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Scalable Quality of Service Solutions for IP Networks and Services

Csaba Simon

Electrical engineer

Summary of the PhD Dissertation

Doctoral School of Informatics

Faculty of Electrical Engineering and Informatics
Budapest University of Technology and Economics

Supervised by

Dr. Tamás Henk

Department of Telecommunication and Media Informatics

Faculty of Electrical Engineering and Informatics
Budapest University of Technology and Economics

Budapest, Hungary

October, 2011

1. Introduction

The telecommunication industry has always been able to incorporate the new technologies. Starting from the early telegraph systems it has continuously evolved to offer richer and diversified services to customers. The recent decades challenged again the telecommunication industry, as the digital telephone systems are integrated with the technologies originally developed to support computer networks. This process led to the convergence of networks and Internet Protocol (IP) emerged as a common platform that forms the basis of future infocommunication systems.

The major appeal of the IP technology relies in its flexibility to implement new services, which is the key to remain competitive in a global market. The quality of these services experienced by the users is a key factor to accept and embrace new services. Although pricing and marketing is becoming more and more important in current infocom markets, there are still open technological questions, which, if properly addressed, give a competitive edge to the service providers.

The most widespread standards that answer these challenges are the Integrated Services [1] and Differentiated Services [2]. The latter scales better with the number of flows and one of its variants is the relative service differentiation [3], introduced as a per hop based service [5][6]. The advantage of this service is its predictability at node level (i.e., per hop), but the operators should offer their services over their domains, which calls for a network-wide service definition. A much detailed insight into these issues is provided in section 4.1.

The extension to global level and the rich set of services comes with a price though, as it requires the introduction and deployment of new features. The management of this architecture and of these services hit the limits of scalability of the established (legacy) management frameworks. In legacy network management, the management-plane is a centralized process [7][8], being responsible for a vast area of functionalities, as described by the FCAPS (Fault, Configuration, Accounting, Performance, Security) structure of TMN (Telecommunication Management Network) [9]. In order to meet the above challenges of future networks [10] the network management systems have to become more dynamic, more self-managing, and they have to participate more actively in (inter)-networking [11], going beyond “classical” or legacy network management paradigms, as detailed in section 4.2.

The Dissertation investigates different aspects of such IP based infocommunication systems, focusing on the scalability issues of such architectures. A short background of these aspects is given separately in the beginning of the sections, which introduce the new results obtained in those research areas. It offers *a scalable QoS framework*, as the role of QoS is paramount in new multimedia (e.g, streaming) services. Nevertheless, the deployment of such services is possible only if *a scalable management framework* can support such flexible, ubiquitous networks and services.

2. Objectives

The main objective of the Dissertation is to pave the way towards a ubiquitous, IP-based, service oriented telecommunication network.

The economies of scale offered by the use of single integrated architectures that span the globe and enable customized services would offer both social and economical benefits to all stakeholders. Several existing technologies can be used as building blocks of such a global picture, but these blocks cannot be glued together as one would do with the pieces of a puzzle. The major difficulty is not the missing interface, but the fact that these building blocks have their intrinsic limitations in terms of scalability or the combination of these will not yield a fully predictable system.

The objectives of this Dissertation are to address two aspects of the above issue.

The first objective is to develop a solution that assures the QoS of a network in a predictable and scalable manner in IP networks.

In order to be able to build a global network that meets user expectations the network must be able to offer predictable Quality of Service (QoS) to the applications, and maintain this offer on a global scale. I introduced a domain-wide proportional service that guarantees that the ratio between the qualities of service experienced by two users served in different classes will remain the same irrespective of the changing network conditions. This approach differs from the widely used Per Hop Behavior (PHB) based ones and I proposed several algorithms that impose the proportional service in the network and validated them through simulations. Thus a user who selected a higher service class will know that she/he always gets a better service and she/he will also know how much better this service will be. With the spread of layered coding of multimedia content the users will be able to adjust their communication costs according to their budget and/or the importance of the content they consume.

The second objective is to offer network management solution for dynamic network composition in self-organizing networks.

The management of large, heterogeneous, global and dynamic networks cannot be kept centralized anymore, because they would not scale. According to today's practice user/operator intervention is required in order to combine networks or services, taking into account user/network context (e.g., her/his needs, preferences, her/his location). I introduced a peer-to-peer hierarchical overlay that can hide local interactions, keeping the overall system scalable. I proposed new interactions called compositions between partitioned overlays that model the combination of two separated network domains. Thus network interaction is automated through online negotiations between devices and network, making it transparent to the services and/or to its users.

The above objectives, once fulfilled, enable the creation of a global, (potentially) mobile, ubiquitous IP based services-oriented network.

3. Methodology

Mathematical modeling

I used the instruments of graph theory to define a new peer-to-peer (P2P) overlay model and to propose new operations that model typical service management operations. I analytically deduced the quality parameters of a novel QoS service type.

Simulations

I used the well-known ns2 packet level simulator tool to model and investigate the properties of the proposed QoS service type. I based my analysis and verification of the proposed algorithms on the simulation experiments effectuated with this tool.

I designed a simulation model and implemented a flow level simulator based on this model in order to validate the proposed algorithm and to investigate the performance characteristics of the proposed hierarchical P2P overlay model.

Prototype implementation and measurements

I used prototype implementations as proof-of-concept of the P2P overlay model.

4. New Results

4.1 Proportional Services

In the last decade two major solutions, namely Integrated Services [1] and Differentiated Services (DiffServ) [2] have been introduced to empower IP networks with Quality of Service (QoS) capabilities. Among these two, DiffServ, which exhibits a better scalability, is based on a per hop shaping of the traffic. The nodes control independently the flows without knowledge about the network state or/and the limitations suffered by the flows at other hops. Besides the advantages, DiffServ has also some drawbacks. One of the drawbacks comes from the static allocation of the internal queuing parameters. For each traffic class a pre-allocated weight is established, which is used by the schedulers in the core routers. If the ratio of the traffic classes changes, then degradation of higher priority traffic classes could happen, while a lower priority class gets a better service.

A possible solution to this problem is the relative service differentiation [3], where a class i gets a better service (or at least not worse) than class $i-1$. Differing from the absolute service guarantees, the proportional differentiation model ‘spaces’ certain class performance metrics proportionally to the differentiation parameters that the network administrator determines [3][4].

Definition 1.1 Let us consider the differentiation between m different service classes, where q_i is a QoS performance measure for class i . The Proportional Service (PropServ) model imposes the following constrains on all pairs of service classes:

$$\frac{q_i}{q_j} = \frac{c_i}{c_j}, i, j = 1, 2, \dots, m \quad (1)$$

where $c_1 < c_2 < \dots < c_m$ are the generic quality differentiation parameters (QDP).

In the case of relative differentiation the QoS parameters are relative to each other, depending on the overall network capacity. It has been shown in the literature that in the

case of the “classical” DiffServ disciplines, if the higher quality classes are overpopulated compared to a lower class, then the flows classified in the higher class might receive worst service than the lower class flows [3][4]. The advantage of the PropServ approach is that the QoS differentiation is a-priori known to the flows, as opposed to the original DiffServ PHB based solution.

Thus even if the actual quality level of each class varies with the class loads, the quality ratio between classes remains fixed (as defined by the QDPs), which remains the same, even if there are changes in the class loads. Additionally, the relative ordering between classes is consistent and predictable from the users’ perspective. In this context there is no admission control and resource reservation; it is up to the users and applications to select the class that best meets their requirements, cost and policy constraints.

Nevertheless the original PropServ proposals in the literature focus on PHB based implementation, where the differentiation is applied on a node level (potentially in each hop) whereas the user perceives the QoS on an end-to-end level [5][6][12]. Complex monitoring and scheduling architecture is required in order to guarantee end-to-end PropServ [13]. Alternatively, there are proposals to apply the PropServ model on a per-flow basis [14] (opposed to the aggregated-flow approach of the original proposal), which flaws the inherent scalability of the original model.

To maintain scalability, algorithmic complexity should be pushed to the edges of the network whenever possible [15]. Therefore in Thesis 1 I propose a novel proportional service model, which acts at network level and a network architecture that follows this model.

Thesis 1. [J1][J2][C1][C2][C3] *I proposed an IP based network architecture that offers proportional differentiation among classes at network level.*

I proposed a novel proportional differentiation model that, opposed to the per-hop models, assures proportional differentiation at network level. I proposed a network architecture that assures this proportional differentiation in the network. I derived analytically the parameters of this system and I gave two algorithms that enforce the network-wide proportional service model both in distributed and centralized manner in the algorithm.

The proposed model is the extension of the DiffServ Proportional Service PHB, therefore the network handles the flows at aggregation level. In order to ease the discussion of the model I introduce the definition below.

Definition 1.2 I use the term *micro flow* for an IP packet-stream that crosses the network domain from a given ingress (entry point into the network) to an egress (exit point from the network), follows a given path and belongs to the same communication session. The aggregation of all the micro flows using the same path between a given ingress and egress and belong to the same QoS class is called *macro flow*.

