ways. This analysis could provide a more complete verification and analysis to support the usage of the proposed mechanism.

In this master thesis, access to the information from the routing system has been designed according to the idea proposed in the I2RS draft document [92]. While the implemented approach provides important features of this interface, future work future work could extend this both for our proposed solution and also for other network systems.

Finally, the mechanism proposed in this thesis is just the beginning of future work that could be done in order to solve (or avoid) congestion problems in a RATN. Once the first step identify where exactly the congestion is happening, is completed; then the next steps should focus reactions to this congestion in order to ensure the expected QoE. For instance, future research could study the impact of this proposed solution on the decision that an eNodeB with UEs connected to it (that is aware of the existence of congestion in its path to internet and has identified this congestion) can take; for example, to perform a handover to another eNodeB that is aware that its path is not congested. Another possible direction is to study load balancing between different eNodeBs and how the user's IP tunnels between the PDN-GW and the UE's current eNodeB should be modified.

6.3 Required reflections

This section discusses the social, economic, and sustainability aspects of deploying the congestion identification mechanism proposed in this master thesis project in Radio Access Transport Networks. The deployment of this mechanism enables mobile networks operators to take actions in order to provide the expected quality of experience to their users. This effect should drive mobile networks operators to procure the promised services, increasing the client satisfaction which is a desirable **social** effect of this master thesis project.

Moreover, the implementation of the mechanism described in this thesis, does not need any change in the current network architecture and any new element has to be included. Just an extension of the existing elements in the RATN allows providing the features described in this report. This implies a relevant **economic** issue to the operators involved in the deployment of the network because this enhancement does not require any extra economic cost. In addition, the real time reaction that this mechanism provides in order to exactly identify the congested link in the network, allows to the mobile network operators to decrease the cost related to solve network congestion problems, which is an important **economic** benefit of this work.

Furthermore, the knowledge of the state of the network allows the properly use of the existing resources and avoid that mobile network operators invest in extra equipment to provide the expected quality of experience to the users. This reduction in terms of investing in resources and equipment implies a more **sustainable** solution in mobile networks.

