COST-257 Final Report

Impacts of New Services on the Architecture and Performance of Broadband Networks

COST-257 Management Committee,
Phuoc Tran-Gia, Norbert Vicari (eds.)

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Preface

This synopsis outlines the results of the COST-257 project, prepared for the final seminar to take place in Würzburg, Germany from 27.–29.9.2000.

COST belongs to a framework programme initiated by the European Union and stands for ”Cooperation in the field of Scientific and Technical Research” of EU, Central and Eastern European countries as well as of institutions from non-COST member states. This series of projects was created in 1970, together with negotiations on common European research and development programmes, which led to the setting-up of well-known programmes such as ESPRIT, EUREKA, RACE and IST. The COST-257 action can be seen as a follow-on project from

- COST-201 on “Methods for planning and optimization of telecommunications networks” (1979-1983),
- COST-214 on “Methods for the design and evaluation of multiservice telecommunication networks” (1985-1988),
- COST-224 on “Methods for the performance evaluation and design of asynchronous and synchronous multiservice networks” (1988-1991),
- COST-242 on “Methods for the performance evaluation and design of multiservice broadband networks” (1992-1996).

COST-257 entitled “Impacts of new services on the architecture and performance of broadband networks” was inaugurated in November 1996. The participating scientific community has grown remarkably during the course of the project. The initial number of participating countries was 12 and grew to 19 in the year 2000. The following countries and institutes participated:

Austria (TU of Vienna), Belgium (University of Antwerp, University of Brussels, University of Ghent), Croatia (University of Zagreb), Cyprus (University of Cyprus), Denmark (TU of Denmark), Finland (TU Helsinki, VTT), France (CNET, ENST Bretagne), Germany (Deutsche Telekom AG, University of Stuttgart, University of Würzburg), Hungary (Budapest University of Technology and Economics), Italy (Fondazione Ugo Bordoni, Politecnico di Torino), The Netherlands (KPN Research, TU Eindhoven, University of Twente), Norway (Telenor Research), Poland (TU Warsaw), Portugal (Instituto Superior Tecnico, Telecommunications Institute), Slovakia (University of Zilina), Slovenia (Institute “Jozef Stefan”), Spain (Telefonica I&D, TU of Catalonia), Sweden (University of Karlskrona, University of Lund, Telia Engineering), United Kingdom (BT Labs, Microsoft Research Limited, University of Cambridge, University of Surrey).

Besides the technical achievements, evidenced by more than 200 technical documents with a publication score of about 1 in 2, the personal networking established on Management Committee meetings contributes significantly to the value of COST actions. Management Committee members and experts having participated regularly (min. 3 times) in the project are as follows:
S. Aalto, A. T. Andersen, A. Arvidsson, J. M. Barcelo Ordinas, C. Blondia, J. M. Brazio, F. Brichet, H. Bruneel, W. Burakowski, C. A. Carvalho Belo, O. Casals, L. Cerda, T. V. Do, K. Dolzer, M. Fiedler, J. Garcia, N. Gerlich, V. Goldman, V. Inghelbrecht, V. B. Iversen, J. M Karlsson, P. Karlsson, P. Key, U. R. Krieger, S. Köhler, P. Kühn, K. Laevens, S.-O. Larsson, P. Lassila, G. Latouche, K. Lindberger, R. Lo Cigno, S. Louca, R. Macfadyen, M. Mandjes, P. Mannersalo, M. Meo, U. Mocci, S. Molnár, B. F. Nielsen, I. Norros, T. Örs, O. Østerbø,

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The MC meetings were also attended by the following ad-hoc contributors and guests:
E. Aarstad, A. Almeida, H. Azmoodeh, A. Bak, K. Basu, A. Beben, J. Biro, T. Bonald, V. Bonin, C. Carranza, J. Charzinski, C. Chrysostomou, Y. Chrysostomou, B. de Schepper, F. de Turck, S. de Vuyst, V. Dobrota, V. Eramo, B. Feng, D. Fiems, M. Fudala, J. Färber, R. Gibbens, M. Grasse, A. Gravey, M. Grqic, S. Grqic, G. Haßlinger, J. Heinänen, P. T. Huth, O. I. Lepe, D. Imriska, T. Jensen, L. Jereb, T. Kalkowski, G. Kandus, J. Kankare, J. Karvo, U. Killat, J. Kilpi, M. J. Kinnunen, M. Klimo, Z. Kopertowski, J.-Y. Le Boudec, R. Lehnert, K. Leibnitz, G. Lennox, J. Levendovszky, F. Lozano, M. Luoma, I. Maricza, M. Ajmone Marsan, L. Massoulie, R. Mathar, C. Matsis, M. Menth, M. Merikallio, Y. Mylonas, B. Naessens, V. Naoumov, T. T. Nguyen, A. Nogueira, P. Olivier, E. Oubagha, S. Peeters, H. Pettersen, M. Peuhkuri, P. Podhradsky, E. Rathgeb, H. Reittu, M. Ritter, L. Rossides, K. Salamatian, K. Seppänen, K. Sevilla, A. Simonian, R. Sobiczewski, J. L. Sobrinho, T. Sommer, K. Spaey, I. Srsic, D. Staehle, A. Svensson, I. Svinnset, C. Szabo, C. Theophilou, A. Tonietti, D. Trcek, T. H. Tuan, K. Tutschku, F. Unghvary, B. van Houdt, D. Vazquez-Cortizo, M. Villen-Altamirano, B. Vinck, J. Walraevens, P. de Sousa, H. R. van As, R. van der Mei, A. Ziedins.

The project period of four years witnessed a dramatic change in network technology and emerging applications. At the beginning of the project, ATM was the dominating technology. The number of ATM-related Technical Documents, both from performance analysis and network dimensioning issues, reflected this common view. During the project, IP-oriented network structures and applications gradually lessened the importance of ATM. In addition the emerging wireless networks in conjunction with their performance issues assumed also a large share of the projects core. Looking at the technology shift one can observe that it is increasingly difficult to define the scope of a four-year research project such as COST-257. On the other hand, considering the number of IP-related TDs as well as the wireless system oriented works, it is clear that the COST-257 researchers have been capable of adjusting their research projects according to the technological developments and changing industrial needs.

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About this booklet and the CD

The booklet is intended to give a readily accessible and comprehensive overview of the results achieved during the COST-257 action. The CD contains an extended version of the booklet text and provides multimedia information for the interested reader. While the text serves as a navigation aid to the results of COST-257, links to the references and to the technical documents of COST-257 provide a more detailed insight into the technical achievements of COST-257. Direct access to the TDs is enabled by either a searchable keyword or full text or author index. The results of COST-257 are structured in the following chapters:

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Contents

Chapter 1
Network Control

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1.1 Introduction

The term congestion refers to an overload status of a system in which the system itself is no longer able to offer the expected QoS to its clients, due to overload of its resources. This situation needs to be controlled, in order to minimise its impact on the client entities and to avoid degeneration into a collapse of the system.

Within this framework, congestion control in a packet broadband telecommunication network should minimise the probability that congestion occurs within the network (preventive congestion control) and to avoid the spread, duration and effects of congestion once it has occurred (reactive congestion control).

Preventive approaches are devoted to limiting the probability of the occurrence of congestion under nominal traffic conditions and are based mainly on the definition of an appropriate connection routing policy and on dimensioning the network according to conservative estimates of the connection level traffic. Reactive approaches are intended to react to unexpected traffic conditions and are aimed at modifying the network behaviour depending on the traffic conditions. For instance, by bandwidth reservation or by making available lightly loaded parts of the network to traffic experiencing congestion or by removing from the network traffic that has a low probability of resulting in successful connection, or by controlling the rate of certain traffic sources.

In broadband multiservice networks with admission control procedures, the resources are shared at two main levels: connection level and packet (or cell) level. If there is no admission control procedure then the resources are shared only at the packet level. Connection level congestion occurs when the resources required by all incoming connection requests exceed the capacity of the network. It has been deeply studied in the context of classical circuit switching networks.

Packet level congestion occurs when the global bandwidth required by the active connections sharing a given network resource (e.g., transmission bandwidth and/or buffer capacity) largely exceeds the capacity of the resource. In this case, the network is no longer able to meet the QoS objectives at packet level, either because of packet queuing delay increases and/or excessive packet losses. Packet level congestion control has the main objective of maximising the utilisation of network resources while meeting the different QoS requirements of the supported connections. It requires the combination of preventive and reactive controls. Preventive controls usually try to limit the number of connections or to enforce each connection to use only a given limited amount of resources. These functions include Admission Control (AC) to accept connections only if there are available resources and Policing Control to enforce the traffic generated by each connection within the limits negotiated at the connection set-up. In particular, AC plays an essential role in the traffic management of ATM networks. In IP networks, the specification of protocols such as RSVP (Resource ReSerVation Protocol) or MPLS (Multi Protocol Label Switching) make admission control methods possible. New internet architectures, such as IntServ and DiffServ could be realized with flow admission control. Reactive controls, on the contrary, intend to reduce the traffic entering the network in case of congestion. For instance, Available Bit Rate (ABR) control functions in ATM allow adaptation of the traffic offered by ABR connections in accordance with network Load, as for example the simple Explicit Forward Congestion Indication (EFCI) used to signal to the receiving terminals that some resources within the network are overloaded. Reactive functions may include selective discard functions to protect higher priority traffic. The present chapter discusses aspects of reactive (ABR) and preventive (AC) traffic controls and their impact on service quality and dimensioning of network elements.

Section 1.2 considers the performance of the ABR reactive control. Section 1.3 considers different Admission Control methods that could be implemented in a multiservice network as preventive network congestion controls. Finally, section 1.4 considers the performance of new congestion control algorithms obtained in this COST-257 action.

1.2 Analysis of the ABR service category

1.2.1 Overview of the ABR service category

The congestion scheme described in the ATM Forum specifications [11] for the ABR service category is a rate-based, closed-loop, per-connection control that uses feedback information from the network to regulate the rate at which sources transmit cells. In this scheme, the transmission rate of each connection is controlled by means of special control cells called Resource Management Cells (RM-Cells). RM-Cells are transmitted embedded in the Data-Cell flow from the Source End System (SES) to the Destination End System (DES). The DES "turns around" the RM-Cells, which return to the SES along the same path carrying congestion information. The main fields of RM-Cells and the initial values set by the source are the following:

- Explicit Rate (ER), set to PCR,
- Current Cell Rate (CCR), set to the Allowed Cell Rate (ACR) of the source. The ACR is a parameter maintained by the source which fixes the maximum rate at which cells may be scheduled for transmission,
- Congestion Indication (CI) Bit, set to 0 (no congestion),
- No Increase (NI) bit, set to 0 (increase rate),
- and Direction (DIR) bit, set to forward.

At the connection set up the source negotiates the maximum and minimum rate at which it may transmit (PCR and MCR). On receiving a backward RM-Cell the SES adjusts the ACR. Whenever a backward RM-Cell is received with CI = 0 and NI = 0, the SES is allowed to additively increase its rate (ACR). However, on receiving an RM-Cell with CI = 1, the SES multiplicatively decreases the ACR. In either case the ACR may not be reduced below the MCR or increased over the PCR.

1.2.2 ABR switch mechanisms

The ATM Forum establishes that a switch shall implement at least one of the following methods to control congestion: set the EFCI bit of the data cells; set CI or NI in forward and/or backward RM-Cells and reduce the ER of forward and/or backward RM-Cells. The switches that set the EFCI or CI bit to indicate a congestion state are known as *binary switches*. It is foreseen that a first generation of ABR switches would adjust the transmission rate of the source by this simple binary indication [236]. Some examples of such switches are given in [292, 183].

A second generation of switches would perform a more advanced algorithm to compute the allowed transmission rate, conveyed to the sources in the Explicit Rate (ER) field of the RM-cells. These are called *ER switches*. ER switch algorithms have been extensively studied in many contributions. Some of the proposed algorithms are based on control theory [115, 169, 130, 41]. Other switch algorithms are directly derived from the fair bandwidth allocation criteria [229, 116, 260, 111, 16, 15]. It is worth mentioning that the ERICA switch algorithm [110] has become popular due to its simplicity and robustness, and has often been used as a point of reference in other proposals. For example Möttönen [182] presents weighted ERICA (W-ERICA), an improved extension to ERICA.

1.2.3 Evaluation of the ABR service category

This Section discusses a number of models and analytical techniques to evaluate the performance of the ABR flow control scheme. We distinguish three approaches: a fluid flow model, a discrete-time Markov model and a model based on control theory. Furthermore, some guidelines are given for the buffer dimensioning of ABR switches.

Fluid flow model

Consider a system consisting of a number of greedy homogeneous ABR end stations and a single bottleneck link. Each of these ABR traffic sources has the same parameters. The propagation delay from the ABR sources to the switch and from the switch to the destination end station are considered in the model. Congestion in the switch is detected by means of two threshold values.

When the queue length exceeds the higher threshold value, the node detects congestion, until the queue length goes below the lower threshold value. Considering the traffic flows as a fluid, the ABR system dynamics are described by means of a set of coupled differential equations with two real-valued continuous variables, the ACR and the ABR buffer occupation. Due to the delay of the feedback information, the system oscillates and control cycles can be isolated. Each cycle consists of a number of phases which are governed by the above set of differential equations. Upperbounds for the buffer occupancy in case of an EFCI Marking mechanism can be found in [222]. The evaluation of the EPRCA mechanism can be found in [199], while a more complete analysis is given in [225].

Discrete-time Markov model

Consider a system consisting of two source end stations and a switch. Time is assumed to be discrete with unit (slot) the time needed to process a cell in the switch. One end station generates CBR/VBR traffic and the other generates ABR traffic according to an explicit rate congestion control scheme. Both the ABR and CBR/VBR traffic are input to the switch and compete for the bandwidth of the same output port in the switch. The switch acts as a virtual destination station for the ABR traffic. The CBR/VBR traffic is assumed to be the superposition of on/off sources. For the ABR traffic two cases are considered: a first case where the ABR source is persistent (or greedy) and a second case where the ABR source is an on/off source. The model takes into account the distance between the ABR SES and the switch. The algorithm used to compute the ER is based on the ERICA scheme, where the ACR is updated on a periodical basis. For this simple model, an embedded Markov chain technique leads to two important measures: the throughput of the ABR traffic (see [25]) and the ABR buffer capacity required to ensure a cell loss less than (see [27]). This model leads to the assessment of the influence of the following system characteristics on these performance measures: (i) the distance between the ABR source and the switch, (ii) the variability of the CBR/VBR traffic, (iii) the frequency by which the Allowed Cell Rate of the ABR source is updated, (iv) the algorithm used to compute the ACR when the ABR traffic is non-greedy and (v) the scheduling discipline used by the switch to serve the various ATM service categories. This analysis leads to simple, but accurate engineering rules for the ABR service category in ATM networks [27].

Control theory model

Control theory seems to be an appropriate approach for the design of closed loop rate based flow control systems. Several studies have been made for a system consisting of a bottleneck link whose input consists of controllable ABR traffic and CBR/VBR traffic, assumed to be non-controllable. There is a delay between the control action of rate modification in the switch and the instant this information reaches the SES. The goal of this control action is to avoid cell loss and to maximize the link utilization. The control loop should be stable over a wide range of traffic patterns and propagation delays. We mention a few approaches. In [165] a control theoretical approach is followed to study a Relative Rate marking scheme, while [293] presents a linear dynamic model for the design of an Explicit Rate mechanism. A dual proportional-plus-derivative controller is used in [215] for an Explicit Rate control loop, and an adaptive model based predictive control approach is used in [208].

Evaluation of maximum ABR queue size

In order to satisfy expectations about cell lossless transmission for traffic using ABR service, some buffer dimensioning rules are needed. In [15, 24] an approximate method for evaluating the upper bound for the required ABR queue size in an ATM switch is described. The analysis assumes the worst case of VBR+CBR (background) traffic in the form of the step unit function. The background traffic is carried with higher priority than the ABR traffic. As a consequence, the available capacity for the ABR traffic is reduced in the shortest possible time (e.g. from the maximum available rate to zero, if the rate of background traffic jumps from zero to the link rate (lcr)). Therefore, the required ABR queue size depends on how quickly ABR sources can reduce their sending cell rates. Notice that the ABR buffer has to absorb the whole ABR traffic submitted to the switch from the moment the background traffic rate was increased. Independently on the ABR switch algorithm, the source can start to reduce its sending rate not earlier than after the round trip time (rtt). Therefore, one can distinguish two components in the resulting buffer size: the constant factor, $rtt * lcr$, and the variable factor, which depends on the applied algorithm. With the above assumptions the deterministic analysis of the system behavior is provided allowing evaluation of required ABR buffer size to avoid cell losses. Appropriate formulas are derived for the ERICA, EFCI as well as ER-PR algorithm.

1.2.4 Conformance definition and policing

A conformance definition may be defined as the formalism used by the network to monitor whether the source transmits according to the traffic contract. The conformance definition may be implemented in the Usage Parameter Control (UPC) for policing. This policing function is used by a network operator to decide whether a connection is compliant or not. Furthermore, the UPC may also mark or discard non-conforming cells.

ITU-T and the ATM Forum have defined the Dynamic Generic Cell Rate Algorithm (DGCRA) as an example conformance definition for an ABR connection. The DGRCA decides the conformance in a cell by cell basis according to the following algorithm. At the arrival of each cell n, the value $y_n = c_n - a_n$ is measured, where a_n is the arrival epoch and c_n is a theoretical arrival time. Then, the cell n is non-conforming if y_n is greater than the cell delay variation tolerance (τ_1) and is conforming otherwise.

The computation of the theoretical arrival time of cell n requires the measuring of the inter-emission time that should be used by the source between cells $n-1$ and n (I_n). I_n is computed based on the feedback conveyed by the backward RM-cell flow up to cell n arrival. The computation of the sequence I_n is not an easy task because a change of rate conveyed by a backward RM-cell received at the measuring point at a given time may be applied to the forward cell flow after a delay equal to the round trip delay between the measuring point and the source. To cope with this problem two time constants τ_2 and τ_3 have been introduced which are respectively an upper bound and a lower bound of this round trip delay. Furthermore, two algorithms “A” and “B” have been defined to determine I_n (see [11] for details).

Despite of the importance of the fact that the UPC has to guarantee the effectiveness of the network services, not much attention has been given to the DGCRA. In [42], the former “A” and “B” algorithms are further analyzed. In [286], the authors study the ability of the DGCRA to restrict ABR sources to the traffic contract by discarding excess traffic. In [131], the impact of violating over non-violating sources is investigated. An analytical evaluation based on an equivalent queueing model of the DGCRA can be found in [45].

1.2.5 Charging

Charging may be an essential condition for ABR sources to adapt their traffic demand. We shall analyze different approaches for a usage-based pricing of

ABR. We can classify these approaches as "static" or "dynamic". A charging scheme is static if the charging parameters are established at the connection set up and do not change afterwards. If they change, the scheme is dynamic. We analyze a static and a dynamic charging scheme already suggested and propose new alternatives which solve some of their drawbacks [43].

A usage-based pricing seems adequate for an ATM network. For sources with guaranteed bandwidth as CBR and VBR, the allocated bandwidth or the generalized concept of the effective bandwidth, and the duration of the connection is a good characterization of their resource usage. Charging of ABR may be more complex as bandwidth is not requested by the sources but assigned by the network. Therefore, it is difficult for the network to know the user appraisal of resources for charging.

