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PERFORMANCE EVALUATION OF QOS ARCHITECTURES FOR PACKET-SWITCHED NETWORKS

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Ph.D Dissertation

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Table of Contents

List of Tables

List of Figures

Abstract

Novel, value-added applications may become widespread in the evolving, multi-service packet-switched networks only if reliable QoS assurances are ensured. As traditional, best-effort IP networks does not contain built-in mechanisms capable of providing transmission quality, additional functions and services have to implemented.

The core element of an efficient QoS provisioning mechanism is an effective resource (i.e. capacity and buffering space) requirement assessment method that is capable of determining resource needs of diverse traffic types reliably and with an adequate accuracy.

In the dissertation capacity and buffer requirement assessment techniques rooted in the Theory of Large Deviations (LDT) are presented. The LDT has proven to be a mathematical apparatus extremely capable of describing the probabilities of rare events, such as a buffer overflow event of a router. These new methods can approximate the bandwidth or buffer size requirement of a certain aggregated traffic flow in case the maximum allowable Workload Loss Ratio is prescribed. Our formulae have lower computational demand than other formulae capable of computing the same measures.

Also a method for coupling the novel resource assessment methods with other call-level techniques to form a combined dimensioning algorithm that is capable of providing multi-level QoS, i.e. packet-level loss and call-level availability, assurances simultaneously is proposed. Furthermore a QoS architecture that can be applied primarily in access aggregation networks to ensure desirable transmission quality for value-added applications in a transparent manner is introduced.

Chapter 1

Introduction

We live the convergence of purpose-specific communication networks. The common principle where these various networks seem to meet is the packet-switched transmission, which - besides its many advantages - exposes several problems that should be dealt with. One of the main challenges in connection with packet-switched network is the provisioning of QoS (Quality of Service) assurances; without proper transmission quality ensuring mechanisms the penetration and marketability of novel, value-added services like the Voice over IP (VoIP) or Video on Demand (VoD) is out of the question. This is an important area, that should receive the distinguished attention of not only academic researchers but also telecommunication network operators and content providers.

Providing satisfactory service quality for certain applications is one of the greatest challenges in the world of telecommunications. This problem has been always actual, but it appeared in different forms and required various modelling techniques in accordance with the evolution of communications networks. As an example let us just think of the plain old telephony system (POTS) where service availability (i.e. the blocking probability) was more important than transmission quality, while in the world of packet-switched network the transmission quality receives the greater emphasis.

There are many who tend to disregard the importance of QoS provisioning, especially those who agree with the "philosophy of large bandwidth". They vote for the over-dimensioning of packet-based networks stating that it implicitly assures that the quality requirements will be met. In many cases they seem to have been right by now, as the evolution of transmission technologies and the increase in data rates have outpaced the demand for bandwidth-consuming applications. This situation, however, may change in the future with the growing dominance of wireless and mobile telecommunication solutions, where capacity over-provisioning can either be impossible (due to the lack of free frequency bands) or too costly to be a competitive alternative. Also, in many cases, the capacity overprovisioning is not a viable option in wired access networks either. Naturally, this field has also seen a rapid development regarding the data rates on the air and copper wire interfaces, nevertheless it is not clear whether available capacities will excess the ever-growing demand for bandwidth on the long run.

Considering the above-mentioned drawbacks of unreasonable overdimensioning, a well-tuned QoS provisioning mechanism, e.g. an admission control policy coupled with a fairly accurate but at the same time simple and reliable resource requirement assessment algorithm, can be less costly than simply expanding the resources in the network. The adequately precise knowledge of the resource requirements of the already admitted traffic flows may also be very useful as it enables the proper handling of critical or unexpected situations, e.g. when huge capacities become suddenly unavailable due to link failures or when there is an abrupt and unexpected increase in service demand.

Despite more than 40 years has passed since the layout of the basics of the Large Deviation Theory (LDT) [1], it has proven to be one of the most efficient mathematical tool for modelling and dimensioning real packet-switching systems of present days. Basically the Theory of Large Deviation is concerned with the exponential decay rate of the probability measures of certain kinds of extreme or tail events, as the number of observations grows arbitrarily large. This implies that the application of the LDT to estimate rare events - e.g. the buffer overflow probability of a router - comes very naturally. Based on this theory it is possible to approximate the amount of lost information and determine the required amount of resources that a certain traffic mix needs to fulfil a predefined quality goal.

In this dissertation efficient resource requirement assessment techniques, that are rooted in the Theory of Large Deviations, are proposed. These new methods are capable of approximating the bandwidth or buffer size need of a certain aggregated traffic flow in case the maximum allowable information loss is prescribed. Also combined dimensioning methods that enable the provision of multi-level QoS assurances, i.e. packet-level loss and call-level availability, simultaneously will be presented. Furthermore a QoS architecture that can be applied primarily in the aggregation segment of access networks to ensure desirable transmission quality for value-added applications in a transparent manner is going to be introduced.

The dissertation is organized as follows. In the next section of this chapter a short summary of the main models and techniques of QoS provisioning in telecommunication networks is provided. After that - in Section 1.2 - the scope of our research activity and our objectives will be composed. In Chapter 2 an overview of the contributions found in the literature that led to our new results and are vital for the understanding of the subsequent parts of this dissertation can be read. In Chapter 3 and Chapter 4 novel LDT-based resource requirement estimators will be presented, that are capable of approximating the minimum transmission capacity (Chapter 3) or buffer space (Chapter 4) of a packet server (e.g. a router) that is required to fulfil the prescribed QoS goal composed in terms of maximum workload loss ratio (WLR). In Chapter 5 a dimensioning technique will be introduced that combines the LDT-based packet server model with call-level models and is capable of determining the resource needs of a certain traffic mix in case the QoS goal is composed in terms of maximum WLR and maximum blocking probability. In the second part of Chapter 5 a new QoS architecture will be introduced, that bases greatly upon the previously discussed dimensioning methods and is capable of providing multi-level QoS guarantees for value-added applications transported through broadband access aggregation networks.

1.1 QoS Guarantee Provisioning in Telecommunication Networks

Assuring the appropriate quality of a certain communication service was always a challenge, but the actual problem was changing all the time in line with the evolution of network technologies used. While in the golden ages of telephony networks ensuring reasonable service availability had the top priority, nowadays - in the era of packet-switched networks - providing appropriate transmission quality have become the main concern of communication service providers.

The quality of service (QoS) can be described with the aid of QoS parameters which can be put into two main groups according to the traffic granularity level they refer to. Availability measures, e.g. the time blocking probability (TBP) describing the fraction of time when the service is unavailable, are typical call- or session level measures. These parameters received considerable attention in the context of line-switched telephony network design, where every admitted connection (call) received a certain amount of dedicated resources and thus the transmission quality was assured. These systems are often modelled with the aid of Markov chains where the number of states refer to the number of resources (often the total number of lines). In most cases the main objective of the investigations of these systems is the calculation of the already mentioned time blocking probability, which is - in this case - the probability of the system being in the highest ranked state of the Markov chain. The most famous contributions on this topic are attached to the name of Erlang and Engset, whose famous formulae are still in use for the design of circuit-switched networks [9][10]. A summary of high standard of the results on this topic can be found in the celebrated book of Kleinrock [11].

The other group of QoS parameters are the so-called packet-level measures. Basically they describe the transmission quality perceived by a certain session or application. They started to receive increasing attention in parallel with the growing domination of packet-switched networks, where network resources are typically shared by all admitted sessions and thus the actual transmission quality of a certain application cannot be predicted in a straightforward manner. The classic packet-switched data networks contain no admission control mechanisms, which means that the blocking probability will be zero (unless some kind of network failure occurs) , however - as a consequence - traffic congestions may arise at the bottlenecks of the network that may severely affect the transmission quality and eventually the performance of the applications.

The most important packet-level QoS parameters are the delay, the delay variation and various loss measures. There are many techniques that deal with the approximation of the expected value of these parameters. These methods often differ in the applied underlying system model, that can be discrete or continuous time, buffered or bufferless just to mention the most important categories.

For the prediction of the expected delay occurring in a single-node packet server one may use the Generalized Processor Sharing (GPS) model. In this framework the server node has multiple queues fed by fluid flows and the server cycle is round-robin with different weights assigned to the queues. As the GPS model assumes fluid flows its perfect realization is not possible due to the packet-based nature of real network traffic, however it is possible to implement such scheduling strategies in real packet servers that the eventual performance of the system will be very close to that of the GPS. A profound study on the GPS model can be found in [19].

The GPS model determines the delay caused by queuing in a single-node server, however, often the end-to-end delay perceived by an application is of interest. This problem can be approached from several directions, one of the most promising of these is the network calculus framework introduced in [20] [21] [22] [23]. Network calculus is a theoretical framework for analyzing performance guarantees in computer networks. As traffic flows through a network it is subject to constraints imposed by the system components, e.g. link capacity, traffic shapers (leaky buckets), congestion control, background traffic. These constraint curves can be combined using convolution under min-plus algebra and thus the end-to-end delay bound perceived by a certain flow can be expressed analytically.

The information loss is often characterized by the saturation probability, which can be either the link or the buffer saturation probability according to the system model being a buffered or a bufferless one, respectively. The saturation probability is in fact the fraction of time in which newly arrived packets are dropped due to the lack of free resources required to handle them. This approach to describe the amount of loss is, however, can be quite inaccurate, as the amount of lost packets may differ significantly for different kind of traffic, while the perceived saturation probability remains the same. To overcome this problem several authors (e.g. the authors of [C4]) favor the use of the *workload loss ratio* (WLR) to assess the information loss instead of the saturation probability. The workload loss ratio is by definition the ratio of lost and sent traffic.

For obtaining a reasonable estimate on the saturation probability or the WLR the Chernoff bounding method is often used especially for the bufferless fluid flow multiplexing framework. For more information about Chernoff bounding methods and the bufferless fluid flow multiplexing framework the interested readers should see [31]. For characterizing the buffer saturation probability or the workload loss ratio under different asymptotical circumstances several LDT-based models have proven to be useful, some of these will be more profoundly introduced in Section 1.4, some papers of high standard on this topic are [3] and [5].

One of the most important steps of providing appropriate transmission quality is the estimation of the expected values of important QoS measures, based on which admission control strategies can be implemented. These admission control mechanisms usually carry out a "what-if" analysis on the network each time a new flow wants to enter the network approximating the values of QoS parameters in case the newcomer flow is admitted. In case the estimated QoS values are below a pre-defined threshold the newcomer flow is indeed granted, in the other case it is rejected. If the QoS measure approximation is too time-consuming this kind of admission control method becomes useless.

To address the above-mentioned problem, several authors started to invent new methods that estimate the resource need of the actual, already admitted traffic flows. The advantage of this approach is that this measure can be periodically updated and it is not necessary to be re-evaluated each time a new flow wants to enter the network. This way in the time instance of a new flow requesting admission only the requested and available resources should be compared and there is no need to do any time-consuming task. The amount of transmission capacity or buffer space a certain traffic aggregate needs to fulfill a predefined QoS goal is called its equivalent capacity or buffer requirement, respectively. High-standard work on this topic can be found in [14].

1.2 Scope and research objectives

During our research activity we were focusing on tractable, Large Deviation Theory-based models that are capable of describing single-node packet server systems with buffered communication links. These models were chosen because they are capable tools to characterize the performance of real-life packet routers. Our main objective was to develop effective methods for dimensioning the main parameters - namely the buffer space and transmission capacity - of such systems in case the maximum acceptable level of information loss is prescribed in terms of the workload loss ratio. Based on these simpler dimensioning techniques we were also aiming at designing a combined - so-called multi-level QoS - dimensioning method that can take not only packet-level (i.e. the WLR), but also call-level (i.e. session blocking probability) QoS requirements into consideration. Our ultimate goal was to incorporate these new contributions into a reliable, easy-to-implement QoS architecture that can provide satisfactory transmission quality for value-added applications (e.g. VoIP, VoD) primarily in broadband access aggregation network.

The main contributions of our research activity are presented in the three subsequent chapters of this dissertation. In Chapter 2 new equivalent capacity estimators are introduced, that are capable of approximating the minimum transmission capacity that is needed to fulfill a certain QoS goal composed in terms of target WLR. These approximation methods can play an essential role in the network capacity dimensioning activity when the network operator wants to provide reliable guarantees on the amount of information loss. The techniques presented in this part differ basically in their expected accuracy and performance, e.g. some of these formulae may give more accurate results than others in case the incoming traffic mix exhibits Gaussian nature, while other formulae may be less accurate but can be evaluated more easily.

In the third chapter of the dissertation efficient buffer requirement estimators are presented that are able to approximate the required size of the buffer in case the capacity is given and when a certain WLR level is prescribed. These formulae may serve as complementary techniques of the formulae presented in Chapter 2 in the sense that they may be used when the transmission capacity cannot be arbitrarily increased (e.g. when the transmission link is a radio channel with limited frequency resources), but still the WLR requirements have to be met. The performance of the formulae presented here may significantly differ in different network scenarios so several hints about which formula to use under which circumstances will be provided.

The fourth chapter of this dissertation contains two novel contributions. One of these is the already mentioned multi-level QoS dimensioning technique, that is capable of estimating the required transmission capacity or buffer space a server node should possess in order to fulfil the prescribed QoS goals composed in terms of WLR and session blocking probability. This dimensioning method serves as a core element of the QoS architecture presented in the second part of this chapter. This QoS architecture was designed to operate in broadband access aggregation networks where providing satisfactory transmission quality for value-added applications is crucial. The main building blocks, the applied admission policy and the exploited protocols of this QoS architecture are thoroughly discussed in this part of the dissertation.

The results in this dissertation were basically obtained through analytical methods. Since the analytical derivation of the formulae contains different simplifying assumptions and approximations, the effects of which are analytically hard to investigate, extensive numerical investigations were carried out to validate the performance of the formulae. Each of the above mentioned sections and chapters contain some numerical investigations through which the performance of the new results are assessed.

Chapter 2

QoS Measure Approximation Techniques

As it was already mentioned in the previous chapter, the main objectives of our research effort were to develop tractable resource requirement estimators for buffered transmission systems to aid QoS-aware dimensioning activity and to design a QoS-architecture that builds upon these dimensioning rules and is capable to guarantee satisfactory availability and transmission quality parameters for QoS sensitive, value-added applications.

In this chapter all the main ideas found in the literature and leading to our new results will be discussed in an adequate depth. First, the different asymptotic analysis methods of the tail probability of buffer length distribution will be presented in Section 2.1 . There the greatest emphasis will be put on the many sources asymptotic method as in many cases it is the one best resembling to real world transmission systems and it was the basis of multiple results of ours.

In Section 2.2 the work of Bahadur and Rao, Montgomery and De Veciana and Courcobetis et al, whose work led to a tractable, refined approximation of the buffer saturation probability, will be introduced. Their work was also incorporated into our new results to a certain extent.

The work of Seres et al, who continued the development of bandwidth and buffer space requirement approximations where Courcobetis and his colleagues finished, is going to be summarized in Section 2.3. These authors managed to transform the previously known formulae into such equivalent forms that are easier to evaluate and thus provided them with effective, rapidly evaluable resource requirement estimators.

In Section 2.4 the ideas publicized by Likhanov and Mazumdar and the corresponding work of Biro et al will be introduced. It is essential to get acquainted with the limitations and drawbacks of these methods to understand the contrast between these results and our contributons.

The Chapter is continued with the introduction of the work of Kim and Shroff and Borsos and György in Section 2.5. Both groups managed to formulate the connection between the buffer saturation probability and the workload loss ratio and these results play key roles in the development of our resource requirement estimators.

In Section 2.6 the work of Mao et al and Heszberger et al will be outlined. Despite they dealt with bufferless systems, their results may indirectly appear in our formulae, and thus it was found essential to briefly summarize their work herein.

In Section 2.7 a short introduction of call-level availability assessment techniques will be provided. The results to be presented in this part were exploited in our multi-level QoS provisioning techniques.

This chapter is going to be concluded with the discussion of a QoS-aware access network architecture imagined by Bouchat et al. Their work served as a baseline in developing our own idea of a multi-level QoS architecture. Their results will be presented in Section 2.8.

2.1 Asymptotic Queue Analysis

The operational parameters of packet server systems can be investigated using a buffered or a bufferless modeling approach. The latter technique may obviously result in simpler, easier-to-evaluate formulae, still the methods developed under this modeling framework are usually capable of giving only conservative estimates of real-life measures. Contrarily, incorporating the available storage capacity into the model gives chance to account for the smoothing effect of the buffer, and thus produces more complex, but at the same time more precise estimations on important system parameters.

The theory of large deviations (LDT) provides a capable mathematical apparatus for taking the beneficial effects of the buffer in the queuing system also into account. The basic concept of LDT lies in the exploitation of the asymptotic behaviour of the probability of buffer overflow in two different regimes. In the large buffer asymptotic (LBA) regime it can be shown that the tail probability of buffer overflow decays exponentially with increasing buffer sizes for the majority of traffic types. The many sources asymptotic regime claims that the decay rate of the tail probability is also exponential when the number of connections and certain system parameters are scaled.