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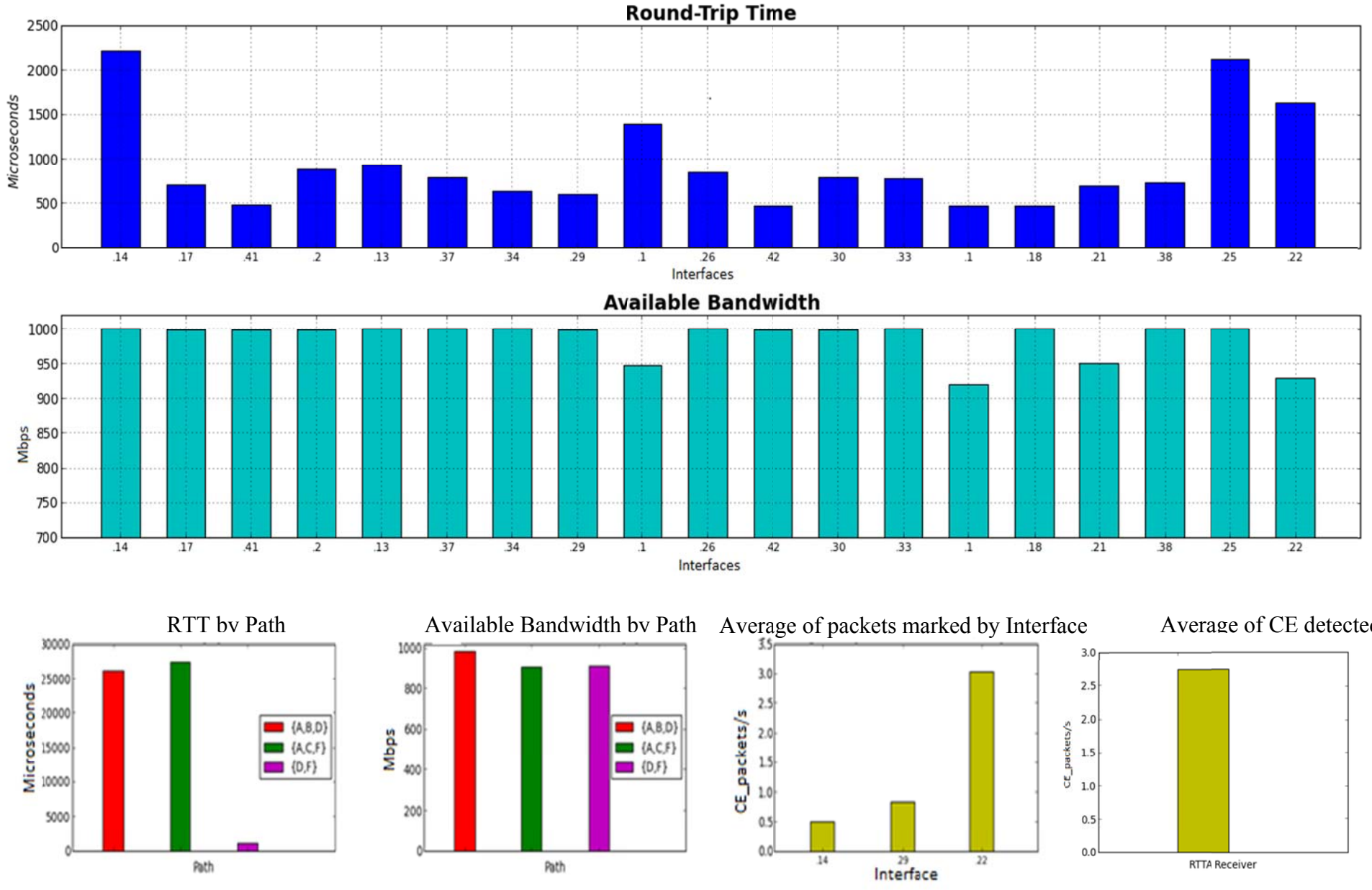
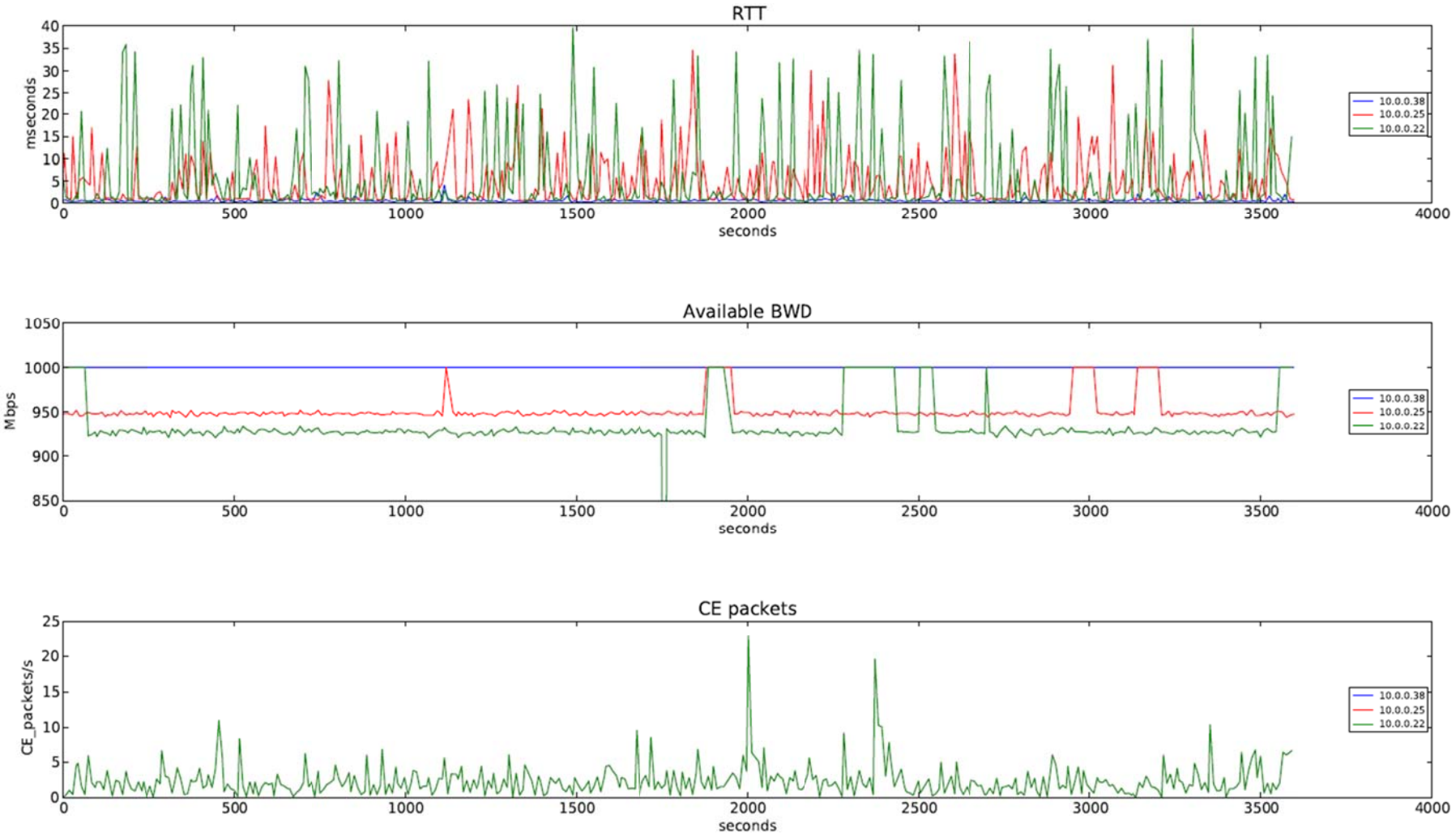
Appendix A Analysis and Visualization: Implementation