Note that over a given path (x) there are lots of micro flows, each micro flow being part of a macro flow. Over the same path there are at most m macro flows, each belonging to a different QoS class. The bandwidth of a macro flow is noted by $F_{i,in}$. If not noted otherwise, the term flow should be interpreted as a macro flow.

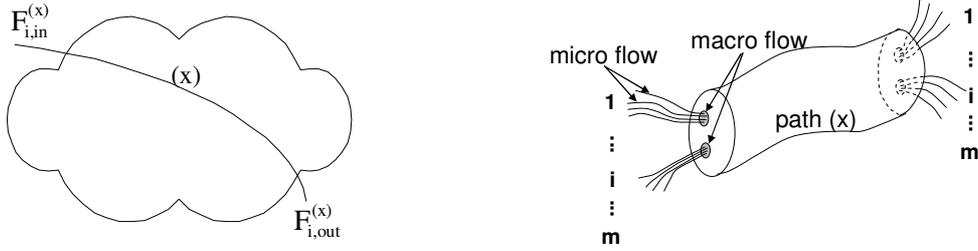


Figure 1.1. Input and output of a flow (on the left) and micro- and macro flows (on the right)

The network-wide proportional services are defined for the goodput performance measure. Since there are slightly different interpretations of the goodput in the literature, I give the definition of my interpretation below.

Definition 1.3 For a given QoS class i the goodput, as quality parameter in the network-wide proportional services model is G_i

$$q_i = G_i^{(x)} = \frac{F_{i,out}^{(x)}}{F_{i,in}^{(x)}}, \quad i = 1, 2, \dots, m, \quad (2)$$

where $F_{i,in}$ is the offered load of a flow F_i , the input bandwidth that appears at the edge (ingress) of the domain. Similarly, $F_{i,out}$ is the bandwidth of the same flow F_i , as it leaves the network domain. The upper index (x) denotes the path of the flows, as depicted in Fig. 1.1. The set of all paths (x) is noted with P .

This definition can be applied to both responsive and non-responsive flows. The literature gives several options to classify the IP traffic according to its capacity to respond to congestion in network. Two of most used definitions call them responsive and non-responsive flows [16], or refer to them as streaming and elastic traffic [17]. I will use the following terminology in this document. Theoretically the IP header should contain enough information to decide, whether it is a responsive or non-responsive flow, but in practice session layer (i.e., TCP or UDP) headers could be consulted, as well.

Definition 1.4 A *responsive flow* is a flow that deploys a congestion control mechanism, thus modifies its sending rate as a function of observed congestion event. A *non-responsive flow* is a flow that does not deploy congestion control.

For non-responsive (e.g., UDP) flows the $F_{i,in}$ is the actual number of bytes arrived at the ingress in a unit time.

For responsive (e.g., TCP) flows the interpretation of $F_{i,in}$ is slightly more complicated, as detailed in Thesis 1.5.

Thesis 1.1 (Network-wide proportional service model and architecture) [J1][C1] – I proposed a network architecture that assures relative differentiation to flows crossing an administrative domain, called network-wide proportional service.

I provide the list of the networking elements and a functional description of each element and their interactions of this architecture.

The network wide proportional service is a QoS differentiation model applied to all the flows of this network and which holds the properties below.

- *A given class i receives the same service differentiation over the whole network domain, thus the per-class QDPs are defined at network level (c_i is the same for all flows from class i , irrespective of their path in the network).*

- This service is based on the goodputs of the flows and it holds the following equation

$$\frac{q_i}{q_j} = \frac{G_i^{(k)}}{G_j^{(k)}} = \frac{\frac{F_{i,out}^{(k)}}{F_{i,in}^{(k)}}}{\frac{F_{j,out}^{(k)}}{F_{j,in}^{(k)}}}} = \frac{c_i}{c_j} \quad (3)$$

for classes $i, j = 1, 2, \dots, m$ and for all paths (k) .

- There is no Call Admission Control – every microflow is accepted in the network and it receives the service according to the QoS class it belongs to.
- The enforcement of the proposed service is performed once along the path of the flow, that is the QoS shaping and/or policing is performed at the ingress (the entry point of the network).
- In the proposed network architecture the loss rates of the flows of a class are no worse than the loss rates of the flows of any lower quality class.
- The network-wide total (including propagation, queuing and processing) delay of the packets of a class is no worse than the delay of the packets of any lower quality class (i.e., if we note with d_i the total delay for class i , then $d_j \geq d_i$, if $j < i$, $i, j = 1, \dots, m$).

Given an algorithm that computes the parameters of the network-wide proportional service (as proposed above in eq. (3)), the network architecture holds the following further properties:

- it is able to differentiate (classify) the flows based on its paths within the networks and its QoS class.
- it is able to treat flow rate and number, flow path, link capacities, flow IDs crossing the links as input parameters to the **algorithms** computing the service parameters.
- it is able to enforce the bandwidth rates at the ingresses, received as the outputs of the **algorithms**
- it is able to separate the handling of the packets of flows with different QoS class
- it is able to support both centralized (at domain level) and distributed algorithms

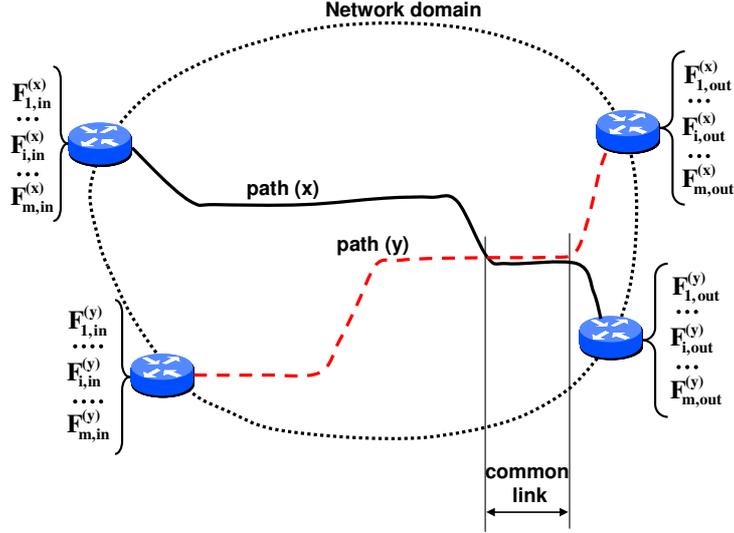


Figure 1.2. Network domain scenario

This novel model/approach, proposes a solution to obtain network level proportional differentiated service, based on the goodputs of the flows. The generic network model is presented in Fig.1.2. The micro-flows with the same ingress and egress points and the same QoS class are aggregated. For each ingress, a , and egress, b , there is a path, and over each path there are m flows, denoted $F_i^{(x)}$. For each flow, F , there is the offered load (input bandwidth, F_{in}) at the ingress and the achieved throughput (output bandwidth, F_{out}) at the egress. In Fig.1.2 there are two paths presented, (x) and (y). Although the two paths share a common link within the network, the class differentiation is applied “pathwise” – as seen in eq. (3), it defines the relations between flows from *different classes* but the *same path*.

Because in the proportional service model unique QDPs were proposed (QDPs of each class are path independent), the notation can be further simplified, introducing the following notation.

Definition 1.5 In the proposed network wide proportional service model the QDPs can be denoted by $\alpha_{i,j}$, as follows:

$$\frac{G_i^{(k)}}{G_j^{(k)}} = \frac{\frac{F_{i,out}^{(k)}}{F_{i,in}^{(k)}}}{\frac{F_{j,out}^{(k)}}{F_{j,in}^{(k)}}} = \frac{c_i}{c_j} = \alpha_{i,j}, \quad \alpha_{i,j} > 1 \text{ if } i > j, \quad i, j = 1, 2, \dots, m, \quad k = 1, 2, \dots, N, \quad (4)$$

where, c_i and c_j are network wide differentiation parameters.

The notation can be further simplified to $\alpha_i = \alpha_{m,i} \quad i = 1, 2, \dots, m$. Note that

$$\alpha_{i,j} = \frac{c_i}{c_j} = \frac{c_m}{c_j} = \frac{\frac{1}{\alpha_{m,i}}}{\frac{1}{\alpha_{m,j}}} = \frac{\alpha_{m,j}}{\alpha_{m,i}}, \quad \alpha_i > 1 \text{ for } i = 1, 2, \dots, m-1 \text{ and } \alpha_m = 1 \quad (5)$$

For the highest (m) and any other QoS class, this equation shows that the goodput of the best quality class (m) is α_i time bigger then the goodput of certain i class.

The novelty of this proposal is that, opposed to the PHBs, it assures this relative differentiation *over a domain, not over a single hop*. Therefore the service received by the user is clearly defined, it is not the result of the combination of per-router traffic manipulation experienced along the path. Note that calculating a priori the effects of independent per-hop processing is not a trivial task. In my proposal the QoS parameters are determined before the traffic enters the network. Additionally, the behavior of the flow can be better communicated to the user, since it can be described by one parameter.

Based on the choice of the network administrators they can select among two implementation alternatives. Although the main principles are the same, there are differences in the building blocks of these two alternatives. The routers at the edge of the network domain have a distinguished role. The ingress and egress functionalities are different, but in practice they might (and most probably will) be collocated. The core routers will have to maintain only information on the aggregate flows, therefore the proposal conforms to what has been called the *core-stateless architecture* [18].