The LBA is concerned with the estimation of the buffer overflow probability when the buffer size gets very large. Let us consider a single-server queuing system with buffer size B and service rate C. Let Q denote the queue length. According to LBA the decay rate of the logarithm of the buffer overflow probability (BOP) is asymptotically linear in B, as B approaches infinity, or more formally [16]:

$$\lim_{B\to\infty} \frac{1}{B} logP(Q > B) = const. \tag{2.1}$$

The LBA approach incorporates the effect of statistical multiplexing only partially. It succeeds in seizing the multiplexing gain arising from the statistical properties of individual sources queued in a large buffer, however, it fails to reflect the economies of scale due to the superposition of many sources.

The method of MSA holds the promise of capturing the whole statistical multiplexing gain. It incorporates all the statistical mechanisms that the LBA builds on, but goes further by taking the gain arising from the multiplexing of numerous flows into account. Therefore, it appears as a remarkably capable tool for designing precise estimators regarding the system parameters of interest.

Let the stochastic process $X[0,t)$ denote the total amount of workload arriving in the time interval $[0,t)$ from N independent flows at a buffered communication link with buffer size B and transmission capacity C. The buffer overflow probability of this finite buffer system can be deduced from the proportion of time over which the queue length $Q(C,N)$ is above level B in a queue of infinite buffer. In the MSA case the decay rate of the logarithm of the buffer overflow probability is asymptotically linear in the number of sources N in a system where the per-source buffer $b = \frac{B}{N}$ and the per source capacity $c = \frac{C}{N}$ are kept constant [16] [17]:

$$\lim_{N\to\infty} \frac{1}{N} logP(Q(N,cN) > bN) = \sup_{t>0} \inf_{s>0} \left\{ \frac{\Lambda(s,t)}{N} - s(b+ct) \right\} \stackrel{def}{=} -I, \tag{2.2}$$

where $\Lambda(s,t) \stackrel{def}{=} logE[e^{sX[0,t)}]$, the logarithmic moment generating function (i.e. cumulant generating function) of $X[0,t)$, s and t are free parameters.

The practical consequence of (2.2) is that for large N one may approximate the buffer overflow probability simply as

$$P(Q(N,C) > B) \approx e^{-NI} \tag{2.3}$$

where $-NI$ can be computed as

$$-NI = \sup_{t>0} \inf_{s>0} \{\Lambda(s,t) - s(B+Ct)\} \stackrel{def}{=} \sup_{t>0} \inf_{s>0} J(s,t) \tag{2.4}$$

In practice there is often a QoS constraint for the probability of buffer overflow ($e^{-\gamma}$), which criterion can be expressed formally as:

$$P(Q(N,C) > B) \approx e^{-NI} \leq e^{-\gamma} \tag{2.5}$$

or

$$\sup_{t>0} \inf_{s>0} J(s,t) \leq -\gamma. \tag{2.6}$$

2.2 Improved Asymptotic Queue Analysis

The MSA-based buffer saturation probability approximations exhibit logarithmic asymptotic properties, which means that the logarithm of the approximation and the logarithm of the actual probability tend to each other as the number of sources approaches infinity. The asymptotic property of these estimators can however be improved by using methods that build upon the Bahadur-Rao theorem [15]. These techniques provide BOP approximations with exact asymptotic properties, which means the probability itself and its estimation tend to each other as N goes to infinity.

The Bahadur-Rao improvement was succesfully incorporated into MSA-based BOP estimators by Likhanov and Mazumdar who developed the following formula that holds as $N \to \infty$ [5]:

$$P(Q(N,C) > B) = \frac{1}{\sqrt{2\pi\sigma^2(s^*,t^*)s^{*2}}} e^{-NI} \left(1 + O\left(\frac{1}{N}\right)\right) \tag{2.7}$$

where $\sigma^2(s,t) \stackrel{def}{=} \frac{\partial^2 \Lambda(s,t)}{\partial s^2}$, $s^*(t) \stackrel{def}{=} \arg\inf_s J(s,t)$, $t^* \stackrel{def}{=} \arg\sup_t J(s^*(t),t)$, $s^* \stackrel{def}{=} s^*(t^*)$.

Formula (2.7) practically means that a reasonable estimate on the BOP can be formulated as

$$P(Q(N,C) > B) \approx \frac{1}{\sqrt{2\pi\sigma^2(s^*,t^*)s^{*2}}} e^{-NI} \tag{2.8}$$

The disadvantage of this method is that for the evaluation of the formula the second derivative of the log-moment generating function is also required. This problem was addressed by Montgomery and De Veciana who proposed a round-about solution by giving a tractable estimate on the second derivative by applying a second-order approximation of $\Lambda_(s,t)$ around s^*, t^* [13]:

$$s^{*2}\sigma^2(s^*,t^*) \approx -2(\Lambda(s^*,t^*) - s^*(B + Ct^*)) = 2NI \tag{2.9}$$

Following the above explained train of thought further, Courcobetis et al were able to recompose the rate function type acceptance region by incorporating the B-R pre-factor into the original constraint [14][3]:

$$-NI \leq -\gamma^{B \cdot R} \stackrel{def}{=} -\gamma + \frac{\frac{1}{2}log(4\pi\gamma)}{1+\frac{1}{2\gamma}} \tag{2.10}$$

2.3 Direct Estimations on Resource Requirements

The main disadvantage of the methods introduced in the previous sections is that they approximate the expected buffer saturation probability instead of the resource requirement of currently present network flows. This way those formulae have to be re-evaluted and compared to the target QoS measure each time when a new network flow wants to enter the transmission link.

A slightly better, but computationally more complex solution is to approximate the required capacity ($C_{equ,BOP}$) or buffer need ($B_{req,BOP}$) of the already present flows by evaluating the following implicit equations

$$C_{equ,BOP} \stackrel{def}{=} \inf \left\{ C : \sup_{t>0} \inf_{s>0} J(s,t) \leq -\gamma \right\} \tag{2.11}$$

and

$$B_{req,BOP} \stackrel{def}{=} \inf \left\{ B : \sup_{t>0} \inf_{s>0} J(s,t) \leq -\gamma \right\} \tag{2.12}$$

The result manifests in a bandwidth or buffer space type measure which can be compared to the value of available network resources. The computations in this approach can be carried out periodically in the background, which means that there is no need to re-evalute the formulae upon the arrival of new flows any more. However this method is computationally more demanding than the previous ones, as it requires an optimization carried out in three dimensions (C, s, and t) instead of in only two dimensions (s and t).

The disadvantages of both introduced methods can be overcome by using the direct capacity and buffer space estimation technique introduced by Seres et al in [3]. Their formulae are proven to be equivalent with Formulae (2.11) and (2.12) and at the same time they have about the same computational complexity as the QoS measure estimations introduced in Sections 2.1 and 2.2. Their basic formulae manifest in the following forms:

$$C_{equ,BOP} = \sup_{t>0} \inf_{s>0} \left\{ \frac{\Lambda(s,t)+\gamma}{st} - \frac{B}{t} \right\} \tag{2.13}$$

and

$$B_{req,BOP} = \sup_{t>0} \inf_{s>0} \left\{ \frac{\Lambda(s,t)+\gamma}{s} - Ct \right\} \tag{2.14}$$

The authors of [3] incorporated the results of Courcobetis et al [14] into their work and proved that Formuale (2.13) and (2.14) holds for the case when the underlying buffer overflow probability approximation applies the Bahadur-Rao improvement prefactor. In this case, however, γ should be substituted with $\gamma^{B \cdot R}$ in the formulae.

2.4 Capacity Requirement Estimators for Workload Loss Ratio

Many techniques in the literature - including the ones presented here by now - focus on the estimation of the tail probability of queue length. The success of this kind of approach to characterize the amount of information loss (which is the real matter), however, is not obvious. What is more the authors of [4] pointed out that the relation of buffer saturation probability and workload loss ratio can be arbitrary. Information loss can be described by the workload loss ratio (WLR), which is by definition:

$$WLR(B) = \frac{E[Q-B]^+}{E[X]} \tag{2.15}$$

where Q denotes the queue length, X is the random variable characterizing the amount of workload arriving in one time unit.

Likhanov and Mazumdar in [5] were able to give an MSA based equation on the workload loss ratio as $N \to \infty$:

$$WLR(B) = \frac{1}{Ms^{*2}\sqrt{2\pi N\sigma^2(s^*,t^*)}}e^{-NI}\left(1+O\left(\frac{1}{N}\right)\right) \tag{2.16}$$

where $M = E[X]$.

As it can be seen this formula incorporates the Bahadur-Rao correction pre-factor that was introduced in [15]. Again the second derivative of the log-moment generation function can be eliminated using the method developed by Montgomery and De Veciana [13]. Eventually the following approximation on the WLR can be obtained:

$$WLR(B) \approx e^{-NI-\frac{1}{2}log4\pi NI-logs^*M} \stackrel{def}{=} \widetilde{WLR} \tag{2.17}$$

In this context the definition of the equivalent capacity in case the QoS requirement is composed in terms of the WLR can be written as:

$$C^{B\cdot R}_{equ,WLR} \stackrel{def}{=} \inf\left\{C : \widetilde{WLR} \leq e^{-\gamma}\right\} \tag{2.18}$$

Based on these results Biro et al [C4] succeeded in developing an equivalent capacity estimator that appears in a very similar form that the formulae invented by Seres et al in [3] :

$$C^{B\cdot R}_{equ,WLR} \approx \sup_{t>0}\inf_{s>0}\left\{\frac{\Lambda(s,t)+\gamma+\tilde{\varepsilon}(\gamma,B,M)}{st} - \frac{B}{t}\right\} \tag{2.19}$$

, where

$$\tilde{\varepsilon}(\gamma,B,M) = \frac{2\gamma log(e^{-\frac{1}{2}-\gamma}+\frac{B+2B\gamma}{4M\sqrt{\pi}\gamma^{3/2}})}{1+2\gamma}. \tag{2.20}$$

Here we note that for the development of Formula (2.19) multiple approximations were carried out. These were necessary to arrive to a closed-form solution, but they naturally affect the preciseness of the estimation. Another disadvantage of this result is that its extension to the estimation of buffer requirement is difficult. Also the counterpart formula (i.e. the one without the B-R pre-factor) cannot be easily obtained.

2.5 The Connection between Workload Loss and Saturation Probability

The results of Biro et al seem to be hardly extendable to buffer requirement estimators so other ideas are needed to construct to reasonable buffer requirement estimators. A more promising approach can be taken according to the results of Kim and Shroff who shed light onto the connection between the buffer saturation and workload loss ratio in [6]. Their findings were mainly based on simulation studies, however, they succeeded to prove their main theorem analitically as well. This main theorem was:

$$WLR(B) = \frac{WLR(B=0)}{P(Q>0)} P(Q>B) \tag{2.21}$$

Independently from the research activity of Kim and Shroff, Borsos and György publicized a similar result in [4]. They arrived to their result following an analytical approach and at the end some heuristics. Their statement was that the following reasonable approximation can be given for the WLR using some bufferless measures and the buffer saturation probability:

$$WLR(B) \approx \frac{WLR(B=0)}{P(X>C)} P(Q>B) \tag{2.22}$$

The relation of these results is obvious: using the formula of Borsos and György one may formulate an upper bound on the actual workload loss ratio that can be computed with Formula (2.21). Formally written:

$$WLR(B) = \frac{WLR(B=0)}{P(Q>0)} P(Q>B) \leq \frac{WLR(B=0)}{P(X>C)} P(Q>B) \tag{2.23}$$

These results clarified the relation of the WLR and the buffer saturation probability and thus provided the means for leading back the problem of resource requirement estimation in case of WLR-based QoS guarantees to the case when the buffer saturation probability is prescribed.

2.6 Bufferless Loss and Saturation Probability Estimators

As we could see in the previous section, for the computation of WLR from the BOP the WLR and BOP of the system with zero-sized buffer are required.

Alternatively, the link saturation probaility (i.e. $P(X > C)$) can also be used instead of $P(Q > 0)$. There are several techniques capable of approximating WLR(0) and $P(X > C)$, but few methods can provide a tractable estimate on $P(Q > 0)$.

The recent results of Heszberger et al on this topic provide some useful and tractable methods for estimating the bufferless saturation probability and workload loss ratio. Their main results will be outlined in the following paragraphs.

Let $X_1, \ldots, X_n$ indicate n independent random variables (e.g. transmission rates of communication sources) with $0 \leq X_i \leq p_i$, $X = \sum_{i=1}^{n} X_i$ and $M = E[X]$. Then for $s > 0$,

$$P(X > C) \leq \left(\frac{M - n_Y p}{C - n_Y p} \right)^{n_Y - \frac{C}{p}} \left(\frac{M}{C} \right)^{\frac{C}{p}}, \tag{2.24}$$

where $p = \max(p_i, i = 1, \ldots, n)$, $n_Y = \lceil \sum_{i=1}^{n} p_i / p \rceil$.

$$WLR(0) \leq \frac{1}{s^* M e^{s^* C}} \left(1 - \frac{M}{n_Y p} + \frac{M}{n_Y p} e^{s^* p} \right)^{n_Y}, \tag{2.25}$$

where s^* is the solution of the following equation.

$$\frac{n_Y p (n_Y p - M)}{n_Y p - M + M e^{ps}} + n_Y p - \frac{1}{s} - C = 0. \tag{2.26}$$

2.7 Call-level QoS Provisioning

As it was stated before among the primary objectives was to develop a QoS architecture that is capable of providing not only packet-level QoS guarantees but proper call (or session) level availability as well. The techniques that have already been discussed in this section were focusing on packet-level resource requirement assessment techniques. In this subsection a short overview on call-level availability provisioning techniques will be provided.

Ensuring satisfactory call-level avalability or in other words limiting the expected time blocking probability (TBP) is a problem that has been investigated by many since the first telephony networks were established. Back then the first priority was to build enough phone lines between exchanges that can service the needs of a certain population.

Probably the greatest pioneer of this specific research area was Agner Krarup Erlang, whose results were widely used for phone line capacity dimensioning throughout the world in the golden era of telephony. One of his main contributions was the proof of that the Poisson distribution applies to random telephone traffic which he publicized in 1909. His greatest achievement, however, was presented in 1917, when he successfully formulated the loss and waiting time in automatic telephone exchanges, i.e. he managed to create his famous "B" and "C" formulae [9]. Erlang's B formula tells the time blocking probability of the queuing system that has N servers (telephone lines) but the connected population is considered to be infinite. Members of the population initiate calls independently with λ intensity and h holding time. The TBP of such system is:

$$TBP = \frac{\frac{(\lambda h)^N}{N!}}{\sum_{i=0}^{N} \frac{(\lambda h)^i}{i!}} \tag{2.27}$$

The fundamental work of Erlang were followed by others who investigated similar problems but usually for more specific queing system scenarios. A well-organized overview of these can be found in [11]. One of these results - the contribution of Tore Olaus Engset - is going to be highlighted here. Engset provided a loss formula for the case when the population (using the queuing system) is not considered to be infinite:

$$TBP = \frac{\begin{pmatrix} P \\ N \end{pmatrix} (\lambda h)^N}{\sum_{j=0}^{N} \begin{pmatrix} P \\ j \end{pmatrix} (\lambda h)^j}. \tag{2.28}$$

where P is the population size.

This contribution have been chosen to be used as a building block in designing our multi-level QoS-aware resource dimensioning technique, which serves as an important element of the QoS architecture we have designed.

2.8 QoS Architecture for Broadband Access Networks

The evolution of access technologies enables the introduction of new, value-added services in networks where they were previously unavailable. However, possible high data rates in the first mile and plentiful physical capacity in the access network by themselves may be inadequate to provide proper QoS level for premium applications and thus ensure subscribers' satisfaction. (Moreover, the above-mentioned network segments often struggle with capacity scarceness, especially in case of radio-based transmission technologies.) Therefore, implementation of an efficient QoS architecture that protects the network from failures and overloads is also needed.

The main requirements that are most often composed in connection with QoS architectures are reliability, simplicity and efficiency. Systems that build upon already available network elements, exploit functionalities of commonly-used protocols, can make quick traffic management decisions and are capable of coping with failures are preferred.

The ENRICO (Enhanced Resource and Information Control) model contains a QoS architecture designed for DSL access networks and more or less fulfils the previously listed goals [2]. The main idea of this framework is that a logical overlay network is defined upon the physical topology. This logical network consists of point-to-point connections - so called QoS pipes - between edge node (DSL Access Multiplexer-Broadband Access Server) pairs for each QoS class. These logical channels can be established e.g. as permanent virtual paths (PVPs), virtual LANs (VLANs) or Label Switched Paths (LSPs) in case the applied transport technology is ATM, Ethernet or MPLS, respectively. The QoS pipes are supervised by a central entity called SRB (Session Resource Broker), which performs traffic load control in a way that is transparent to both the subscriber and the application service provider. For gathering the needed information for admission control decisions certain functions of currently available protocols such as the RSVP or the DHCP are also exploited. The acquired resource requirements of incoming service sessions are sent directly to the SRB which checks resource availability and grants access accordingly.