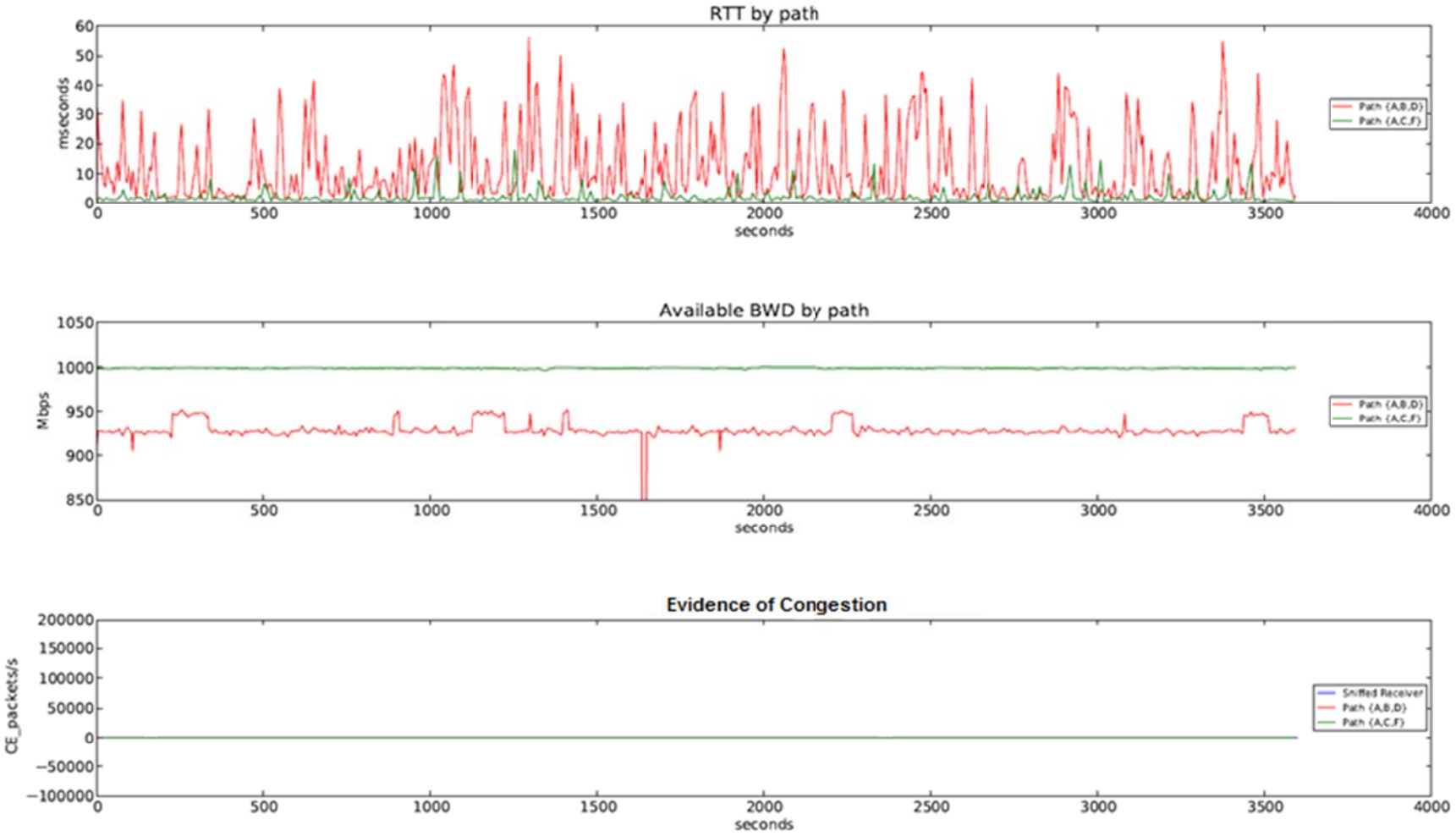
The purpose of this appendix is to help the understanding of the Analysis and Visualization module of the Implementation chapter. The first diagram corresponds to a capture to the Real-Time Capture in a certain moment, and the two seconds graphs correspond to the analysis performed by the module Time-Capture for router interfaces of a certain router of the test-bed and for two paths of the test-bed, respectively.



Appendix Figure A-1: Capture of the Real-Time Visualization in a certain moment.