Static pricing schemes

Songhurst and Kelly [247] propose a pricing model where the charge of a connection is given by the expression:

$$\text{Total Charge} = a(x)T + b(x)V + c(x) \tag{1.1}$$

where T is the duration of the connection, V the volume submitted by the connection, x the tariff choice and $c(x)$ is a fixed subscription fee. An expression for $a(x)$ and $b(x)$ based on the effective bandwidth can be obtained for VBR connections. For ABR sources the authors propose to use the above expression with $a(x)$ proportional to the MCR, i.e. $a(x) = \gamma \times \text{MCR}$, and $b(x)$ is assumed to be much lower than γ or even zero. In order for the sources to select an MCR according to their bandwidth appraisal, the authors assume that the network divides the free bandwidth proportionally to the MCR.

The static model previously described has the drawback that many data sources may not be able to choose an adequate MCR due to their bursty nature. For such sources a pricing model which charges the transmitted volume rather than the duration of the connections would fit better. We propose a static model for the ABR Service, which consists of several prices per cell tariffs p_i. Users select a tariff p_i at connection set up in order to charge the cells transmitted at the shared bandwidth. We assume that the network allocation algorithm divides the free bandwidth proportionally to the chosen tariff p_i. Some sources may need a guaranteed MCR, which is likely to be charged based on the duration of the connection. According to these considerations, the following charging equation has been deduced:

$$\text{Total Charge of a connection} = \\ \gamma \times \text{MCR} \times T + p_i \times \max\{V - \text{MCR} \times T, 0\} \tag{1.2}$$

Note that if the connection transmits always at a rate $r \geq$ MCR, then MCR $\times\, T$ is the volume of traffic submitted at the guaranteed MCR. Therefore, the first term of the right side of equation (1.2) charges this traffic at γ [unit of price/cell]. The second part of equation (1.2) is intended to charge the volume of traffic transmitted above the MCR at the price p_i chosen by the user, $p_i < \gamma$. Clearly, when the source rate $r \geq$ MCR, the cells given by $\max\{V - \text{MCR} \times T, 0\}$ are the cells transmitted above the MCR. This is not a drawback, however, because it would penalize users who choose a guaranteed MCR higher than their needs.

Dynamic pricing schemes

Based on the principle of social welfare optimization [150], a dynamic charging for the ABR Service has been proposed by Courcoubetis et al. [50]. The method consists of using the forward RM-Cells of the ABR service to convey the source demand to the switches, and the backward RM-cells to convey the prices to the sources. Both demand and prices are adjusted by an iterative algorithm which in equilibrium satisfies the source demand and maximizes the network revenue (maximum social welfare). In the following we briefly describe this method.

Let C_l be the capacity available for ABR traffic traversing link l and a_l the price per transmitted cell charged to each connection traversing link l. Denote R_c the route of connection c and w_c the charge of connection c. Clearly $w_c = \sum_{l \in R_c} a_l$. Let assume that each connection c has a bandwidth demand equal to $D_c(w_c)$. Finally denote x_c the actual rate of connection c.

In [50] it is shown that the maximum social welfare is satisfied by the following relations:

$$x_c = D_c(w_c), \quad \text{for all } c \tag{1.3}$$

$$\sum_{c:l \in R_c} x_c \leq C_l, \quad \text{for all } l \tag{1.4}$$

$$a_l(C_l - \sum_{c:l \in R_c} x_c) = 0, \quad \text{for all } l \tag{1.5}$$

Equation (1.3) says that the social welfare is maximized when the connection rates equal their demands. Note that equations (1.4) and (1.5) imply that $a_l = 0$ for the non congested links ($\sum x_c < C_l$), and $a_l \neq 0$ for the congested links ($\sum x_c = C_l$). In [50] an iterative scheme which converges to this equilibrium is proposed. The algorithm consists of the links updating the price a_l^n at fixed intervals n of duration δ (referred as charging intervals). Prices at interval n are decreased or increased if $\sum D_c^{n-1} < C_l$ or $\sum D_c^{n-1} > C_l$ respectively.

In order to integrate this charging scheme in the ABR flow control, the authors in [50] propose to add two new fields to the RM-Cells: a request bandwidth (RB) field and a price per unit of bandwidth (PB) field. Based on the prices w_c, the sources set the RB field with the demand function $D_c(w_c)$. These values are used by the switches to compute the prices a_l^n applying an iterative algorithm. The switches increase the PB field of all the backward RM cells by a_l^n. Thus, when the backward RM cell arrives to the source it conveys $\sum_{l\in R_c} a_l$ in the PB field. Note that a billing unit located at the network edge keeping track of the PB field of the backward RM-Cells should be needed to compute charges.

A dynamic pricing model based on the switch loads

A problem of the pricing model previously described is that users do not have an incentive to specify their true demand. Users could set a misleading demand value in order to modify the prices (for example, the demand field could be set always to zero). To solve this problem we propose to use the offered load ($\sum_{c:l\in R_c} x_c^{n-1}$) instead of the source demand ($\sum_{c:l\in R_c} D_c^{n-1}$) to decide when to increase or decrease prices. Obviously, the RB field of the RM-cells conveying the request bandwidth of the sources is not needed anymore with this algorithm. Just the PB field would be used as described in the previous section.

Assume that the control algorithm of the switch adjusts the source rates such that the offered load of link l converges to a certain target cell rate TCR$_l$. We define the overload as $O_l^{n-1} = \sum_{c:l\in R_c} x_c^{n-1}/\mathrm{TCR}_l$. Based on the overload we propose the following algorithm to compute the prices:

$$a_l^n = \begin{cases} \max\left\{\left(1 + h\,\mathrm{sgn}\left[O_l^{n-1} - \alpha\right]\right) a_l^{n-1}, 0\right\}, & \text{if} \quad a_l^{n-1} \neq 0 \\ a_0, & \text{if} \quad a_l^{n-1} = 0 \end{cases} \tag{1.6}$$

where

$$\operatorname{sgn}[x] = \begin{cases} +1, & \text{if } x \geq 0 \\ -1, & \text{if } x < 0 \end{cases}$$

and $\alpha <\approx 1$ is a constant parameter. Note that now prices are adjusted such that the link load converges to $\alpha \times \mathrm{TCR}_l$. An analytical analysis of this algorithm can be found in [43]. This analysis shows that the previous algorithm is always stable and easy to tune.

1.2.6 ABR support to TCP

Internet traffic supported by the TCP/IP set of protocols is currently the biggest part of non real time traffic. Therefore, ABR may be one of the Service Categories chosen to give support to the Internet traffic in ATM networks.

Since the ABR Service Category was specified, numerous studies have been done to analyze its inter-operation with the transport protocol used in Internet (TCP). Earlier studies of TCP over ATM can be found in [72] and [230]. These studies show that poor performance may result from the fragmentation of the TCP packets in the ATM network. The authors explore cell dropping policies to improve the performance. In [73] the authors show that the performance achieved with ABR may depend on the selection of the ABR parameters. In [117] the authors analyze the effect of the buffer size on the loss ratio and the performance achieved using ABR. In [166] the authors show by simulation that higher goodput may be obtained using TCP over ABR than over the ABT Transfer Capability standardized by the ITU-T [105]. In [160] a comparison is done between UBR and ABR for the interconnection of shared media LANs. In the scenarios analyzed in this study, the authors find that UBR outperforms ABR in terms of goodput. In [238] a proposal is done to improve the performance of ABR when the ABR control loop does not extend to the TCP end source. This proposal consists of doing a rate-to-window translation at the gateways.

A simulation comparison of TCP over ABR, UBR, and the newer GFR Service Category can be found in [44]. These simulations show that fairness problems easily arise in UBR. This problem is minimized in GFR and eliminated using ABR.

1.3 Admission control methods

Admission control (AC) is a preventive traffic control which Aims to admit an arriving new traffic source if and only if its quality of service (QoS) as well as that of the already accepted sources is guaranteed. The admission control procedure should also ensure a high utilisation of network resources through efficient statistical multiplexing.

For stream traffic, one possibility for the network to guarantee QoS constraints is to keep the aggregated incoming bit rate less (with a given high probability) than the capacity of each link. This is the Rate Envelope Multiplexing (REM) [228, p.86]. By analogy with a fluid flow system, this way of multiplexing flows has also been denoted *bufferless multiplexing*. In Section 1.3.1 of this document, several methods of AC for stream traffic based on declared parameters and using bufferless multiplexing are proposed.

Parameters used to describe traffic sources are typically restricted to the peak bit rate, the tolerance to the peak rate, the mean cell rate and the tolerance to the mean cell rate or maximum burst size. The difficulty of the users for predicting them, mainly in the case of the mean cell rate, leads to their overestimation and consequently to network under-utilisation. An alternative solution is to use traffic measurement in the AC procedure. Measurement-based Admission Control (MBAC) methods may be distinguished depending on which parameters are measured (mean rate and variance, globally or per class), and which parameters are declared by sources (peak rate, mean rate). These MBAC methods may allow having a profitable network even if the users only declare the peak rates. They are described in Section 1.3.2.

For elastic traffic, it is useful to distinguish between open loop and closed loop controls. When open loop control is assumed, so-called rate sharing multiplexing scheme is performed [228, p.147 and ff.], i.e. buffering is used to share out the available rate amongst flows. AC criteria guaranteeing that buffer size is enough for handling elastic flows with negligible loss are discussed in Section 1.3.1. When elastic traffic is controlled with closed loop control, by means of the Available Bit Rate (ABR) service category in ATM or by TCP (Transport Control Protocol) in IP networks, the transmission rate of each source is adjusted to current traffic levels. The quality of service offered to such flows is manifested mainly by their response time or equivalently by their realized throughput. A minimum throughput can be guaranteed by limiting the maximum number of flows. AC rules for closed loop controlled elastic flows are presented in Section 1.3.3. Finally, Section 1.3.3 contains also a study on the integration of stream and elastic flows with open loop control.

1.3.1 Admission control for stream traffic based on declared parameters

When no traffic measurements are performed, an admission control criterion for stream traffic must rely on declared traffic descriptors and the possible use of shaping at the network ingress.

Admission control for peak rate allocation

If sources are only described by their peak bit rate (as with CBR in ATM and Premium service in IP), a naive approach is to admit flows while the sum of peak bit rates is less than link capacity. However, packet flows are never strictly periodic and we have to account for jitter. Evaluation and propagation of jitter in the network have been intensively analysed in the context of ATM networks, where packets have constant length. In this context, the notion of *negligible jitter or negligible CDV* (Cell Delay Variation) has been introduced [32]: a stream is said to have negligible jitter if it is *"better than Poisson"*, that is, if its impact on the queue of a network element is better than a Poisson stream with the same load. In [32], it is notably conjectured that flows which have negligible jitter at the network ingress retain this property as long as they are multiplexed with other flows with negligible jitter. Consequently, the following admission control can be used:

- Determine the maximum admissible load ρ of a multiplexer of rate C and buffer size B assuming Poisson arrivals (i.e, the loss rate of an M/D/1/B queue of load ρ is less than the target loss rate);
- ensure incoming flows have negligible jitter (by spacing to their peak rate at network ingress);
- admit flows while load is less than ρ.

A key point in defining negligible jitter is that cells have constant length. In an IP environment where packets have varying length, the above notion of negligible jitter can not be simply extended. For instance, consider routers having queues with two priority levels. The non-preemptive high-priority queue handles stream traffic while the low-priority queue is reserved to elastic traffic with traditional *best effort* service. Considering this simple queueing system, we have to take account of an additional jitter with respect to that associated with constant length packets because of two effects (i) stream packets being multiplexed in the same queue, a jitter due to the varying packet length is introduced, (ii) stream traffic packets may arrive in an empty high-priority queue while a best effort packet is currently being served. Therefore, the stream packet has to wait during the residual service time of the best effort packet, which is typically a varying value.

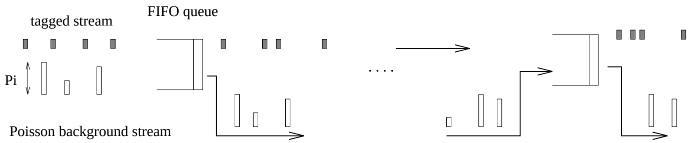

Figure 1.1: Tandem routers with varying length, background packet stream.

If we consider effect (i) in isolation, it is possible to provide a condition guaranteeing that a given stream remains *better than Poisson*. Consider a tandem of simple FIFO queues fed by a tagged stream with fixed packets of size d_1 and by a Poisson background traffic with rate λ and with random packet size of distribution Π (see figure 1.1). We denote $\mathbb{E}(\Pi) = d_2$, $\mathbb{E}(\Pi^2) = x^2$, and C is the link capacity (supposed to be identical for each link). The load due to background traffic is denoted by $\rho_2 = \lambda d_2/C$. We consider *saturated queues*. It can be shown that the tagged stream remains *better than Poisson* (in the meaning that the coefficient of variation is less than one) if and only if the following inequality is satisfied,

$$x^2 \leq d_1 d_2 \left(1 + \frac{1}{\rho_2}\right). \tag{1.7}$$

Note that if $d_2 = d_1 = x$, inequality (1.7) is satisfied, which is consistent with the notion of *negligible CDV* for constant length packets.

Considering effects (i) and (ii), we may argue that, at least in a first stage, IP premium service for real-time applications is expected to represent a small fraction of the total IP traffic (around 10%) [187]. The load generated by stream traffic will therefore be very low. A key condition for having an efficient Premium service is to guarantee that the bit rate of each source is a small fraction of the link capacity [228]. If this latter condition is satisfied and if the load dedicated to this service is low, then the resulting jitter could be very small. On the other hand, the growing transmission speed reduces the difference between the varying packet length environment and the ATM world.

Admission control using bufferless multiplexing

The Rate Envelope Multiplexing (REM) strategy has been developed in the context of ATM networks. However, we believe that the notion of REM and, consequently, the methods proposed here can apply to IP networks offering a certain level of QoS to real-time applications, specifically when using the Controlled-Load service. In the following, we consider *cells* rather than *packets*, keeping the terminology of ATM networks. We indicate when some caution is needed when attemping to extend the proposed methods to IP networks.

We consider a node with small buffer size able to absorb only cell scale congestion and an output link having capacity C. It is known [228] that cell losses can be produced either because congested queue exceeds buffer capacity when the input rate is less than C (cell scale congestion) or because the input rate exceeds the link capacity (burst scale). The previous section focused on cell scale congestion while this section deals with cell loss due to rate overload. The quality of service objective, measured by the Cell Loss Ratio (CLR), can be obtained by providing a buffer large enough to deal with cell scale congestion (ca. 100 cells) and performing AC to ensure negligible burst scale congestion. Let CLR_{bs} denote the cell loss ratio due to burst scale congestion.

The CAC methods based in the REM strategy aim to keep CLR_{bs} less that a given threshold ε_{bs}. Denoting by Λ the overall incoming bit rate, they try to fulfill $\mathrm{P}\left(()\,\Lambda \geq C\right) \leq \varepsilon$, where ε is a certain target saturation probability. The relation between ε_{bs} and ε is not generally well defined. In the ATM context where packets have small fixed size and target loss ratio are around 10^{-9}, a rough approximation is $\varepsilon = \varepsilon_{bs} \times 100$.

Three AC methods, based on the concept of equivalent bit rate (or equivalent bandwidth) for a variable bit rate stream i, have been developed in the context of this COST-257 action. The equivalent bandwidth e_i is such that if the sum of the e_i over all sources is less than C, the condition $\mathrm{P}\left(()\,\Lambda \geq C\right) \leq \varepsilon$ is satisfied. The first method is based on an empirical expression for the effective bandwidth while the second relies on the use of the Chernoff bound. These two methods have already been presented in the COST-242 final report [228]. The third method is new and it is briefly explained below.

This third method relies on the computation of a single parameter α which depends on the characteristics of the current (or representative) set of connections [278]. We distinguish two cases. In the first case, the algorithm computing α is very simple. This algorithm can be applied in a real time environment since it is based on an expression for the total equivalent bandwidth which can be easily updated at each connection request or release. Then, a non-linear acceptance boundary is obtained which fits precisely the real acceptance boundary. A second case assumes that the equivalent bandwidth of each connection is not updated at each connection request or release. This method is designed to provide conservative results (although less accurate). Intermediate solutions such as time-driven or threshold-driven updating method can also be implemented [280]. The resulting effective bandwidth formulas are relatively simple, which makes the implementation of the method feasible. For details about this third method we refer to [278] and [280].

1.3.2 Measurement-based admission control for stream traffic

It is known that traffic descriptors such as the mean bit rate and the peak bit rate are hard to predict, particularly for video streams. Thus, allocating resources on the basis of these parameters can lead to significant under-utilization of the network. To solve this problem, it has been proposed to base the AC procedure also on the measured mean rate (either the global rate or the rate per class).

During the past years a significant number of papers on MBAC have appeared. Below we review the most significant among them. Two notions play a crucial role.

- Jamin *et al.* [112] introduced the separation between the *admission criterion* and the *measurement procedure*. The admission criterion determines on the basis of a number of traffic characteristics (of the existing flows and the new flow) whether or not to accept the new flow. The measurement procedure captures the required traffic characteristic from the flows that are currently present (possibly in combination with the a priori traffic descriptor).

- Grossglauser and Tse [96] introduced the concept of a *certainty equivalent* (CE) method. These methods use a static AC algorithm, but insert measured quantities rather than a priori known traffic descriptors (and these measured quantities are assumed to be the 'real ones').

Certainty equivalent methods. The attractive feature of CE methods is to reuse existing static ACs. However, CE methods will give too optimistic results [96]. This is due to the fact that the measured quantities themselves are random variables and therefore incur additional uncertainty. Inserting these quantities into the static AC as if they were the 'real ones' might therefore lead to QoS degradation. Quite a number of MBACs that appeared in the literature are in fact CE methods. Mostly the above mentioned aggressive properties are compensated in an indirect way by a conservative measurement procedure. An example of such a method is Jamin *et al.* [112]. These measurement procedures are parametrized by a couple of tuning knobs, like window sizes and the exponential weight of historic data. Casetti *et al.* [39] advocate an algorithm that adjusts the tuning parameters on the basis of the traffic offered.

An interesting conjecture was postulated by Jamin and Shenker [113]. They consider the flow blocking/packet loss curve — obviously if the packet level criterion is stringent (low packet loss), substantial blocking will occur; similarly high packet loss will go with low blocking. In [113] it is stated that every *well-tuned* MBAC has (for given input traffic) the same flow blocking/packet loss curve. Notice that this does not mean that all MBACs are equivalent, as there are more performance measures than just packet loss and blocking (delay for instance), and apart from that the MBACs may differ in measurement effort required.

Also in Duffield *et al.* [63] a CE procedure is presented: it is tried to capture the aggregate traffic's cumulant function, enabling to calculate the asymptotics of the loss ratio. They however do not compensate by a conservative measurement procedure, and therefore the approach seems to be quite 'dangerous'. A similar remark holds for the method of Courcoubetis *et al.* [49]. They measure the buffer occupancy, and extrapolate the resulting empirical distribution function. Apart from that, an implicit assumption is exponential decay of the buffer content distribution, which does not hold with long-range dependent input. Gibbens and Kelly [91] provide a family of admission criteria (all of them based on Chernoff bound arguments), but do not aim to provide measurement procedures.