The disadvantage of this solution lies in that the centralized architecture is sensitive to failures and the SRB entity should possess huge computational capacity to process all the incoming requests. Also in [2] the actual way of dimensioning the logical overlay network is not discussed at all. To heal the imperfections of ENRICO and to fill these gaps we have proposed the QoS architecture which is going to be introduced in Section 5.2.

Chapter 3

Novel Equivalent Capacity Estimation Techniques

Novel, value-added applications may become widespread in the converged multi-service packet-switched networks only if reliable QoS assurances are provided. The truth of this statement cannot be questioned, however, the ultimate, universal method answering the corresponding "how" question is still sought. The main concern here is - as usual - the cost effectiveness of the system to be implemented: the optimal system operating point on the curve characterizing the relation between resource savings and operational cost (or system complexity) is to be found.

The "heart and soul" of an efficient QoS provisioning mechanism is an effective resource requirement assessment method that is capable of determining resource needs of diverse traffic types with an adequate accuracy. Basically the resource requirement of a network service can be characterized by two parameters: bandwidth (or transmission capacity) and buffer size need. Obviously there is a tradeoff between these two measures, but in general there is always a principle or a constraint that helps to decide which parameter should be considered as given. In many cases the given parameter is the allocated buffer size, constrained by the maximum tolerable delay requirement of the application in question or simply by the technical parameters of the device under configuration. Thus in the majority of cases the minimum service rate of the server that satisfies the loss requirement of the corresponding applications is to be determined.

The transmission capacity requirement of a certain application is usually obtained analytically, but some empirical approaches can also be imagined.

The analytical method regards the analytical model of the traffic source in question. These models tend to be either overly simplistic to be realistic or too complex to be tractable, so it is very important to choose the right model for the right task. For network dimensioning tasks - where the running time of the process is not mission-critical - it may be wise to choose more detailed model in the hope of improved accuracy, while in case of admission control related activities a more general, simple model would be satisfactory. More sophisticated analytical models usually require more parameters known a priori which may forestall their usability and promote the application of the so-called parsimonious models (i.e. models requiring few parameters) instead. For acquiring the required model parameters it may be wise to apply measurement based methods wherever possible as these methods can provide more accurate and adaptive values, than purely theoretical considerations.

Numerous articles and papers deal with the problems of obtaining a reasonable approximation on the capacity requirements of certain applications. Celebrated papers and studies on this topic are delivered by Kelly [7] and Roberts [8]. Courcobetis et al in [14] developed so-called equivalent capacity formulae for the case when the buffer overflow probability is constrained. Seres et al in [3] developed methods that produce the same results as the formulae of Courcobetis, however have lower computational demand. Likhanov and Mazumdar [5] tried to approximate the WLR under the certain asymptotic regimes in order to form the basis of an improved equivalent capacity formula, while Jamin et al [12] and Courcobetis et al [27] were focusing on the measurement-based parameter acquisition methods that are associated with the equivalent capacity formulae.

In this chapter several capacity requirement estimation methods that are able to obtain the minimum transmission rate needed by a certain application to fulfil its pre-defined QoS goal composed in terms of the workload loss ratio are going to be introduced. The main difference distinguishing our methods from all others found in the literature is that they account for the workload loss ratio (while other usually regard the saturation probability of the buffer) and the reduced computational load in comparison with the straightforward application of the definition of the equivalent capacity presented in (2.11). The approximation techniques to be shown in this chapter basically differ from each other in their accuracy, computational demand and scope of applicability.

The forthcoming sections of this chapter are organized as follows. In the first section the basic terms and definitions required to understand the new results to be presented later will be introduced. In the subsequent section a new asymptotically precise capacity estimation technique will be presented. In the next section a practical, easier to implement, but in general less accurate version of the basic method is to be described. Later in this chapter a third equivalent capacity approximation method is to be introduced that may outperform the other two in terms of accuracy especially in case the arrival process is well-characterized with a Gaussian process. In Section 3.5 the possible techniques that can be applied to acquire the parameters needed by the capacity requirement estimator are to be investigated. After that - in Section 3.6 - some cases when the capacity requirement estimation can be solved with closed-form formulae will be shown. This chapter is concluded with numerical examples illustrating the efficiency and behavior of our novel methods.

3.1 Terms and Definitions

In this section a short background on the basic terms upon which our new contributions are built will be provided and the definitions, terms and notations to be used throughout this chapter will also be introduced.

Let the stochastic process $X[0,t)$ denote the total amount of workload arriving in the time interval $[0,t)$ from N independent flows at a buffered communication link with buffer size B and transmission capacity C. N is regarded as a scaling factor and in that sense we can identify a per-source buffer $b = \frac{B}{N}$ and per source capacity $c = \frac{C}{N}$. Let us also assume that $X[0,t)$ has stationary increments.

The buffer overflow probability of this finite buffer system ($P(Q > B)$) can be deduced from the proportion of time over which the queue length $Q(C,N)$ is above level B in a queue of infinite buffer. In the many sources asymptotic (MSA) regime the decay rate of the logarithm of the buffer overflow probability is asymptotically linear in the number of sources N in a system where the per-source parameters are kept constant [16] [17]:

$$\lim_{N\to\infty} \frac{1}{N} logP(Q(N,cN) > bN) = \sup_{t>0} \inf_{s>0} \left\{ \frac{\Lambda(s,t)}{N} - s(b+ct) \right\} \stackrel{def}{=} -I, \quad (3.1)$$

where $\Lambda(s,t) \stackrel{def}{=} logE[e^{sX[0,t)}]$ is the cumulant generating function of $X[0,t)$, s and t are free parameters. I is often referred to as the rate function.

The practical consequence of the above equation is that for large N one may approximate the buffer overflow probability (BOP) simply as

$$P(Q(N,C) > B) \approx e^{-NI} \tag{3.2}$$

where $-NI$ can be computed as

$$-NI = \sup_{t>0} \inf_{s>0} \{\Lambda(s,t) - s(B + Ct)\} \tag{3.3}$$

The QoS constraint on the workload loss ratio can be expressed formally as:

$$WLR \stackrel{def}{=} \frac{E[Q-B]^{+}}{E[X]} \leq e^{-\gamma} \tag{3.4}$$

where X is the random variable characterizing the workload arriving in a time unit.

The minimal transmission capacity requirement - i.e. the minimal service rate at which the prescribed WLR threshold is still not exceeded - is called equivalent capacity and denoted by $C_{equ,WLR}$. Its definition is:

$$C_{equ,WLR} \stackrel{def}{=} \inf \left\{C : WLR \leq e^{-\gamma}\right\} \tag{3.5}$$

3.2 Theoretical Equivalent Capacity Estimation

In this section an equivalent capacity estimation method that greatly builds upon the buffer saturation probability approximation techniques in the many sources asymptotic regime described by the Large Deviation Theory is going to be introduced. Our novel method manifests in the same, explicit form as the similar contributions of Seres et al [3], but is capable of estimating the equivalent capacity of the traffic mix served by the queuing system in case the QoS constraint is composed in terms of the WLR instead of the buffer saturation probability.

Theorem 3.2.1. *The minimal transmission rate $C_{equ,WLR}$ that ensures that the expected WLR will not exceed the prescribed threshold ($e^{-\gamma}$) can be approximated by the solution of the fix-point equation*

$$c = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + F(\gamma,c)}{st} - \frac{B}{t} \right\} \stackrel{def}{=} f(c). \tag{3.6}$$

with respect to c, where $F(\gamma,c) = \gamma - logP\{Q > 0\} + logWLR(0)$.

Proof: Kim and Shroff revealed the following connection between the buffer saturation probability and workload loss ratio in [6]:

$$WLR(B) = \frac{WLR(0)}{P(Q > 0)} P(Q > B) \tag{3.7}$$

This implies that the prescription $WLR(B) \leq e^{-\gamma}$ can be transformed into the following equivalent prescription on BOP :

$$P(Q > B) \leq e^{-\gamma + logP(Q>0) - logWLR(0)} \tag{3.8}$$

Defining $F(\gamma, C) \overset{def}{=} \gamma + logP(Q > 0) - logWLR(0)$ (3.8) can be written as

$$P(Q > B) \leq e^{-F(\gamma,C)} \tag{3.9}$$

According to Seres et al the minimal transmission capacity that ensures that $P(Q > B) \leq e^{-\delta}$ manifests in the following form [3]:

$$C_{equ,BOP} = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + \delta}{st} - \frac{B}{t} \right\} \tag{3.10}$$

Using $\delta = F(\gamma, C)$ in (3.10) the statement of the theorem follows. Q.E.D.

The $c = f(c)$ fix point equation is to be solved iteratively, i.e $c_{n+1} = f(c_n)$, where n denotes the iteration step. The iterative process should be continued until the c and $f(c)$ values have a relative difference that is less than a predefined threshold, denoted by ε. This "stop condition" of the algorithm can be formally written as:

$$\frac{|c_n - f(c_n)|}{max \{c_n, f(c_n)\}} \leq \varepsilon \tag{3.11}$$

Practically c_0 should be an arbitrary value between the aggregate mean and aggregate peak rate of the investigated traffic mix. These values can usually be regarded as given or can be obtained easily e.g. with the aid of measurements.

For the evaluation of (3.6) a double optimization task should be performed in each iteration step. This approach may yield an accurate enough solution faster than the definition-based technique, where finding the C value for the next trial is not so straightforward. Also our experiments showed that the fix-point equation method produces an equivalent capacity estimation with less than 1% relative difference (i.e. $\varepsilon = 0.01$) in a couple of iteration cycles.

3.3 Practical Approximation for Equivalent Capacity

The equivalent capacity approximation procedure discussed in the previous section requires the computation of $P(Q > 0)$ in each iteration step. Estimating $P(Q > 0)$ however is a highly non-trivial task and usually it is advisable to be avoided. Some authors (e.g. Shroff et al in [6]) state that certain buffer saturation probability approximation techniques produce quite accurate results even for $B = 0$, nevertheless those techniques usually require quite a few extra parameters. Thus it seems wise to seek a round-about solution and eliminate $P(Q > 0)$ from the formulae in a way that yields a reliable, reasonably accurate and easily evaluable expression of the equivalent capacity.

In this section a modified version of the formula used in the fixed-point equation (3.6) is going to be shown. this formula applies $P(X > C)$ as a substitute for $P(Q > 0)$. It is stated that solving the resulting fixed-point equation a reasonable, conservative estimate of the the solution of the fix-point equation (3.6) can be obtained. Let us denote the solution of the orginal fix point equation (3.6) by $\widetilde{C}_{equ,WLR}$, formally written as:

$$\widetilde{C}_{equ,WLR} \stackrel{def}{=} \lim_{n\to\infty} c_n \tag{3.12}$$

where c_n, n positive integer denotes the result of (3.6) in the nth iteration step.

Theorem 3.3.1. *A practical upper bound can be obtained on the equivalent capacity approximation $\widetilde{C}_{equ,WLR}$ by the solution of the following fix-point equation*

$$\hat{c} = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + \hat{F}(\gamma,\hat{c})}{st} - \frac{B}{t} \right\}. \tag{3.13}$$

with respect to $\hat{c}$, where $\widehat{F}(\gamma,\hat{c}) = \gamma - logP\{X > \hat{c}\} + logWLR(0)$.

Proof: According to Borsos and György [4]

$$WLR(B) \approx \frac{WLR(0)}{P(X > C)} P(Q > B) \tag{3.14}$$

Regarding (3.7) and that $P(X > C) \le P(Q > 0)$ we get

$$WLR(B) \le \frac{WLR(0)}{P(X > C)} P(Q > B) \tag{3.15}$$

Combining (3.15) with the original prescription $WLR(B) \leq e^{-\gamma}$ the following stricter prescription can be formulated:

$$\frac{WLR(0)}{P(X > C)} P(Q > B) \leq e^{-\gamma}, \tag{3.16}$$

Prescription (3.16) can be transformed into the following equivalent form:

$$P(Q > B) \leq e^{-\gamma + logP(X>C) - logWLR(0)} \tag{3.17}$$

Let us define $\widehat{F}(\gamma, C) \stackrel{def}{=} \gamma + logP(X > C) - logWLR(0)$ and rewrite (3.17) to the following form

$$P(Q > B) \leq e^{-\widehat{F}(\gamma, C)} \tag{3.18}$$

According to Seres et al the minimal bandwidth demand that ensures the fulfilment of the target BOP level, i.e. $P(Q > B) \leq e^{-\delta}$ manifests in the form presented in (3.10). Using $\delta = \widehat{F}(\gamma, C)$ in (3.10) the statement of the theorem follows. Q.E.D.

Let us denote the solution of (3.13) by $\widehat{C}_{equ,WLR}$, formally written as:

$$\widehat{C}_{equ,WLR} \stackrel{def}{=} \lim_{n \to \infty} \hat{c}_n. \tag{3.19}$$

where $\hat{c}_n$, n positive integer denotes the result of (3.13) in the nth iteration step.

For the above-described method $P(X > C)$ and $WLR(0)$ has to be computed in each iteration step. As the calculation of $P(X > C)$ has been extensively studied in the literature it is easy to find an appropriate estimation method among them that works with the given parameters and computationally favorable. While the computation of $WLR(0)$ has been studied by only few authors, there still exist some techniques that require very few parameters and provide quite accurate result with moderate computational complexity. Such techniques will be discussed later in this chapter in Section 3.5.

It was stated in the previous section that the fix-point equation-based method generally produces an accurate enough solution in fewer iteration steps than the equivalent capacity based technique. Naturally this statement still holds for the algorithm discussed in this section, however the accuracy of the final result may be worse than that of the the original fix-point equation or the definition based approach due to the applied $P(Q > 0) \approx P(X > C)$ approximation.

3.4 Modified Equivalent Capacity Approximation

The underlying MSA approximation applied in the previous equivalent capacity estimation formulae (3.6) and (3.13) can be refined by applying the Bahadur-Rao theorem [15]. The resulting estimator would produce asymptotically more accurate estimations at the cost of increased parameter demand, i.e. for the Bahadur-Rao pre-factor the second derivative of the logarithmic moment generating function should be known. However, this new parameter can be eliminated from the formulae by applying a second order approximation of $\Lambda(s,t)$ around the optimal s, t values [13].

Based on the results of Courcobetis et al [14] and Seres et al [3] two equivalent capacity estimators have been designed. These formulae have the same parameter demand and only a slightly higher computational complexity as our previous estimators, but are in theory capable of outperforming the accuracy of those. This improved accuracy is primarily expected when the aggregate arrival process has Gaussian nature as in this case the second order approximation of $\Lambda(s,t)$ is accurate.

Theorem 3.4.1. *A $C_{equ,WLR}$ estimate - that may outperform $\widetilde{C}_{equ,WLR}$ and $\widehat{C}_{equ,WLR}$ can be acquired by solving the following fix-point equations, respectively*

$$c^{B\cdot R} = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + G(F(\gamma, c^{B\cdot R}))}{st} - \frac{B}{t} \right\}. \tag{3.20}$$

and

$$\hat{c}^{B\cdot R} = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + G(\hat{F}(\gamma, \hat{c}^{B\cdot R}))}{st} - \frac{B}{t} \right\}. \tag{3.21}$$

where $G(x) \stackrel{def}{=} x - \frac{\frac{1}{2} log 4\pi x}{1+\frac{1}{2x}}$

Proof: In case the following approximation is used to estimate the BOP

$$P(Q(N,C) > B) \approx \frac{1}{\sqrt{4\pi NI}} e^{-NI} \tag{3.22}$$

instead of (3.2), then - according to Seres et al - the equivalent capacity for QoS constraint $P(Q > B) \leq e^{-\delta}$ manifests in the following form

$$C^{B\cdot R}_{equ,BOP} = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + \delta^{B\cdot R}}{st} - \frac{B}{t} \right\}. \tag{3.23}$$

where

$$\delta^{B\cdot R} \stackrel{def}{=} \delta - \frac{\frac{1}{2}log(4\pi\delta)}{1+\frac{1}{2\delta}} \tag{3.24}$$

Combining the $P(Q > B) \leq e^{-\delta}$ prescription (used as starting point by Seres et al) and the prescriptions composed in (3.8) and (3.17) the statements of the theorem follow if the substitution $\delta = F(\gamma, C)$ or $\delta = \widehat{F}(\gamma, C)$ is applied. Q.E.D.

Let us denote the solutions of (3.20) and (3.21) by $\widetilde{C}^{B\cdot R}_{equ,WLR}$ and $\widehat{C}^{B\cdot R}_{equ,WLR}$, respectively. Formally written as:

$$\widetilde{C}^{B\cdot R}_{equ,WLR} \stackrel{def}{=} \lim_{n\to\infty} c_n^{B\cdot R}, \tag{3.25}$$

and

$$\widehat{C}^{B\cdot R}_{equ,WLR} \stackrel{def}{=} \lim_{n\to\infty} \hat{c}_n^{B\cdot R}. \tag{3.26}$$

where $c_n^{B\cdot R}$ and $\hat{c}_n^{B\cdot R}$, n positive integer denotes the result of (3.20) and (3.21) in the nth iteration step, respectively.