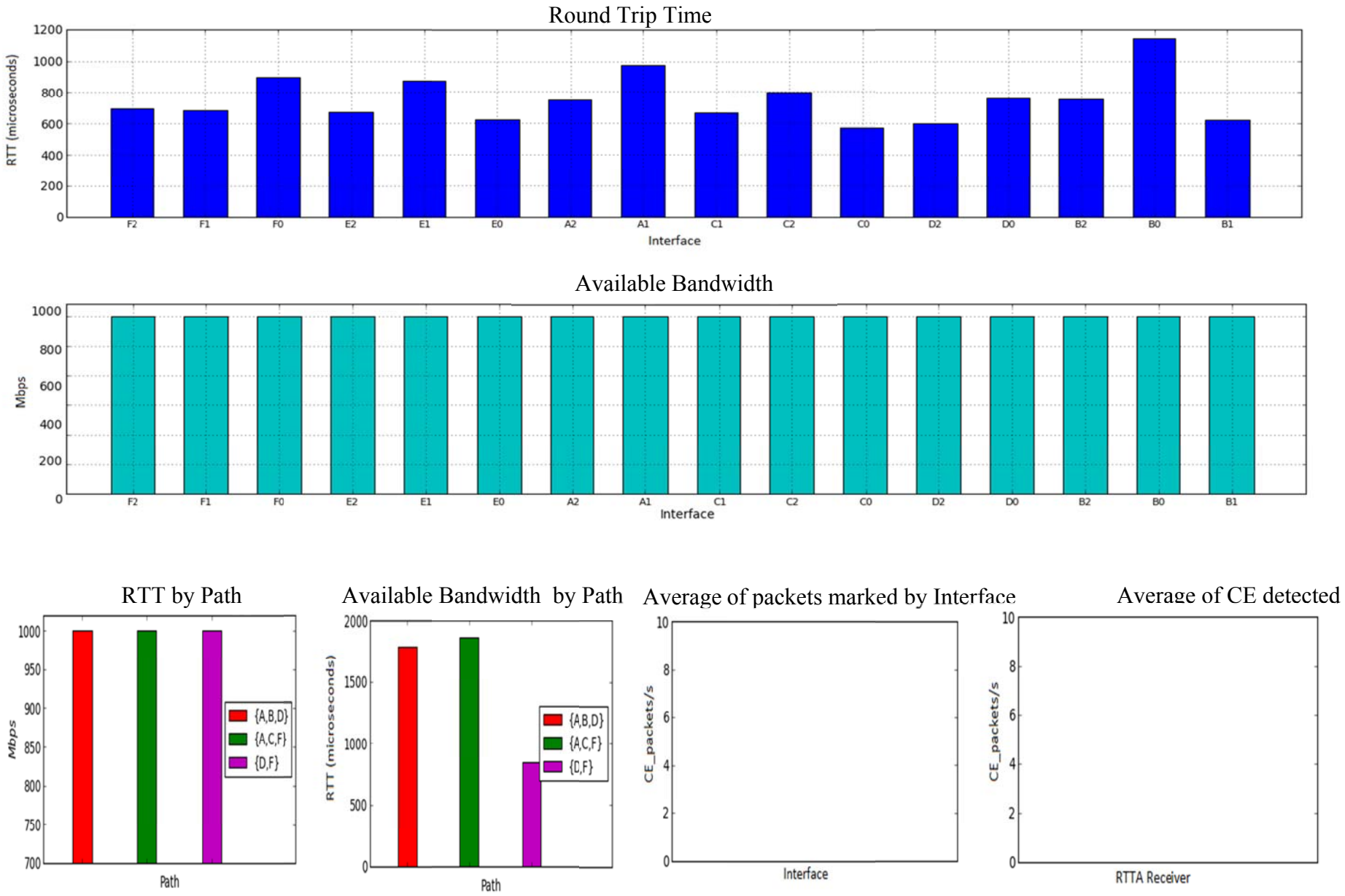
Appendix Figure A-2: Time-Capture Analysis for the interfaces of the Router .22 in 60 minutes.