The architecture of a *centralized* network-wide proportional service network is depicted in Fig.1.3. In this case a new networking element is needed, called Central Broker. In several DiffServ-related work the authors introduced a Bandwidth Broker which is responsible with network-wide traffic engineering tasks [19][20][21].

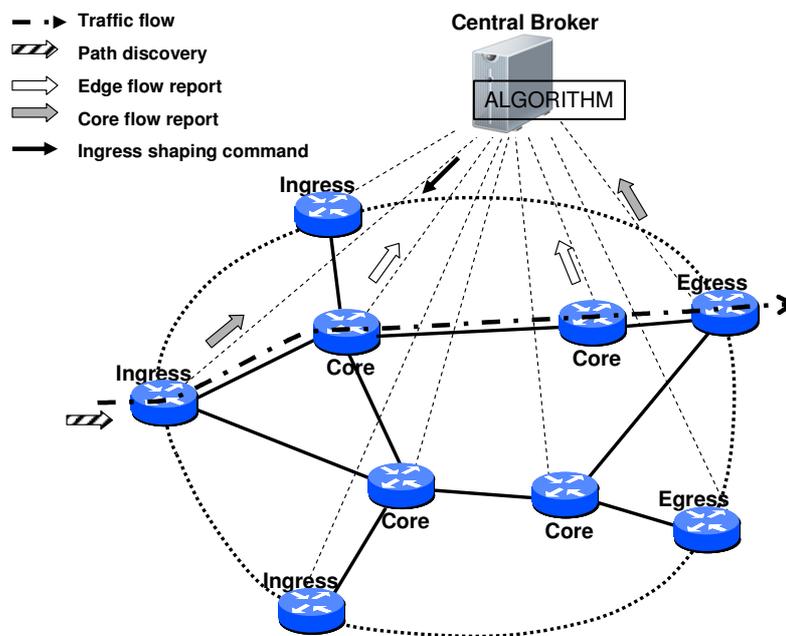


Figure 1.3. Architectural elements of the network-wide proportional service domain (centralized version)

The architecture of a *distributed* network-wide proportional service network is depicted in Fig.1.4. The distributed version of the network architecture is simpler, since there is no need for the Central Broker. On the other hand the functionalities of the CB are distributed among the routers in the network. The signaling data is transmitted along the path of a flow and there is no communication towards a centralized entity.

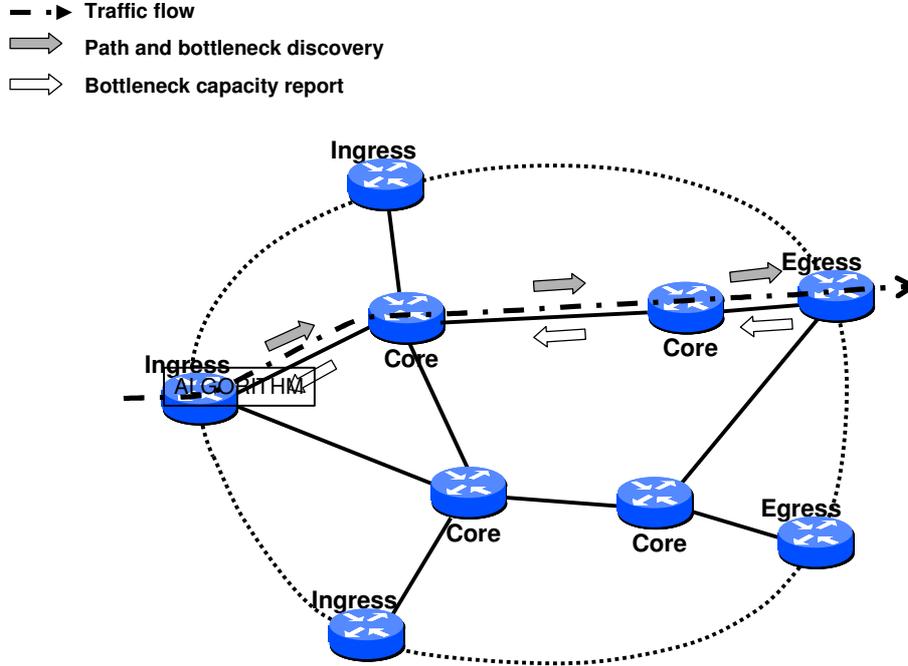


Figure 1.4. Architectural elements of the network-wide proportional service domain (distributed version)

In a classical IP network, when congested, the flows loose packets and suffer various delays at the intermediate (core) routers resulting in a reduced throughput at the egress. The achieved throughput is the result of a complex interaction among flows crossing the same links inside the networks, thus it is hard to predict. However, if this network deploys the network-wide proportional service model the bandwidth of a flow at the egress of the domain depends on the traffic matrix (the $F_{i,in}^{(x)}$ offered loads) and the QDPs, as conditioned by the eq. (3). The open question is whether the *throughput* (the capacity share of the flow in its bottleneck) can be analytically determined. If yes, it can be later used in algorithms to police the traffic at the ingress and reduce the load inside the network.

Thesis 1.2 (Throughput, delay and loss rate in the network-wide proportional service model) [C1][C2] – *I derived in the network-wide proportional service model the capacity share of a flow ($F_{i,out}^{(z)}$) that crosses a bottleneck within the network, where μ_{l_b} is the capacity of this bottleneck link l_b .*

$$\dots F_{i,out}^{(z)} = \frac{\mu_{l_b}}{\sum_{\forall(y):l_b \in (y)} \left(\frac{F_{i,in}^{(x)}}{F_{i,in}^{(z)}} \sum_{j=1}^m \alpha_{j,i} \frac{F_{j,in}^{(x)}}{F_{i,in}^{(x)}} \right)} \quad (6)$$

I showed that if the traffic of all classes has the same stochastic properties, in the proposed network architecture the loss rate in a class will be less than or equal to the loss rate in any worse (having lower QDP) class.

I showed that if the traffic of all classes has the same stochastic properties, in the proposed network architecture the delay experienced by packets of a class will be less than or equal to the delay of packets from any worse (having lower QDP) class.

In the followings I present the main steps taken to derive the closed form formula in eq. (6) of Thesis 1.2. Given a network that deploys a network-wide proportional service model, as depicted in Fig.1.2, it can be written the eq. (3) for all the paths that share the common bottleneck link l_b :

$$\left\{ F_{i,out}^{(z)} = \alpha_{i,j} F_{i,in}^{(z)} \frac{F_{j,out}^{(z)}}{F_{j,in}^{(z)}} \right\}, i,j = 1,2, \dots, m \text{ and } \forall(z) : l_b \in (z) \quad (7)$$

The problem is that this set of equations is not fully independent, for every path there are only $m-1$ independent equations. On the other hand, since the flows cross a bottleneck, it can be written the work conserving law for the bottleneck link:

$$\sum_{\forall(z): l_b \in (z)} \sum_{i=1}^m F_{i,out}^{(z)} = \mu_{l_b} \quad (8)$$

Now, if we note with R the total number of paths that cross the bottleneck link l_b

$R = |\forall(z) : l_b \in (z)|$, then there are $R(m-1) + 1$ equations, but Rm unknown variables, thus the resulting system of equations still can not be solved.

Eq. (3) defines the relation among different priority classes *within the same flow path*. It says nothing about what happens when two or more flows from the same priority class but from different paths share the same link(s).

It is a natural requirement that flows from the same class should share the common resources equally (in other words the same class flows are not differentiated). This desirable property can be considered a *fairness criterion* which specifies the relation among *concurrent flows*.

Definition 1.6 (Fairness criterion) Let flows from the same class i but different paths (z) and (y) sharing the same bottleneck link have the same goodput

$$\frac{F_{i,out}^{(z)}}{F_{i,in}^{(z)}} = \frac{F_{i,out}^{(y)}}{F_{i,in}^{(y)}}, \forall(z) : l_b \in (z), \forall(y) : l_b \in (y), i = 1, \dots, m \quad (9)$$

(Note that this criterion is reasonable when the whole network domain is considered and the same c_i QDPs are used over the whole domain.)

For convenience let us order the R paths crossing the bottleneck link l_b . Now there are $(1), (2), \dots, (R)$ paths. Furthermore, without losing the generality of the notation, I consider the $(z-1)$ path for $z=1$ to be the same as (R) . Following this notation and using eq. (9), it can be shown that over a bottleneck link l_b

$$F_{i,out}^{(z)} = F_{i,out}^{(z-1)} \frac{F_{i,in}^{(z)}}{F_{i,in}^{(z-1)}}, \forall(z) : l_b \in (z) \quad (10)$$

if, for convenience, for $z = I$ we consider $z-I = R$. Note that, similar to the logic followed at eq. (7), this equation yields only $R-I$ independent equations.

Based on eq. (10) and selecting an arbitrary QoS class i , $R-I$ new equations can be added to the system of equations. Now there are $R(m-I) + I + R-I = Rm$ independent equations and Rm unknowns $\{ F_{i,out}^{(z)} \}_{i=1,\dots,m}, \forall(z) : l_b \in (z)$ and it can be derived the eq. (6).