The (3.20) and (3.21) estimators are recommended to be used primarily when the aggregate traffic exhibits Gaussian nature, as in that case they can provide more precise estimate of the equivalent capacity. Traffic mixes generated by many independent traffic sources with identically distributed data rates (and finite variance) generate Gaussian-like traffic.

3.5 Obtaining the required parameters

For the evaluation of the formulae presented in this chapter so far $\Lambda(s,t)$ and $F(\gamma, C)$ - or rather $\hat{F}(\gamma, C)$- need to be known for a given γ, C pair and for each s, t values. As regards the cumulant generating function of $X[0,t)$ (i.e. $\Lambda(s,t)$) it can be obtained either analytically or using measurement-based methods. For the analytical approach an appropriate model of the traffic sources should be chosen in a way that the moment generating function of $X[0,t)$ would be expressible. The main advantage of this method is that non-existing systems can be analyzed prior their establishment, and choosing an analytical source model whose moment generating function is easily expressible and handleable may result in easily evaluable equivalent capacity approximations. On the other hand choosing irrelevant or overly simplistic source models may produce unacceptable, inaccurate estimations on the required capacities.

Courcobetis et al in [27] introduced a method for obtaining a strictly measurement-based estimation on the moment generating function. They recommended the online maintenance of the following estimates on the effective bandwidth:

$$\hat{\Lambda}(s_i, t_j) = log\left(\frac{1}{K}\sum_{k=1}^{K} e^{s_i X[(k-1)t_j, kt_j)}\right) \tag{3.27}$$

where s_i and t_j are discrete values of s and t, i and j are non-negative integers, K denotes the number of available measurements.

Using this method real systems can be investigated without the need of creating accurate, analytical source models. However, as with all measurement-based techniques, the accuracy and reliability of the estimates may be an issue.

For the evaluation of $F(\gamma, C)$ or $\hat{F}(\gamma, C)$ $P(Q > 0)$ or $P(X > C)$ and the workload loss ratio of the equivalent bufferless system (i.e. WLR(0)) have to be calculated or approximated. As the calculation of $P(Q > 0)$ is a highly non-trivial task in general, the possible practical methods dealing with the approximation of $P(X > C)$ will be focused on instead.

To obtain $P(X > C)$ and the expected WLR(0) additional information on the admitted traffic flows is needed. Obviously, the amount and accuracy of this additional information affects the preciseness of the estimates, but as a general rule it is wise to keep the number of required parameters to a minimum in order to make the whole resource requirement assessment technique easier to realize in practice.

Heszberger at al have recently developed some so-called parsimonious techniques that are capable of providing reasonable estimates on the expected saturation probability and workload loss ratio under the bufferless fluid flow multiplexing framework in case only the the aggregate mean and the individual peak rates of the admitted flows are known [31] . Their most promising, closed-form estimators are also recalled here.

Let $X_1, \ldots, X_n$ indicate n independent random variables (e.g. transmission rates of communication sources) with $0 \leq X_i \leq p_i$, $X = \sum_{i=1}^{n} X_i$ and $M = E[X]$. Then for $s > 0$,

$$P(X > C) \leq \left(\frac{M - n_Y p}{C - n_Y p}\right)^{n_Y - \frac{C}{p}} \left(\frac{M}{C}\right)^{\frac{C}{p}}, \tag{3.28}$$

where $p = \max(p_i, i = 1, \dots, n)$, $n_Y = \lceil \sum_{i=1}^{n} p_i/p \rceil$.

The right-hand side of (3.28) is the exact saturation probability of the aggregated flow consisting of n_Y independent on-off sources with homogeneous p peak data rate and M aggregated mean arrival rate. According to Mao et al [28] substituting the actual traffic mix consisting of N sources with these n_Y on-off sources with identical peak rate in a manner that both the aggregated peak and mean rate remains the same, the calculated saturation probability of the substitute traffic mix will surely be greater than or equal to the saturation probability of the original mix.

Following a similar process as above one may construct the corresponding WLR(0) bound, obtaining:

$$WLR(0) \leq \frac{1}{s^* M e^{s^* C}} \left(1 - \frac{M}{n_Y p} + \frac{M}{n_Y p} e^{s^* p} \right)^{n_Y}, \tag{3.29}$$

where

$$s^* = \frac{1}{p} \log \frac{C(M - n_Y p)}{M(C - n_Y p)}. \tag{3.30}$$

Here we note that it is possible to use a more precise s^* value instead of (3.30), but that approach does not yield closed-form solutions and only slightly improves the accuracy of the WLR estimation. Therefore we recommend to use (3.29) with 3.30 to approximate the bufferless WLR.

3.6 Closed-form solutions

The equivalent capacity estimation methods presented so far require the solution of a double-optimization task in several subsequent iteration steps. This of course can be time-consuming and affect the applicability of this technique for example as a CAC (call admission control) method. For such fields of application computation complexity (speed) is more important than accuracy, and therefore closed-form formulae are favorable.

In this section we are going to show a special case when the formulae presented in the previous sections can be written in closed-forms. In order to be able to do that the aggregate traffic mix should be substituted or modelled with a fractional Brownian motion (fBm) traffic.

The stochastic process $[Z_t, t \in \Re]$ is a normalised fBm with self-similarity (or Hurst-) parameter $H \in (0, 1)$, if it has stationary increments and continuous paths, $Z_0 = 0$, $E[Z_t] = 0$, $Var[Z_t] = |t|^{2H}$ and if Z_t is a Gaussian process.

Let us define the process $X[0,t) \stackrel{def}{=} mt + Z_t$ for $t > 0$. It is a fractional Brownian traffic and can be regarded as the the amount of traffic offered to a multiplexer in time interval $[0,t)$. The fBm model is a so-called self similar model, that has been suggested by Norros [29] and Gibbens [30] for describing Internet traffic aggregates.

Using this model the cumulant generating function manifests in the following form: $\Lambda(s,t) = stm + (1/2)s^2\sigma^2 t^{2H}$. Substituting the cumulant generating function of the fBm traffic into (3.13) and performing the optimization we get the following closed-form fix-point equation:

$$\hat{c} = M + H\left(\sqrt{2\widehat{F}(\gamma,\hat{c})}\sigma\right)^{(1-H)}\left(\frac{1-H}{B}\right)^{(1-H)/H} \tag{3.31}$$

where $\widehat{F}(\gamma,\hat{c}) = \gamma - logP\{X > \hat{c}\} + logWLR(0)$. It was shown in the previous section, that $\widehat{F}(\gamma,\hat{c})$ can simply be computed using (3.28) and (3.29).

Not surprisingly the Bahadur-Rao version of the above presented closed-form fix-point equation manifests in almost the same form, as the presence of the B-R prefactor does not affect the optimization procedure. Therefore the closed-form version of (3.21) looks as follows:

$$\hat{c}^{B\cdot R} = M + H\left(\sqrt{2G(\widehat{F}(\gamma,\hat{c}^{B\cdot R}))\sigma}\right)^{(1-H)}\left(\frac{1-H}{B}\right)^{(1-H)/H} \tag{3.32}$$

where $G(x) \stackrel{def}{=} x - \frac{\frac{1}{2}log4\pi x}{1+\frac{1}{2x}}$.

The advantage of the formulae presented in this section is that in each iteration step only a closed-form expression has to be evaluated, which can be done quickly. There is no need to perform any optimization procedure any more. The price of this that the actual traffic mix has to be modelled as an fBm traffic.

Courcobetis et al in [27] present a complex traffic substitution method with the aid of which an unknown aggregated traffic can be substituted with fBm traffic at the operating point (i.e. at the optimal s and t values) of the system. For the application of this method continuous online traffic measurements should be carried out. The authors also state that if the traffic substitution is performed according to their method the Large Deviation Theory-based approximations will give the same results for both (i.e. original and substituted) traffic mixes.

3.7 Numerical investigations

In this section the behavior of the equivalent capacity formulae presented in this chapter will be studied through numerical examples. The aim of these investigations is to assess the convergence speed of the algorithms and their relative performance. Also the computed equivalent capacity values will be compared to the peak rate reservation case and the equivalent capacity values computed in the bufferless fluid flow multiplexing framework using (3.29). Furthermore the relation between the equivalent capacity, the level of multiplexing and the strictness of the loss criterion will be studied.

The numerical examples of this section can be put into two groups containing four scenarios each. The first scenario group focuses on the multiplexing of MPEG-4 video traffic, while the other scenario group deals with aggregated VoIP speech flows encoded with the ITU-T G.729 codec.

In the first scenario group 50, 100, 200 and 500 identical MPEG-4 flows were mixed together. The number of flows is denoted by N. The parameters of the MPEG-4 sources are obtained from [32]. Among the possible video traces the Jurassic park high-quality MPEG-4 trace was chosen. The analysis of this trace file showed that the peak data rate during playing this video is 3.3 Mpbs, while the average data rate of the source is 770 kbps. The buffering capacity of the queuing system was chosen in a manner that the maximum delay and delay variance caused by buffering will not exceed 100 ms. Practically it was computed as $B = M * 0.1s$, because the equivalent capacity is surely not less than the average data rate of the aggregated traffic (i.e. $M = N * 770kbps$). The σ value of the aggregated flow was calculated based on the standard deviation of one source (i.e. σ_1), using the $\sqrt{N} * \sigma_1$ formula. A rather strict QoS criterion on the WLR was set: $\gamma = 20$, i.e. a WLR in the order of magnitude of 10^{-9} is tolerable.

The equivalent capacity values were computed using (3.13) and (3.21). The iterations were continued until the relative difference (ε, see (3.11)) decreased below 1%. The calculated values for this scenario group can be observed in Table 3.1. The first six columns contain the scenario-specific input parameters of the numerical computations, while the computed values can be found in the last three columns. The last column contains the equivalent capacity value approximated in the bufferless fluid flow multiplexing framework with (3.29).

The numerical results show that the per flow capacity requirement mono-

Table 3.1: Capacity requirements of aggregated Jurassic Park MPEG-4 flows

N	M [Mbps]	N*p [Mbps]	B [MByte]	H	σ [Mbps]	$\widehat{C}_{equ}$ [Mbps]	$\widehat{C}_{equ}^{B \cdot R}$ [Mbps]	$\widetilde{C}_{equ}^{B=0}$ [Mbps]
VoD ($p = 3.3$ Mbps, $m = 0.77$ Mbps, $\gamma = 20$)								
50	38.5	165	0.481	0.795	3.17	58.833	52.398	103
100	77	330	0.963	0.774	4.484	96.481	94.663	165
200	154	660	1.925	0.737	6.341	177.880	175.561	275
500	385	1650	4.813	0.653	10.026	413.032	409.952	570

tonously decreases as the level of multiplexing (i.e. the number of sources) increases. It can also be observed that the equivalent capacity formula exploiting the B-R pre-factor produces smaller - but not essentially more precise - estimates on the $C_{equ,WLR}$. From the numerical results the benefits of applying a buffered system model is obvious. Comparing the last two columns of Table 3.1 it can be seen that significant amount (30%-50%) of capacity can be saved by using the formulae based on the buffered model.

The capacity requirements of 100 and 200 aggregated VoD flows have also been investigated as a function of γ, the prescribed WLR level. The buffer sizes were set the same way as in the previous scenarios. The γ range was set in a manner to approximately cover the interval $(10^{-6}, 10^{-9})$. The resulting graphs are depicted in Figure 3.1. Note that the calculated equivalent capacity values are shown as the fraction of the capacity that would be allocated in a peak rate reservation scheme (i.e. $N * p$).

Looking at Figure 3.1 one may observe that the relative capacity requirements monotonously increase as the WLR prescription gets stricter (i.e. γ increases). Also the formulae exploiting the B-R pre-factor produce smaller estimates for the whole range of investigated γ values. (This, however, does not essentially mean that these approximations are more precise.) Another noticeable property of the graphs is that they seem to be nearly linear. This means that the relation between the equivalent capacities and the WLR prescription $e^{-\gamma}$ is logarithmic, which implies that providing a bit more capacity for an aggregated flow has a substantial beneficial impact on the expected WLR level.

In the second scenario group 50, 100, 200 and 500 identical G.729 coded VoIP speech flows were mixed together. The parameters of the sources were obtained from real traffic traces. The analysis of the trace files showed that the

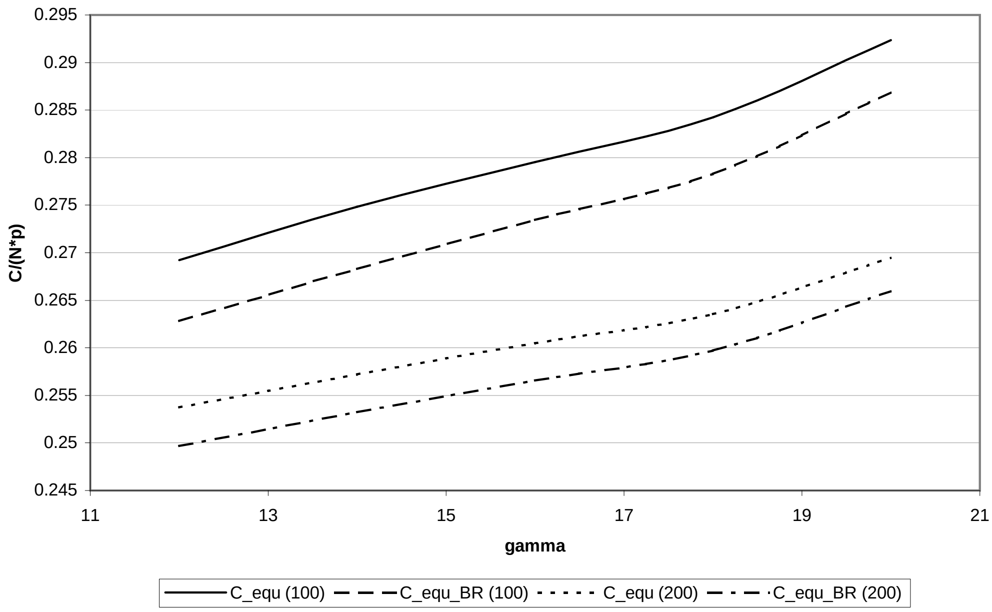

Figure 3.1: Capacity requirements of aggregated Jurassic Park MPEG-4 flows relative to peak rate reservation

peak data rate during a G.729 coded VoIP speech is 24 kbps, while the average data rate of the source is 20 kbps. The buffering capacity of the queuing system was chosen in a manner that the maximum delay and delay variance caused by buffering will not exceed 50 ms. This means that the buffer size was computed as $B = M * 0.05s$, where $M = N * 770kbps$ is the aggregated mean rate. The σ value of the aggregated flow was calculated using the $\sqrt{N} * \sigma_1$ formula, where σ_1 is the standard deviation of a single source obtained from processing the trace files. A rather strict QoS criterion on the WLR was set, $\gamma = 20$, i.e. a WLR in the order of magnitude of 10^{-9} is tolerable.

Here we note that the strict WLR criterion with the very small buffer and low level of multiplexing caused our algorithms giving inaccurate approximations on the equivalent capacity. Therefore the criteria were loosened in those problematic scenarios by setting $\gamma = 12$ (i.e. approximately 10^{-6} WLR). The equivalent capacity estimations where the loosened WLR criterion was applied are printed with italic fonts in Table 3.2. In one case even this loosened criteria was not enough for our formulae to work properly, so the corresponding field of Table 3.2 have been filled with "n/a". The explanation of these phenomenon (i.e. our formulae failing to give reasonable approximation) is that

Table 3.2: Capacity requirements of aggregated G.729 VoIP flows

N	M [Mbps]	N*p [Mbps]	B [kByte]	H	σ [kbps]	$\widehat{C}_{equ}$ [Mbps]	$\widehat{C}_{equ}^{B\cdot R}$ [Mbps]	$\widetilde{C}_{equ}^{B=0}$ [Mbps]
VoIP ($p = 24$ kbps, $m = 20$ kbps, $\gamma = 20$)								
50	1	1.2	6.25	0.81	63.24	*n/a*	*1.175*	1.189
100	2	2.4	12.5	0.789	89.44	*2.249*	*2.201*	2.29
200	4	4.8	25	0.793	126.491	4.498	4.443	4.59
500	10	12	62.5	0.778	200	10.678	10.6	10.985

the real value of the equivalent capacity is so close to the aggregate peak rate that our formulae - due to the applied underlying conservative approximations - overestimate the equivalent capacity and thus give a required capacity value greater than the aggregate peak. This is of course not a valid result, but the approximation given by our formulae is still quite accurate as it stays close to the aggregate peak.

The equivalent capacity values were computed using (3.13) and (3.21). The iterations were continued until the relative difference decreased below 1%. The calculated values for this scenario group are shown in Table 3.2. The first six columns contain the scenario-specific input parameters of the numerical computations, while the computed values can be found in the last three columns. The last column contain the equivalent capacity value approximated in the bufferless fluid flow multiplexing framework with (3.29).

It can be seen that the per flow capacity requirement monotonously decreases as the level of multiplexing (i.e. the number of sources) increases. It is also noticeable that the equivalent capacity formula exploiting the B-R pre-factor produces smaller - but not necessarily more precise - estimates on the $C_{equ,WLR}$ than the other estimator building solely upon the basic many sources asymptotic equality. The numerical results show the benefit of buffering, however it is much less significant as it was for the bursty VoD traffic. Comparing the last two columns of Table 3.1 it can be seen that a moderate amount (1%-4%) of capacity can be saved by using the formulae designed for the buffered model.