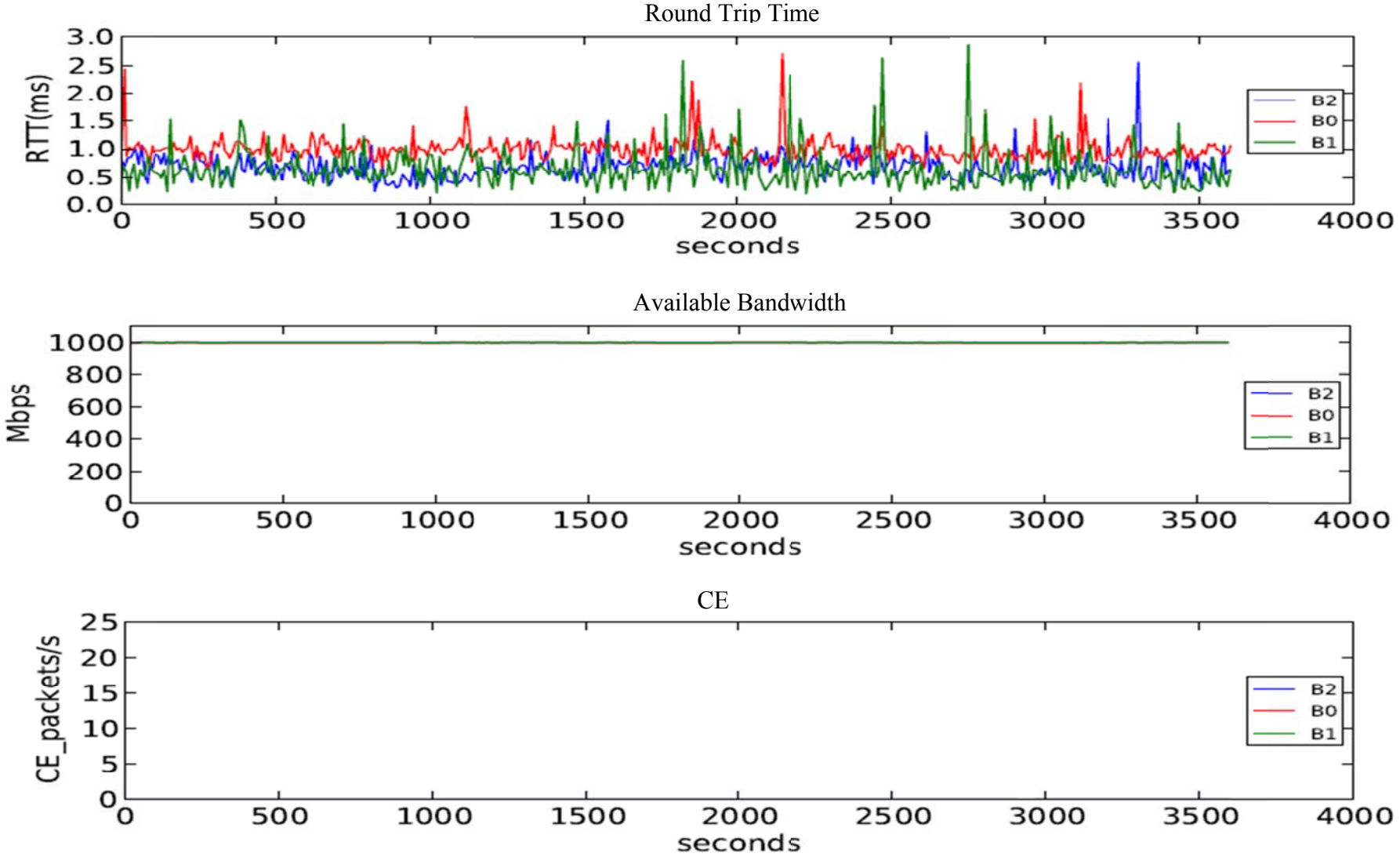
Appendix Figure A-3: Time-Capture Analysis for two paths of the test-bed in 60 minutes.