Methods that take into account the randomness of the measurements. The papers of Grossglauser and Tse [96, 97], Duffield [62] and Gibbens, Kelly, and Key [92] are essentially different from the above in that they depart from using CE methods. They recognize that overload is due to jointly occurring 'misleading measurements' (giving an overly optimistic impression of the momentary load) and rare behavior in the period after the measurement. CE methods neglect the first type of error.

Under weak assumptions on the asymptotic regime (heavy load, a single connection's bandwidth being small compared to the link rate) [96] characterizes the error made by using a CE method. The same is done in [62] for the regime where loss is rare and large deviations theorems are applicable; it is shown that the presence of correlation between the samples may degrade the performance of the CE-based MBAC even more.For a more extensive overview and evaluation of MBAC algorithms, we refer to [265].

Equal slope algorithm when global measurements are performed

In many cases, the proposed MBAC methods are based on upper bounds of the Chernoff bound (usually a single tangent to the source effective bandwidth/mean rate curve), thus providing more conservative results [228].This section presents a new approach for MBAC.

Consider a link of rate C. Let M be the measured overall mean cell rate. Assume connections have declared traffic parameters h_i and r_i (for a DBR connection $r_i = h_i$). Suppose connection i has an actual mean rate of m_i^r and adopt the worst case assumption that its rate variations are of on/off type. Using the equivalent cell rate formulae of Chernoff bound (see [228]) or Polynomial bound (see references [278] and [280] discussed in Section 1.3.1, it is possible to estimate the bandwidth requirement of the existing connections. To ensure the admission decision errors on the safe side, the m_i^r must be set such that the sum of equivalent rates is maximal subject to the constrains: $0 \leq m_i^r \leq r_i$ and $M = \sum m_i^r$. This problem can be solved as detailed in [279] resulting in an estimate of the overall rate requirement of existing connetions: $E = \sum_i e_i(m_i^r)$. The equivalent cell rate e_0 of a new connection of parameters h_0 and r_0 can be evaluated assuming its mean rate is equal to r_0. The connection will be accepted if $E + e(r_0) \leq C$.

A conservative approximation obtained by ignoring the mean rate bound $0 \leq m_i^r \leq r_i$ yields a simple formula for m_i^r, see [279]. It is shown that the resulting AC is only moderately conservative compared with the exact one which takes into account the condition $0 \leq m_i^r \leq r_i$.

Equal slope algorithm when per-class rate measurements are performed

Members of a given class could be defined as those with the same value of the traffic parameters used in the admission control. For instance, a class of sources could be defined either as:

a) sources with the same value of h_i.
b) sources with the same value of h_i and r_i.
c) sources with the values of h_i within a certain range.

In the three cases, the method described in the last subsection could be used applying the *equal slope* algorithm in an independent way to each class. For example, in any formula, M should refer to the measured rate for a certain class. In cases (b) and (c) the method becomes simpler: The *equal slope* algorithm becomes the *equal mean rate* algorithm. The theoretical basis of this algorithm can be found in [279].

1.3.3 Admission control for elastic traffic

Using open loop control and rate sharing multiplexing

A well known AC for elastic traffic is the one proposed by Elwalid, Mitra, and Wentworth (EMW) [68]. This method is based on effective bandwidths. It is attractive from both a theoretical and a practical point of view, due to its novel, insightful construction and simple resulting formulas. It is assumed that a source declares the peak rate h, the sustained rate r and the maximum time that the source is allowed to send at peak rate (or maximum burst length) T. The method assumes 'worst-case' on-off sources. Below, we sketch the method and comment on its efficiency. For a more extensive evaluation of EMW's AC method we refer to [265] and [56]. Also in [149] admission control for elastic flows is discussed, based on processor sharing models.

Lossless multiplexing. In the case of so called 'lossless' multiplexing, the AC rule accepts a new connection only when it can be guaranteed that there cannot be any loss at all (see [68] and [228, pp. 130-132]). In this case, the following "additive" CAC rule is developed: given traffic classes

(r_i, h_i, T_i) there exist parameters c_i such that $\sum_{i=1}^{n} N_i c_i \leq C$. This implies that N_i sources of type i can be accepted without the possibility of generating any loss. These *effective bandwidths* c_i are computed as follows: $c_i = \max\{r_i, (h_i T_i)/(B/C + T_i)\}$, and it can be shown that if the bandwidth is allocated according to the above rule then the maximum reached buffer level will be below B for any traffic mix. Therefore, we conclude that all N_i satisfying $\sum_i N_i c_i \leq C$ are contained in the "lossless area" (i.e. all combinations $N_1, ..., N_n$, that cannot yield any loss). The authors of [68] mention that the identification of this entire lossless area requires the solution of a rather complex optimization problem. However, in [56] we show that it is possible to approximate the lossless area very accurately by a method of low complexity. From this study, it also appears that the lossless area is mostly considerably larger than the area resulting from the condition $\sum_i N_i c_i \leq C$.

Statistical multiplexing. In [68] the solution of the lossless case is used to find an extension to the case in which small loss rates are allowed. The important observation is that the lossless effective bandwidth c_i is not required all the time, but only during a fraction r_i/c_i. Thus, new connections can be accepted as long as the probability $\mathrm{P}(()\sum_{i=1}^{N} N_i C_i > C)$ remains below the required maximum loss probability (here, C_i equals c_i with probability r_i/c_i and is zero otherwise). In [68] a relatively easy method for deriving the corresponding admissible region is provided.

In [56] (also [265]) it is shown that loss probabilities derived by EMW are often overestimations of the real loss probabilities because they do not cover all kinds of multiplexing effects. This is concluded by performing the asymptotics proposed by Simonian and Guibert [244] and Botvich and Duffield [29].

Using closed loop control and fair sharing

We assume here that elastic traffic is controlled by means of a flow control protocol like ABR/ATM or TCP/IP, enabling transmission rates to adjust to the maximum allowed by current traffic levels. Both protocols aim to fully exploit available network bandwidth while achieving fair shares between contending flows. A reasonable starting point to evaluating the impact of random closed loop controlled traffic is to consider an isolated link and to assume that new flows arrive according to a Poisson process. Assuming that the closed loop control achieves exact fair shares, this system constitutes an M/G/1 processor sharing queue [147, 227], for which a number of interesting results are known. Let the link capacity be C and its load ρ (= arrival rate $\times$ mean size $/C$). The number of transfers in progress N_t is then geometrically distributed with parameter ρ, and the average throughput of any flow is equal to $C(1-\rho)$. I.e. the mean transfer time $\mathbb{E}[T(x)]$ for a document of size x is linear in x and given by (see for instance Kleinrock [126]):

$$\mathbb{E}[T(x)] = \frac{x/C}{1-\rho}$$

Higher moments of the conditional transfer time $T(x)$ and expressions for its distribution are also available, see for instance Yashkov [291], Ott [203], Schassberger [241], and Van den Berg and Boxma [267]. These results are insensitive to the document size distribution.

A generalization of the "standard" M/G/1 processor sharing queue is obtained if the flows are guaranteed a certain minimum and maximum throughput r^- and r^+ respectively, e.g. as is the case in ABR/ATM. The resulting model is an M/G/1/K processor sharing queue in which the service rate r_n each flow receives depends on the number n of flows in progress. For this queueing model, explicit performance results are available in Cohen [48]. In particular, explicit formulas for the mean file (document) transfer time $\mathbb{E}[T(x)]$ of a file of given size x and for the blocking probability of a newly arriving file (flow) can be obtained. Again, they only depend on the mean value of the file size, and are insensitive for higher moments of the file size distribution.

If flows have weights ϕ_i corresponding to differential service rates, the corresponding generalization of the above model is discriminatory processor sharing. Expected throughput depends on object size and is no longer insensitive to the object size distribution. The sharing parameters ϕ_i ensure effective discrimination for the transfer time of *short* documents but throughput for all classes tends to the limit $C(1-\rho)$ as document size increases. Results in [227] show that discrimination is more effective as the document size distribution variability increases. Overall throughput can be improved by giving priority to short objects. In [226], it is therefore suggested to use the SRPT (Shortest Remaining Processing Time) scheduling discipline.

Admission control, by limiting the number of flows using any given link, ensures that throughput never decreases below some minimum level. As recalled in [226], the notion of minimum acceptable throughput is not clear. In the latter reference, it is suggested that admission control does not necessarily imply a complex flow set-up stage with explicit signalling exchanges between user and network nodes, since this would not be acceptable for elastic flows with very short duration.

Integration of stream and elastic traffic. In the case of integrated stream and elastic traffic, the classical processor sharing model proposed above has to be extended in order to model the impact of the presence of a varying number of stream traffic calls. For the analysis of the resulting extended model (i.e. processor sharing system with varying service capacity) we refer to [197]. In particular, in [197] we analyze and compare the performance of different admission strategies for stream and elastic traffic calls.

Integrating stream and elastic traffic with open loop control

This section addresses the AC problem, for the case where different service levels for stream and elastic traffic are realized through a priority queueing architecture in the switches, see Figure 1.2; one (small) buffer of size B_1 is used for stream traffic with stringent delay requirements and the other (larger) one of size B_2 is used for elastic traffic. The "elastic buffer" is served if and only if the "stream buffer" is empty. It is assumed that both the elastic and stream traffic flows are controlled by a dual leaky bucket, which enforces a certain peak rate, a mean rate and a maximum burst size for each flow. The QoS guarantees are the maximum allowed data loss ratios for the stream and elastic traffic denoted by ε_1 and ε_2, respectively.

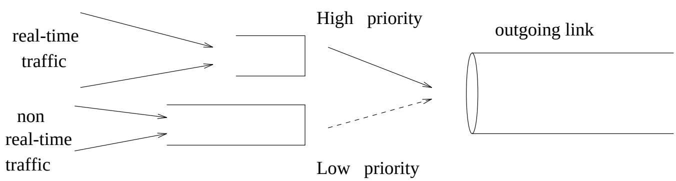

Figure 1.2: Two-buffer architecture at the switch output port for integration of stream and elastic traffic.

In the literature only a few papers are concerned with the AC problem for integrated stream and elastic traffic. The most relevant paper, both from a theoretical and practical point of view, is by Elwalid and Mitra [67]. Its principal result works as follows. As observed above, the high-priority traffic is not disturbed at all by the low-priority traffic and, hence, the loss performance of the hp-buffer can be obtained from known theory for statistical multiplexing in a single buffer. The key observation for the analysis of the loss performance of the lp-buffer is that the amount of service the lp-sources get equals $C -$ (total rate of hp-sources) if the hp-buffer is empty and is equal to zero if the hp-buffer is non-empty (here C denotes the link rate). To put it in another way: the lp-buffer contents in the priority system equals the buffer contents in a queueing system with service rate C continuously available and fed by the superposition of the lp-sources and the output of the hp-buffer. This simple relationship enables mathematical analysis; a prerequisite, however, is the knowledge of the output process of a queue fed by Markov fluid input. Elwalid and Mitra rely on an approximation of the output process of a Markov fluid queue that was found by them before in [66].

A manageable CAC method. It is noticed that in the practically relevant case where the stream buffer is very small (say zero), its output process is very similar to its input process. These processes disagree only during the (very small) time intervals when the input rate caused by the lp-sources is larger than C (i.e. when cell loss occurs). Now suppose that the loss ratio in the stream buffer is significantly smaller than the loss ratio in the elastic buffer. Then only a minor error is made when the loss probability in the elastic buffer is computed by means of the loss probability in a (virtual) buffer of size B_2 fed by the aggregate of both stream and elastic traffic sources, served with speed C. Hence, the CAC problem for the situation with integrated real-time and non-real-time traffic can be simply reduced to two 'standard' CAC problems for a single traffic type (for which e.g. the method proposed in [68] can be used). See [263] and [265] for more details.

1.4 New congestion control trends and algorithms

It is generally accepted that the problem of network congestion control remains a critical issue and a high priority. Especially given the growing size, demand, and speed (bandwidth) of the increasingly integrated services network. Furthermore congestion may become unmanageable unless effective, robust, and efficient methods for congestion control are developed. The existing congestion control solutions deployed in the Internet Transport Control Protocol (TCP) [108, 252] are increasingly becoming ineffective, and it is generally accepted that these solutions cannot easily scale up, even with various proposed ”fixes” [120, 251, 109], new approaches [87], and architectures [30, 22]. Also, recently there is increasing interest in non-TCP based applications. As demand for multimedia (streaming) applications increases, it becomes increasingly important to ensure that these applications can co-exist with current TCP-based applications. It is becoming widely accepted that streaming media should be subjected to similar rate controls as TCP traffic, and recently a number of researchers advocate that they should also exhibit TCP-friendly behavior [154] [85]). For TCP, the newly developed (also largely ad-hoc) strategies [214, 30, 213] are also not proven to be robust and effective. For non-TCP, examples include the model based [204] and the equation based [86] approach. Even though these methods are based on a model, the derived control strategy is ad-hoc and not proven with regard to its properties. Asynchronous Transfer Mode (ATM) has also witnessed a similar approach, with various congestion control schemes proposed, see e.g. [229, 110], for the solution of the Available Bit Rate (ABR) problem [11] not proven analytically.

New approaches are encouraged to analyze and control congestion. Next we present an example of active queue management (analysis of RED), then an example of ECN using a single bit feedback, and finally an approach based on fuzzy logic that uses queue length feedback to calculate a global congestion control feedback signal.

1.4.1 Performance evaluation of random early detection

Random Early Detection was proposed by Floyd and Jacobson [87] as an active queue management approach for the Internet. RED actively drops packets (probabilistically) based on the average queue length, which makes it applicable for non-cooperating transport protocols. It allows faster sensing of congestion (using the average queue size estimate), than the classical Jacobson algorithm [108] (using timeouts of lost packets).

Peeters and Blondia [207] have analyzed the influence of RED on the packet loss ratio and the throughput of a system consisting of a number of homogeneous responsive traffic sources, which share a RED switch. The responsive character of the sources is modeled by means of three load levels: each service time a source can change its load level, but when it is subjected to loss, it always enters the lowest load level. The system state is described in terms of a discrete-time Markov Chain. A matrix-analytical method was used to derive the steady state loss ratio and throughput of the system.

1.4.2 ECN based congestion control using 1 bit messages

The implementation of binary congestion control mechanisms, both at the sources and at intermediate routers, is a suitable approach for heterogeneous network environments where different technologies may co-exist. In particular, reactive mechanisms that provide explicit congestion notifications based on binary signals are easily implemented in such network environments, as congestion signals may be easily translated at network boundaries. Such reactive mechanisms may be based on algorithms that keep per-flow information, i.e. exercise control by keeping information about all flows traversing the node, or else, may be based on stateless procedures in the sense that congestion control is exercised without keeping per-flow information. In either case, we consider here algorithms that provide binary notifications to the sources; in particular exploring the transmission of precise values of rates (or other semantics) to the sources using 1-bit encoded messages.

In [4] a congestion control scheme is presented where precise control information computed at the network nodes can be sent to the sources using 1-bit messages. The mechanism is based on adaptive differential encoding techniques (used in classic transmission systems), applying in this case the adaptive Delta modulation. The implementation involves encoders and decoders located in intermediate routers and in the sources, introducing a small complexity overhead to existing algorithms that perform congestion control keeping per-flow information. A number of scenarios have been tested with algorithms originally proposed for explicit rate and price-based congestion control. The 1-bit encoded scheme exhibits similar performances as compared with the algorithms that explicitly send the rates to the sources using multi-byte messages, while significantly reducing the control bandwidth. In [6], a congestion control scheme is presented, using the same 1-bit message approach, where nodes exercise control with stateless algorithms. Given that the algorithms perform control without keeping per-flow information, this mechanism is suitable for implementation in core network nodes where a large number of flows are expected.

1.4.3 Fuzzy congestion control

Fuzzy Logic Controllers (FLCs) may be viewed as an alternative, non-conventional way of designing feedback controllers where it is convenient and effective to build a control algorithm without relying on formal models of the controlled system and control theoretic tools. (Note that Fuzzy logic control belongs to the general class of Computational Intelligence [206]). The control algorithm is encapsulated as a set of common-sense rules. FLCs have been applied successfully to the task of controlling systems for which analytical models are not easily obtainable or the model itself, if available, is too complex and highly nonlinear. While these techniques are not a panacea (and it is very important to view them as supplementing proven traditional techniques), we are beginning to see a lot of interest not only from the academic research community [98], but also from telecommunication companies [13].

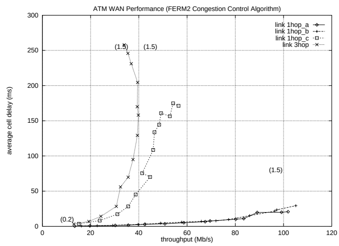

Figure 1.3: Average end-to-end ABR cell delay vs. useful throughput of simulated ATM WAN under Fuzzy Congestion Control.

In [210] a fuzzy based congestion control approach is proposed to address the congestion control problem. The performance of the proposed controlled system is evaluated via simulation. Very good performance was demonstrated using a four node network representing a WAN with inter-switch distances of 1500 km each and 20 sources sharing each switch (10 local and 10 are part of a 3-hop path) (see figure 1.3). The details can be found in [209].

Since the proposed scheme is generic, it is expected that the control of congestion in the Internet can be successfully solved using the fuzzy logic based approach (e.g. for ECN and RED based control). Current work is underway to apply Fuzzy Control in RED queues ([233]).

For wireless ATM networks, the fuzzy logic flow-control approach has been successfully applied in [240, 135] to cope with the impact of handoffs on the cell transport of ABR connections.

Chapter 2
Traffic Measurement and Data Analysis

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2.1 Introduction

Traffic measurements and reliable off-line as well as on-line estimation techniques are required to determine accurately the traffic load and resource usage in current high-speed networks with their different service classes. For this purpose new operational data collection procedures have to be developed and implemented in the network elements that use both measurements scheduled at a regular time basis and special high-resolution measurements triggered by network management actions. These schemes use different types of counting processes for distinct traffic objects and network elements (cf. [17, 114, 205, 220]).

Normally, measurement facilities count events of interest, e.g. incoming or outgoing calls, sessions, frames, packets or cells, at a specific network element in consecutive time intervals of fixed length. Considering, for instance, advanced packet-switched networks, the distribution of interarrival times between the events of the measurement, e.g. between sessions or transferred pages of a Web session generating IP packets in the Internet and between ATM cells or IP packets arising from packetized voice or video transmission, may belong to the exponential, light-tailed or even the subexponential class (cf. [205, 220]). For planning and control purposes, it is very important to estimate and to reconstruct the resulting traffic load accurately and by statistically thorough techniques.

In this chapter, statistical issues of traffic measurement, nonparametric and parametric estimation of the corresponding traffic models as well as measurement studies and data analysis of Internet dial-up and WWW traffic and the traffic characterization of new Internet services are presented.

2.2 Statistical methods of data analysis

Considering modeling and analysis of traffic loads in high-speed networks, a huge set of arrival processes with different short- and long-term correlation structures have been developed and numerous light- and heavy-tailed distributions describing the underlying random variables of the load models have been identified. Recently, considerable attention has been devoted to the estimation of the parameters of such models by real data.