In the case of the differentiation of classed based on *loss rates* (loss differentiation) or *delay* (delay differentiation) the QoS service parameters are the inverse of the loss rates or ingress-to-egress delays. The reason behind this is that the better the QoS of a class, the lower the loss rate (or delay) should be, whereas for the goodput the better service means higher goodput. In order to assure a unique framework I keep the definition of better classes from Def. 1.1., I use the following notations.

Definition 1.7 The QoS parameter for a loss-differentiation is $q_i = 1/l_i$, where l_i is the loss-rate of flow i experienced from ingress-to-egress. Similarly $q_i = 1/d_i$, where d_i is the ingress-to-egress delay of flow i .

Similarly to Def.1.5 I can simplify the notation for the *loss-differentiation* if I introduce the differentiation parameter, as follows.

$$\beta_i = \frac{q_m}{q_i} = \frac{l_i}{l_1} \quad (11)$$

Starting from eq. (3) and knowing that the loss rate over a path (x) for class i is

$$l_i^{(x)} = \frac{F_{i,in}^{(x)} - F_{i,out}^{(x)}}{F_{i,in}^{(x)}} \quad (12)$$

I showed that the $\beta_i^{(x)}$ differentiation parameter for the loss rates for a given class i over path (x) is:

$$\beta_i^{(z)} = \frac{1 - G_m^{(x)}}{1 - G_m^{(x)}} \alpha_i \quad (13)$$

Remember from Def.1-3 that $G_m^{(x)}$ is the goodput of the best QoS class flow over path (x).

By definition I have $G_m^{(x)} < I$, $\alpha_i > I$, thus $\frac{G_m^{(x)}}{\alpha_i} < G_m^{(x)}$, and: $\beta_i^{(x)} = \frac{1 - G_m^{(x)}}{1 - G_m^{(x)}} \alpha_i > 1$

This also means that if $\alpha_i > \alpha_j$, then $\frac{G_m^{(x)}}{\alpha_i} < \frac{G_m^{(x)}}{\alpha_j}$, and thus $\beta_i^{(x)} > \beta_j^{(x)} \quad \forall(x) \in P$.

This means that if I order the classes based on the goodput based QDPs, then *the loss rate in a class will be less than the loss rate in any worse class (i.e., having lower QDP)*.

In what follows I summarize the argumentation behind the statement from Thesis 1.2 on the *delay* differentiation.

I showed that if the distribution of the arrival rates of the flows in different classes are the same, then the delays experienced by the packets of the flows from a higher class are no worse (lower) than those experienced by the packets from a lower one. Opposite to the

goodput (capacity) and loss differentiation, the delay differentiation also depends on the load of the queues at the ingress.

I based my proof on the fact that the buffer sizes in the core routers are small and therefore they are only used to buffer out the effect of short term bursts. Therefore the delays experienced within the network (the queuing and processing delays at the core routers and the propagation delays along the links of the core) are the same.

$$d_i^{(x)} = d_{i,ingress}^{(x)} + d_{i,core}^{(x)}$$

$$d_{i,core}^{(x)} = \sum_{l_i \in (x)} (d_{i,link}^{(x)} + d_{i,queue}^{(x)} + d_{i,processing}^{(x)}) \leq d_{i,ingress}^{(x)} \quad (14)$$

The difference in delay is due to the queuing delays experienced at the ingress. On the other hand in a class based queuing (CBQ) discipline, the load of the queue of a better class is lower than in queue of a lower class.

Therefore for the delay based QoS service parameters I have

$$q_m \geq \dots \geq q_2 \geq q_1 \quad (15)$$

where the equation holds for the case where ingress queues are empty.

Thesis 1.3 (Distributed algorithm) [C1] – *I proposed a distributed algorithm that shapes the traffic at the ingress in such a way that the flows of the network exhibit the properties of a network-wide Proportional Services model. The flowchart diagram of the algorithm is depicted in Fig.1.5. The algorithm for each ingress router keeps track of the bottlenecks over the paths originating from that ingress. Then it computes the shares of each flow crossing the bottleneck, using eq. (6). I showed with extensive simulations that this algorithm deployed in a distributed manner (fitting the distributed network architecture proposed in Thesis 1.1) enforces the flows to fulfill eq. (3).*

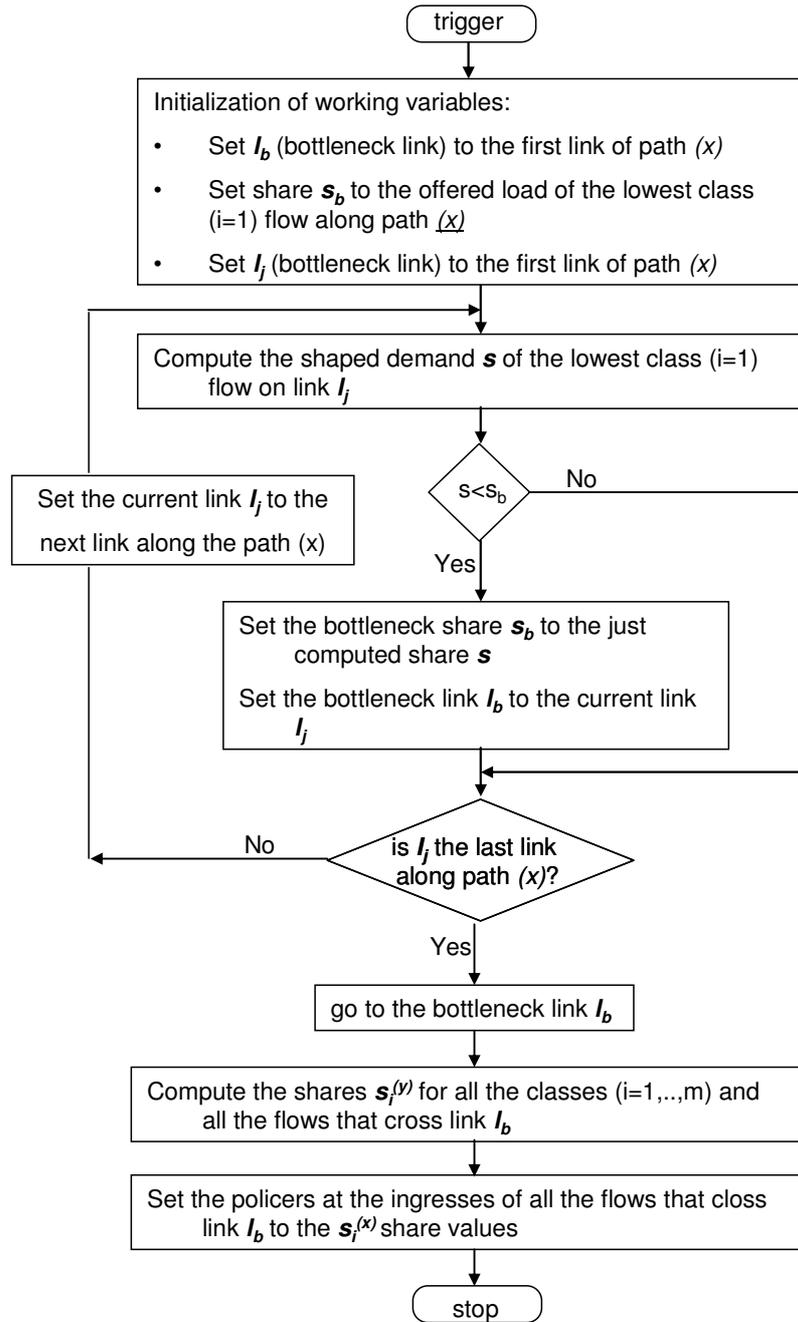


Figure 1.5. Flowchart diagram of the distributed algorithm

The trigger that starts the algorithm is a change in offered load (interrupt generated by measurement unit at ingress) and the bottleneck capacity is set to the offered load. Then, the algorithm computes the flow's share on the outgoing link. The share is determined using the analytical result from eq. (6) of Thesis 1.2. If the share on the link is less than the offered load, this share will be the flow's new bottleneck capacity. Then the algorithm moves to the next hop in the route, and computes the share again on the outgoing link. If the share is less than the previous one, the algorithm changes the bottleneck capacity to the new value, and steps to the following node.

This procedure is repeated until the egress node is reached. When arrived to the egress node I have the bottleneck capacity of the whole path. Now from the bottleneck node (the one the bottleneck link is originating from) can compute the shares of all the flows crossing the bottleneck link and notify the ingress node to set its policers to enforce these computed throughputs. The intermediate nodes update the flow bottleneck information as well, reducing the share at bottlenecks accordingly.