The capacity requirements of 100 and 200 aggregated VoIP flows was also investigated as a function of γ, the prescribed WLR level. The buffer sizes were set the same way as in the previous VoIP scenarios. The γ range was set in a manner to approximately cover the interval $(10^{-6}, 10^{-9})$. The resulting

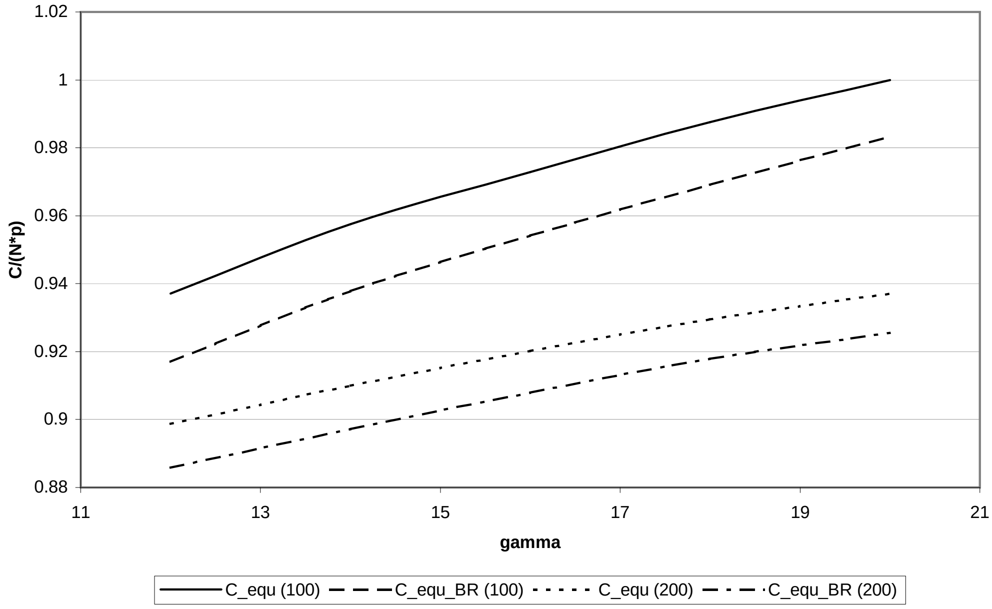

Figure 3.2: Capacity requirements of aggregated G.729 VoIP flows relative to peak rate reservation

graphs are depicted in Figure 3.2. Note that the calculated equivalent capacity values are shown as the fraction of the capacity that would be allocated in a peak rate reservation scheme.

Looking at the curves one may observe that the relative capacity requirements monotonously increase as the WLR prescription gets stricter (i.e. γ increases). Also the formulae exploiting the B-R pre-factor produce smaller estimates for the whole range of investigated γ values. Another noticeable property of the graphs is that they seem to be nearly linear. This means that the relation between the equivalent capacities and the WLR prescription $e^{-\gamma}$ is logarithmic, which implies that providing a bit more capacity for a G.729 coded VoIP flow aggregate has magnified beneficial impact on the expected WLR level.

As regards the performance of the iterative formulae the iteration steps (for both scenario group) required to fulfil the $\varepsilon \leq 1\%$ stop condition never exceeded two steps, i.e. the relative difference of the equivalent capacity approximations obtained in the first and the second cycle was always less than 1%. This means that the convergence of our method was very fast for these scenarios, and even the first iterations gave accurate results. In the first iteration we always used $\hat{c}_0 = M + \frac{N*p-M}{2}$, but also experimented with other starting values from the interval (M,N*p) and experienced similarly fast convergence.

Chapter 4

Novel Buffer Requirement Estimation Techniques

Capacity dimensioning techniques are not always the sole, appropriate means of ensuring proper transmission quality for premium traffic. In many cases it is not possible or economic (e.g. in radio networks) to handle the increase in traffic demand only by the expansion of physical transmission capacity. More often - especially in case of bursty traffic sources - considerable transmission capacity can be saved by applying a larger buffer instead of raising the transmission rate.

Naturally, the buffer size enlargement has its own limitations: implementing a buffer of appropriate size and speed may have its financial and/or technological limits, but - and that is more important - the enlarged buffer also means higher transmission delay that may eventually result in unacceptable degradation of service quality. This problem becomes especially grave in case of delay-sensitive, value-added applications such as a VoIP or VoD service. These limitations have to be considered upon deciding which parameter (i.e. transmission rate or buffer capacity) of the queuing system is to be tweaked and to what extent that parameter can be changed.

The queue length distributions of various queuing systems have been extensively studied in the literature. In [24] the authors investigate the connection between cell loss probability and output buffer size in ATM switches. The book of Appenzeller et al [25] takes a more abstract, more general approach to this problem; they summarize the main contributions obtained in the field of queue length distribution characterization applying the results of the Large Deviation Theory. In the celebrated paper of Ganesh et al [26] the authors focus on formulating new rules of thumb for sizing router buffers fed by many congestion-controlled (basically TCP) flows.

In this chapter techniques capable of approximating the minimal buffer space of a queuing system with fixed transmission rate, that is needed to fulfil the prescribed quality level composed in terms of the workload loss ratio will be presented. The first approximation method is based on (2.2), the MSA equality discussed in Section 2.1 and the results of Seres et al [3] and Shroff et al [6]. The second result is an easily evaluable upper bound of the measure computed with the first method and it builds upon the result of Borsos and György [4]. The third contribution incorporates the Bahadur-Rao pre-factor (see Section 2.2) in the underlying QoS measure approximation and thus it may yield a more precise result than the previous formulae, especially in case when the aggregated arrival process is Gaussian.

The chapter is organized as follows. In the next section the main terms and definitions that will be used throughout this chapter are going to be introduced briefly. In Section 4.2 a theoretical buffer requirement estimator method will be introduced. After that a more feasible buffer size approximation method is going to be shown, which yields an upper bound on the theoretical result. In Section 4.4 the buffer space estimation methods incorporating the Bahadur-Rao improvement will be presented. In Section 4.5 we are going to show that for fBm traffic the buffer requirement approximation formulae manifest in closed-form. The chapter is concluded with numerical investigations.

4.1 Terms and Definitions

Let the stochastic process $X[0,t)$ denote the total amount of workload arriving in the time interval $[0,t)$ from N independent flows at a buffered communication link with buffer size B and transmission capacity C. N is regarded as a scaling factor and in that sense we can identify a per-source buffer $b = \frac{B}{N}$ and per source capacity $c = \frac{C}{N}$. Let us also assume that $X[0,t)$ has stationary increments.

The buffer overflow probability of this finite buffer system ($P(Q > B)$) can be deduced from the proportion of time over which the queue length $Q(C,N)$ is above level B in a queue of infinite buffer. In the MSA regime the decay rate of the logarithm of the buffer overflow probability is asymptotically linear in the number of sources N in a system where they are kept constant [16] [17]:

$$\lim_{N\to\infty} \frac{1}{N} logP(Q(N,cN) > bN) = \sup_{t>0} \inf_{s>0} \left\{ \frac{\Lambda(s,t)}{N} - s(b+ct) \right\} \stackrel{def}{=} -I, \quad (4.1)$$

where $\Lambda(s,t) \stackrel{def}{=} logE[e^{sX[0,t)}]$ is the logarithmic moment generating function of $X[0,t)$, s and t are free parameters.

The practical consequence of the above equation is that for large N one may approximate the buffer overflow probability simply as

$$P(Q(N,C) > B) \approx e^{-NI} \quad (4.2)$$

where $-NI$ can be computed as

$$-NI = \sup_{t>0} \inf_{s>0} \{\Lambda(s,t) - s(B+Ct)\} \quad (4.3)$$

The QoS constraint on the workload loss can be expressed formally as:

$$WLR \stackrel{def}{=} \frac{E[Q-B]^+}{E[X]} \leq e^{-\gamma} \quad (4.4)$$

where X is the random variable characterizing the amount of workload arriving in a time unit.

The minimal buffer space requirement - i.e. the minimal buffer size at which the prescribed WLR threshold is still not exceeded - is denoted by $B_{req,WLR}$ and can be defined as:

$$B_{req,WLR} \stackrel{def}{=} \inf \left\{ B : WLR \leq e^{-\gamma} \right\} \quad (4.5)$$

4.2 A Theoretical Buffer Requirement Estimation Method

In this section a buffer requirement calculation method that applies an underlying approximation of the expected buffer saturation probability in the many sources asymptotic regime described by the Large Deviation Theory is going to be introduced. The new method to be presented manifests in the same, explicit form as the similar formulae of Seres et al [3], but is capable to estimate the buffer requirement of the queuing system in case the QoS constraint is composed in terms of the WLR instead of the buffer saturation probability.

Theorem 4.2.1. *Let the stochastic process $X[0,t)$ denote the total amount of workload arriving in the time interval $[0,t)$ from N independent flows at a buffered communication link with transmission capacity C. Let us also assume that $X[0,t)$ has stationary increments. In this queuing system the minimum buffer space that enures that the expected WLR will not exceed the prescribed threshold ($e^{-\gamma}$) can be approximated with the following formula:*

$$B_{req,WLR} \approx sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + F(\gamma, C)}{s} - Ct \right\} \stackrel{def}{=} \widetilde{B}_{req,WLR}. \qquad (4.6)$$

where

$$F(\gamma, C) = \gamma - logP\{Q > 0\} + logWLR(0).$$

Proof: Kim and Shroff observed the following connection between the buffer saturation probability and workload loss ratio [6]:

$$WLR(B) = \frac{WLR(0)}{P(Q > 0)} P(Q > B) \qquad (4.7)$$

This implies that the prescription $WLR(B) \leq e^{-\gamma}$ can be transformed into the following equivalent prescription on BOP :

$$P(Q > B) \leq e^{-\gamma + logP(Q>0) - logWLR(0)} \qquad (4.8)$$

Defining $F(\gamma, C) \stackrel{def}{=} \gamma + logP(Q > 0) - logWLR(0)$ (4.8) can be written as

$$P(Q > B) \leq e^{-F(\gamma, C)} \qquad (4.9)$$

According to Seres et al the minimal buffer space that ensures the fulfilment of the target BOP level, i.e. $P(Q > B) \leq e^{-\delta}$ manifests in the following form [3]:

$$B_{req,BOP} \approx sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + \delta}{s} - Ct \right\} \qquad (4.10)$$

Using $\delta = F(\gamma, C)$ in (4.10) the statement of the theorem follows. Q.E.D.

For the evaluation of (4.6) a double optimization has to be carried out with respect to the free parameters of s and t. This method is equivalent (i.e. it yields the same result) with the the so-called implicit method, where the buffer requirement is sought based on its definition presented in (4.5). By following the implicit method, however, a triple optimization should be carried out as the expected WLR has to be approximated through solving a two-dimensional optimization task in each iteration step. As a conclusion the explicit buffer requirement estimator presented in (4.6) is computationally more feasible than the implicit estimation technique.

4.3 A Practical Buffer Requirement Estimation Technique

Finding a tractable estimate on $P(Q > 0)$ for the evaluation of (4.6) may be a problem. Therefore another method that eliminates the need for obtaining $P(Q > 0)$ while still providing a reasonable and computationally tractable approximation on the buffer requirement of the queuing system have been proposed. The main idea of this method is that $P(Q > 0)$ is substituted by its lower bound, $P(X > C)$ which results in a formula that gives a conservative upper bound on the buffer requirement estimated with (4.6).

Theorem 4.3.1. *A reasonable upper bound can be obtained on the buffer space requirement approximation presented in (4.6) using the following formula:*

$$\widetilde{B}_{req,WLR} \leq sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + \widehat{F}(\gamma, C)}{s} - Ct \right\} \stackrel{def}{=} \widehat{B}_{req,WLR}. \quad (4.11)$$

where

$$\widehat{F}(\gamma, C) = \gamma - logP\{X > C\} + logWLR(0).$$

Proof: According to Borsos and György [4]

$$WLR(B) \approx \frac{WLR(0)}{P(X > C)} P(Q > B) \quad (4.12)$$

Regarding (4.7) and that $P(X > C) \leq P(Q > 0)$ we get

$$WLR(B) \leq \frac{WLR(0)}{P(X > C)} P(Q > B) \quad (4.13)$$

Combining (4.13) with the original prescription $WLR(B) \leq e^{-\gamma}$ the following stricter prescription can be formulated:

$$\frac{WLR(0)}{P(X > C)} P(Q > B) \leq e^{-\gamma}, \quad (4.14)$$

Prescription (4.14) can be transformed into the following equivalent form:

$$P(Q > B) \leq e^{-\gamma + logP(X>C) - logWLR(0)} \quad (4.15)$$

Let us define $\widehat{F}(\gamma, C) \stackrel{def}{=} \gamma + logP(X > C) - logWLR(0)$ and rewrite (4.15) to the following form

$$P(Q > B) \leq e^{-\widehat{F}(\gamma,C)} \quad (4.16)$$

According to Seres et al the minimal buffer space that ensures the fulfilment of the target BOP level, i.e. $P(Q > B) \leq e^{-\delta}$ manifests in the form presented in (4.10). Using $\delta = \widehat{F}(\gamma, C)$ in (4.10) the statement of the theorem follows. Q.E.D.

For the evaluation of the buffer requirement estimator presented in (4.11) the bufferless saturation probability (i.e. $P(X > C)$) needs to be obtained. This measure can be approximated much easier than $P(Q > 0)$, which makes this method easier to be applied in practice. There are numerous, computationally feasible methods to obtain $P(X > C)$, while $P(Q > 0)$ can easily be calculated only under special circumstances.

4.4 Modified Buffer Requirement Estimation Methods

All previously presented methods use an underlying $P(Q > B)$ estimation directly rooted in the basic MSA equality. This WLR estimation can however be further improved by applying the Bahadur-Rao theorem.

In this section two buffer requirement estimators that apply improved underlying $P(Q > B)$ approximation techniques that incorporate the Bahadur-Rao pre-factor will be presented. Using the B-R pre-factor may yield more precise estimation on $P(Q > B)$ and as a consequence a more precise estimation on the buffer requirement.

An important property of the formulae to be presented is that they still manifest in the same, simple, explicit forms as the other ones already introduced in this chapter, and therefore their computational complexity remains in the same order of magnitude as of those.

Theorem 4.4.1. *The underlying MSA approximation applied in the previous buffer space requirement estimation formulae (4.6) and (4.11) can be refined by applying the Bahadur-Rao theorem. The formulae manifest in the following forms:*

$$B_{req,WLR} \approx sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + G(F(\gamma, C))}{s} - Ct \right\} \stackrel{def}{=} \widetilde{B}^{B-R}_{req,WLR}, \tag{4.17}$$

and

$$\widetilde{B}^{B\cdot R}_{req,WLR} \leq sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + G(\widehat{F}(\gamma, C))}{s} - Ct \right\} \stackrel{def}{=} \widehat{B}^{B\cdot R}_{req,WLR}. \tag{4.18}$$

where $G(x) \stackrel{def}{=} x - \frac{\frac{1}{2}log4\pi x}{1+\frac{1}{2x}}$

Proof: In case the following approximation is used to estimate the BOP

$$P(Q(N,C) > B) \approx \frac{1}{\sqrt{4\pi NI}} e^{-NI} \tag{4.19}$$

instead of (4.2), then - according to Seres et al - the buffer space requirement, $B_{req,BOP}$ for QoS constraint $P(Q > B) \leq e^{-\delta}$ manifests in the following form

$$B_{req,BOP} \approx sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda(s,t) + \delta^{B\cdot R}}{s} - Ct \right\} \tag{4.20}$$

where

$$\delta^{B\cdot R} \stackrel{def}{=} \delta - \frac{\frac{1}{2}log(4\pi\delta)}{1 + \frac{1}{2\delta}} \tag{4.21}$$

Combining the $P(Q > B) \leq e^{-\delta}$ prescription (used as starting point by Seres et al) and the prescriptions composed in (4.8) and (4.15) the statements of the theorem follow if the substitution $\delta = F(\gamma, C)$ or $\delta = \widehat{F}(\gamma, C)$ is applied. Q.E.D.

It can be observed that Formulae (4.17) and (4.18) - while in theory being more precise than the buffer requirement estimation formulae (4.6) and (4.11) - manifest in a very similar form as those. This also implies that the computational complexity of these formulae is very close to each other. This advantageous property has a price, however. To be able to incorporate the original B-R pre-factor into the buffer saturation probability estimation a second-order approximation had to be carried out resulting in (4.19). This second-order approximation is only precise in case the arrival process is Gaussian. Therefore the formulae presented in this section should primarily be used in case the aggregated arrival process is well-characterized by a Gaussian process, for example a fractional Brownian motion.

4.5 Closed-form solutions

The buffer requirement estimation methods presented in this chapter require the solution of a double-optimization task. This can be computationally demanding and has a negative effect on the speed of any (e.g. an admission control) algorithm using these formulae. However, this double optimization can be avoided if the aggregated traffic is substituted or modelled with fractional Brownian motion (fBm).