In this Section, some nonparametric and parametric estimation techniques are summarized. They are developed in [162, 163] and [277] to estimate the basic characteristics of a nonhomogeneous Poisson process, a renewal process as well as a fractional Brownian motion (FBM) by empirical data. In [164] the nonparametric estimation of heavy-tailed distributions is addressed.

2.2.1 Estimation of a nonhomogeneous Poisson process

In [162] a nonparametric method to estimate the intensity of a nonhomogeneous Poisson process and the renewal function of a renewal process by empirical data of a limited number have been developed.

The estimation of the Poissonian intensity requires the observations of the total number of events in successive intervals of fixed length whereas the estimation of the renewal function is based on the observation of the interarrival times between the events of interest. Due to the noise in the data both estimation tasks are formulated as statistical ill-posed problem by means of related Volterra integral equations. Then Tikhonov's regularization technique for their solution and the stabilization of the estimates is applied (cf. [256, 269]). This method essentially uses the noise level and provides reliable estimates even if the variance of the noise is large. For this purpose, a regularization parameter may be selected by cross-validation methods (cf. [176]). This selection corresponds to the approximate minimum of the mean error of the estimation.

Two estimation approaches are considered, histogram-type estimates and expansions by orthogonal polynomials. In the case of the polynomial expansion the stabilization is performed by the choice of the regularization parameter to provide the convergence of the coefficients of the expansion to zero. Regarding the histogram-type estimate in the case of the estimation of a Poissonian intensity this task is done by the selection of the appropriate number of disjoint subintervals. All these parameters are selected depending on the amount of empirical data.

2.2.2 Estimation of the renewal function

Considering traffic measurements in advanced telecommunication networks determined by simple models of uncorrelated arrivals of calls, sessions or frames, some basic estimation problems related to renewal processes are studied in [163].

Using independent observations of the interarrival times between the recorded events of interest, new methods to estimate the renewal function of an underlying renewal process are proposed. Two types of estimates are constructed: histogram-type estimates and mixed parametric-nonparametric estimates. The parametric part of the latter reflects the asymptotic behaviour of the renewal function on the tail and the nonparametric one improves the behaviour inside the range of an empirical sample. In the proposed approach the estimation task is formulated as a stochastically ill-posed problem and procedures for the regularization of the estimates are applied (cf. [256, 269]).

To estimate the optimal number of terms in the histogram-type estimate a Bayesian principle is applied. Furthermore, small samples of normal and Gamma distributions are used as interarrival-time distributions to illustrate the power of the proposed nonparametric estimation approach.

2.2.3 Estimation of heavy-tailed distributions

The analysis of existing measurements of IP and WWW traffic by statistical methods has shown that the characteristic random variables of the latter are often long-tail distributed or even follow mixtures of long-tailed distributions (see [17, 53, 184, 205, 271, 272] and references therein). In [164] a nonparametric approach is discussed to estimate the probability density function (pdf) of long-tailed distributions and to cope with the data analysis in a highly variable environment such as the Internet.

Two nonparametric estimates, a Parzen-Rosenblatt (P-R) kernel estimate and a histogram with variable bin width called a polygram, are considered. The consistency of these estimates for heavy-tailed distribution densities is discussed. To provide the consistency of the estimates in the metric space L_1, the transformation of the initial random variable of a sample to a new one distributed on the interval $[0, 1]$ is proposed.

It is shown by a simulation study that a polygram and the P-R estimate are preferable for the application to real data if the true pdf is not available. If one knows that the pdf is heavy-tailed, then a polygram is recommended as simple nonparametric estimate.

Then the proposed estimates are applied to analyze real data of WWW sessions (see Section 2.5). The latter are characterized by the sizes of the responses and inter-response intervals as well as the sizes and durations of sub-sessions. By these means the effectiveness of the nonparametric procedures in comparison to parametric models of the WWW traffic characteristics is demonstrated.

2.2.4 ML-estimation of the FBM parameters

The fractional Brownian motion (FBM) is a popular model for long-range dependent traffic. Norros [190] suggested the following model $X(t) = mt + \sqrt{a}Z(t)$ where $X(t)$ represents the amount of traffic arrived in $(0, t)$. The model has three parameters, m is the mean input rate, a is a variance parameter and $0.5 < H < 1$ is the self-similarity parameter of the normalized fractional Brownian motion $Z(t)$.

In [277] a maximum likelihood estimation of the parameters of a fractional Brownian motion by means of geometrical sampling is developed. It is assumed that the traffic has been observed at n time instants. Since $X(t)$ is Gaussian, the joint distribution of the observed traffic values $(X_1, X_2, \ldots, X_n)$ is n dimensional Gaussian with mean mt and covariance matrix G. Thus, a Gaussian Maximum Likelihood Estimation (MLE) can be applied (in the time domain) to estimate the model parameters m, a and H. Explicit expressions for the ML estimates of m and a in terms of H are given as well as the expression for the log-likelihood function. By a maximizing argument the estimate of H is derived from the latter.

2.3 Spatial traffic estimation

In [262] a new method for the estimation and characterization of the expected teletraffic in mobile communication networks is presented. This method considers the teletraffic from the network point of view. The traffic estimation is based on the *geographic traffic model*, which obeys the geographical and demographical factors for the demand for mobile communication services. For the spatial teletraffic characterization, a novel representation technique is introduced which uses the notion of discrete *demand nodes*. It is shown how the information in geographical information systems can be used to estimate the teletraffic demand in an early phase of the network design process. Additionally, it is outlined how the discrete demand node representation facilitates the application of demand-based, automatic mobile network design algorithms.

2.4 Measurement and analysis of the Internet access

Nowadays, when Internet access is a popular consumer application even for ”normal” residential users, some telephone exchanges are congested by customers using modem or ISDN dial-up connections to their Internet Service Providers (ISPs). To estimate the number of additional lines and the switching capacity required in an exchange or a trunk group, Internet access traffic has to be characterized in terms of holding-time, call interarrival-time distributions and other characteristics. In [74, 75, 125, 124, 273] and [274] studies on the characterization of Internet dial-up traffic are presented.

2.4.1 Internet dial-up traffic analysis

In [74, 75] the authors have analyzed log files of Internet dial-up traffic tracing the usage of the central modem pool and ISDN access line pool at the University of Stuttgart for a period of six months. After identifying the busy periods, they have characterized the session behaviour by the first two moments of the session holding time and the session interarrival time. Poisson arrivals were not observed. A multi-server loss system with a renewal arrival stream and hyperexponential distributions was chosen to describe the traffic at the session level. By an evaluation of subsets of the logged user group, the authors were able to derive a simple scaling rule for the model stating that the session arrival rate increases linearly with the number of users.

2.4.2 User-traffic dependent on access speed

For the development of generalized parametric Internet traffic models, knowledge of invariants and dependences of measured traffic on the access speed are required. The results of this investigation form the basis for developing a generalized Internet user model. The latter becomes increasingly important in the presence of the variety of future access technologies, which range from slow wireless IP access to high-speed HDSL connections.

The data studied in [273] was measured at the dial-in access of the computing center of the University of Würzburg, Germany. In a two-week measurement the accounting data were related to a packet trace. Thus, the dial-in speed and all TCP characteristics could be evaluated. A total of 62000 sessions was captured containing a data volume of 82 GByte.

To relate the characteristics of the connections measured at the dial-in access, that is, access speeds ranging from 9.6kbps to 64kbps, to high-speed Internet access the first measurement [273] was compared to a measurement of ADSL Internet access [274]. This second data set represents 'unlimited' Internet usage since the high speed access was provided free of charge.

The measured traffic was evaluated both at the session and the TCP level. Several general properties of Internet traffic that are almost independent on the access speed were identified. Among them are the session duration, the shape of the up- and downstream session volume, the shape of the TCP-connection volume and the TCP connection data rate. The correlation of the up- and downstream data rates is influenced only marginally by the access speed. The traffic volume and the obtained data rates were found to increase proportionally to the access speed. Nevertheless, the similar shape of the distributions should be helpful when developing a parameterized Internet traffic model. Special attention should be turned to interactive services stagnating and to those new services enabled by increasing access speed.

2.4.3 Validation of self-similarity

The presence of self-similarity in data traffic is undisputed, and many papers within the action COST-257 have gone further in studying the broader theory of multi-fractals. The simplicity and parsimony of self-similarity lends itself well to practical applications in network dimensioning and traffic analysis.

For most data networks, network traffic statistics are recorded on a 15-minute timescale. These coarse measurements significantly underestimate the actual short-term peaks in the traffic. The natural question is therefore, how can we use these coarse statistics to estimate the traffic behaviour at a 1-second timescale.

In [153] a simple tool for estimating short-term peaks from coarse network statistics is presented. It is shown from measurements on two of BT's trial ADSL networks that the timescales over which the theory of self-similarity is valid allows its practical use in real networks.

Knowing the mean traffic rate m, and the variance v_t at any timescale t, one estimates the peak p_t over that timescale as the mean plus a multiple of the standard deviation $p = m + \alpha\sqrt{v}$. From the slowing decaying variance property of self-similar traffic, $v_t = v_1 t^{(2H-2)}$, one can estimate the 1-second peak from the 15-minute peak using the equation: $p_1 \approx m + (p_{900} - m)900^{(1-H)}$. In [153] this assertion was validated using traffic measurements.

The obtained results, based on the theory of self-similarity, suggest a very simple tool for estimating short-term traffic peaks in data traffic from coarse network statistics. Initial fine-timescale measurements are still required, however, to estimate the Hurst parameter. Using traffic measurements from two IP/ADSL trials, it has been shown in [153] that this technique gives remarkably accurate results, subject to network statistics being no coarser than every 15 minutes.

2.5 WWW traffic measurement and analysis

The description of WWW client traffic for simulation and analysis purposes requires simple and abstract models capturing different levels of abstraction and different time scales. Such models are derived in [46] and [271] based on measurement studies and a related data analysis.

In [119] WWW server traffic measured in an Ethernet segment connected to the Internet is analyzed by a similar hierarchical model. Its characteristics at the session and packet level are determined and the parameters of the corresponding distributions are estimated by means of the gathered data. Applying a simulation approach, the derived traffic model is used to study the WWW data transfer over an ATM network.

2.5.1 Measurement and modeling of WWW traffic

The model presented in [272, 271] is derived from measured IP traffic. The measurement study is based on a two-week measurement of WWW traffic in an Ethernet segment of the Department of Computer Science at the University of Würzburg. About 20 workstations including 1 file server and 2 WWW servers were connected to this segment. A packet trace was captured with the tool TCPDump and analyzed to identify the components of WWW sessions. The data are described by two basic characteristics and four related random variables: the characteristics of sub-sessions, i.e. the size of a sub-session in bytes and the duration of a sub-session in seconds, as well as the characteristics of the transferred WWW pages, i.e. the size of the response in bytes and the inter-response time in seconds were measured. They exhibit a heavy-tail behaviour. A timeout mechanism was used to discern sub-sessions (timeout 15min) and transferred WWW pages (timeout 3sec).

Responses to WWW requests are identified as the main part of the transferred data and the time between these responses is used to model the relation between the responses. A truncated Pareto distribution is derived as basic statistical model and its parameters are estimated to describe these real data.

2.5.2 Data analysis of WWW traffic

To illustrate the power of the nonparametric estimation approach sketched in Section 2.2.3, the real data of WWW client traffic described previously were analyzed in [164]. A maximum likelihood as well as a Hill estimator (cf. [101]) are applied to estimate the parameters of an exponential and a Pareto distribution as models of the basic characteristics, e.g. the response sizes and inter-response times of WWW responses. However, both models fail a Kolmogorov-Smirnov test. Therefore, polygram and Parzen-Rosenblatt estimates are applied to reconstruct the densities of the underlying basic random variables (see Section 2.2.3 - cf. [164]).

2.6 Traffic characterization of Internet services

To determine the traffic demand and quality of service requirements of existing and new Internet services, a thorough understanding of the nature of the corresponding user traffic and its characterization by stochastic models is an important teletraffic issue.

2.6.1 Hierarchical modeling of Internet services

In [151] the user traffic associated with new Internet services such as multimedia encyclopedia, teleconferences, teleteaching, video-playback and virtual-world services are investigated by a hierarchical event-oriented model. The latter distinguishes between the levels of sessions, connections, transactions and packets. It classifies the traffic of each dominant protocol used by a service according to the two categories of elastic or stream traffic-flows. The proposed methodology provides a traffic characterization at all four levels by corresponding metrics, e.g. the number of active connections, the mean bit rate per connection, the interarrival times of transactions or packets and the transaction or packet sizes. Their related distributions with the associated parameters are derived from measurements.

In [151] the proposed methodology is illustrated by a data analysis of the user traffic measured in a LAN environment. Using these data of each service type, the traffic category, the rate variation of the resulting flows as well as the light- and heavy-tailed distributions and related parameters of each traffic characteristic at the different levels of the service model are determined, e.g. Pareto distributed transaction sizes for the elastic flows of encyclopedia and virtual-world services.

2.6.2 Measurements and analysis of interactive video traffic

Interactive video traffic is generated by video conferencing applications which offer a variety of settings via an interface to their users. Among those settings are parameters like quality, frame rate or rate control. They are used to determine the visual quality, the target number of frames per second and the maximum bandwidth of the application. In [60] the influence of those settings on the traffic characteristics like the frame length distribution, the coefficient of autocorrelation and the Hurst parameter as well as the number of frames per second really generated are evaluated and compared to the well-known results [231] of streaming video (MPEG). The outcome of those evaluations can be used as a basis for network dimensioning as done in [60, 61] or in order to develop a traffic model.

Chapter 3
Traffic Characterization

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3.1 Introduction

Understanding the nature of traffic, identifying its characteristics and building practical models are vital for the teletraffic engineering of today's packet switched networks. New observations of measured traffic call for new approaches (e.g. multifractal characterization) and the ever changing services and protocols of the Internet trigger particular models (e.g. WWW models).

This chapter overviews the traffic characterization and modeling activities of the COST-257 project. Section 3.2 presents different models developed for packet traffic including discrete time models, Gaussian models and multifractal models. Models related to particular services (internet dial-up traffic) and protocols (WWW traffic, TCP traffic) are summarized in section 3.3. Section 3.4 discusses the statistical methods related to traffic models and, finally section 3.5 addresses the issue of traffic generation.

For more details see the chapter on "Source Characterization in Broadband Networks" in the COST-257 Interim Report [178] and the referred COST TDs.

3.2 Models for packet traffic

3.2.1 Discrete time models

Discrete time queueing models are of inherent interest in modelling communication systems. Many such systems are timed or synchronous with an advance of certain information units at every time slot or clock cycle. Models of this type are especially important when modelling ATM systems but are also relevant for the modelling of other systems. A general reference to these types of systems can be found in [34]. The work within this area in the COST-257 project has partly focused on general models and partly focused on models dedicated to the modelling of highly variable packet traffic.

In [136] a framework for the modelling and analysis of a superposition of ON-OFF sources in discrete time is considered. The sources are allowed to have a general distribution while successive ON and OFF periods are independent. Second order descriptors for the process of generated cells and results for the superposition of N of these ON-OFF sources are derived. The superposition of N sources of this kind is then analysed under the assumption that the duration of the OFF periods is geometric.

The work in [136] primarily addresses light-tailed sources. In [137] the case of sources with heavy-tailed ON or OFF sources is examined. A special form of hyper-geometric distribution was applied to model the heavy tails of these distributions. When considering a certain limiting construction with heavy-tailed ON periods and geometrically distributed OFF periods the model can be seen as a discrete analogue of the continuous M/G/∞ system which has been used for the modelling of long-range dependence in a continuous setting. An outline of a possible queueing analysis is given but no numerical results are obtained.

The paper [55] is related. The approach followed in this paper leads to a different way of obtaining a discrete version of the M/G/∞ arrival model. In each time slot there is a Poisson distributed number of arrivals. Each of these arrivals consists of a batch which has a size described by a Pareto distribution. This process is fed into a single server queue and it is shown that the queue length distribution asymptotically behaves like a power law distribution. It is shown by numerical simulation that the asymptotic behaviour is a very good approximation over the whole range of values for the buffer occupancy.

A process obtained by a superposition of an infinite number of simple ON-OFF sources is analysed in [54]. It is shown that the resulting process exhibits long-range dependence and by matrix analytic methods it is argued that the queue is unstable for certain values of the parameters in the sequence of ON-OFF sources.

Another arrival model, not directly related to highly variable traffic is described in [174]. This work characterises the output process of a discrete time queue with deterministic service fed by a batch renewal process. A joint transform for the duration of the idle and busy periods in successive busy cycles is derived.

3.2.2 Gaussian traffic models

By the Central Limit Theorem, the sum of a large number of "small" independent random variables has an approximately Gaussian (normal) distribution. Thus, one can expect that an aggregated teletraffic stream consisting of a large number of individual communications can be reasonably well modelled by a Gaussian stochastic process.

A continuous time Gaussian process with stationary increments is a process $(A_t : t \in \mathbb{R})$ such that for any $t_1, \ldots, t_k$ and s, the random vector $(A_{t_1} - A_s, \ldots, A_{t_k} - A_s)$ has a multivariate Gaussian distribution independent of s. For normalization, we set $A_0 \overset{\text{a.s.}}{=} 0$. Such a process is characterized by a number m and a symmetric non-negative function $v(t)$ such that

$$\mathrm{E}A_t = mt, \quad \mathrm{Var}\,(A_t) = v(t).$$

The covariance function of A is then given by

$$\mathrm{Cov}\,(A_s, A_t) = \Gamma(s,t) = \frac{1}{2}(v(s) + v(t) - v(t-s)). \tag{3.1}$$

In teletraffic modelling, m is called the mean rate, and $v(t)$ is the cumulative variance function. If A and B are two independent processes, we have $v_{A+B}(t) = v_A(t) + v_B(t)$. This means that it is very simple to deal with complicated traffic mixes with different dependence structures, if the Gaussian approximation is good enough.

The general case includes processes whose paths have unbounded variation. Basic examples include Brownian motion, used in diffusion approximations to queues, and fractional Brownian motion, a self-similar process used as a model for long-range dependent traffic. On the other hand, it is often interesting to model traffic by a Gaussian process with smooth paths, i.e. as a Gaussian fluid process. In that case, A can be written as

$$A_t = \int_0^t \Lambda_s \,\mathrm{d}s,$$

where Λ, the rate process, is a stationary Gaussian process. For more details, see [3].

Queueing theory with Gaussian input is discussed in subsection 6.3.4.

Other COST-257 work related to Gaussian traffic models: [277] presents a new estimation technique in the case of fBm traffic. A new simulation technique for general Gaussian traffic is described in [195]. A "handbook" of fBm formulae was compiled in [196]. A summary of COST-257 work with Gaussian traffic is included in the hypertext version of this document.

3.2.3 Multifractal analysis and modelling

Multifractal analysis is a mathematical discipline that studies a singular measure μ through local Hölder exponents

$$\alpha(x) = \lim_{\epsilon \downarrow 0} \frac{\log \mu(B(x,\epsilon))}{\log \lambda(B(x,\epsilon))} \quad \text{(if the limit exists)}, \tag{3.2}$$

where λ denotes Lebesgue measure and $B(x, \epsilon)$ is an open ball. Multifractal analysis is also a data analysis method that studies the statistics of approximate, finite resolution Hölder exponents computed from the data.