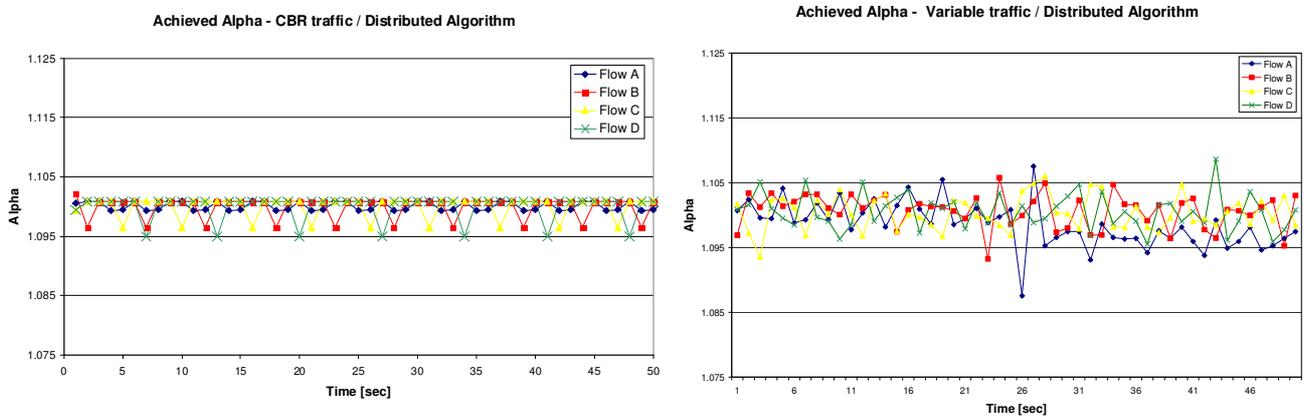


Figure 1.6. Distributed algorithm for constant bit rate (on the left) and varying UDP flows (on the right)

I verified the distributed algorithm with simulations. In order to understand and evaluate easier the results, I did the simulation experiments with $m = 2$. The differentiation among the two classes is defined by α_{21} , the only QDP in the system, which I refer to as *alpha* in the rest of this document. All the simulations presented in this sections were run with the target value of $\alpha = 1.1$. This alpha value is considered a worst case scenario: if my algorithm works fine for such a fine grained differentiation, surely will work for higher alpha values. I prepared three network topologies, as detailed in the Dissertation. To illustrate the accuracy of the algorithm, here I present the results with 4 flows (paths) in the network. On the left side of Fig.1.6. I present the achieved alpha ratios for static CBR (constant bit rate) flows, while on the right side of Fig.1.6. I present the achieved alpha ratios for varying non-responsive (i.e., UDP in the simulations) flows. The results show that the required differentiation is achieved indeed, meaning that the analytically computed results are successfully enforced by the proposed algorithm.

Thesis 1.4 (Centralized algorithm) [J1][C3] – I proposed a centralized iterative algorithm that computes the bandwidth of every flow at the ingresses according to eq. (6). I gave the flowchart diagram of the proposed iterative algorithm in Fig.1.7. This algorithm fits in the centralized network architecture proposed in Thesis 1.1. and enforces the network-wide Proportional Services model. I also showed that the algorithm can be used combined with a traffic prediction mechanisms, using a drop-tail queuing discipline for UDP flows.

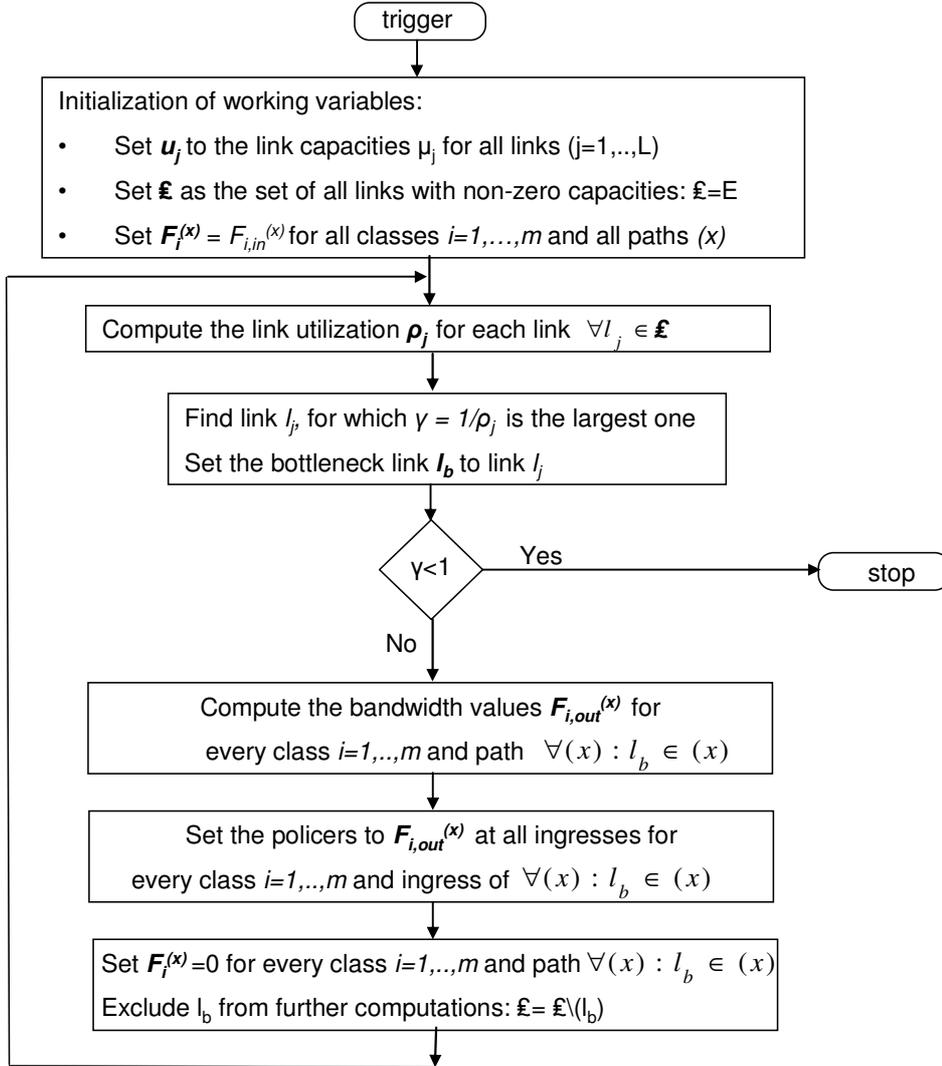


Figure 1.7. Flowchart diagram of the central algorithm

The major difference of this iterative algorithm, compared to Thesis 1.3 is that it eliminates *all* extra traffic (i.e., at network level) at the ingress, while the one presented in Thesis 1.3 eliminates only the extra traffic over the *path*. It achieves a global optimum, whereas the solution presented in Thesis 1.3 may yield local optimum in transient periods (that is, until all the paths are updated).

The algorithm is an iterative one and is based on the enforcement of both eq. (6) and the proportional differentiation eq. (3) over the flows sharing the bottleneck link. If the traffic is shaped at the ingresses of a network according to this algorithm then there will no losses *within* the network and the output bandwidth will be as given by eq. (6).

The algorithm always searches for the most congested bottleneck in the network, and it does so based on an overload factor.

Definition 1.8 The *overload factor* γ^l of a link l is the fraction of the flow that should cross the bottleneck, if the condition from eq. (3) holds for all these flows.

I showed analytically that γ^l can be expressed as a function of the offered load of the flows and the differentiation parameters

$$\gamma^l = \frac{\mu_l}{\sum_{(x)_{cross_l}} \sum_i \frac{F_{i,in}^{(x)}}{\alpha_i}} \quad (16)$$

The most congested bottleneck is the link where the overload factor is the smallest one.

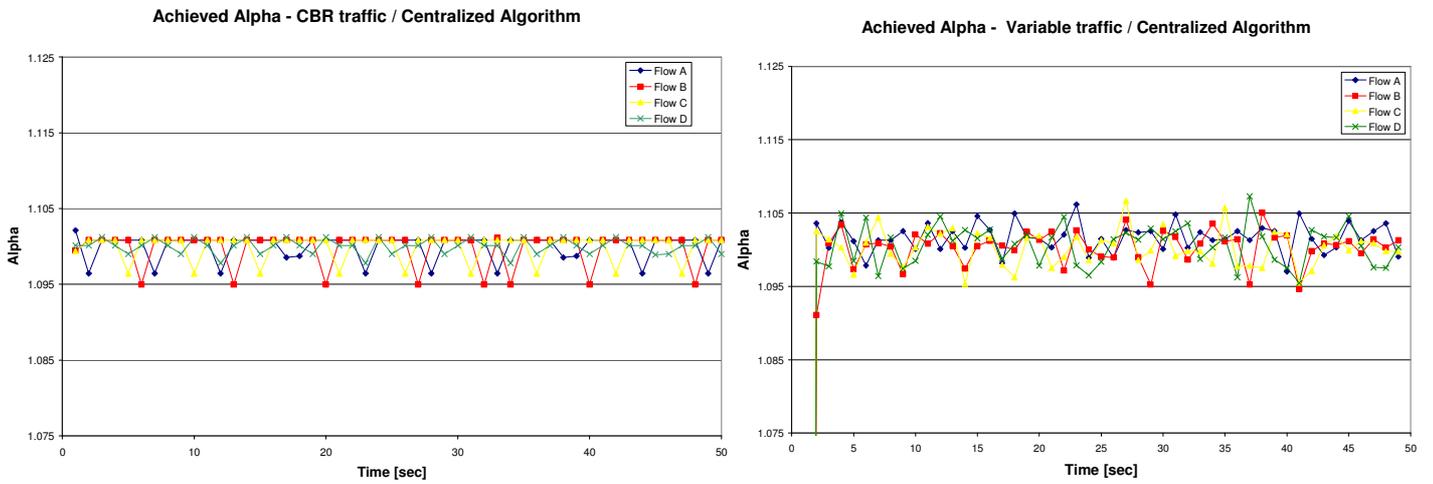


Figure 1.8. Centralized algorithm for constant bit rate (on the left) and varying UDP flows (on the right).