In this section the closed-form versions of the previously introduced buffer requirement estimator formulae that are applicable for fBm modeled aggregated traffic are going to be shown. In Section 3.6 the main characteristics of fractional Brownian traffic have already been introduced, but for convenience of the reader we repeat that here.

The stochastic process $[Z_t, t \in \Re]$ is a normalised fBm with self-similarity (or Hurst-) parameter $H \in (0,1)$, if it has stationary increments and continuous paths, $Z_0 = 0$, $E[Z_t] = 0$, $Var[Z_t] = |t|^{2H}$ and if Z_t is a Gaussian process. Let us define the process $X[0,t) \stackrel{def}{=} mt + Z_t$ for $t > 0$. It is a fractional Brownian traffic and can be regarded as the the amount of traffic offered to a multiplexer in time interval $[0,t)$.

Using this model the cumulant generating function manifests in the following form: $\Lambda(s,t) = stm + (1/2)s^2\sigma^2 t^{2H}$. Substituting the cumulant generating function of the fBm traffic into (4.11) and performing the optimization we get the following closed-form expression for the buffer requirement:

$$\widehat{B}_{req,WLR} = \left(\frac{H}{C-M}\right)^{H/(1-H)} \left(\sqrt{2\widehat{F}(\gamma,C)}\sigma\right)^{1/(1-H)} (1-H), \qquad (4.22)$$

where $\widehat{F}(\gamma,C) = \gamma - logP\{X > C\} + logWLR(0)$. It was shown in Section 2.5, that $\widehat{F}(\gamma,C)$ can simply be computed using (3.28) and (3.29) approximations.

The Bahadur-Rao version of the above presented closed-form equation manifests in almost the same form, as the presence of the B-R prefactor does not affect the optimization procedure. Therefore the closed-form version of (4.18) looks the following:

$$\widehat{B}^{B \cdot R}_{req,WLR} = \left(\frac{H}{C-M}\right)^{H/(1-H)} \left(\sqrt{2G(\widehat{F}(\gamma,C))}\sigma\right)^{1/(1-H)} (1-H), \quad (4.23)$$

where where $G(x) \stackrel{def}{=} x - \frac{\frac{1}{2}log4\pi x}{1+\frac{1}{2x}}$.

The advantage of the formulae presented in this section is that only a closed-form expression has to be evaluated, which can be done very fast. There is no need to perform any optimization procedure any more. The price of this that the actual traffic mix has to be modelled as an fBm traffic.

Courcobetis et al in [27] present a complex traffic substitution method with the aid of which an unknown aggregated traffic can be substituted with fBm traffic at the operating point (i.e. at the optimal s and t values) of the system. For the application of this method continuous online traffic measurements should be carried out. The authors also state that if the traffic substitution is performed according to their method the Large Deviation Theory-based approximations will give the same results for both (i.e. original and substituted) traffic mixes.

4.6 Numerical investigations

In this section the operation of the buffer requirement estimators presented in this chapter will be investigated through numerical methods. The aim of these experimentations is to compare the performance of the formulae applying the Bahadur-Rao improvement and the formulae relying upon the basic many sources asymptotic equation. The relation between the buffer requirement, the level of multiplexing and the strictness of the loss criterion is also of interest and will be studied.

The numerical examples to be presented in this section can be put into two groups containing four scenarios each. The first scenario group focuses on the multiplexing of video (namely MPEG-4) traffic, while the other scenario group deals with aggregated VoIP speech applying ITU-T G.729 codec.

In the first scenario group 50, 100, 200 and 500 identical MPEG-4 flows were mixed together. The number of flows is denoted by N. The parameters of the MPEG-4 sources are obtained from [32]. Among the video traces the Jurassic park high-quality MPEG-4 trace was picked. According to the trace file the peak data rate during the playback of this video is 3.3 Mbps, while the average data rate of the source is 770 kbps. The service rate of the queuing system was chosen in a manner that the per-flow capacity remains 900 kbps. The σ value of the aggregated flow was calculated based on the standard deviation of one source (σ_1), which was obtained from [32]) using the $\sqrt{N} * \sigma_1$ formula. A rather strict QoS criterion on the WLR was set, $\gamma = 20$, i.e. a WLR in the order of magnitude of 10^{-9} is tolerable.

The buffer requirement values were computed using (4.11) and (4.18). The calculated values for this scenario group can be observed in Table 4.1. The

Table 4.1: Buffer requirements of aggregated Jurassic Park MPEG-4 flows

N	M [Mbps]	n*p [Mbps]	C [Mbps]	H	σ [Mbps]	$\widehat{B}_{req}$ [Mbyte]	$\widehat{B}_{req}^{B\cdot R}$ [Mbyte]	delay [ms]
VoD ($p = 3.3$ Mbps, $m = 0.77$ Mbps, $\gamma = 20$)								
50	38.5	165	45	0.795	3.17	12.249	8.630	1530
100	77	330	90	0.774	4.484	4.014	2.846	253
200	154	660	180	0.737	6.341	1.500	1.105	49
500	385	1650	450	0.653	10.026	0.916	0.719	12

first six columns contain the scenario-specific input parameters of the numerical computations, while the computed values can be found in the last three columns. The last column contains the maximum delay value caused by the buffer of size $\widehat{B}_{req}^{B\cdot R}$.

It can be seen that the buffer requirement monotonously decreases as the level of multiplexing (i.e. the number of sources) increases and the per-flow capacity is kept constant. It can also be observed that the buffer requirement formula exploiting the B-R pre-factor produces smaller - but not necessarily more precise - estimates on the $B_{req,WLR}$. Taking a close look at the last column it can be observed that 900 kbps per flow capacity is not enough in case the level of multiplexing is low, because the maximum delay caused by the buffer may reach 1.5 seconds. Obviously, this high delay and possibly high delay variation is undesirable for this kind of streaming multimedia application, and for lower flow aggregation levels a higher per-flow capacity should be allocated.

The buffer size requirements of 100 and 200 aggregated VoD flows were also investigated as a function of γ, the prescribed WLR level. The transmission capacity values were set the same way as in the previous scenarios. The γ range was set in a manner to approximately cover the interval $(10^{-6}, 10^{-9})$. The resulting graphs are depicted in Figure 4.1.

Looking at the figures one may observe that the buffer requirements monotonously increase as the WLR prescription gets stricter (i.e. γ increases). Also the formulae exploiting the B-R pre-factor produce smaller estimates for the whole range of investigated γ values. Another noticeable property of the graphs is that they are almost linear, which in this case means that the relation between the buffer requirement and the WLR prescription (i.e. $e^{-\gamma}$) is logarithmic. The consequence of this behavior is that the WLR level can be

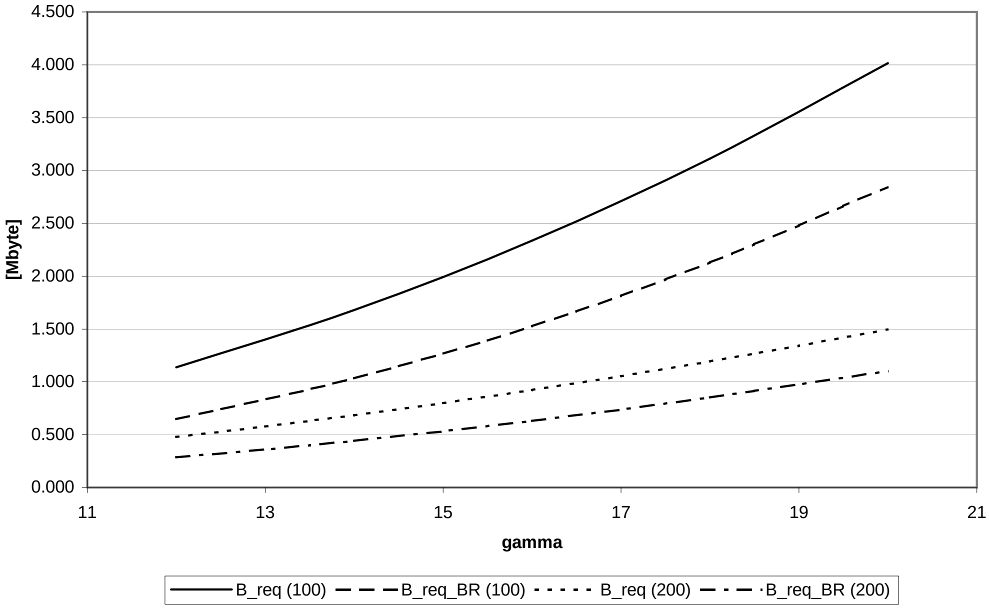

Figure 4.1: Buffer requirements of aggregated Jurassic Park MPEG-4 flows

positively affected easily by providing a bit more buffering space for a certain traffic aggregate, however one must also keep in mind the drawbacks of larger buffer, i.e. possibly larger delay and delay variation.

In the second scenario group 50, 100, 200 and 500 identical G.729 coded VoIP speech flows were mixed together. The parameters of the sources were obtained from traffic traces. According to the traces the peak data rate of a G.729 coded VoIP speech is 24 kbps, while the average data rate of the source is 20 kbps. The per-flow capacity was kept constant again, in this scenarios it was set to 22 kbps. The σ value of the aggregated flow was calculated using the $\sqrt{N} * \sigma_1$ formula, where σ_1 is the standard deviation of a single source obtained from the trace files. A rather strict QoS criterion on the WLR was set, $\gamma = 20$, i.e. a WLR in the order of magnitude of 10^{-9} is tolerable.

The buffer requirements were computed using (4.11) and (4.18). The calculated values for this scenario group can be observed in Table 4.2. The first six columns contain the scenario-specific input parameters of the numerical computations, while the computed values can be found in the last three columns. The last column contains the maximum delay value caused by the buffer of size $\widehat{B}_{req}^{B \cdot R}$.

It can be seen that the buffer requirement monotonously decreases as the

Table 4.2: Buffer requirements of aggregated G.729 VoIP flows

N	M [Mbps]	n*p [Mbps]	C [Mbps]	H	σ [kbps]	$\widehat{B}_{req}$ [kbyte]	$\widehat{B}_{req}^{B\cdot R}$ [kbyte]	delay [ms]
VoIP ($p = 24$ kbps, $m = 20$ kbps, $\gamma = 20$)								
50	1	1.2	1.1	0.81	63.24	860.875	549.5	3996.36
100	2	2.4	2.2	0.789	89.44	172	113.125	411.36
200	4	4.8	4.4	0.793	126.491	61.625	39.5	71.82
500	10	12	10.5	0.778	200	15.25	9.875	7.52

level of multiplexing (i.e. the number of sources) increases and the per-flow capacity is kept constant. It can also be observed that the buffer requirement formula exploiting the B-R pre-factor produces smaller - but not necessarily more precise - estimates on the $B_{req,WLR}$. Looking thoroughly at the last column it can be observed that 22 kbps per flow capacity is not enough in case the level of multiplexing is low, because the maximum delay caused by the buffer may reach even 4 seconds. Obviously, this high delay and possibly high delay variation is unacceptable for this kind of real-time interactive application, and for lower flow aggregation levels a higher per-flow capacity should be allocated.

The buffer requirements of 100 and 200 aggregated VoIP flows were also investigated as a function of γ, the prescribed WLR level. The transmission capacities were set the same way as in the previous VoIP scenarios. The γ range was set in a manner to approximately cover the interval $(10^{-6}, 10^{-9})$. The resulting graphs are depicted in Figure 4.2.

Looking at the figures one may observe that the buffer requirements monotonously increase as the WLR prescription gets stricter (i.e. γ increases). Also the formulae exploiting the B-R pre-factor produce smaller estimates for the whole range of investigated γ values. Another noticeable property of the graphs is that they are close to linear, which in this case means that the relation between the buffer requirement and the WLR prescription ($e^{-\gamma}$) is logarithmic. Consequently the WLR level can easily be reduced by providing more buffering space.

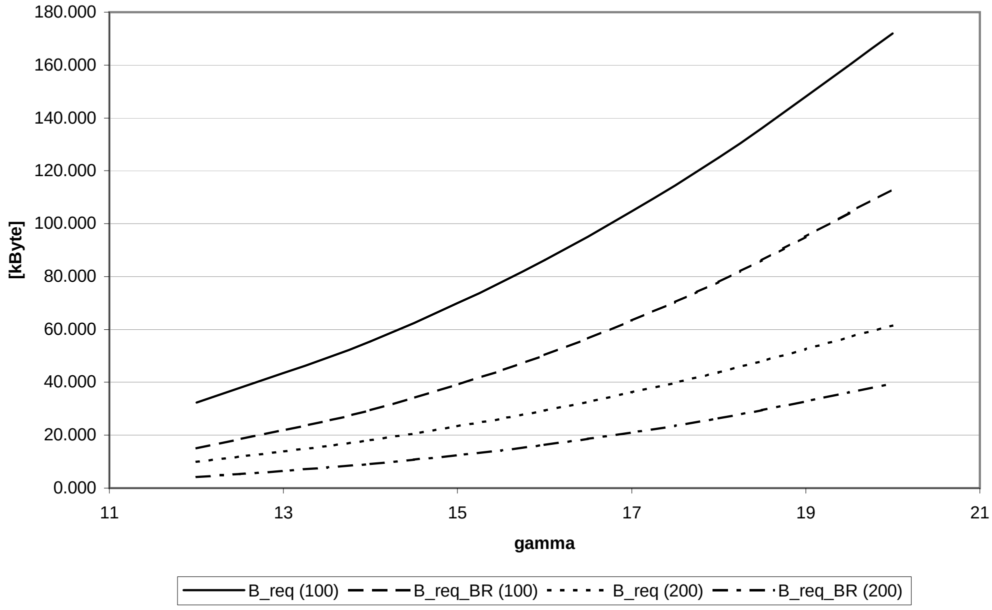

Figure 4.2: Buffer requirements of aggregated G.729 VoIP flows

Chapter 5

Application of the Presented Results

The results presented in the previous chapters can be applied to solve capacity planning and buffer dimensioning problems in case the QoS requirement of the network flows is composed in terms of the workload loss ratio. Real applications, however, compose their QoS needs not only in terms of loss, but often also in terms of delay and/or delay variation. This statement is especially true in case of value-added, multimedia services such as a Video-on-Demand (VoD) service, which is becoming popular nowadays. This implies that more sophisticated methods - that are capable of taking the diverse QoS needs of traffic flows into consideration - should be applied for real network dimensioning activities.

Another important aspect worth to be considered upon dimensioning is the call-level behavior of the expected traffic, i.e. the understanding of the birth-death process of network flows. Taking this kind of traffic dynamics into account makes the whole dimensioning process more precise and economical and also guarantees that the service availability will reach the desired level.

Dimensioning methods found in the literature usually investigate traffic demand on only one, distinct granularity level. For instance papers and books dealing with the dimensioning issues of telephony (or in general circuit switched) networks focus on the call-level granularity level and aim to formulate bounds on the expected availability measure, often on the time blocking probability (TBP). Famous basic results in this field are attached to the names of Erlang, Engset and Kleinrock [9][10][11].

Another group of researchers focus on the inner dynamics of traffic flows and investigate traffic behavior on the packet granularity level. Their aim usually is to obtain bounds or approximations on the expected loss occurring in the system under study. Celebrated papers dealing with this topic can be coupled with the work of Kelly, Courcobetis and Roberts just to name a few pioneers in this research area [7][14][8].

Characterizing the expected delay perceived by traffic flows usually requires yet another approach. To characterize this measure often the famous GPS model is used [19]. Many papers extend the scope of the GPS model with the aid of network calculus and aim to determine the end-to-end delay of the traffic flows traversing the network under investigation. Pioneer papers and books was written on this topic by Le Boudec [21], Cruz [22] and more recently Fidler [23].

In this chapter a combined dimensioning method aiming to ensure simultaneous call- and packet level QoS guarantees is going to be shown. This technique is capable of providing guaranteed availability composed in terms of time blocking probability, and at the same time attaining the desired loss level composed in terms of workload loss ratio. Also this method takes the maximum tolerable delay into consideration during the dimensioning process, and thus it is truly a capable tool to provide multi-level QoS guarantees in packet-switched networks.

Based on the combined dimensioning technique we are also going to introduce a QoS architecture designed for being implemented in broadband access networks. The basic building blocks, exploited protocols and the recommended admission control scheme will also be thoroughly discussed in this chapter.

The forthcoming sections of this chapter are organized as follows. In the next section our dimensioning algorithm that aims to provide both call-level and packet-level QoS guarantees are going to be explained. Then the QoS architecture that greatly builds upon this dimensioning technique will be presented.