The data, interpreted in this case as a measure μ on time interval $[0, t]$, are said to possess multifractal scaling, if the partition function

$$S_k(\eta) = \sum_{i=1}^{2^k} \mu\left(\left(\frac{i-1}{2^k}, \frac{i}{2^k}\right]\right)^{\eta}$$

is linear with respect to k in log-log-plot for η in some relatively large interval $[\eta_{\min}, \eta_{\max}]$. Riedi and Levy Vehel [221] observed that network traffic has, at least in their data, very good multifractal scaling properties. They computed so-called multifractal spectra, which showed that the character of short time scale burstiness of real traffic differs from models used that far, including the self-similar ones that can be called "monofractal". This also inspired some work in COST-257.

The multifractal scaling property was indeed found in data from the Finnish University Network, see [158, 157]. The latter TD shows also that popular traffic models yield in multifractal analysis different spectra than real traffic.

In hope to find a logically simple prototype model for traffic with multifractal characteristics, the following class of processes was defined and analysed in [159]. Let Λ_t be a stationary Markov process taking only finitely many

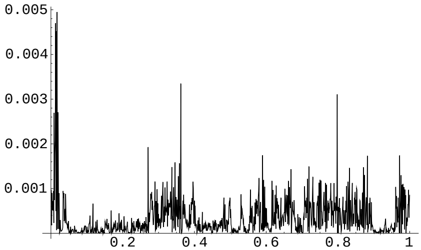

Figure 3.1: Realization of increments of $A^{(7)}$ when Λ is a two-state Markov process with transition intensities $\nu_1 = 2$, $\nu_2 = 1/2$, transmission rates $S_1 = 1/3$, $S_2 = 7/6$, and $b = 4$. The resolution is 0.001.

values that are all positive and such that $\mathrm{E}\Lambda_t = 1$, and let $b > 1$. Define the increasing processes

$$A_t^{(n)} = \int_0^t \prod_{i=0}^{n} \Lambda_{b^i t}^{(i)} \,\mathrm{d}t. \tag{3.3}$$

If there exists a non-degenerate limit process as $n \to \infty$, this process has very well-behaving multifractal properties, besides having stationary increments. As regards the existence, it is proved in [159] that $A^{(\infty)}$ exists as an L^2 limit of $A^{(n)}$ if and only if $b > \mathrm{E}\Lambda_t^2$. As an example what these processes look like, figure 3.1 shows the increments of a realization of $A^{(7)}$.

3.3 Models for particular services and protocols

3.3.1 Models for Internet dial-up traffic

A popular way of accessing the Internet is to use the public telephone network with modems or ISDN dial-up connections. In [74, 75] scalable models were developed for dimensioning purposes based on comprehensive traffic measurements and analysis study of dial-up traffic. It has been found that dial-up sessions are much longer than classical telephone sessions, and their length depending on the tariffing scheme. The session holding time and session inter-arrival time exhibit high variability over the whole day. However, for modeling purposes during busy hours a renewal process with a hyperexponential distribution with only the first two moments matched is an adequate model with respect to loss behaviour.

3.3.2 Models for Internet user traffic

The modelling of Internet traffic is one of the main targets of recent teletraffic research. The contribution of the World Wide Web (WWW) has experienced an extreme increase in the last decade and its expected future dominance called for a number of studies in COST-257 [119, 271, 151].

The characteristics of recorded WWW traffic was analyzed in [119] considering the IP and session arrival statistics. This work also studies the impact of WWW traffic when ATM is the underlying transport mechanism. Based on the traffic characterization results a simulation study has been carried out to investigate the queueing behaviour. The main finding of this study is that variations on long time scales are not present in the queueing system if dynamic capacity allocation is used. More precisely, if the capacity of the server is increased with an amount proportional to the mean rate of the new TCP connection, the long-term correlations disappear when this TCP connection is opened. From a traffic modelling point of view this is advantageous since the application of short-range dependent models (e.g. Markovian models), which are analytically more tractable than long-range dependent models (e.g. fractal models), is adequate in these cases. However, the practical application of such a scheme is still an unsolved issue since the information needed for optimal capacity allocation is unknown until the connection is closed.

In [271] a simple model for a WWW session is constructed based on a measurement and analysis study on a local Ethernet segment. It turned out that both the distribution of the response size and the distribution of the inter-response time have the heavy-tailed property and both quantities are well modelled by the Pareto distribution. Moreover, the samples of both values are found to be independent. The model is validated by the simulated data transmission over an ATM link utilizing VBR service category. The validation of this model also shows that it exhibits stronger short-term dependencies and lacks the long-range dependencies.

A study of different Internet services is presented in [151]. The traffic, which can be categorized to be stream or elastic traffic, generated by different services (e.g. encyclopedia, tele-conference, tele-education, video player, virtual world) is characterized based on the characteristics of relevant events (sessions, connections, transactions, packets). The study provides guidelines about the appropriate choice of distributions and parameters for building appropriate traffic models.

3.3.3 Models of TCP traffic

Teletraffic theory has been quite succesful in modelling communication systems during the 20th century. The success is partly due to the possibility of separating the modelling of telecommunication systems as a number of arrival streams that can be treated as input to rather complex service systems. The very nature of TCP invalidates this approach due to the inherent feedback from the network model to the source model. This together with the increasing importance of the Internet has created a demand for traffic models specifically addressed to TCP traffic. Ideally such a model should reflect the feedback mechanism which is introduced by the flow control algorithm. This research field is still quite immature in spite of its importance.

Some work in the area has been performed within the COST-257 project. The work reported in [40] partially addresses the feed back problem by modelling first order effects. The models consist of a number of Markovian TCP source models and a TCP network model. The latter is a single server queue with limited buffer space and deterministic server. The probability of loss is calculated with a Poisson process as input process. The TCP source model is an approximate model of TCP Tahoe with slow start and congestion avoidance. The calibration of the two models is done by adjusting the segment loss probability. The segment loss probability influences the total load offered by the TCP model and that load in turn specifies the loss probability in the network model. The two models are solved iteratively until the segment loss probability in the two models agree.

3.4 Statistical methods related to traffic models

The applicability of source models is ultimately tested by comparison with measurements. The statistical problem of actually fitting parameters to source models is generally quite challenging. For this reason a significant amount of work within this area focus on special properties of the traffic processes typically but not restricted to first and second order properties of the measured traffic and the abstract models.

Several methods for estimation of the important parameter H related to self-similar traffic models are discussed in [181]. These methods are validated through the analysis of a number of traces obtained by measurements in the ATM network of Telecom Finland. The conclusion of the paper is that the estimation of the Hurst parameter is highly sensitive but more important is that the Hurst parameter probably is meaningful only when considering pure self-similar traffic (that is traffic well described by fractional Brownian motion).

The study [180] addresses the importance of long-range dependence with respect to network engineering. The study is performed by several simulation experiments based on data obtained from measurements of the ATM network of Telecom Finland. The conclusion of the paper is that there is a certain time scale of importance, the size of which is related to the buffer size of the system to be engineered. The actual scaling factor is left for further research.

A related study [8] questions the usefulness of certain random permutations that have been used [71] for establishing the effect of long-range dependence for queueing systems. Two kinds of random permutations, the so called internal and external shuffling are considered. It is shown in [8] that the former more or less preserves both short and long term correlations while the latter reduces short term correlations significantly. The results indicate that it is very difficult to use shuffling as a means to investigate the importance of long-range dependence on queueing performance.

One important second order descriptor of an arrival process is the peakedness functional [64, 161]. This descriptor is analysed in detail for discrete time processes in [179]. The refinement necessary for discrete time is performed and peakedness is calculated for a number of important discrete time models. The potential for the application of peakedness as a powerful descriptor is illustrated by several examples of real traces.

A standard method in performance modeling is to analyse a system with an aggregated stream, which is obtained from a superposition of simple ON-OFF sources. An inherent problem of this approach is the dimension of the aggregated system. This problem has been addressed in [248]. The approach is to fit the spectral properties of the superposition with a model of lower dimension. The approach is well known in continuous time but this work modifies the method for discrete time. The method is applied to the superposition of MPEG video sources and the results obtained are validated through experiments. The results of this experiment are quite promising.

Most traffic models have an emphasis on second order properties of the counting process. There are several equivalent ways of describing second order properties like the Index of Dispersion for Counts, the autocorrelation function of the intensity, the Palm function, peakedness, and spectral analysis of the autocorrelation function. The results of [7] analyse the sensitivity of other properties when the point process is used as the input process to a queue. It is shown that neither second order properties of counts nor of intervals are sufficient to predict queueing behaviour. It is concluded that in order to rely on second order information it is necessary to have more knowledge as can be obtained from measurements.

The investigation of modem users of a dial-in modem pool is carried out in [273, 274]. It is found that session duration and asymmetry between up and down stream traffic volumes are surprisingly independent for varying modem speed. The traffic volume and the obtained data rates were found to increase proportionally to the access speed. This study is relevant in the design of asymmetric access methods like ADSL, since it shows that the degree of asymmetry must not be too high.

3.5 Traffic generation algorithms

A fast algorithm called RMD$_{mn}$ for generating Gaussian traffic was presented in [195] and implemented as a freely distributable C program in the case of fractional Brownian motion. The idea is to improve the well-known Random Midpoint Displacement algorithm (see figure 3.2), exact only for Brownian motion, by conditioning to the (at most) m nearest already generated neighbor values on the left and n on the right. Experiments showed that m and n can be quite small, for example 4 gives excellent traces of fBm. For a general

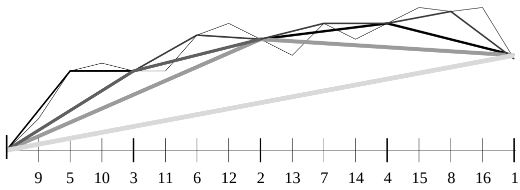

Figure 3.2: Random Midpoint Displacement algorithm for generating the path of a Gaussian process top-down. The numbers show the order of generated points.

Gaussian process with stationary increments, the algorithm takes as input the variance function (see page 45). Although the algorithm has a top-down character, a version proceeding indefinitely in time has been designed and made available. The memory requirement grows only logarithmically w.r.t. the number of generated points.

Chapter 4
Internet Performance

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4.1 Introduction

This chapter gives an overview of the Internet activities in the COST-257 project. Section 4.2 presents a short overview of the analysis of Internet traffic. Section 4.3 describes the different aspects of Quality of Service (QoS) in the context of traffic streams. The section starts by defining traffic aggregates such as stream based or elastic traffic. First the traffic aggregates themselves are investigated, followed by the consequences of their integration for network and resource dimensioning.

The impact of different pricing schemes for the QoS of different traffic classes is studied in Section 4.4, followed by a detailed investigation of the differentiated services (DiffServ) concept of the IETF in Section 4.5. Switching and routing with all the important aspects such as DWDM, MPLS and QoS routing is addressed in Section 4.6. The Chapter ends with an overview of the COST-257 contribution to the integration of IP and ATM networks and some comments on server dimensioning.

4.2 Measurement, characterization and analysis of Internet traffic

Network traffic measurement, characterization and analysis has been considered as a necessary activity since the early days of networking. One main objective of these activities is the dimensioning of networks. Dimensioning the Internet is a hard task because of network size, large traffic volumes, different transmission technologies and distributed administration. Nevertheless traffic measurement and characterization is definitely needed for traffic engineering and controlling to ensure the appropriate QoS for future applications such as IP telephony. The interested reader is referred to Chapter 2, where the contribution of COST-257 to this field of research is presented in more detail.

Generally speaking the authors in [74, 273, 75, 274, 125, 124, 151] characterize the user behaviour and the traffic distribution of different applications for today's access networks. The most used application in the Internet – WWW – is investigated in [119, 271, 46]. [119] describes a study of WWW traffic and investigates a special scenario where ATM is used as the underlying transport mechanism. A more general approach for deriving an abstract model from measured WWW traffic, which could be used to describe WWW-traffic for analysis and simulation purposes, is presented in [271] and [46].

4.3 Quality of service aspects

4.3.1 Traffic aggregates

Considering that the even-growing, large number of flows in the Internet cannot be handled any more by mechanisms dealing with per-flow awareness, the description of aggregated traffic and its performance becomes more and more important. Developing QoS architectures such as DiffServ and MPLS which inherently support traffic aggregation, together with the demand for a small number of classes, further increase the need to describe losses, jitter and delay variation of various traffic aggregates.

General traffic

The most general evaluations of aggregated traffic characteristics [194] discussed here yield performance formulae for a single queue serving Gaussian traffic. A method to estimate the parameters needed for fractional Brownian traffic is described in [277] and introduced in Section 2.3. The cumulative arrival process to the introduced queue is characterized by continuous Gaussian traffic. According to the large deviation principle for Gaussian processes which is known as (generalized) Schilder's theorem, the most probable path for a given storage occupancy is calculated using a approximation. By recalling it in order to accurately approximate the idle probability of the buffer, the results are close to simulation results of the Gaussian counterpart of the Anick-Mitra-Sondhi model [9] over the whole range of observed buffer levels. Norros [3] extended this approach by introducing classes of Gaussian traffic being served by priority queues.

Elastic traffic

Taking into account the large proportion of elastic traffic in the Internet, it is important to focus on specific attributes of aggregated TCP traffic. The question arising hereby is whether it is possible to abstract the complicated mechanisms of the TCP protocol without loosing the ability to describe the main effects happening in the network.

Bonald and Roberts [28] evaluate a simple link level model of elastic flows in an access (resp. backbone) link environment, where different scheduling mechanisms are applied for different classes of TCP traffic. The authors assume an offered load below 1 and fair bandwidth sharing between all flows and evaluate the throughput of single flows as well as the traffic aggregate dependent on the offered load. The analytical models to obtain the results are the M/G/1-PS and M/G/m-PS queues, for an access link whose bandwidth can be totally used by a flow, and a backbone link whose bandwidth can only be partly used by a single flow, respectively. The results show that on an access link, it is best to use a fair queueing scheduler, as heavy-tailed distributed file lengths can significantly degrade the service of the low priority classes. For a backbone link, the authors state that a simple priority queueing mechanism is sufficient to provide QoS for all flows of both classes.

Another approach to describe the traffic of aggregated elastic traffic can be found in [40] where Casetti and Meo modeled the stationary behaviour of several classes of aggregated TCP connections by dividing the system in two parts: TCP sources and the network. The model of each part is analysed separately while the impact of the two models on each other is considered in an iterative way. The TCP part of the model is hierarchical with an application level characterized by negative-exponentially distributed on-off activity periods and a Markov chain which represents the TCP window dynamics. Using queueing network analysis, the network is modeled as an M/D/1/B queue and tandem queue, respectively. Several key performance issues of TCP, such as throughput, segment loss probability and queueing delay are obtained and compared with simulations. The accurate results under most circumstances justify the large number of approximations and yield a simple methodology to estimate several key performance issues of aggregated TCP traffic.

The effect of transporting aggregated TCP flows accessing an ATM network over xDSL modems is evaluated by Elizondo-Armengol, Gavilanes-Plasencia and Sanchez-Canabate [65]. The influence of sharing an output link of an ATM switch is evaluated by simulation studies. Thereby especially mean delay, cell delay variation, cell loss probability and goodput are obtained.

Stream traffic

Stream traffic summarizes applications with stringent timing requirements, such as audio or interactive video traffic. Whereas audio traffic can be easily modeled, interactive video (e.g. video conferencing) depends on a variety of parameters that can be adjusted by the user [60]. As the generated traffic as well as the QoS requirements are different from elastic traffic, other traffic models have to be applied to evaluate the QoS.

Mandjes, van der Wal, Kooij and Bastiaansen [155] carried out an analysis of stream traffic modeled as CBR sources, focusing on the end-to-end performance in large IP networks dependent on the access protocol. The network part is modeled as a convolution of several N*D/D/1 or M/D/1 stages, respectively. It turns out, that the first network representation tends to be optimistic (lower bound) while the latter is pessimistic and, thus, represents a statistical upper bound. The results show the gross bit rate achieved for different interactive applications in dependency if the end-to-end delay with a variety of access protocols as parameter. Dependent on the speed of the access network, different protocols are recommended, but the critical end-to-end delay bound of interactive applications can be met in any case.

Menth [173] analysed performance characteristics of wireless traffic that is multiplexed over IP using the real time protocol. The author carried out a discrete-time analysis using a framework for solving discrete and finite Markov chains with some new extensions. It is shown that the bandwidth requirements can be significantly reduced when applying multiplexing instead of tunneling. Furthermore there exists an optimum for the timer value that determines how long to wait for arriving traffic. Finally comparisons with known results from AAL2 in ATM are presented.

An analysis of aggregated real-time CBR sources is carried out by Burakowski, Fudala and Tarasiuk [35]. To separate stream and elastic traffic a priority queueing (PQ) resp. weighted fair queueing (WFQ) scheduler is used. Results for mean and maximum packet delay, as well as packet loss ratio are obtained. The authors show that the delays are smaller with PQ while the losses stay about the same.

Integration of stream and elastic traffic

The evaluation of a whole network requires the consideration of both stream as well as elastic traffic. The interesting question arising in this context is the overall bandwidth need in order to satisfy the QoS requirements of both classes.

Dolzer and Payer carry out simulation studies (see [61] and [60]) in order to find the optimum number of classes needed to isolate flows with different timing requirements, while keeping the network simple and still obtaining a good multiplexing gain. As the focus of the studies is on a realistic environment with realistic sources, traces of real-time video traffic - whose the characteristics are further discussed in Section 2.6.2 - together with audio and worst case data sources, are used as input traffic. The authors show that in the evaluated enterprise network environment, the separation into two classes – stream and elastic – satisfy the QoS requirements of all traffic types if the weights at the fair queueing scheduler is allocated according to the effective bandwidths.

Nunez-Queija, van den Berg and Mandjes [127] evaluate different strategies for the integration of stream and elastic traffic in a single multiservice network. The authors show that it is network cost beneficial to either integrate the two classes with certain guarantees or just to mix them, compared to the segregation into two different networks. In the segregation case, the Erlang loss model and the M/G/1/K-PS model were used respectively for stream and elastic traffic. In case of a single multiservice network the performance characteristics are obtained by a Markov process as Poisson arrivals and exponentially distributed file length are assumed.

4.3.2 Network and resource dimensioning

As mentioned in the previous section, appropriate dimensioning of the network is a key issue for all QoS aware networks. A widely applied approach is to calculate the bandwidth that should be allocated to a set of flows in order to satisfy their QoS requirements. This bandwidth is called effective bandwidth and has been extensively studied in the past (e.g. [9]). Nevertheless it is important to apply known methods and formulae to new problems and applications in QoS aware IP networks.

Fiedler [77] presents a closed form formula for direct evaluation of the required capacity for a given QoS per connection which is obtained from the well-known loss probability formula of the bufferless fluid flow model. The sources are modeled as on-off sources which are characterized by mean and peak cell rates. As the state space can get huge, several possibilities to cope with this problem are discussed. Finally a comparison with the formula for the required capacity presented in COST-242 [228] shows that under certain circumstances the direct evaluated formula should be applied to avoid underestimation of bandwidth.