The proposal uses an adaptive traffic estimation method to predict the traffic at the ingress. I showed through simulations that in case of UDP traffic the algorithm successfully results in flow parameters that conform to the proposed model. The left hand side of Fig.1.8. shows the achieved alpha ratios for CBR flows, while the right hand side of Fig.1.8. shows the achieved alpha ratios for varying UDP flows. It can be seen that the algorithm can enforce the desired alpha rates. The Dissertation contains a detailed analysis of the simulation results.

Thesis 1.5 (Network-wide proportional service model for responsive flows) [J2] – *I showed that the network-wide Proportional Services Model can be applied to responsive (e.g, TCP) flows. I gave an interpretation of the offered load for responsive flows and I proposed a method with an active queue management mechanism that shapes the traffic according to the model. I validated this proposal by means of simulations.*

In order to be able to apply the eq. (3) to TCP flows, I adapted the definition of the goodput to TCP flows. Due to the congestion avoidance mechanisms of TCP, the source does not have an “offered load”; the load has elastic adaptation to the state of the network. My proposal is to make this unpredictable offered load proportional with the number of micro-flows within every class. In this approach the entering load of a certain class will be as follows:

Definition 1.9 The offered load at the ingress for TCP flows is as follows

$$F_{i,in}^{(x)} = n_i^{(x)} D \quad i = 1, 2, \dots, m \quad \forall (x) \in P \quad (16)$$

where $n_i^{(x)}$ is the number of micro-flows in a certain flow-class over path (x) ,

D is a network wide constant,

$F_{i,in}^{(x)}$ is the offered load for a certain, i , class over path (x) .

I proposed to use congestion-avoidance active queuing management (AQM) mechanisms to shape the TCP traffic. “Classical” drop tail queues are not suitable to shape TCP traffic, because a packet drop will lead to rate-limitation at the sender. Practically the TCP rate control mechanism “overreacts” the drop events: therefore I had to apply a much finer grained shaping mechanism. I selected an active queue management algorithm which uses packet loss and link utilization history to achieve the same goals, BLUE [22][23]. The FIFO queuing discipline does not assure fairness among micro flows within the same class, some of the micro flows being starved, while others take advantage. I compared by means of simulation the behavior of FIFO and BLUE and the latter outperformed the FIFO, avoiding starvation at micro-flow level. I also examined the shaping qualities of the BLUE AQM mechanism, by evaluating the losses (drops) at the ingress routers and concluded that drops occur only when significant changes occur (e.g., more microflows are in the slowstart phase), compared to the well-known RED, where packet drops are normal part of the policing mechanism.

Based on the above consideration BLUE is a proper mechanism to assure fairness within the flow-classes. There are no starving flows; the micro-flows share equally the available bandwidth. Additionally, the consequence of using BLUE is that congestion control can be performed with a minimal amount of buffer space, reducing the end-to-end delay over the network.

I used the BLUE active queue management implementation in ns2 [24] in the ingress nodes to avoid network congestion, and to assure the calculated bandwidths (throughputs) for the different micro-flows. I showed by simulation experiments that in case of TCP traffic the algorithm successfully results in flow parameters that conform to the proposed model. The left hand side of Fig.1.9. shows the achieved differentiation among flows if the number of TCP micro flows remains constant. Note that this corresponds to the CBR scenarios investigated in Thesis 1.3 and 1.4. The right hand side of Fig.1.9. shows the achieved differentiation among flows in the case when the number of micro flows has been changed at every 1 sec according to a similar pattern that was used in the varying flow scenarios investigated in Thesis 1.3 and 1.4.

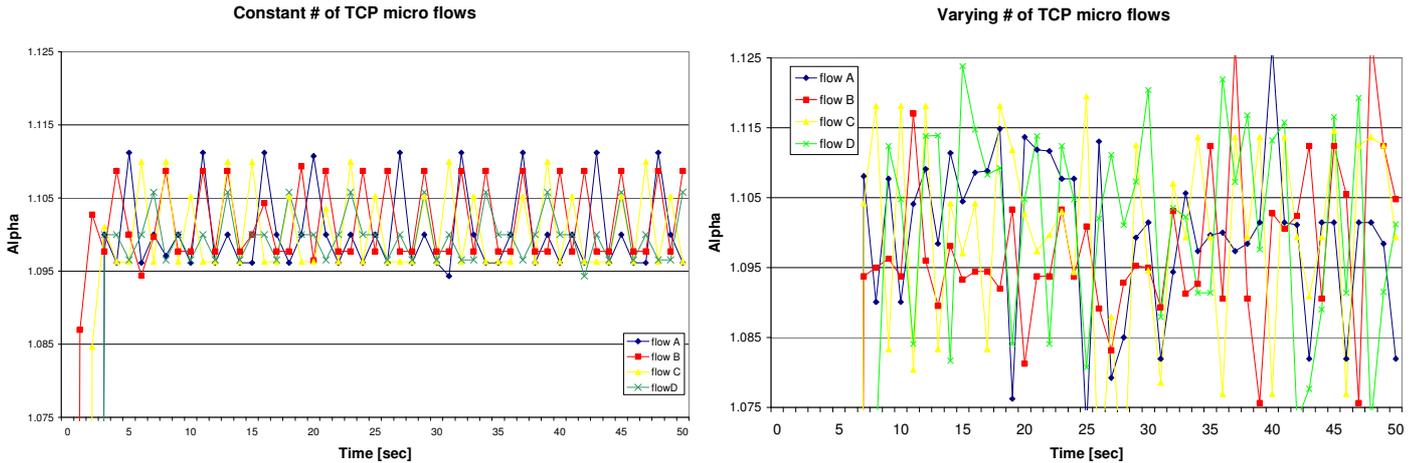


Figure 1.9. Constant (on the left) and varying (on the right) number of TCP micro flows, policed with BLUE at the ingress

I also showed that both TCP and UDP differentiation can be deployed in the same ingress router. This means that in practice the TCP flows should be shaped as introduced in this Thesis and the BLUE-based policers will enforce that value, while the UDP flows will be policed according to one of the algorithms proposed in Thesis 1.3 or 1.4.

4.2 Distributed Management Framework for Self-Organizing Networks

Today, the Internet has become essential for enabling data information flow exchanges all over the world enabling in turn a wide range of applications and services. As the current Internet, designed in the 1970s, grows beyond its original expectations (a result of increasing demand for performance, availability, security, and reliability) and beyond its original design objectives, it progressively reaches a set of fundamental technological limits and is impacted by operational limitations imposed by its architecture.

The works on future networking scenarios emphasizes two key issues [25]. Firstly, networks will become more technologically heterogeneous, accommodating old and new access systems as well as applications and services – indeed, migration and feature rollout will not be a ‘one-off’ but a constant activity. Secondly, networks will become organizationally heterogeneous: today’s cellular systems will be complemented by a diverse mixture of other network types – personal, vehicular, sensor, hot-spot and more.

Some of these networks may even be useful in isolation, but the true potential of this infrastructure is only obtained when they become interconnected in a way which allows their resources to be shared, and new communications patterns to be established between their users and services. This translates into the need that such network of networks will have to form and re-form dynamically in response to changing conditions. At the same time these interactions must be achieved automatically because the dynamic nature of the set of network-network relationships rules out time-consuming or complex manual configuration. What was the vision of “All-IP” networks developed at the turn of the millennium during the last years is clearly heading towards this direction.

The new concepts towards Future Networks burden existing networking infrastructure with more and more tasks that sooner or later will not be manageable by human

intervention alone. In the context of the future networking environments the network management and control should become more dynamic and flexible. The concept of self-managing networks is widely accepted as a solution that overcomes the rapidly growing complexity of the Internet and/or other networks and enables their further growth [26][11]. Operators, as key responsible actors for shaping the future networking infrastructure, are willing to have new functionalities and mechanisms that allow the network to self-manage with as little of human intervention as possible, providing higher availability of services, lower the provisioning time of new customers and reduce the time to market of new services [26][27].

Throughout this section I suppose that the networks hold self-organizing capabilities. This means that the network elements have enough computational capacity to run the self-organizing logic and have proper interfaces to manipulate the network elements.

Definition 2.1 *Composition* is the process through which two management domains / authorities interact, negotiate and decide their cooperation intentions, combine their resources as decided during negotiation. Composition refers to domains, therefore this process is also referred to as *network composition*, where the network is the logical network controlled and managed within/by the management domain / authority. The composition process of the management domains is governed by some composition manager functions and is realized by the control plane functions of the interacting networks. Both the management plane and the control plane functions of the networks might be affected (changed, enhanced or limited) throughout the composition process [B1][J3][28][29].