Appendix B Analysis and Visualization: Verification for both scenarios

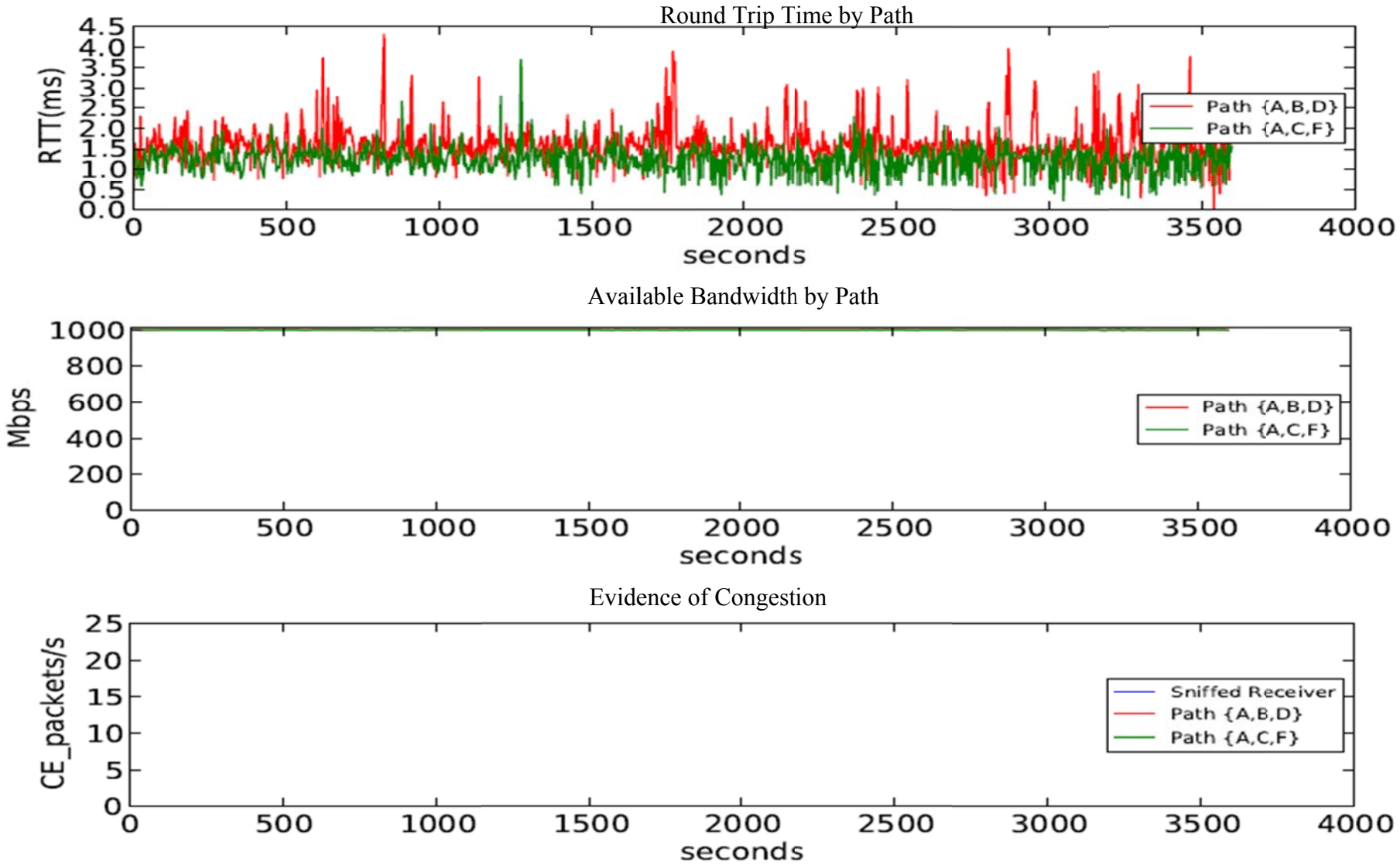
The purpose of this appendix is to provide the full visualization for the two scenarios, A and B, proposed in the Verification chapter.



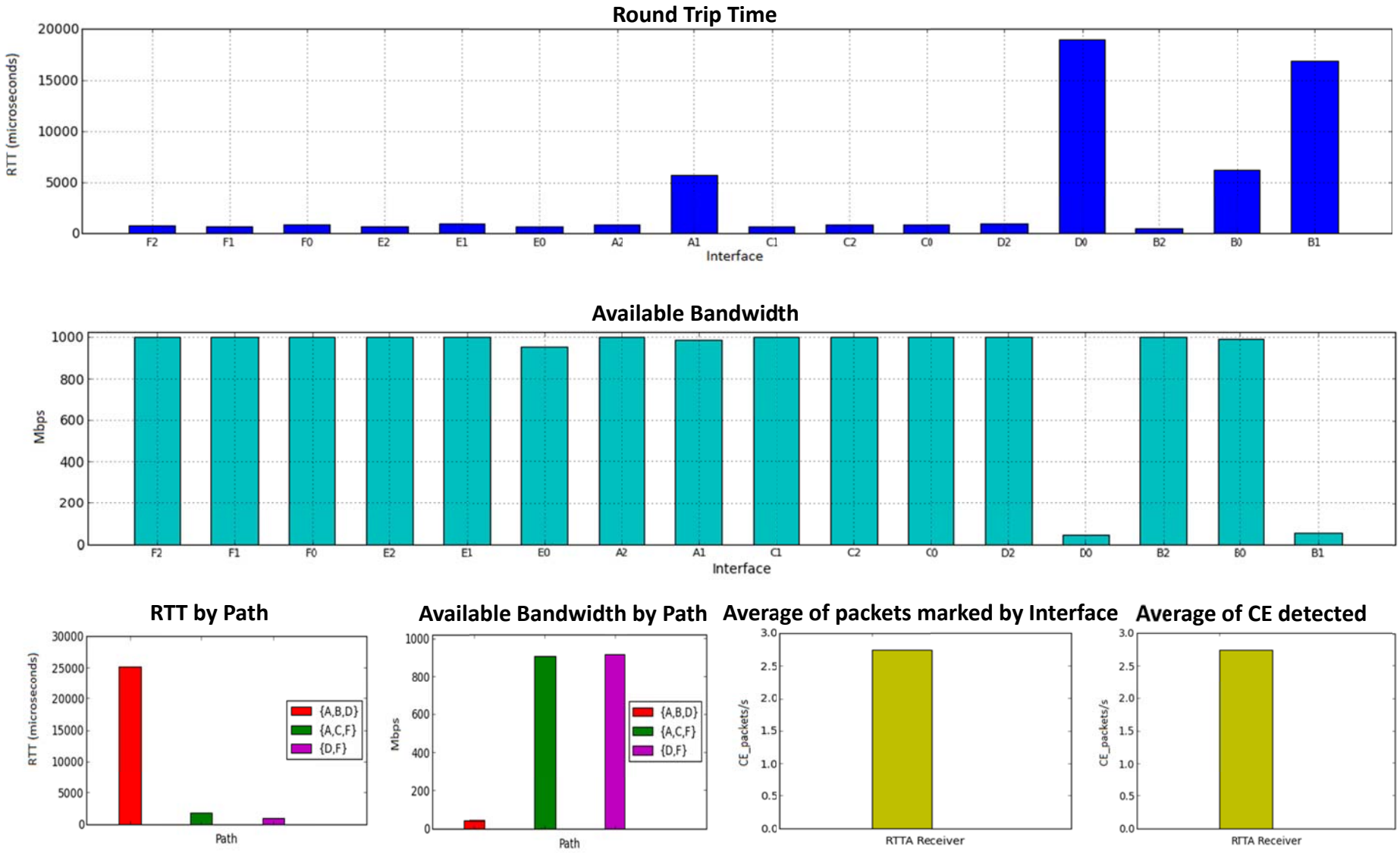
Appendix Figure B-1: Capture of the Real-Time Visualization of the Scenario A in a certain moment.