In [82] Fiedler and Voos compare different possibilities for overcoming the numerically instability of the finite buffer stochastic fluid flow model. They conclude, that also for large buffers the system can be handled if a set of rules presented in the paper are considered.

The “fun factor” is introduced by Charzinski [47] as a simple and intuitive measure of the QoS that is achieved in a TCP connection relative to a theoretically possible value. Hereby two equivalent formulae considering time and bandwidth, respectively, are presented. A comparison is carried out with a variety of existing models for dimensioning of the required bandwidth. The models studied are the Engset fluid flow, fractional Brownian motion, M/G/r-PS and rate convolution. The results show that the fun factor is not only an easily understood parameter because of its direct impact on the perceived QoS, but is also suitable to be used for network dimensioning as $1 - (funfactor)$ behaves roughly like a target loss probability.

4.3.3 Admission control

In the network control Chapter 1 a lot of interesting approaches and methods for admission control are presented. Most of theses methods can be taken into consideration for a future internet with more than one traffic class, e.g. stream and elastic traffic. In particular, admission control concepts based on measurements are applicable for an internet environment with different classes. In [279] Vargas and Sanchez-Canabate investigate the effects of measurement-based admission control (MBAC) particularly on stream based traffic. A more general overview is given in [264]. Starting with this overview, van den Berg and Mandjes selected two MBAC algorithms and studied their performance under various traffic scenarios.

4.4 Performance through charging (pricing)

Providing QoS guarantees concerning transparency, accessibility and throughput by means of transaction pricing is addressed in Roberts [226]. Two in principle disjunct traffic classes – stream and elastic traffic – are considered. Stream traffic flows have an intrinsic duration and rate, e.g. IP telephony, whereas elastic traffic flows consist of digital objects which must be transferred without error from one place to another like WWW or FTP traffic. To calculate the blocking probability of the streams in the proposed model, the notion of effective bandwidth for stream traffic is used, while a simple processor sharing model is used for elastic traffic. In [227] three broad categories of charging schemes are distinguished: flat rate pricing, congestion pricing and transaction pricing. With flat rate pricing users pay a fixed charge dependent on their network access line. In the congestion pricing model the users pay for the level of service they judge necessary and in the transaction pricing model users pay per transaction as in a telephone network. These three different pricing schemes are investigated considering the demands of an internet service provider with the conclusion that a service model without systematic use of admission control precludes the use of transaction pricing and may consequently jeopardize the solvability of the network provider.

Following the concept of differentiating between stream and elastic traffic, Lindberger [149] discusses dimensioning methods for each class separately, including processor sharing models for elastic traffic. The possible gain by integrating the two classes, giving stream traffic higher priority, is shown.

A completely different approach for assessing the effectiveness and stability of flow control schemes by constructing a distributed game is presented by Key and McAuley [122]. Arguing that it is possible to provide differential quality of service to users by means of a simple pricing scheme, a distributed game is constructed. The network sends back the correct congestion prices to the user, and the users are supposed to react to these prices. The exact mechanisms and protocols to play the game are presented in detail in the paper.

4.5 DiffServ mechanisms

In the IETF the DiffServ model is an intensively discussed framework, proposed to ensure some kind of statistical guarantees for differentiated traffic flows. Unlike the Integrated Service (IntServ) concept, the DiffServ concept aggregates traffic streams in classes. Nowadays three different traffic classes (Expedited Forwarding (EF), Assured Forwarding (AF) and Best Effort) are defined through per hop behaviour (PHB) in the routers. Inside these classes a further differentiation between flows based on drop probabilities is possible.

One of the predecessors of DiffServ is the Simple Integrated Media Access (SIMA) concept. According to the SIMA concept each customer shall define only two parameters before a connection establishment: a nominal bit rate (NBR) and the selection between real-time and non-real-time services. Kilkki [123] introduces in detail the SIMA concept and compares the obtained service with the IntServ and the ATM service concept.

Köhler and Schäfer [129] present a simulation study of the TCP behaviour, considering that in a DiffServ environment the sender determines the class of the data packets, while the receiver independently chooses the class of the acknowledgements. The authors investigate to different scenarios. First, different classes for data and acknowledgements (ACKs) are investigated and secondly the assignment of different drop precedence for data and ACKs inside a single class is considered. The results show that the throughput of a TCP connection depends not only on the data class, but also on the correct choice for the ACKs. Some combinations of classes for data and ACKs could even lead to an unfair use of bandwidth. In the second scenario, the results show that for a high throughput the selection of the drop precedence is in most cases important for data packets only.

In the DiffServ framework random early detection (RED) queues have been considered as a way to efficiently perform active queue management. Peeters and Blondia [207] present a discrete-time analysis of a RED queue with responsive traffic sources and derive the loss ratio and throughput of the system. Köhler, Menth and Vicari [128] expand this analysis and evaluate the performance of TCP in conjunction with the RED algorithm in order to get insights in the behavior of TCP at a RED queue, as well as hints on proper dimensioning of RED queues. By modeling salient features of TCP and RED with discrete-time analysis, the authors obtain performance measures like the distribution of the TCP congestion window size or the averaged RED queue occupancy. The results show that the parameters of a RED queue have a strong impact for the TCP performance and a wrong setting may lead to drawbacks instead of benefits from this mechanism.

IP link dimensioning models in the context of the DiffServ scheme are presented by Olivier and Roberts [200]. The models are based on the simplifying assumption of fixed resource sharing among classes. For the EF-PHB a flow-based admission control to translate the peak rate guarantee of the EF aggregated traffic into a rate guarantee for each connection is proposed. A multi-rate model is used for performance evaluation and the per class flow blocking probabilities are estimated by a generalization of the Erlang loss function. For the AF class an allocated portion of bandwidth through an appropriate queueing discipline is assumed which leads to a lower bound for the obtained performance indicators. Additionally a preliminary analysis at packet level based on a Markovian M/M/1/1+K model is undertaken.

4.6 Switching and routing

In the last two years switching in the form of MPLS has seen a revival in the Internet community. Based on the speed of ATM switches, the PNNI routing concept and new ideas of the MPLS working group (e.g. label stacks), the connectionless principle of the IP routing paradigm is changing. All these changes lead to a very interesting question in network research: How should the conjunction of MPLS, routing and security in IP networks be defined to reach some kind of QoS? In the following the contributions of COST-257 to the topic are presented.

4.6.1 DWDM and switching

Mannersalo, Hämäläinen and Norros [156] present a simulation study on IP switching, using the ALIAS simulator. They study how switching IP over ATM proceeds along a network and how the performance of the corresponding router network is improved. The results show that IP switching is efficient when the network is highly loaded. Additional studies of IP packet reordering, the queue length in the switch and the congestion behaviour of TCP are presented.

The problem of wavelength assignment in multifiber wavelength division multiplexing (WDM) networks is investigated in [102]. In WDM networks several optical signals are transferred in a single optical fiber. Hyytiä and Virtamo point out that for a single fiber case the wavelength assignment could be mapped into a graph coloring problem once the routes are fixed. With multiple-fibers the problem could not be mapped to a coloring problem and thus, the authors study an iterative algorithm for choosing the routes.

4.6.2 QoS by routing

QoS routing or constrainet-based routing is intended to find paths in the IP network meeting the QoS requirements of a flow or an aggregation of flows, while using the network resources efficiently. The QoS routing task is very complex for several reasons, which are mainly related to the strong uncertainty on the network topology and state, the extension of the network and the heterogeneity of the network nodes.

However, to make the implementation of QoS routing feasible it is mandatory to keep complexity at a low level. For this purpose, one preliminary question concerns the possibility of using existing Internet protocols for managing QoS. In [99] an overview is given of the existing routing schemes and their properties concerning the introduction of QoS in IP networks. The overview addresses both the Interior Gateway Protocol and the Exterior Gateway Protocol, with particular attention to the (E)IGRP, a proprietary CISCO routing protocol. Since the most promising routing protocols are OSPF and EIGRP an optimization procedure is presented for setting their parameters in order to share the network load homogeneously [250].

The information needed by the routing algorithm is essentially the network topology and the resource availability; in particular the available bandwidth of all links is required. This latter set is a huge amount of information because the available bandwidth is a very variable quantity. To reduce the amount and the frequency of link state advertisements (LSAs), topological aggregation methods and frequency reduction criteria have been proposed [10]. Obviously these measures introduce a further uncertainty whose negative effects have to be balanced with the reduction of the advertised information.

The complexity of route selection depends on the metrics chosen in relation to the required QoS. Common metrics are bandwidth, hop count, cost, delay, etc. Routing algorithms selecting paths considering more than one metric are called multiple metric algorithms. Metrics can be additive, e.g. hop count, multiplicative, e.g. blocking probability and concave, e.g. available bandwidth.

Among the variety of metrics usable in Internet QoS routing, hop count and bandwidth are considered at the moment to be the most important metrics, since they implicitly take care of delay requirements. In general, the optimal routing problem with multiple constrains is an NP-complete problem. However, if constraints concern metrics that are not independent it is possible to find polynomial algorithms for the solution.

”Shortest distance path” algorithms select the shortest path in a graph where the link length is equal to the inverse of the available link bandwidth. With this link metric, the algorithm favors paths with a minimum number of hops when the network is under-loaded and prefers maximum available bandwidth paths when the load increases. Furthermore the method can be implemented using existing Internet routing protocols and routers.

In a different approach the connection blocking probability can be used as a QoS parameter, under the premise that CAC procedures are applied for connections requiring a given QoS and it is possible to define a required bandwidth for the same connections. If it is possible to apply a classical Poisson model to the connection arrival process, the algorithm proposed in [70] can reach the high performance of algorithms based on Markov decision processes with a lower monitoring effort and computational complexity. The algorithm proposed in [69] is more general and does not assume any specific traffic model yet still reaches good performance on the base of stochastic comparison theory.

4.6.3 Multicast routing

Besides unicast routing the importance of multicast routing algorithms and protocols is growing. In an IP multicast environment, IP multicast traffic for a particular (source, destination group) pair is transmitted from the source to the receivers via a spanning tree that connects all the hosts in the group. Once the tree is constructed, all multicast traffic is distributed over it. Because most of the multicast traffic is produced by multimedia applications, the determination of QoS parameters is important. In COST-257 different approaches [121, 198, 1] for calculating blocking probabilities in a typical multicast network are presented.

4.7 IP and ATM interaction

The ADSL transmission technology based on ATM technology is considered to play an important role in future access network scenarios. Also many existing WAN backbones are based on ATM technology. Thus the transport of IP packets over ATM is investigated.

Several ATM service categories with various parameters could be used for the transport of TCP data over ATM. The performance of the SBR.3 service category to transport TCP data with different bandwidth requirements is studied in [65]. The results show, that the standard CAC method used is not well suited for the characteristics of the TCP traffic and that the parameters of the TCP users have to be revised.

A comparison of different ATM service categories for the transport of TCP/IP over ATM is presented in [44]. While UBR and GFR rely on the feedback control of TCP a second control loop is added with ABR. The advances and problems, depending on the actual network configuration, are showed for each of the transmission capabilities. A method to provide explicit congestion notification (ECN) for packet switched traffic is presented in [5]. The algorithm is designed to operate in a homogeneous Internet environment as well as in the IP over ATM scenario using the ABR transfer capability.

The guaranteed frame rate (GFR) service category guarantees the transmission of a minimum portion of the traffic and hereby prevents the loss of single ATM cells that are fragmented IP packets. A discrete-time analysis is presented in [224, 275] giving insight to the dimensioning of GFR parameters.

4.8 Server dimensioning

One of the important tasks in the process of designing a good network, is the dimensioning of the server environment. Van der Mei, Hariharan and Reeser [268] present an end-to-end queuing model for the performance of HTTP Web servers. The authors encompass in the model the impacts of client workload, server software and hardware configuration and communication protocols. A comparison of the performance prediction based on the model with a Web server in a test lab is given. In this environment the model matches very well and thus the model can be used for configuration, tuning and sizing of Web servers.

A detailed modeling and experimental characterization study of TCP/IP software implementations is given by Lepe and Garcia [145]. Using software probes, in-kernel software profiling and Intel's PerfMon counters an investigation of the TCP stack in different operation systems is performed.

Chapter 5
Wireless Networks

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5.1 Introduction

Extending network-based services to wireless mobile subscriber access was an important milestone in the global telecommunication network evolution. Spectacular success of the cellular telephone networks (e.g. GSM - *Global System of Mobile Communications*) has proven that all, today's and future, network technologies should offer wireless and mobile capabilities.

Enhancement of services supported by the wireless networks is planned to follow an evolutionary scenario comprising four main phases [185]. However, introduction of new services is conditioned by the available bandwidth of the radio access channels. Today's digital cellular mobile network belongs to the second generation ($G2$) of wireless networks and offers very low up to low bit rate communication services for voice and data. Further network evolution is on the way towards including packet data services ($G2+$), Narrowband Integrated Services Data Network (N-ISDN) capabilities ($G3$), until, from today's perspective, Broadband ISDN (B-ISDN) services are offered ($G4$).

A general wireless network architecture assumes three main levels, which are the access network, the core transport network and the service network [90]. In the access network, the base transceiver station (BTS) provides radio connectivity to the mobile stations (MS). For this purpose, a specialized radio access protocol is implemented in both the base and mobile station. Recognized solutions for such protocols are based on TDMA or CDMA schemes.

Optionally, the access network can contain concentrators (C) to connect a number of BTS to the core network. One can distinguish between two types of access networks, indoor and outdoor. In the indoor network the distance between MS and BTS is relatively small and typically limited up to 100 meters. On the contrary, the outdoor network covers rather large areas of up to few kilometers. The size of this area strongly depends on such parameters as the number of MS and signal propagation conditions.

The core network is a wide area network that is designated for transporting user and signaling information, and is typically built on cable links (although some wireless links can also be applied, like satellite or radio relay). This network could support a variety of communication services, depending on the involved technology. In general, the access and the core networks need not necessarily to be built on the same telecommunication technology. However, for supporting end-to-end network services with adequate quality of service, a common technology is strongly desirable. For instance, the complete set of broadband services requires ATM in both the access and the core network.

The service network contains highly specialized servers that support mobility management and provide application specific processing, e.g. for transcoding of speech, voice mail, message, and facsimile handling.

In wireless networks the Quality of Service (QoS) provisioning problem is more challenging than in the fixed network. This is due to the wireless channel characteristics, mainly caused by limited bandwidth and relatively high (and temporary variable) bit error rate. In order to cope with the transmission errors some protection mechanisms are required at the physical or data link layer. However, the implementation of such mechanisms produces additional overheads decreasing the effective radio link capacity. As a consequence, availability of resources dedicated by the network to a given connection could be variable. Additionally, in the access network a number of connections share the same radio link, and this requires the application of special MAC protocols. The problem becomes more complicated when the network additionally supports user mobility. When a customer requires handover, the network should also change the resource allocation to the new BTS.

5.2 Narrowband networks

Present digital cellular systems are considered to be part of the second generation of wireless networks. Since these systems were primarily designed to offer voice services, they provide only limited capabilities for data transmission at low and medium bit rates. Most prominent examples for narrowband networks are GSM in Europe as well as IS-54, IS-95, and Personal Digital Cellular (PDC) in North America and Japan. Additionally, systems like Digital Enhanced Cordless Telecommunications (DECT) and Personal Handyphone Systems (PHS) have been introduced to offer wireless services in residential and office environments.

All second generation wireless systems support circuit switched voice and data services with basic rates typically ranging around 9.6 kbps. With the growing needs for higher rates, new services are currently being developed that permit multiple allocation to the physical resource. Extensions to GSM like the High Speed Circuit Switched Data (HSCSD) or GPRS (General Packet Radio Service) are considered to be in a transition phase (G2+) towards third generation systems allowing bit rates up to 170 kbps. In addition, enhancements to the physical layer of GSM like the introduction of a new modulation scheme under the name of EDGE (Enhanced Data Rates for GSM Evolution) further indicate the evolution of narrowband systems.

5.2.1 Planning of narrowband wireless networks

Due to today's tremendous customer demand and rapid growths of mobile networks, the need for a systematic planning methodology has become essential. Furthermore, knowledge about customer behavior and measured traffic data allow for detailed planning procedures. In conventional cellular planning, the planning process is driven by radio coverage considerations, i.e., selection of cell site locations, frequency planning, antenna design, etc. Basically, the algorithm used for cellular planning consists of four phases.

In the *Radio Network Definition* phase, a radio planning engineer chooses cell site locations based on his knowledge and planning experience. Then, the *Propagation Analysis* evaluates the radio coverage of the area using field strength prediction methods. If the coverage requirements are met, the expected number of traffic channels is calculated and a *Frequency Allocation* is performed. If the frequency plan can be computed, the network performance is evaluated in the *Radio Network Analysis* phase by computing some QoS measure in the cells.

From the above description it is apparent that the consideration of the customer traffic influences the network planning only to a very limited extent. Thus, a new approach to performing a truly *demand oriented* network planning has been introduced that takes as input a characterization of customer demand and incoming traffic by discrete points, the *demand nodes* [261].

The core components of the new integrated design concept are the automatic network design algorithm *Set Cover Base Station Algorithm* (SCBPA, see [261]) and a traffic characterization procedure which generates a set of demand nodes. The first step in this approach is to create a distribution of demand nodes based on estimations of the demand in a certain coverage area. Then, the SCBPA uses a greedy heuristic which selects the optimal set of base stations that maximizes the proportion of covered traffic, i.e., the number of demand nodes which measure a field strength level above a certain threshold value.

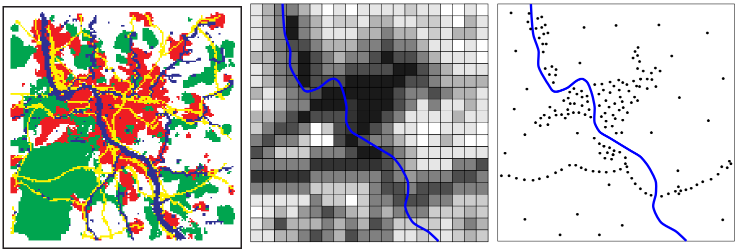
One of the key components of the system model for network performance analysis is a model of the demand node pattern. The other major issue is to characterize the behavior of the users at call level and to obtain a measure of the system performance in terms of subjective QoS of the user. Both items will be described in greater detail in the following sections.

5.2.2 Description of the spatial user demand

In this section, we will describe some methods for characterizing the spatial user distribution in a cellular network.

The demand node concept A *demand node* represents the center of an area that contains a quantum of demand from teletraffic viewpoint, accounted in a fixed number of call requests per unit, see [258, 262]. Such demand nodes represent the user demand discretized both in space and demand.

Empirical distributions The sequence of generating the discrete traffic estimation is shown in Figures 5.1(a), 5.1(b), and 5.1(c). Figure 5.1(a) depicts the input of the estimation algorithm. It contains data that is usually available for the specific area. The colors represent different categories of land usage, e.g. urban, suburban, forest, water, or open areas. Each class is assumed to



(a) Geographical data (b) Traffic matrix (c) Demand node distribution

Figure 5.1: Example for generating demand nodes

generate a certain amount of traffic and the traffic matrix is obtained by superimposing the traffic from the different land usage classes active on one area element (Fig. 5.1(b)). The darker an element of the traffic matrix is, the higher the anticipated demand in the corresponding area element. Then, the matrix representation is being transformed into a spatial discrete point pattern by using a clustering algorithm. A partitioning method has been described in [262]. The result of this sample partitioning is shown in Fig. 5.1(c).