When two separate networks compose, one crucial challenge is to combine their network management systems into a consistent network management system for the composed networks. Similarly, when a network splits into two (or more) networks, the network management system has to dissolve into corresponding pieces in a consistent and predictable way. I proposed a network management framework and network management processes that alleviates this problem.

Thesis 2. [B1][J3-J7][C4-C9] *I proposed a Distributed Management Framework (DMF) for self-organizing networks (SON) to determine how two or more network management domains (authorities) should interact with each other to establish cooperation. I described absorption and gatewaying as two types of network cooperation. I defined a hierarchical peer-to-peer (p2p) overlay network structure that enables the realization of the management plane functions. I mapped the p2p overlays to the structure of the distributed network management framework. I modeled the DMF structure with a SON graph together with the basic graph operations to describe absorption and gatewaying. I proposed and described an algorithm to identify which abstraction level networks should cooperate.*

The proposed Distributed Management Framework is applicable to any networks that have layer 3 addressing and connections and computationally can support the self-organizing functionalities. Any packet-based network that has layer 3 addressing and is a Next Generation Network (NGN) according to the ITU-T definition [30] is suitable to implement the DMF for SONs. Naturally, the protocols that implement the DMF and self-organizing functionalities should be deployed on the nodes of a generic NGN. A typical example for such networks are the Ambient Networks [B1][31].

Thesis 2.1 (Distributed Management Framework for Self-Organizing Networks) [J3][J7] – *I proposed a Distributed Management Framework (DMF) for self-organizing networks (SON) to determine how two or more network management domains (authorities) should*

interact with each other to establish cooperation based on policies. In my DMF I defined a self-contained, iterative, distributed and scoped process to locate and disseminate the necessary information to establish cooperation at the management plane. The architecture of the DMF for SON is shown in Fig. 2.1.

Remark: The DMF controls the behavior of the control plane functionalities of the management domains.

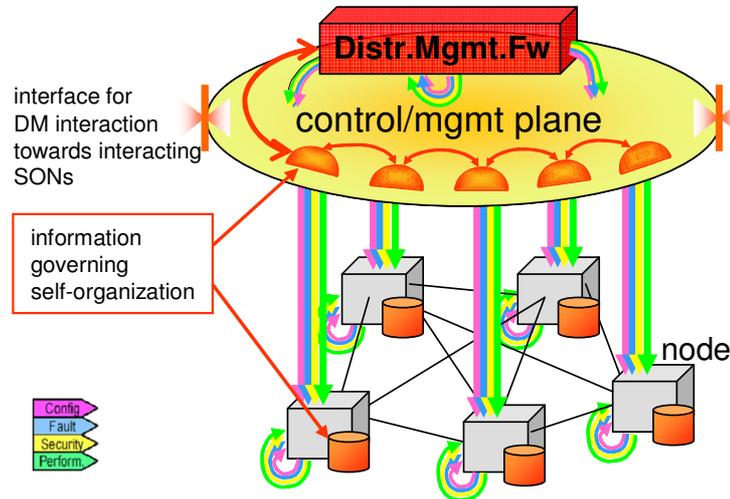


Figure 2.1. Distributed Management Framework - overview

Example: In Ambient Networks [B1][7][31], the Ambient Network Management controls the Functional Entities (FE) of the Ambient Control Space (ACS).

Thesis 2.2 (Network Composition) [J4][J6][C6][C7] – I defined and described absorption and gatewaying as two types of network compositions, which together with bootstrapping (initialization) and de-composition (termination) can fully describe a network composition model and is conform to the framework defined in Thesis 2.1.

Absorption defines the unconstrained information sharing of the composing network management domains, resulting in one single management plane as the union of the former ones (both resource and service wise). If absorption is not possible then gatewaying defines how to create a new management plane for the composed network and how to disseminate information from the embedded networks. Throughout gatewaying full control of resource and service sharing is possible.

Bootstrapping defines the basic self-configuration process that initiates a stand-alone management plane¹. Decomposition defines the process of division of a management domain into two or more standalone management domains, which implies the identification of the self organizing properties and the proper separation of them during this process.

Note that I assumed throughout the rest of my results, that during the network composition *control* over servicers and resources is restrictive, i.e., for each gatewaying composition the resulted network bears with less and less control over its embedded resources.

In order to realize the DMF architecture it must be mapped to a distributed and hierarchical structure:

¹ E.g., powering up a stand-alone networking element.

Thesis 2.3 (Peer-to-peer Management Overlay) [J5][J6][C4][C5] – I mapped the Distributed Management Framework’s abstract model to a hierarchical peer-to-peer network. The peer-to-peer network at its lowest layer consists of the physical network nodes. For each of the network management domains in the DMF an overlay is created, whose members are the elected representatives (e.g. superpeers) of the embedded p2p overlays corresponding to their network management domains.

I further created a SON graph model together with six operations to model my DMF. This SON graph gives a direct mapping between the distributed management framework and the peer-to-peer overlays as follows: each vertex in the SON graph corresponds to one management domain and one overlay network (see Fig. 2.2). This overlay network is run by the representatives of the overlays corresponding to the children vertex.

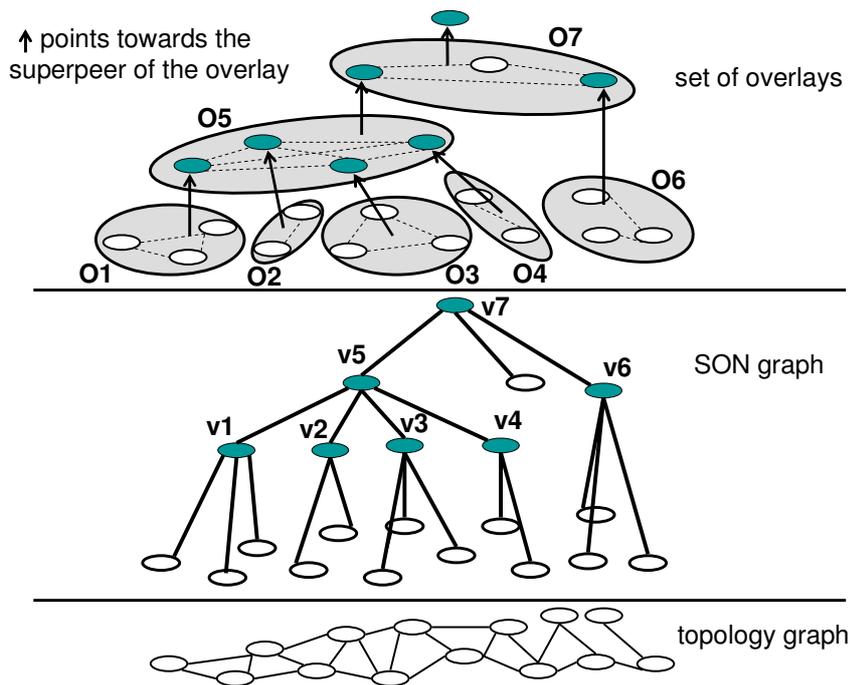


Figure 2.2. SON graph and the p2p overlays

I mapped the constructors for the network compositions as defined in Thesis 2.2 into the SON graph model using the six operations: joining, leaving, merging, splitting, expanding and collapsing.

Join: The root node of the joining SON graph is attached to an existing node of the other SON graph (see Fig. 2.3.i).

Leave: A complete subtree is detached from the original rooted tree. The result is two disjoint SON graphs over the same topology (see Fig. 2.3.ii).

Expand: If the node had a parent before, then a new node between the parent and the child node is created. Otherwise the node will create a new parent of its own (see Fig. 2.3.iii).

Collapse: A node that has a parent and has exactly one child is deleted and a direct edge is created between its former child and its former parent (see Fig. 2.3.iv).

Merge: All the vertices belonging to the parents will be placed under a single parent, the other parent being deleted (note that only the leafs represent physical nodes, the rest of

the nodes are virtual ones). There exists a single node in the SON graph into which all child nodes of the root node of the other SON graph will be attached as children nodes (see Fig. 2.3.v).

Split: Some children detach from a parent node and create a new parent node to group them all as children nodes (see Fig. 2.3.vi).