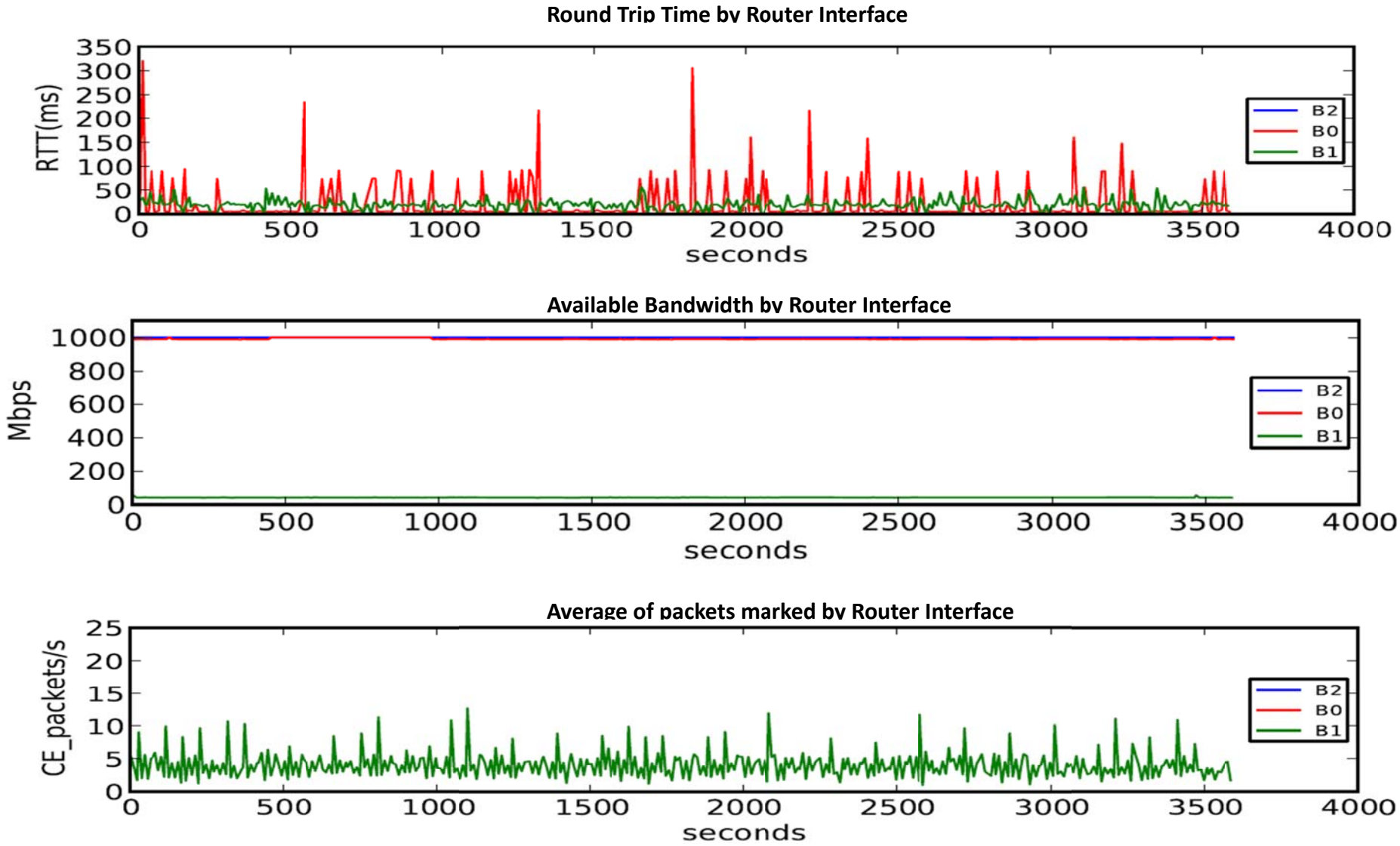
Appendix Figure B-2: Time-Capture Analysis for the Router B in the scenario A in 60 minutes.