Spatial stochastic processes While it is most desirable to have empirical demand node patterns for network planning, in some cases the data is not available to create the traffic matrices. Then, theoretical stochastic processes are used to describe the user distribution. Such a process is in general a random variable, which takes random choices of mappings from a Borel set to a counting measure, the number of points in a set.

The simplest distribution of demand nodes is described by a spatial Poisson point process. In this case, the number of points in any Borel set on the plane follows a Poisson distribution, depending on the area of the set and the intensity of the process. If the intensity measure of the process is independent of the location, it is called *homogeneous*.

The *Isotropic Phase Planar Point Process* [218, 219] allows for characterization of a large number of spatial distributions and can be considered as more general case of spatial process. Due to its construction, this process can introduce cluster effects and dependence between sets of points.

5.2.3 Modeling of QoS at call level

In this section, we present basic models for describing the user behavior at call level. As mentioned in Section 5.2.1 this plays an important role in the proper dimensioning and planning of the network. In the following, we will focus on answering the question: How do the cluster structure and different classes of service (voice, low bit rate data, and multimedia) influence the QoS of a customer in a mobile network? If for example the network planning does not take into account any customer clustering, the subjective QoS in terms of *call blocking probability* experienced by a specific test customer will decrease in areas with customer concentration.

Customer traffic process with retrials Let us consider the cell to be given by a finite source model in which the number of sources is obtained from the spatial locations of the customers as described in Section 5.2.2. The customers can either be in idle or in active states. After call termination or rejection, the user will remain idle until generating the next call. The model is therefore a standard loss system with K servers (number of channels) and X sources (random number of customers). The novelty of this approach is to derive the distribution of X from the spatial traffic description. In [258], it is shown that there is a degradation of the QoS when the customer population is clustered and the planning did not take such structure into account.

This model can be extended [259] to include a phenomenon quite common in daily life: A user whose call is blocked will immediately retry calling instead of giving up. This causes the network load and thus the blocking probability to rise even further — a snowballing effect of call arrival processes can occur, which leads to dramatic degradations of the call completion performance of single switching systems, and subsequently of the whole network. Figure 5.2 shows the model of the cell, where Θ denotes the *retrial probability*. The analysis is carried out by solving two-dimensional Markov chains in a recursive way [259].

Figure 5.3 illustrates the impact of the retrial probability on the blocking probability of first attempts for different ratios of mean call holding time and

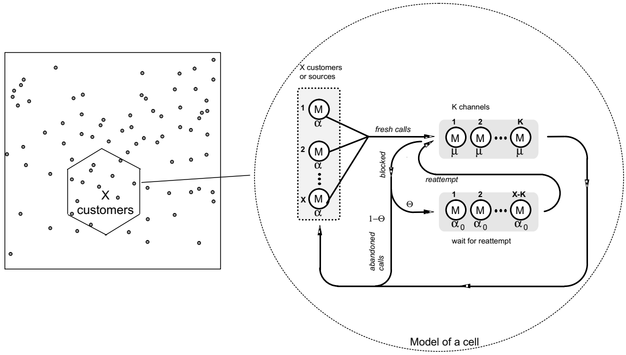


Figure 5.2: Single cell with finite number of customers and modeling of the retrial phenomenon.

mean wait-for-reattempt time α_0/μ. It can be seen clearly that the blocking probability of the first attempt is higher than without repeated attempt. Compared to the case without retrials ($\Theta = 0$), it is clear that we cannot neglect the repeated attempt phenomenon. This phenomenon also holds in a model in which hand-over calls are prioritized over fresh calls by the use of guard channels [259].

Integrated services call level models So far the location of the users was assumed to be fixed. In this section we will consider scenarios where voice and data services are offered to users with mobility.

A service request from a mobile user to a base station may be due either to the generation of a new call or to a handover request. Since handover failures force the termination of a call in progress, these events are considered to be worse than new calls blocking, whose effect is just to force the user to repeat his access request at a later time.

Two mechanisms are implemented in order to favor handover requests over new call requests under heavy traffic conditions [168]. First, priority is given to handover requests by reserving a small number of channels for them; second,

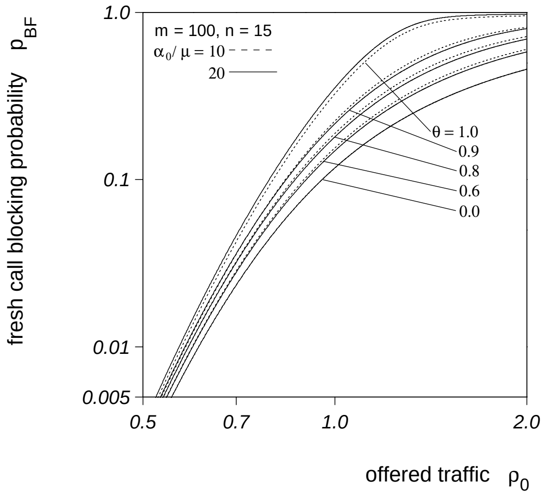


Figure 5.3: Impact of retrial probability on fresh call blocking.

when the handover of a multimedia call is requested, the base station tries to accommodate the entire call, but, if this is not possible, the two components of the call can be decoupled: the voice connection is accepted, if possible, while the data connection is temporarily suspended, waiting to be resumed as soon as enough channels are available in the cell to serve the entire multimedia call. Decoupling a multimedia call because of a partially served handover can improve the quality of service, mainly because it allows the user to continue at least partly his communication. To facilitate recombination of decoupled multimedia calls, new calls are not accepted as long as some active decoupled connections exist in the cell.

The cell state, in any instant, is determined by the number of currently active connections for each class of traffic. A continuous-time Markov chain is derived where transition rates are determined from the rates of the arrival processes of new call and handover requests, and from those of the processes of call completion and outgoing handover requests.

As a basic scenario for validation, a configuration with two classes of service was used, comprising voice calls and data calls which require the allocation of 4 channels. Results showing the average carried traffic for the two classes of service, the average number of busy channels and the average blocking probability for new call and handover requests, were produced using both the analytical model and a simulator. The comparison of the curves clearly indicates that the approximations introduced in the model development do not alter the numerical results significantly.

Other results were also obtained with different configurations of the cell under investigation. Scenarios with voice, data and multimedia connections were considered, with different values of the number of channels required for each data call; the performance of a system in which some channels are reserved to data connections was evaluated, assessing the performance improvements for systems in which the decoupling of multimedia connections is introduced.

A similar approach is taken in [175], where the task of the design and planning of cellular communication networks with overlapping macrocells and microcells is formalized in terms of call blocking probability.

The main advantage of the use of microcells in a GSM network lies in an improved spatial reuse of frequencies, hence in substantial capacity increases, with the consequent possibility of offering, in addition to telephony, data services at medium-high rates (up to hundreds of kb/s), and even multimedia services through the integration of voice and data traffic flows. The main disadvantage lies in the fact that, given the user mobility pattern, the number of handovers during a connection increases for decreasing cell size. This can be a critical factor, since the handover failure probability has a strong impact on the QoS that the user receives from the system.

Different resource planning strategies are presented in [175] and compared by means of the system dynamics. The objectives of QoS provisioning are achieved by equipping the system with the proper number of radio resources and by reserving some of these resources to specific classes of users.

5.2.4 Dimensioning of voice traffic links

Another important aspect in the planning of cellular networks is the appropriate dimensioning of the links transporting the voice traffic. This is especially important when migrating towards future mobile network generations where also data, video, and multimedia services will be supported. In such systems the CDMA access method with variable bit rate vocoders will be employed, resulting in the problem of efficiently transporting streams of low and variable bit rate over a high speed transport network, usually operating with ATM technology.

The application of ATM Adaptation Layer Type 2 (AAL2) is foreseen, supporting the multiplexing of several narrowband connections on one ATM connection. After encapsulating the voice packet in a CPS (Common Part Sublayer) PDU, several such CPS packets are fitted into a single ATM cell. In case that the remaining space in the ATM cell is not sufficient to include a CPS packet, its payload is split up in two cells. Additionally, a timer serves to keep the packetization time within a certain delay requirement.

In [89] and [88] studies were performed that investigated the suitability of AAL2 for narrowband CDMA speech connections. Here, the packetization and traffic shaping was modeled as a discrete time Markov chain. The transitions were given as recursive state transition equations. From the analysis the distribution of the delay of the speech packet was derived. The results for an E1 link showed that AAL2 is capable of transporting about 80% of the traffic compared to the traffic without the overhead of AAL2. The reason for this loss in capacity is attributed to the additional CPS packet headers. This example illustrates the need for detailed modeling of the transport network in order to achieve a correct dimensioning of the system.

5.3 Broadband WATM networks

In 1996, the ATM Forum Wireless Working Group started activities to develop standards for wireless ATM networks (WATM). Recall that ATM was originally designed for fibre-optic transmission links characterised by extremely low error rate (10^{-12} or less). Therefore, adaptation of ATM to wireless requires updating existing standards by adding radio access layer as well as user/terminal mobility management. These new standards should allow the WATM network to provide the same set of communication services as in the wired network (with similar QOS requirements).

The reference configurations for the WATM network, including all WATM application scenarios are defined in [59]. They differ in terms of terminal mobility, wired or wireless access, mobile or fixed switches, internal ability for establishing ad-hoc networks and co-operation level with PCS (*Personal Communication Systems*) systems.

The WATM network requires an enhanced protocol stack compared to cable ATM networks. The protocol stack, which should be implemented in the terminals, access points and/or ATM switches, depends on the assumed network configuration. These enhancements are related to wireless access and mobility management. Wireless access provisioning requires some modifications on the physical layer while supporting mobility demands new functions implemented at the ATM layer and in the signaling protocol.

5.3.1 Data link layer

Considering the transport of messages in an advanced packet-switched network at the data link, network interface and network layers over an error-prone communication channel, e.g. a wireless ATM network, the transport characteristics are additionally influenced by handover procedures according to the implemented terminal mobility management (cf. [38]). Furthermore, the hierarchical structure of the existing protocol stack, the segmentation of messages and the correlation of errors at the wireless physical layer as well as the error recovery at the wireless LLC layer have a strong impact on the transport performance.

In [134] error control protocols in such an environment are studied. The network interface layer, called block layer here, coincides with the ATM adaptation layer and provides a service-specific bearer service over an error-prone communication channel for the network and transport layers on top of it, for example LLC/SNAP encapsulated IP frames (cf. [93, Sec. 4.1.2, p 202f]). An Automatic-repeat-request (ARQ) protocol such as Selective Repeat (SR) or Go-back-N (GBN) and their ramifications can be applied both at the wireless data link layer as well as at the service-specific convergence sublayer (SSCS) or at the common part convergence sublayer (CPCS) of the ATM adaptation layer (AAL) to cope with the defective transport of ATM cells and to guarantee a secure delivery of messages to the higher layers (cf. [146, 152, 235]).

Regarding the efficiency of the data transport a large block length is required while error considerations demand short blocks. Therefore, we suppose that the size of the block PDUs can be varied to some extent, that the transport channel of cell PDUs is error prone due to noise, failures of the transmission equipment, data loss, as well as customer and terminal mobility and that a

GBN ARQ protocol with window size is applied at the block layer to guarantee a correct delivery of the block PDUs free from losses (cf. [235, p. 129f]).

The question arises how the performance of a GBN protocol subject to the segmentation of messages and the correlated errors of the PDUs, particularly the additional load due to retransmissions, can be determined by analytic means and how the corresponding parameters should be selected. Then it may be possible to perform a self-tuning of the block length and window size processes under the control of the error detection process at the cell level. In [134] this questions have been answered by a hierarchical performance model of a GBN ARQ protocol. This hierarchical model can be used as first simple approach to study the impact of varying block length, segmentation and correlated errors on the performance of ARQ protocols.

5.3.2 Satellite MAC protocols

The Adaptive Random-Reservation Medium Access Control (MAC) protocol was designed [202, 201] to allow statistical multiplexing of ATM traffic over the air interface to GEO satellites with on-boarding processing capabilities, especially for the independent and spatially distributed terminals. It is shown that the potential user population which can be served is considerably increased by statistically multiplexing bursty traffic over the air interface.

The MAC protocol has to be designed to allow statistical multiplexing of ATM traffic over the air interface, especially in the uplink for the independent and spatially distributed terminals. The following design objectives are taken into consideration: a) maximize the slot utilization, especially for bursty traffic, b) guarantee the QoS requirements for all service classes, and c) maximize frame efficiency by minimizing overheads.

The minimization of overheads is not an easy task, especially for ATM which was designed for channels with very good error characteristics. To minimise cell loss over the satellite link, channel coding has to be used to make the transmission more robust. The LLC header to facilitate error recovery mechanisms is optional and not in scope of this study. Finally, a satellite specific header with satellite routing and wireless resource management fields is added to form a MAC packet.

In [202] we analyse the performance of the proposed protocol. It is shown that maximum throughput can be achieved by using this access scheme. A

TDMA access protocol combining both Random Access and Demand Assignment Multiple Access (DAMA) is particularly suited for a scenario with a high number of terminals with very bursty UBR traffic (e.g. web browsing). UBR sources with short burst lengths access the slots remaining after the reservation procedure by random access which drastically reduces the slot access delay, at the expense of a lower utilisation. However, for UBR sources with burst sizes consisting of several ATM cells, reservation access provides higher throughput, but the access delay is considerably longer.

5.3.3 Performance of ATM service categories

The ATM service categories CBR, VBR, ABR, and UBR are defined in [11]. They should be supported by WATM networks with similar QoS objectives as specified for fixed ATM networks, see [59].

Evaluation of CBR, VBR and UBR services

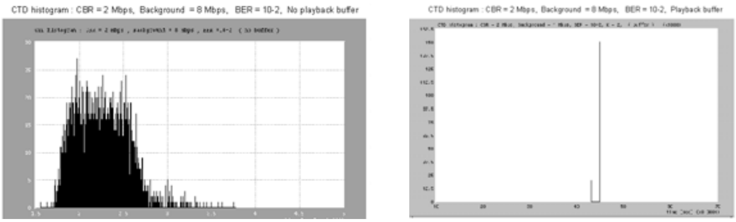
In this section we summarise the exemplary numerical results presented in [14, 19] concerning the performance of the ATM network services in the fixed wireless environment. Simulation studies were provided assuming a bottleneck network topology with a variety of tested connections established between two terminals attached to the network by different RAP (Radio Access Point) units, each of them governed by the MEDIAN protocol [172]. Additionally, transmission errors occurred independently.

Transmission errors The radio channels are usually characterised by a relatively high BER value, even up to 10^{-2} [18]. The overall situation gets worse when the terminal is mobile and then the transmission conditions are worse due to fading and shadowing [18, 245, 246]. As a consequence, even though a powerful protection mechanism is used, whole bursts of cells could be lost.

MAC protocol A number of new MAC protocols was recently submitted for the purpose of wireless ATM networks, among them are most recognised MASCARA [18], SAMBA [237], and MEDIAN [172]. These protocols were designed to operate under the TDMA (*Time Division Multiple Access*) scheme, where the time slot allocation for a given connection is made dynamically. In

MEDIAN, the time slots are assigned to the active connections on the basis of polling information and the priorities of the connections. The priority strictly corresponds to the maximum permitted cell waiting time Δ_{max}. A cell is discarded when it is waits longer than Δ_{max}.

On the basis of simulation experiments the following conclusions can be derived.



(a) CTD before the buffer

(b) CTD after the buffer

Figure 5.4: The CTD characteristics measured before and after the playback buffer

CBR service Additional mechanisms should be implemented in order to support CBR service. The commonly recommended approach to meet the CBR service requirements in ATM networks is to use the playback buffer mechanism implemented at the AAL layer. This solution was originally proposed for fibre optics ATM networks, where the cell loss ratio is negligible (e.g 10^{-9}). However, for the wireless ATM network concept the designers have to take into account the fact that an essential part of cells can be simply lost due to transmission errors and these losses can also occur in bursts (despite applying error protection mechanisms). Moreover, the variation of cell transfer delay is expected to be significantly larger and this is mainly due to the radio access protocol behaviour. As a consequence, the above leads to a need to modify the playback buffer mechanism in order to improve its functionality in the wireless ATM network environment. The solution proposed in [19] is based on some enhancements of the dummy cell insertion (instead of lost cells) algorithm and K times extension of AAL1 PDU numbering range by assigning the same number to K consecutive cells. The effectiveness of the recommended approach is illustrated in Fig. 5.4(a), 5.4(b).

The Fig. 5.4(a) and 5.4(b) show exemplary CTD characteristics of a CBR connection measured on the ATM layer and on the AAL1 layer with recommended playback buffer algorithm, respectively. As it was expected, the final CTD characteristic is of a minor deviation. This is due to the fact that some cells are delivered earlier to the destination.

VBR service Satisfying the requirements for this service [11, 59] seems to be difficult to reach. This is mainly caused by the behaviour of the MEDIAN protocol, which introduces significant cell delays. Anyway, from the application viewpoint, these relatively large CTD values (about 23 ms) are still acceptable for currently available applications, like voice, video conference, etc [188]. However, a similar mechanism as was proposed for the CBR service to reduce CDV is extremely desirable.

TCP over UBR service The effectiveness of the UBR service was evaluated in terms of traffic emitted by a single greedy TCP source. Included cell transfer characteristics referring to throughput, window size, and RTT say that even assuming heavy conditions, the effectiveness of cell transfer is still acceptable. In the running experiments the bit error rate was 7.5×10^{-3} (for higher bit error rates the TCP segment loss probability was too high). The cell loss ratio (CLR) for this BER was 10^{-2}.

Rate-based and fuzzy logic ABR flow-control

Connections of the ABR service class are designed to use the remaining capacity of an ATM network after the CBR and VBR service-class connections have received their current cell rate. To avoid congestion in an ATM network and to use the whole capacity efficiently, the source of every ABR connection has to be informed regularly about this remaining capacity. For this purpose, the ATM Forum has standardized algorithms for rate-based ABR flow-control.

Considering the high-speed data transfer by the ABR service class in a WATM network, the impact of forward and backward hard handover protocols on the performance of the two rate-based ABR flow-control schemes Explicit Rate Indication for Congestion Avoidance improvement (ERICA+) and Fuzzy Explicit Rate Marking Adaptation (FERMA) has been investigated (cf. [239, 135, 240]). In [239, 135] the influence of different handover protocols on the congestion control scheme ERICA+ has been considered. Then the original FERM algorithm (cf. [209]) has been adapted in [240].

In this study, the ABR service class that is specified in the ATM Forum document on traffic management is used (cf. [11]). The underlying WATM architecture and the used protocol stack of this study are derived from NEC's prototype implementation WATMnet and the ATM forum documents (cf. [217, 212], [51]). It includes a simplified version of the TDMA/TDD structure and the associated wireless MAC protocol of this system. Using a model of a basic client-server scenario in a wireless ATM network, the impact of the error-prone wireless communication channel and of mobility-management techniques determined by backward and forward hard handover protocols has been studied (cf. [217, 12, 211, 216, 257]).