5.1 A Combined Resource Dimensioning Method Ensuring Multi-level QoS Guarantees

In this section a combined resource dimensioning method that is capable of providing availability and loss guarantees simultaneously when coupled with a simple admission control mechanism is going to be introduced. This technique

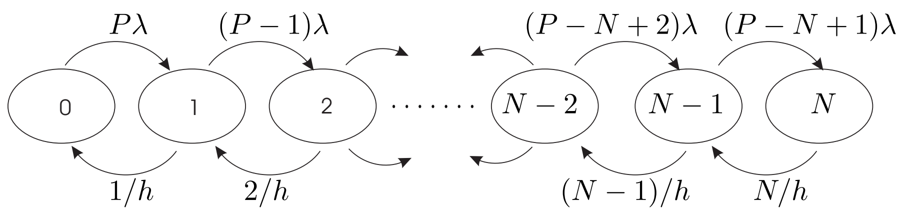

Figure 5.1: The applied call-level (Engset) model

can be further improved to ensure delay requirements as well by setting the buffer size of the packet-server node appropriately. The dimensioning method to be presented here differs mainly from others found in the literature in the sense that it aims at providing simultaneous quality guarantees composed in terms of QoS measures belonging to different traffic granularity levels.

The main idea of the resource dimensioning technique to be shown is that certain dimensioning methods providing call-level, availability guarantees are combined with other methods designed to ensure packet-level loss-type QoS measures. Many formulae were designed to calculate the expected availabilty of a queuing system, the main contributions amongs these are very well summarized in the book of Kleinrock [11]. From the many possible call-level methods the one that seemed to be the most suitable for applying to practical cases have been chosen.

The blocking probability formula that have been chosen is attached to the name of Engset [10], a a Norwegian mathematician and engineer who did pioneering work in the field of telephone traffic queuing theory. He developed the Engset formula in 1915 before the breakthroughs of A. K. Erlang (1917). The main difference between the Engset and the more famous Erlang formulae is that Erlang supposed that the population of telephony subscribers is infinite, while Engset worked with finite populations in his model. The state diagram with corresponding transition probabilities of the Engset model can be described with the Markovian chain depicted in Figure 5.1.

In the Engset model N denotes the server capacity (originally the number of line cards or phone lines), P denotes the size of the population (i.e. telephony subscribers attached to the exchange), λ is the service (or call-) initiation intensity and h is the holding time of a call. The system accepts calls when the number of ongoing calls is less than N, and rejects every call that arrives during the period when all linecards are busy.

The measure of importance in this model is the time blocking probability, i.e. the fraction of time when all server resources (i.e. linecards) are occupied, or - in other words - the steady-state probability of call rejection due to lack of free resources. Let this measure denoted by TBP. It can be computed with the following (Engset) formula:

$$TBP = \frac{\begin{pmatrix} P \\ N \end{pmatrix} (\lambda h)^N}{\sum_{j=0}^{N} \begin{pmatrix} P \\ j \end{pmatrix} (\lambda h)^j}. \tag{5.1}$$

Simply spoken Formula (5.1) tells us that if the population size is P and calls arrive with intensity λ and lifetime h to the system that is capable to handle maximum N simultaneous flows (or calls), then we should expect that arriving calls will be rejected in TBP fraction of time.

The problem depicted above can be approached from the opposite direction as well: Formula (5.1) may be used to indirectly obtain the minimal server capacity (N^*) that ensures that the TBP level will not exceed a certain threshold (β). Formally written as:

$$N^* = inf \left\{ N : \frac{\begin{pmatrix} P \\ N \end{pmatrix} (\lambda h)^N}{\sum_{j=0}^{N} \begin{pmatrix} P \\ j \end{pmatrix} (\lambda h)^j} \leq \beta \right\}. \tag{5.2}$$

By combining the above-discussed Engset model with the resource requirement assessment methods introduced in Chapter 3 and Chapter 4, one may construct a dimensioning method that can account for both availability and loss-type QoS prescriptions. This combined dimensioning technique is composed formally in the following theorems.

First define the combined equivalent capacity (C^*) as the minimum transmission capacity (for a given buffer size) that ensures that both the expected WLR level remains under a certain threshold $e^{-\gamma}$ and the TBP level does not exceed β.

$$C^* \stackrel{def}{=} inf \left\{ C : WLR \leq e^{-\gamma}, TBP \leq \beta \right\} \tag{5.3}$$

Similarly, define the combined buffer requirement (B^*) as the minimum buffer space (for a given transmission capacity) that ensures that both the expected WLR level remains under a certain threshold $e^{-\gamma}$ and the TBP level does not exceed β.

$$B^* \stackrel{def}{=} inf \left\{ B : WLR \leq e^{-\gamma}, TBP \leq \beta \right\} \tag{5.4}$$

Theorem 5.1.1. *The minimal transmission rate C^* that ensures that neither the expected WLR nor the TBP will exceed the prescribed thresholds $e^{-\gamma}$ and β, respectively, can be approximated by the solution of the fix-point equation*

$$c^* = sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda_{N^*}(s,t) + K}{st} - \frac{B}{t} \right\} \tag{5.5}$$

with respect to c^*, where B is the buffer size, s and t are free parameters, $K \in \{F(\gamma, c^*), \hat{F}(\gamma, c^*), G(F(\gamma, c^*)), G(\hat{F}(\gamma, c^*))\}$ and $\Lambda_{N^*}(s,t)$ is the cumulant generating function of $X_{N^*}(0,t]$, the random variable denoting the workload arriving from N^* sources in time interval $(0,t]$. The number of sources (N^*) should be computed with (5.2)

Proof: If a transmission channel admitting maximum N^* simultaneous flows is provided then according to Formula (5.2) the TBP level will not exceed β by definition. On the other hand, according to Theorems 3.2.1, 3.3.1 and 3.4.1, N^* simultaneous flows require at least c_n^*, $n \to \infty$ transmission capacity in order to keep the WLR level below the $e^{-\gamma}$ threshold, where c_n^*, n positive integer denotes the result of (5.5) in the nth iteration step. Q.E.D.

A similar theorem can be composed for the combined buffer requirement, B^*.

Theorem 5.1.2. *The minimal buffer space B^* that enures that neither the expected WLR nor the TBP will exceed the prescribed thresholds $e^{-\gamma}$ and β, respectively, can be approximated by*

$$B^* \approx sup_{t>0} inf_{s>0} \left\{ \frac{\Lambda_{N^*}(s,t) + L}{s} - Ct \right\} \tag{5.6}$$

where C is the transmission capacity, s and t are free parameters, $L \in \{F(\gamma, C), \hat{F}(\gamma, C), G(F(\gamma, C)), G(\hat{F}(\gamma, C))\}$ and $\Lambda_{N^*}(s,t)$ is the cumulant generating function of $X_{N^*}(0,t]$, the random variable denoting the workload arriving from N^* sources in time interval $(0,t]$. The number of sources (N^*) should be computed with (5.2)

Proof: If a transmission channel admitting maximum N^* simultaneous flows is provided then according to Formula (5.2) the TBP level will not exceed β by definition. On the other hand, according to Theorems 4.2.1, 4.3.1 and 4.4.1, N^* simultaneous flows require at least B_n^* buffer space in order to keep the WLR level below the $e^{-\gamma}$ threshold. Q.E.D.

The above-described dimensioning methods work properly in case the models characterizing the call and packet level dynamics of the offered traffic load are precise enough and the packet server applies a simple admission control policy admitting the service of only N^* simultaneous flows. Here we note that these dimensioning techniques are somewhat conservative in the sense, that the equivalent capacity value was computed for the worst case scenario, when N^* (the maximum allowed) number of flows are admitted to the system. This scenario, however, happens only on rare occasions. A more accurate approach to this problem would be to calculate (5.6) and (5.5) for $\Lambda_N(s,t)$, where N is a random variable with Engset distribution. Nevertheless, the computational demand of this solution would be significantly higher and thus we recommend the above-depicted methods instead.

5.1.1 Numerical investigations

In this section the performance of the capacity dimensioning method introduced in Theorem 5.1.1 is to be assessed with the aid of numerical examples. Through these investigations the required capacity value computed with (5.5) will be compared to two different capacity allocation schemes, namely the peak rate reservation and the bufferless fluid flow multiplexing schemes.

Let us consider the access aggregation network of three service areas that differ in their density of subscriber population. The "City" area is the most densely populated and the aggregator nodes (e.g. DSLAMs) found there serve 250 subscribers each. The "Suburb" zone is less densely populated: one multiplexer node serves 150 subscribers in this area. The "Rural" area is relatively sparsely populated; only 100 users are served by each aggregator node.

Let us suppose that three QoS classes are defined in these access networks. The first class is created for providing appropriate transmission quality for broadband VoD (Video on Demand) applications. The popular IP telephony applications belong to the second class, while the third class provides best-effort service for any other flows in this example. Let us suppose that the traffic flows belonging to different QoS classes do not interfere with each other, i.e. they have dedicated resources (transmission capacity and buffer space).

Table 5.1: Maximum number of simultaneous VoD/VoIP channels and required pipe capacities.

	P	λ [1/hour]	N^*	C^{peak} [Mbps]	$\widetilde{C}_{equ}^{B=0}$ [Mbps]	B [kByte]	σ [Mbps]	C_{equ} [Mbps]	C_{equ}^{BR} [Mbps]
VoD ($h = 90$ min, $p = 3.3$ Mbps, $m = 0.77$ Mbps, $\beta = 0.01$, $\gamma = 12$)									
City	250	0.1	44	145.2	56.0	424	4.44	43.358	41.920
Suburb	150	0.1	28	92.4	40.1	270	3.54	29.445	28.148
Rural	100	0.05	14	46.2	25.0	135	2.51	17.220	16.230
VoIP ($h = 5$ min, $p = 0.024$ Mbps, $m = 0.02$ Mbps, $\beta = 10^{-6}$, $\gamma = 8$)									
City	250	1	42	1.008	0.965	5.25	0.058	0.964	0.927
Suburb	150	1	30	0.720	0.700	3.75	0.049	0.694	0.660
Rural	100	1	24	0.576	0.574	3	0.044	0.571	0.551

In Table 5.1 the additional input parameters required for the dimensioning process are summarized. The second column shows subscriber population size per multiplexer node, while the third column contains the service initiation intensities belonging to different services and areas. For example, the initiation intensity of VoD sessions in the "City" and "Suburb" zones is 0.1 1/hour , which means that subscribers in these areas initiate a VoD request (independently from each other) in a one-hour interval with probability 0.1.

Service-related information and target QoS levels can be found in the 3rd and 7th row of the table for VoD and VoIP service, respectively. Accordingly, a video session has 90 minutes holding time on average, regardless of the location of the subscriber that initiated the VoD service. The same applies to VoIP sessions, but these have an average holding time of 5 minutes. The peak and mean date rates of video sessions are 3.3 Mbps and 0.77 Mbps, respectively. VoIP sessions use ITU-T G.729 audio codecs, so the peak date rate of these sessions is 24 kbps, the mean data rate is 20 kbps. Target TBP for VoD sessions is set to 1%, while for VoIP sessions a more stringent target TBP (10^{-6}) is prescribed. Target PLR level is set to e^{-12} and e^{-8} for VoD and VoIP, respectively.

The required numbers of servers computed with (5.2) can be found in the fourth column of the table. The fifth column of the table contains the capacity needs of logical channels transmitting N^* simultaneous sessions without the exploitation of the multiplexing gain (i.e. the peak data rates of the sessions are allocated). The sixth column of the table contains the required channel

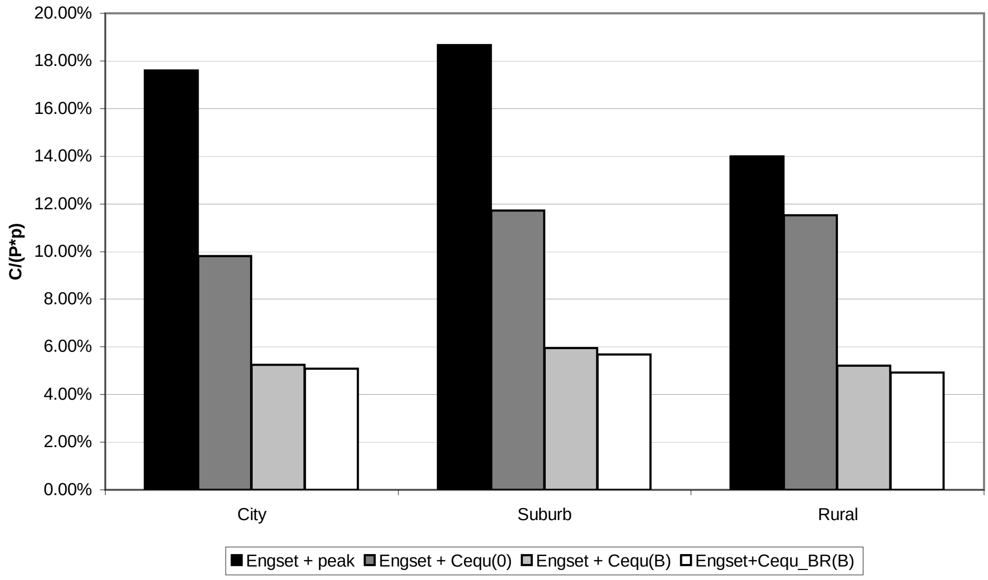

Figure 5.2: Capacity requirements of aggregated Jurassic Park MPEG-4 flows relative to peak rate reservation for the full subscriber population

capacities computed with (3.29). This formula takes multiplexing gain into consideration, but fails to capture the benefits arising from buffering. The seventh and eighth columns contain the buffer and sigma values used for computing equivalent capacity under the many sources asymptotic framework. The buffering spaces were set in a manner that the maximum delay and delay variance caused by buffering will not exceed 100 ms and 50 ms for VoD and VoIP traffic, respectively. The σ values of the aggregated flows was calculated using the $\sqrt{N^*} * \sigma_1$ formula, where σ_1 is the standard deviation of a single source obtained by processing the corresponding trace file.

By comparing the values in the 5th and 6th columns, it can be seen that reasonable amount of capacity could be saved, if the multiplexing gain is exploited. This especially applies to the bursty video traffic. However, if the benefits of buffering is also taken into account by using Formulae (3.13) and (3.21), further capacity can be saved. The savings can be very significant in case of the VoD traffic, while it is rather modest in case of the VoIP traffic.

In order to have a clear view about the savings offered by the novel dimensioning method, let us take a look at Figure 5.2 and Figure 5.3. In these diagrams, the 100% value belongs to the bandwidth need of the aggregated traffic at which neither call-, nor packet level (i.e. multiplexing) gains have

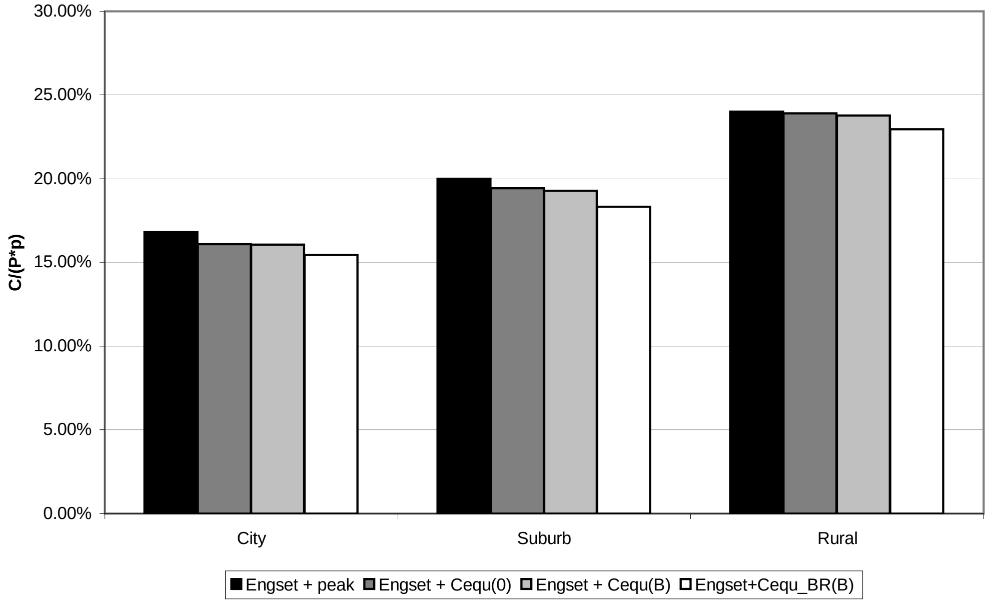

Figure 5.3: Capacity requirements of aggregated G.729 VoIP flows relative to peak rate reservation for the full subscriber population

been considered. In other words, the peak rate of each session for each subscriber has been allocated for this traffic mix. Accordingly, the aggregator nodes in the City would allocate $250 * 3.3 = 825$ Mbps for VoD traffic, if any of the possible gains are not exploited. However, if the multiplexing gain is taken into consideration, 56 Mbps is just enough for ensuring the prescribed QoS level. Going further and exploiting the gain from buffering too, the capacity need drops to around 42 Mbps which is yet another 25% saving. This way a total of 95% capacity can be saved compared to the "peak rate reservation for the whole population" scheme.

It is also noticeable, that savings increase as the population size increases. This is of course due to the increasing multiplexing gain[1].It can also be observed that there is only a marginal difference between the capacity requirement computed with (3.13) and (3.21). It is also worth noting that hardly any capacity saving arises from packet-level multiplexing in case of the VoIP scenarios, this is due to the non-bursty nature of VoIP traffic and the low level of multiplexing.