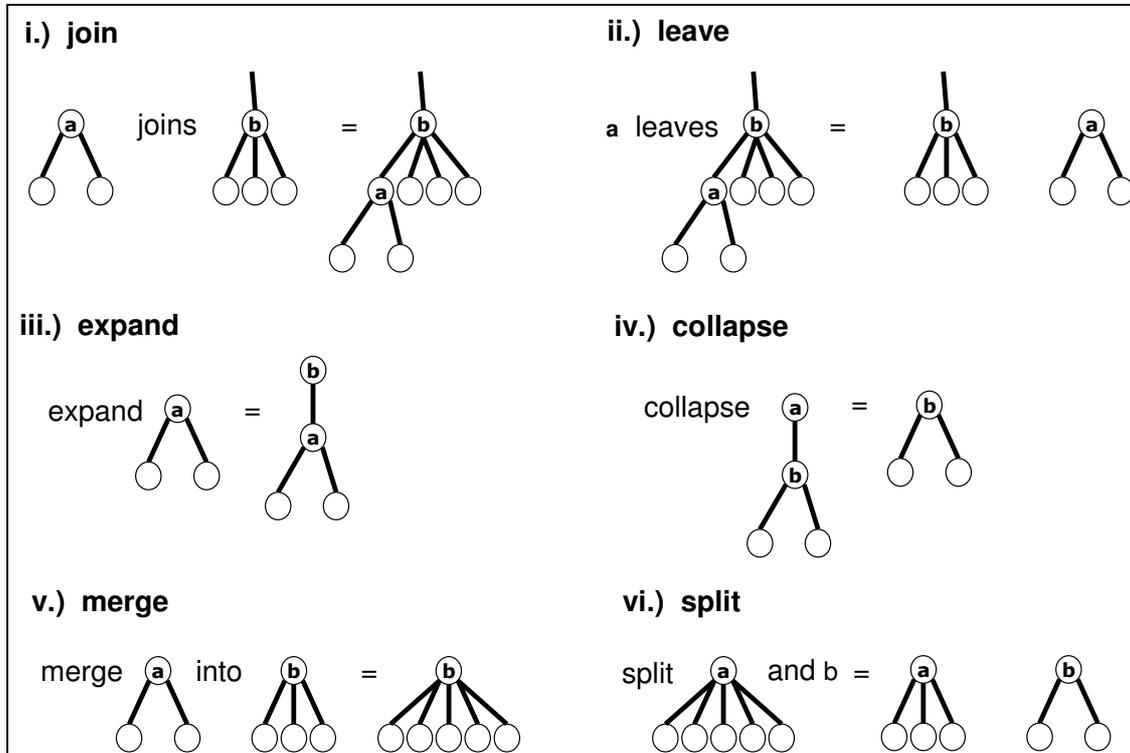


Figure 2.3. The six graph operations

Note that applying these operations to the SON graph it keeps its properties, and it remains a rooted tree. The SON graph will split in two parts if the underlying topology is partitioned.

In order to determine how a network of networks shall be connected to another network of networks (composition) I defined an algorithm as follows.

Thesis 2.4 (Bottom-up composition algorithm) [C8][C9] – *I proposed a bottom-up composition algorithm, which walks through the SON graph from the leaves (physical nodes) to the higher levels (wider and wider scale composed networks) and checks for possible absorptions. If no absorption can be realized then the point of gatewaying is determined (see Algorithm 2-1). I showed by numerical evaluations that the configuration time and signaling complexity of the bottom-up composition algorithm is $(O \log(N))$.*

Remark: The bottom-up composition algorithm can explicitly be realized by the super peers of the p2p management overlays.

Algorithm 2-1: Composition algorithm for physical nodes a and b : Without the loss of generality, let's assume that the SON hierarchy level of node a is lower than for node b . Let's denote with a the composing SON and b the target SON. Let's define a utility function (U), which counts the population of two SON nodes as follows:

$$U(a,b) := \#[A(a) \cap A(b)] \quad (17)$$

where $A(.)$ is the set of attributes at the overlay layer. Let $p(x)$ denote the parent node of x , if x is the root node than $p(x) = 0$. The description of the bottom-up algorithm in pseudo code is given in Fig.2.4.

```

a := root node of the source SON;
b := layer 2 adjacent leaf node of the target SON;
u :=  $U(a,b)$  //  $a$ =root;  $b$  = leaf;
while ( $U(a, p(b)) == u$ ) do //
    // check for matching attributes
    if ( $A(b) - A(a) == \text{empty}$ ) then  $ABSORPTION(a, b)$ ; exit;
     $b := p(b)$ ;
end while;
 $GATEWAYING(a, b)$ ;

```

Figure 2.4. The bottom-up composition algorithm

The absorption and gatewaying functions used in the algorithm are given in Fig.2.5.

```

 $ABSORPTION(a,b)$ 
    merge( $a,b$ );
end;

 $GATEWAYING(a,b)$ 
     $b := \text{expand}(b)$ ;
    join( $a,b$ );
end;

```

Figure 2.5. The absorption and gatewaying functions of the algorithm

The bottom-up composition principle states that the composition or de-composition is a succession of peer-to-peer interaction (represented by nodes of the SON graph) starting from the leaves (i.e., bottom-most node) of the interacting SON graphs. Every interaction ends up with a decision. The decisions are based on a utility function, which is a decreasing function over one SON from bottom to up. If an attribute match is found between the source and the target network level then absorption is executed. If no proper match is found then a new level (abstraction) is created to accommodate the best possible mix of the two networks.

I implemented a simulation model and with extensive simulations I showed that the configuration time of a SON that deploys the bottom-up algorithm is of complexity of $O \log(N)$, where N is the number of nodes in the network (see Fig. 2.6).

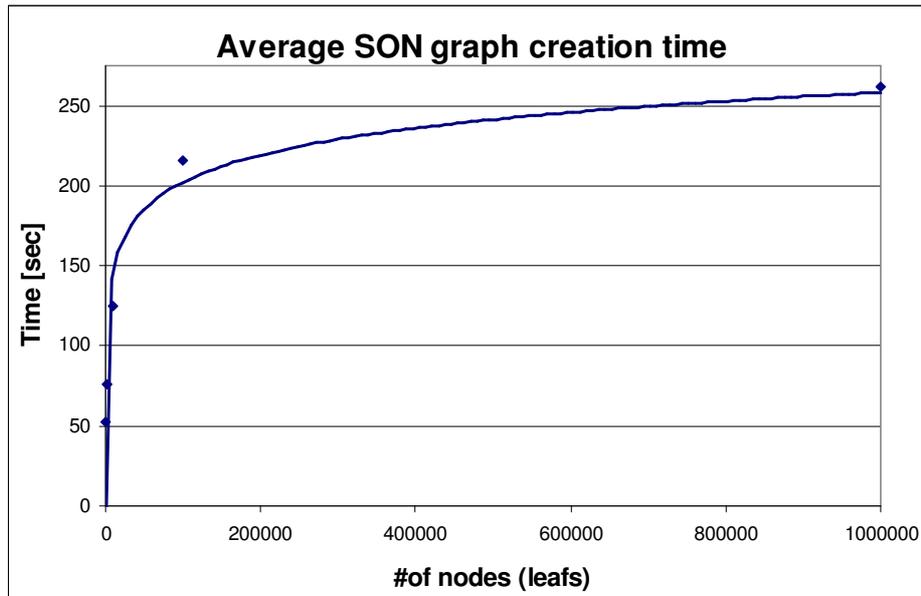


Figure 2.6. Scalability of the bottom-most composition algorithm

5. Applicability of the new results

Nowadays several network operators simply overprovision their capacities to cope the growing traffic demands. On the other hand, peer-to-peer file exchange and streaming media applications already generate the majority of internet traffic. The growing number of legal enterprises offering online media content will challenge these business models. Further on, the spread of the Internet of Things model will increase the number of networked end-hosts, but with a distinct QoS profile. Also, the high speed cellular access networks will generate significant amount of traffic from/to mobile users. All these factors lead to a diversified demand of services with a fast growing traffic volume. Therefore both vendors and operators are seeking for more cost-effective and QoS agnostic solutions to cope with these changes. My proposals enable operators to achieve these goals; the results can be deployed in their networks. Feasibility of the deployment depends on the business strategies of the operators, and is much likely to favor the application of my results in the case of green field investments or/and especially during the migration to future networks (e.g., 3GPP cores, IPv6 islands).

The advantages of the proposed mechanisms would allow the operators to deploy cheaper devices, save operational (management) costs and marketing costs (promises on QoS levels are more easily communicated). The implementation of the proposals does not require development of new mechanisms in routers. The network-wide proportional services can be implemented by defining a new signaling and control logic over existing router mechanisms, as detailed in the dissertation. I demonstrated the feasibility of the Distributed Management Framework through a prototype implementation, which can serve as the basis of its implementation in a vendor specific enterprise system. This would not imply the development of new mechanisms but it should be adapted to the particular data management implementations.

The results in Thesis 1 enable the network operator to deploy QoS traffic control only at the ingress, which confers scalability to their network. This is a challenging issue in new global networks. Additionally, using the proposals, the operators achieve network wide

QoS, not only per hop. Further advantage of the proposal is that the queues within the network domains, at the core routers are much less occupied. This can be used either to deploy cheaper routers (with lower memory need) or to gain competitive edge on the service market (offering better services). The new “All IP” core networks are a typical case of operator controlled domain with ingress filtering, and the proposal fits naturally in this architecture, as the control is deployed at the ingress, only.

The result of Thesis 2 served as a basis of an EU funded research project, where our partners were major vendors and operators of the global telecom market. As such the output of this work was aligned to the standardization work on the next public wireless networks. Particularly, the proposed DMF is the management plane extension of current control-plane 3GPP composition. In Future Internet the proposed network self-organizing methods can serve as the basis for the dynamic attachment of the smaller networks (e.g., personal area networks – PAN) or newly installed islands in the migration process from IPv4 to IPv6 Internet.

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