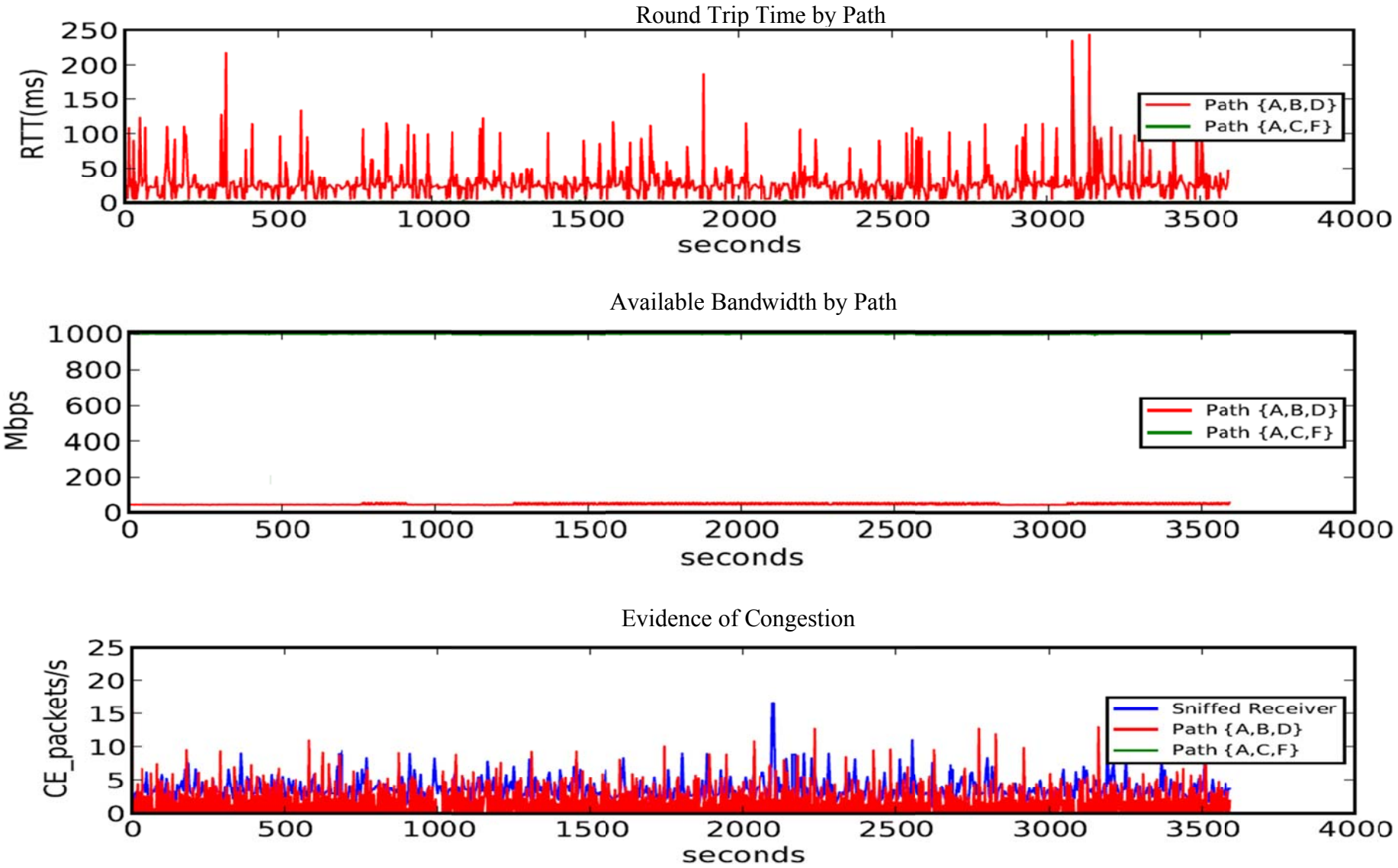
Appendix Figure B-3: Time-Capture Analysis of the scenario B for paths in 60 minutes.



Appendix Figure B-4: Capture of Real-Time Visualization for the scenario B in a certain moment.



Appendix Figure B-5: Time-Capture Analysis for the Router B for the scenario B in 60 minutes.



Appendix Figure B-6: Time-Capture Analysis for the scenario B for paths in 60 minutes.

TRITA-ICT-EX-2014:15

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