[1]The Rural+VoD scenario is an exception, as in this case the service initiation intensity was set to a lower level!

5.2 A QoS Architecture Providing Multi-level QoS Guarantees in Broadband Access Networks

In this section a QoS architecture aiming to provide multi-level quality assurance for value-added applications - primarily in access aggregation networks - will be presented. The whole architecture greatly builds upon both the ENRICO concept [2] and the combined dimensioning method discussed in the previous section.

The ENRICO concept has been designed for QoS provisioning primarily in Ethernet or ATM-based access aggregation networks (ANs). In this concept an interface is provided between the NAP (Network Access Provider) and the ASP (Application Service Provider) through which QoS- and resource reservation-related information may be exchanged in a manner that is fully transparent to subscribers. In this model there is a central entity, the Session Resource Broker (SRB), which defines a logical overlay network upon the physical topology and manages resource allocation and call admission control. The predefined overlay network consists of logical point-to-point trunks between the edge nodes of the AN. These logical channels are called QoS pipes, as there are one such channel for each QoS class between each access node - edge node pair. To gather the needed information for the operation of the QoS architecture certain functions of currently available protocols such as the RSVP or the DHCP are also exploited.

The ENRICO model - when it was first published - left several question unanswered. For example it did not specify the admission controlling scheme to be used or the dimensioning strategy to be applied for the creation of QoS pipes. Also its centralized structure - i.e. all management and controlling related tasks are carried out by the single SRB entity - poses an operational risk and demands huge computational capacity from that central entity.

5.2.1 Main Concepts, Building Blocks and Functions

The architecture we have developed strives to heal the imperfections of ENRICO (see Figure 5.4). On one hand we have specified the implementation of certain functions and also have changed the centralized admission control strategy. Our architecture contains four elements which are the following. The core of the architecture is still a central entity - called CME (Central Management Entity) - however its functionality is significantly reduced. It has to

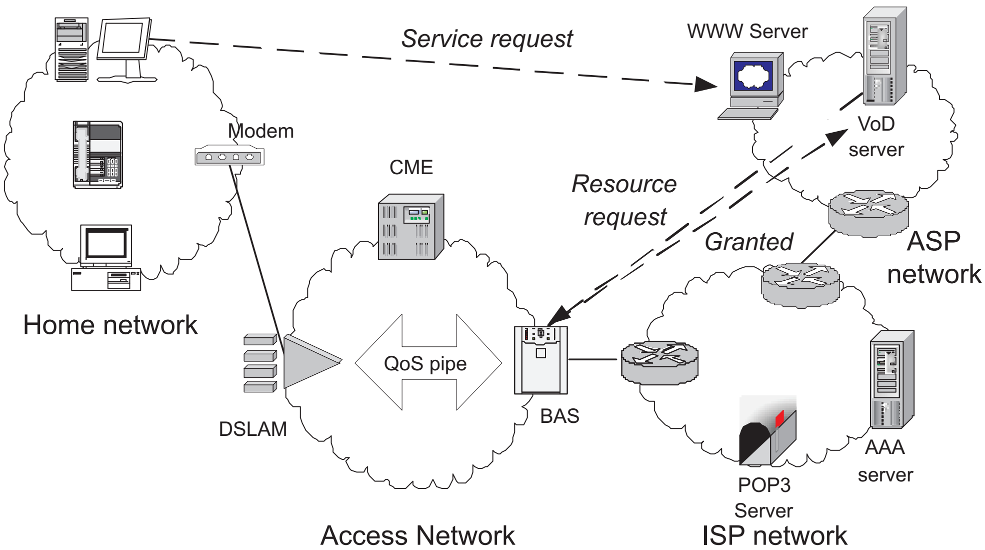

Figure 5.4: The model of the recommended QoS architecture

perform only two tasks: the (re-)configuration of the logical overlay network and taking reactions upon failures of network elements.

The majority of traffic management related tasks - including admission control, traffic metering, shaping, policing and pricing - are delegated to the Broadband Access Servers (BASs). Broadband Access Servers are the nodes located on the border between the AN operator's and the ISPs' networks, and so all traffic addressed to a certain subscriber should travel through these network node. Therefore they are less in numbers (i.e. it is supposed to be one BAS for each ISP that is contarcted with AN operator) than the DSL Access Multiplexers (DSLAMs), are usually located at the premises of the AN operator and have increased intelligence compared to DSLAMs. These properties make them the proper candidate for accomplishing extra functions, such as the ones mentioned above.

The nodes located in the inner region of the AN are ATM or Ethernet switches with limited intelligence and function sets. Their task is to forward traffic incoming at a certain port to another outgoing port. Also they should be configurable to handle traffic according to their priority implicitly indicated by their QoS class. The scheduling mechanism of these switches should support bandwidth and buffer space allocation on PVP or VLAN ID basis. This latter functionality ensures the realization of the independent QoS pipes with dedicated resources discussed earlier, as these pipes are to be identified by their PVP or VLAN ID.

The fourth element of the architecture consist of the already mentioned DSLAMs. These ATM or Ethernet switches are located at the boundary of the AN close to the subscribers' home networks and have only one task, i.e. to aggregate (distribute) and forward traffic arriving from (traveling to) the subscribers. These nodes should also be capable of handling different QoS classes according to their priority. They also serve as end-points of the QoS pipes.

The main concept of operation of the architecture is the following. The CME entity creates the initial logical overlay network by establishing QoS pipes between the edge nodes of the AN. The creation of QoS pipes is in fact the configuration of DSLAM, switch and BAS nodes so that each of these would be aware of how to handle traffic belonging to different PVPs or VLANs and how much capacity (fraction of service cycle) and buffer space is dedicated to a certain PVP or VLAN. The initial amount of resources allocated to each QoS pipe can be determined for example with the aid of the combined dimensioning method discussed in Section 5.1.

Once the logical overlay network is created traffic management related activities took part primarily at the BAS nodes. These nodes continuously monitor the load of the QoS pipes (that are starting at them) and make admission control decisions in a manner that is transparent to both the subscriber and the Application Service Provider (ASP).The traffic monitoring and metering is used for measurement-based equivalent capacity and/or buffer requirement estimations, i.e. the aggregated peak and mean data rate of the traffic is measured and the cumulant generating function of the aggregated traffic is approximated with the method discussed in Section 3.5. The measured parameters accompanied by the prescribed QoS goals are then fed to one of the resource requirement estimator algorithms discussed in Chapter 3 and Chapter 4.

The transparency of the admission control process is ensured through additional functionality of the BAS, namely DHCP traffic monitoring and RSVP message capturing. DHCP monitoring is needed to couple assigned IP addresses and QoS pipes groups, in other words to determine through which DSLAM the subscriber with a certain IP address can be reached.

The RSVP capturing is used for impersonating the subscriber according to the following process. The BAS captures RSVP messages addressed to the subscriber initiating the service and checks whether the resource requirements described in the RSVP message can be fulfilled in the QoS pipe that would transport the service. Then, in accordance with the result of this check, the BAS answers in lieu of the subscriber whether the request can be granted. For the admission control decision the BAS need to couple the destination IP address of the RSVP message with the corresponding QoS pipe and this is why the DHCP monitoring functionality is needed in the BAS nodes. Also the BAS needs to be aware of the current load of that pipe and this is done by the monitoring and resource requirement assessing function. The whole admission process with decision strategy and the protocol related issues will be discussed in depth later in this chapter.

As a summary it can be stated that the above-discussed QoS architecture - due to its distributed nature - is more resilient to failures. It is also advantageous that the computational and data processing load is distributed among the BASs and there is no need of one super-intelligent and super-fast central entity that handles all service requests.

5.2.2 The Admission Control Algorithm

In the QoS architecture introduced in the previous section we recommend to use the following flexible CAC algorithm. Let us first assume that the logical overlay network has already been created by the CME and the BASs are aware of the sizes of QoS pipes that are connected to them. Furthermore, the maximum sizes of the QoS pipes have also been determined and this information is also known by the BASs. This latter activity can be carried out for example by proportionally distributing free capacities among the QoS pipes sharing a certain link, but there exist more sophisticated approaches to this problem (see [J1]).

Now, let us overview the admission control process carried out by the distributed QoS architecture step-by-step:

- *Step 1* The subscriber in its home network would like to connect to the Internet. Therefore, it requests IP address from its ISP (Internet Service Provider) using the widespread DHCP protocol.

- *Step 2* The DHCP communication is monitored by the BAS that is located on the boundary of the AN operator's and the ISP's networks.

The BAS stores the IP address that has been assigned to the subscriber in its local database and also the identifier of the DSLAM which the subscriber is connected to. (These pieces of information are needed later to determine the identifier of the QoS pipes through which the subscriber can employ a certain type of service.)

- *Step 3* The subscriber wants to make use of a premium service provided via the Internet. She sends a service requests to the web server of the ASP by clicking on the appropriate hyperlink for example.
- *Step 4* The web server forwards the requests to the content server that is aware of the statistical characteristics and QoS requirements of the required service.
- *Step 5* The content server requests network resources by sending an RSVP message to the subscriber.
- *Step 6* The RSVP message is captured by the BAS. It looks up the DSLAM identifier of the subscriber based on the IP address and determines the appropriate QoS class which the new session would belong to based on the required QoS parameters in the RSVP message. Finally, the BAS checks resource availability in the appropriate QoS pipe.
- *Step 7* The BAS makes its admission decision. According to actual resource availability three types of actions may be taken at this point by the BAS. These are

  - If there is enough capacity available in the appropriate pipe, the access should be granted. The BAS informs the content server about the positive decision in an RSVP message. In parallel, its database that tracks available capacities in the QoS pipes is refreshed (e.g. available capacity decreased by the peak rate of the newcomer flow).
  - If there is not enough capacity available in the appropriate pipe, but the pipe can be expanded to a size that can cope with the new request, the BAS requests the expansion of the pipe from the CME. The CME performs the expansion and informs the BAS about it. The BAS then sends an RSVP message to the content server telling that its resource request is granted. In parallel, the database of the BAS that tracks available capacities in the QoS pipes is refreshed.

– If there is not enough capacity in the appropriate QoS pipe and it can not be expanded to cope with the newcomer flow, the resource request is rejected and the content server is informed about the decision in an RSVP message. The database of the BAS that tracks available capacities in the QoS pipes - of course - remains unchanged.

5.2.3 Protocols

The QoS architecture exploits the information carried by and functions offered by three well-know, widely used protocols, the Dynamic Host Configuration Protocol (DHCP), the Resource Reservation Protocol(RSVP) and the Simple Network Management Protocol (SNMP). In this section a short overview on these protocols and their operational mechanism will be provided and the actual way of their exploitation by the QoS architecture will be briefly described.

The Dynamic Host Configuration Protocol is a protocol used by networked equipments to obtain the parameters necessary for operation in an Internet Protocol network. It is widely used by Internet subscriber hosts (clients) for obtaining IP addresses from the ISP's IP Address pool upon connecting to the Internet. Its working mechanisms are described in RFC 2131 of the Internet Engineering Task Force [34].

The IP address acquisition by DHCP protocol is carried out in four steps. First, the client broadcasts a DHCP Discovery packet on the physical subnet to find available servers. Then, the DHCP server(s) receiving the Discovery packet allocate an IP Address from their address pool and answer to client with a DHCP Offer packet. The client accepts the offer of one of the DHCP servers and broadcast a DHCP Request packet containing the requested (chosen) IP Address. The DHCP server(s) receive(s) this Request packet and the DHCP server whose offer was requested by the client answers to the client with a DHCP Acknowledge message. (The other DHCP servers put their offered IP Address back to the pool of valid IP Addresses.) The BASs of the previously discussed QoS architecture should capture and process the DHCP Acknowledge messages as these contain the assigned IP Address of the host.

The Resource ReSerVation Protocol (RSVP), described in RFC 2205, is a Transport layer protocol designed to reserve resources across a network for an integrated services Internet [35]. It has two main concepts: flowspec and filterspec. RSVP reserves resources for a flow that is identified by the destination address, the protocol identifier and optionally the destination port. The flowspec contains all the information needed for QoS guarantee assurances; it describes the QoS class of the flow, the required resorces/QoS parameters and traffic characteristics of the traffic flow. The filterspec defines the set of packets that should be treated according to a certain flowspec.

The resource reservation process carried out through the RSVP protocol is realized in two steps. First, the RSVP host that wants to send a data flow with specific QoS transmits an RSVP Path message that will travel along the unicast or multicast routes pre-established by the working routing protocol. When the destination router receives the path message it will make a reservation based on the request parameters (flowspec information). Then the destination router sends back to the source an RSVP Resv message. Each node in the path back to the source can either accept or reject the request.

The QoS architecture provides transparent admission control through capturing RSVP Path messages at the BASs that answer in lieu the subscriber node with an RSVP Resv message.

It was mentioned at the Admission control scheme that the QoS pipe sizes may be reconfigured on demand. For that the the BASs should negotiate their request with the CME. this can be done e.g. through the widely used SNMP protocol [36].

In a typical SNMP managed system, there are a number of elements to be managed (DSLAMs, switches, BASs), and one or more entities (CME) that manage those. A software agent runs on each managed nodes and reports information via the SNMP protocol to the management entity.

SNMP agents send management data on the managed systems as variables (such as "available capacity"), but the protocol permits active management tasks as well, like modifying and applying a new configuration. The management entity can query information through the GET, GETNEXT and GETBULK protocol operations or the agent can send data on a regular basis (without being queried) using TRAP or INFORM protocol operations. The management systems can send configuration updates and controlling requests through the SET protocol operation to actively manage a system.

In the QoS architecture the SNMP protocol can accomplish the configuration-related tasks, that may be triggered by BASs receiving higher traffic load than expected.

Chapter 6

Summary and Conclusions

Novel, value-added applications may become widespread in the evolving, multi-service packet-switched networks only if reliable QoS assurances are ensured. As the best effort, IP-based packet switched networks do not contain built-in mechanisms capable of providing transmission quality guarantees, additional functions and services have to implemented.

The core element of an efficient QoS provisioning mechanism is an effective resource requirement assessment method that is capable of determining resource needs of diverse traffic types reliably and with an adequate accuracy. Basically the resource requirement of a network service can be characterized by two parameters: bandwidth (or transmission capacity) and buffer size need.

In the dissertation we have presented capacity and buffer requirement assessment techniques, that are rooted in the Theory of Large Deviations, a mathematical apparatus extremely capable of describing the probabilities of rare events, such as a buffer overflow event of a router. These new methods are capable of approximating the bandwidth or buffer size need of a certain aggregated traffic flow in case the maximum allowable Workload Loss Ratio (WLR) is prescribed. Our formulae have lower computational demand than other formulae found in the literature and capable of computing the same measures. Also we have shown how to couple our resource assessment methods with other call-level methods to form a combined dimensioning technique that is capable of providing multi-level QoS, i.e. packet-level loss and call-level availability, assurances simultaneously. Furthermore we have designed a QoS architecture that can be applied primarily in access aggregation networks to ensure desirable transmission quality for value-added applications in a transparent manner. Our novel contributions have also been investigated through numerical examples.

6.1 Research Contributions

The contributions of the dissertation are divided into three main parts. In Chapter 3 iteration-based equivalent capacity approximation methods have been shown that are capable of estimating the transmission capacity need of a certain traffic aggregate in case the maximum level of WLR is prescribed. The main asset of the results lies in the fact that they provide an accurate enough estimate in fewer iteration steps than other similar formulae found in the literature.

In Chapter 4 buffer requirement estimators are presented that are able to explicitly compute the minimum buffer space that an aggregated traffic needs in order not to violate the QoS prescription composed in terms of the WLR. These explicit formulae have clearly lower computational demand than other, implicit buffer requirement formulae that have been publicized.

In Chapter 5 two new contributions are presented. The first one is a combined dimensioning method that exploits the resource assessment techniques discussed in previous chapters along with other known call-level dimensioning formulae. This new dimensioning approach enables the provisioning of simultaneous, multi-level (i.e. packet-level loss and call-level availability) QoS guarantees. The other contribution in this chapter is a QoS architecture that was designed to ensure desirable transmission quality for value-added service like VoIP or VoD in a transparent manner in broadband access aggregation networks.

6.2 Future Research Directions

The resource requirement assessment methods introduced in this dissertation were studied through numerical investigations, however simulations and experiments on real systems would also be needed to justify the efficiency and reliability of these techniques. Also the the author was not able to prove general convergence criteria for the iterative equivalent capacity estimation methods discussed in Chapter 3. Extensive numerical experimentations with wide range of parameters, however, showed very fast and reliable convergence in every case.

The resource assessment formulae presented here are rooted in the many sources asymptotic equality of the Theory of Large Deviations. This equation - as its name shows - only holds asymptotically, when the number of sources approaches infinity. Therefore it is an open question that what the minimal number of flows that ensures adequate accuracy for the estimations is.

In Section 3.5 the parameter need of the estimation methods was discussed in depth. It was shown that the required parameters can be obtained via measurements on the real system according to [27]. However, that method has not been proven to be the optimal one, so there is still room to develop novel methods that may outperform that in terms of accuracy or speed.

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