5.3.4 Handover protocols

Regarding data communication in a WATM network the connections must be permanently maintained during the communication phase since a mobile terminal (MT) moves within a certain coverage area of a base station (BS) and may cross its boundary (cf. [217, 12, 211, 216, 257]). In this case, the connections must be handed over to a new transmission cell whereby a new radio channel is seized and the QoS requirements of the corresponding virtual channels (VCs) must be satisfied in addition to the existing ones within the new cell. The corresponding hard handover protocol has to guarantee the sequence integrity and loss-free delivery of the ATM cells during this transition process. Moreover, interworking with the rate-based ABR flow-control algorithms is required to guarantee the effectiveness of the approach (see [135] and references therein, also section 5.3.3).

To achieve these goals, an improved backward hard handover protocol has been developed for handovers between the transmission cells of the current BS (CBS) and a new BS (NBS) of the MT, called better hard handover for a simplified error- and loss-free message transfer (cf. [135]).

Derived from a congestion-awareness concept the new handover protocol is a minimal enhancement of a standard hard handover protocol. It includes only a few new signaling messages using special RM cells. Independent of handover events they are exchanged between the base stations and the mobile switching center (MSC) which is an ATM switch with mobility support enhancements for the wireless part of the network. Whenever there is a substantial change of the ABR bandwidth or an alteration in the number of active ABR connections in the transmission area of a BS, the BS informs the MSC immediately by sending an MSC information message with the current ABR target cell rate TCR, the number of ABR connections and the sum of their minimum cell rates. This very low additional bandwidth requirement in the path between each BS and the MSC gives the MSC the knowledge about the current possible explicit rate (ER) in each BS transmission area.

The improved backward hard handover is combined with forward handover as a fallback procedure in the case of a too short announcement period before a handover. This means that normally every mobile terminal announces its expected new handover t_a milliseconds before the real handover event occurs. If there is not enough time for such an announcement, the mobile terminal has to perform a forward handover. (cf. [135]).

5.3.5 Call admission control

In [20] we focused on some difficulties in performing the CAC function in the case of wireless ATM networks with wireless access points controlled by a dynamic TDMA (Time Division Multiple Access) class protocol. The assumed medium access protocol is MEDIAN [172]. Specific characteristics of this protocol, like frame structure, different handling of up and down cell traffic, allowable bandwidth limitations (overheads, guard time, error protection, redundancy, etc.) leads to the review of a recognised approach [228] to admission control. In the paper a conservative method for performing the CAC, with distinguishing between up and down calls, is proposed. This method is based on the declaration of the PCR (Peak Cell Rate) only. The exemplary numerical results are included.

5.4 Wireless LANs

In ad-hoc WLANs, where every station has a similar functionality, Carrier Sense Multiple Access (CSMA) based protocols may be advantageous because of their simplicity and because no station is compelled to assume special functions. Recently proposed standards such as the IEEE 802.11 or HIPERLAN I use variations of the basic CSMA scheme.

However, if we are to integrate delay-sensitive and delay-insensitive traffic in the same WLAN, we must keep end-to-end delay and delay variations below certain bounds for delay-sensitive traffic. Unfortunately, it is well known that in general CSMA protocols suffer from large access delay quantiles (the basic CSMA algorithm is actually unstable) due to the existence of packet collisions.

This section describes FIFO by Sets CSMA (FS-CSMA) [270], a MAC protocol for high-speed ad-hoc WLANs. FS-CSMA is a Collision Resolution Algorithm that builds groups (Transmission Sets, TS) with the packets arriving during fixed-length time intervals. Packets belonging to the same Transmission Set are transmitted using CSMA with random backoff whereas the different Transmission Sets are served following a FIFO discipline.

FS-CSMA generates a collection of (possibly empty) Transmission Sets $\{S_k\}$, each of them containing the packets that arrive during successive disjoint grouping intervals of constant length W slots. These Transmission Sets are served (i.e. their included packets are transmitted using CSMA) with a FIFO discipline, so that S_k is served before S_j if $k < j$. If every S_k had no more than one packet to be transmitted we would be facing a FIFO-MAC algorithm, with packets being sent in order of arrival.

The number of packets per TS is closely related to the integer W, which determines the length of the grouping interval, and to the network load ρ. The smaller W or ρ are, the more the system behaves like a FIFO system. On the contrary, the protocol will perform as pure CSMA if W is made large enough. The value of W must be chosen as a trade-off between the desired properties of a FIFO scheduling and the overhead introduced by the algorithm.

An FS-CSMA ad-hoc network may be seen as a distributed FIFO queueing system. The elements of this distributed queue are the Transmission Sets and their service consists of the successful transmission via CSMA with random backoff of all the packets included in the Transmission Set.

Chapter 6
Queueing Theory

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6.1 Introduction

This chapter gives an overview of the queueing problems addressed in the COST-257 project. Typically, these problems arise in modeling the behavior of the queues appearing in various subsystems of communication networks, such as multiplexers, traffic shapers, resequencing units, switching elements, rate adapters, etc. A profound study of the behavior of these buffers and the mechanisms that lead to this behavior is extremely important, because the performance of a communication network, and hence the quality of service experienced by the users, may be very closely related to it.

Section 6.2 presents some useful general relationships between performance quantities. Section 6.3 summarizes the work of COST-257 on queueing and storage models. First, the results on a number of specific discrete-time and continuous-time models are presented. Second, some new developments with respect to fluid flow models are given. The last part of this section addresses the work on Gaussian storages. Section 6.4 is devoted to the analysis of loss network models. Finally, in Section 6.5, the study of some specific network elements and some applications, related to queueing, are briefly discussed.

6.2 Some general relationships

In this section, we present some general relations between performance quantities. The relations are *general* in the sense that no detailed assumptions concerning the arrival process need to be made.

6.2.1 Queue length versus delay

The results presented in this subsection hold for a discrete-time single-server queueing system with a first-in-first-out (FIFO) service discipline, a general arrival process and either a finite or an infinite storage capacity. The service times of the customers are independent and identically distributed (i.i.d.) random variables with common probability generating function (pgf) $B(z)$ and mean service time $1/\sigma$. For infinite-capacity queues, it is assumed that the equilibrium condition $\lambda/\sigma < 1$ (where λ is the average arrival rate) is satisfied.

In [281], it is shown that the following simple relation exists between the pgf $D(z)$ of the delay of an arbitrary customer and the joint pgf $P_0(x, y) = E[x^{v_0} y^{r_0}]$ of the queue length v_0 during an arbitrary non-empty slot (during which the system is non-empty) and the remaining service time r_0 of the customer in service at the start of this slot:

$$D(z) = \frac{P_0(B(z), z)(z-1)}{\sigma z[B(z) - 1]}, \tag{6.1}$$

When the service times are either geometrically distributed or deterministic, an invertible relationship between the pgf of the delay and the pgf of the queue length can be derived [281].

6.2.2 Global versus per-type characteristics

In this subsection, we focus our attention on the derivation of performance characteristics per source type. A stable discrete-time infinite-capacity single-server queueing system with deterministic service times of one slot and a FIFO service discipline is considered. The buffer is fed by N independent heterogeneous, possibly time-correlated, sources of T types. Customers generated within the same time slot enter the buffer in random order.

In [288], the following expression is obtained for the pgf $D_i(z)$ of the delay $d(i)$ of an arbitrary customer of source type i:

$$D_i(z) = \frac{z}{N_i \sigma_i (z-1)} \int_{1/z}^{1} \frac{\partial}{\partial x_i} R(x, \dots, x, z) dx. \qquad (6.2)$$

Here N_i denotes the number of sources of type i, σ_i is the average load of a type i source, and the function $R(x_1, \dots, x_T, z)$ is the joint pgf of the vector $(e(1), ..., e(T), v)$, where $e(t)$ represents the number of customers of type t generated during an arbitrary slot and v is the total system contents at the start of the next slot. From (6.2), all the moments of $d(i)$ can be obtained in terms of the moments of the numbers of customer arrivals of the different source types and the system contents.

6.3 Queueing and storage models

6.3.1 Discrete-time queueing models

This subsection is devoted to a number of discrete-time queueing models, studied in the COST-257 project. Throughout this subsection, time is divided into constant-length intervals, called slots. Two different kinds of shorthand notations will be used to describe the discrete-time models, namely the notation A/B/c and the notation A'-B-c. Here the A-descriptor refers to the interarrival-time distribution, the B-descriptor characterizes the service-time distribution, the c-descriptor indicates the number of servers, and the A'-descriptor refers in some sense to the probability distribution of the number of arrivals per slot.

The Geo/D/c queue

The Geo/D/c queue is a discrete-time infinite-capacity queueing system with c servers, deterministic service times of D slots and geometrically distributed interarrival times between customers. In [276], an iterative algorithm is presented to analyze the probability mass functions (pmf's) of the system contents at the start of an arbitrary slot and the waiting time (in case of a FIFO service discipline). The algorithm is a discrete-time extension of the method of Crommelin ([52]) for the analysis of the M/D/c queueing system, and is of low complexity with respect to computation time and memory usage.

The GI^G/D/1 queue

The discrete-time GI^G/D/1 queueing system consists of one server and an infinite-capacity buffer. The arrival process is a so-called batch renewal process, characterized by a sequence of i.i.d. batch interarrival times a_i (in slots) and a sequence of i.i.d. batch sizes b_i (in cells). In [138], the output process of the queue is analyzed under the assumption that the pgf's of the random variables a_i and b_i are both rational functions. The output process consists of alternating busy periods and idle periods. Each busy/idle-cycle is independent from the others, but a busy period and the following idle period may be correlated. In [138], the joint pgf of the duration of a busy period and the following idle period is obtained, and from this the pgf, first moments and accurate approximate tail probabilities.

The $\sum_i \text{onoff}_i$-D-1 queue

The arrival process of the $\sum_i \text{onoff}_i$-D-1 queue consists of the superposition of N independent heterogeneous on/off-sources of T traffic types. Each source alternates between on-periods, during which one packet is generated per slot, and off-periods, during which no packets are generated. The lengths of the on- and off-periods are modeled as independent random variables with general distributions. In [287], an analytical method, based on the use of generating functions, an infinite-dimensional Markovian state description and the concept of the limiting arrival process when the buffer contents is extremely large, is developed for the analysis of the queue. As a result, an accurate, explicit approximation for the tail distribution of the system contents is obtained.

The Cortrain-D-1 queue

The Cortrain-D-1 queue is a discrete-time single-server infinite-capacity queue with deterministic service times of one slot and a so-called correlated train arrival process. Messages consisting of a variable number of fixed-length packets arrive to the buffer at the rate of one packet per slot (primary correlation) and the distribution of the number of leading packet arrivals in a slot depends on some environment variable (secondary correlation). The environment is assumed to have two possible states, each with geometric sojourn time. The distribution of the message lengths (in terms of packets) is general. By means of a generating-functions approach, closed-form expressions are derived for the mean value, the variance and the tail distribution of the system contents in [290] and for the mean message delay and the mean message transmission time in [58].

The GI_1+GI_2-D-1 queue with HOL priorities

In [284], a stable discrete-time single-server queue with infinite buffer space, constant transmission times of one slot, and head-of-line (HOL) priority scheduling is considered. There are 2 types of traffic arriving in the system. Both types of arrivals are assumed to be i.i.d. from slot-to-slot; the numbers of arrivals from different classes (within a slot) can however be correlated. Class 1 cells are assumed to have absolute priority over class 2 cells, and within one class the service discipline is FIFO. An analysis based on generating functions leads to closed-form expressions for the pgf's, the means and variances of the system contents and the cell delay, and the correlation coefficient of the system contents of both classes.

The N*MMBP-D-1 queue with deterministic vacations

The N*MMBP-D-1 queueing system has a single server and an infinite storage capacity. A fixed number c of contiguous slots are periodically allocated for service every R slots (service cycle). Cells are generated by N independent homogeneous sources, each modeled as a 2-state Markov modulated Bernoulli process (MMBP). That is, a source has two possible states (active or passive), each with a geometric sojourn time; when the source is in the active (or passive) state, there is an arrival with probability g (or f). In [254, 118], a new method is introduced for the analysis of the system, resulting in closed-form expressions for the steady-state pgf's and the expected values of the system contents at the start of an arbitrary slot and the delay, as well as asymptotic approximations for the tail probabilities.

The GI-G-1 and GI-D-1 queues with random server interruptions

In [84] and [103], a discrete-time single-server infinite-capacity queue with a general uncorrelated message arrival process and random server interruptions is considered. Each message consists of an integer number of packets and one packet can be transmitted from the buffer during a slot. Two operation modes are distinguished: CAI (continue after interruption), where the processing of an interrupted message resumes with the next packet of this message, and RAI (repeat after interruption), where the complete message is retransmitted after a server interruption. In [84], the message lengths (in terms of packets) are modeled as a series of positive i.i.d. random variables and the server interruption process is uncorrelated. Using a generating-functions approach, explicit expressions for the means and variances of the buffer occupancy and the message delay (in case of a FIFO service discipline for messages) are derived for both transmission modes. Fixed-length messages and Markovian server interruptions are considered in [103]. Closed-form expressions are obtained for the pgf's, moments and tail distributions of the buffer contents and the unfinished work for both CAI and RAI.

The N*MMAP-G-1 queue

The N*MMAP-G-1 queueing system is a discrete-time single-server infinite-capacity queue with N independent homogeneous sources generating packets of variable length. Each source is modeled as a 2-state Markov modulated arrival process. The service of a packet requires a positive integer number of slots, described by a general service-time distribution. For this model, three important quantities are studied in [253], namely the number of packets in the buffer, the unfinished work and the packet delay (for a FIFO service discipline). In particular, for each of these quantities, an expression is derived for the pgf, the mean value, variance, and tail distribution.

Queues with long range dependent traffic

A simulation study of the effects of long range dependent traffic on the performance of a queueing system can be found in [180].

In [54], the behavior of a discrete–time single–server infinite–capacity queue, where the arrival process consists of a superposition of an infinite number of ON-OFF-sources with increasing mean ON-periods and OFF-periods, is investigated. The analysis of the queueing system is seen as the limit of a sequence of D-BMAP/D/1 queues, which can be analyzed by means of matrix-analytic techniques ([23]). It is shown that the mean queue length is diverging if and only if the arrival process is long range dependent.

The discrete-time M-G-∞ input process is defined as follows. During a slot, a number of new so-called sequences are generated according to a Poisson distribution, independently from slot to slot. The customers belonging to a sequence arrive in consecutive slots. The length τ_A of a customer sequence has a heavy tail, so that the M-G-∞ process is long range dependent. In [136, 137, 139], two possible approaches to the analysis of a single-server infinite-capacity queue with deterministic service times equal to one slot and as arrival process the M-G-∞ process defined above, are briefly discussed: a Beneš approach and a slot-to-slot generating-functions approach. In [55], the sequence length τ_A is assumed to be asymptotically Pareto distributed with parameter s with $2 < s < 3$, i.e., $\text{Prob}\,[\tau_A = i] \sim ci^{-s}$ for $i \to \infty$, with $c > 0$. The asymptotics describing the tail distribution of the buffer occupancy are obtained by means of a generating-functions approach ([289]) and the elementary form of the Tauberian theorem for power series ([76]). It is shown in [55] that the buffer-occupancy distribution has a power-law decay with exponent $2 - s$. Besides the slope of the distribution, also the constant which determines the asymptotic behavior, is obtained.

Queues with heavy-tailed service-time distributions

Wiener-Hopf factorization provides a method for the waiting time analysis of discrete-time queueing systems, whose workload evolves according to Lindley's equation [100]. It is shown in [100] how Wiener-Hopf factorization can be applied in case of single-server infinite-capacity GI-GI-1 and SMP-GI-1 queueing systems, with arbitrary heavy-tailed service-time distributions, and either a general independent or a semi-Markovian arrival process (SMP). The analysis exploits the fact that the tail of the service-time distribution beyond a certain bound has a negligible effect on the idle-period distribution. This implies that the distribution of the idle periods can be determined with a service-time distribution truncated at an appropriate bound.

6.3.2 Continuous-time queueing models

The purpose of this section is to give an overview of continuous-time queueing models studied in the COST-257 project.

The $\sum_i \mathrm{MAP}_i$/PH/1/K and $\sum_i \mathrm{MAP}_i/\mathrm{PH}_n$/1/K queues

In [132], the finite-capacity $\sum_i \mathrm{MAP}_i$/PH/1/K queueing system with a single server, $\mathrm{K} - 1 > 0$ waiting positions and a FIFO service discipline, is considered. The arrival stream is a Markovian arrival process (MAP) resulting from a superposition of L independent general MAPs. The service times of the customers are i.i.d. random variables governed by a phase-type distribution, and are independent of the arrival process. It is shown in [132] that the system can be analyzed by an improved matrix-geometric technique. It is derived from Naoumov's general results obtained for finite continuous-time Markov chains with quasi-birth-and-death-structured generator matrices [186]. The most important performance characteristics of the system, such as the distribution of the number of customers in the system, the stream-dependent loss probabilities and the waiting- and sojourn-time distributions of a customer of a specific stream, are obtained.

In [133], the analysis is extended to the $\sum_i \mathrm{MAP}_i/\mathrm{PH}_n$/1/K queue, where the service times are phase-type distributed random variables whose distribution functions depend on the actual number of customers in the system. The distribution of the number of customers in the system, as well as the mean waiting and sojourn times of the customers are derived.

The $\mathrm{M_1} + \mathrm{M_2}/\vec{\mathrm{M}}$/1 and $\mathrm{M_1} + \mathrm{M_2}/\vec{\mathrm{M}}$/1/K queues with non-preemptive priorities

These queueing systems have two distinct service classes. The arrival streams of the customers of each class $i \in \{1, 2\}$ are assumed to be independent Poisson processes with rates λ_i. The service times are independent exponentially distributed random variables with class-dependent mean service rates μ_i and are independent of the arrival processes. The behavior of the finite-capacity $\mathrm{M}_1 + \mathrm{M}_2/\vec{\mathrm{M}}$/1/K system with one server and K-1 waiting places, is analyzed in [283]. Non-preemptive *HOL priority scheduling* is used, where customers of class 2 have priority over those of class 1. Within a priority class, the service discipline is FIFO. In [283], a new recursive algorithm is developed to solve the balance equations and the normalization condition. The steady-state distribution of the number of customers of each service class in the system, including the blocking probabilities, are derived. Furthermore, an explicit representation of the waiting-time distributions of both classes as well as recursive formulas for the Laplace-Stieltjes transforms of the waiting times of the low priority class are presented.

The single-server infinite-capacity $\mathrm{M}_1 + \mathrm{M}_2/\vec{\mathrm{M}}/1$ queue is studied in [36]. A (non-preemptive) priority is assigned for class 1 based on a *queue place reservation mechanism*. When a customer of class 1 arrives to the system, he reserves the last place in the current queue for the service of the next customer of class 1. The reserved place is kept by the system even when it has reached the top of the queue before the arrival of a new customer of class 1. A FIFO service discipline is used within a class. For this system, the mean waiting times of customers of both classes are derived.

6.3.3 Fluid flow models

The fluid flow model constitutes an abstraction of a queueing system: the amount of work being delivered by arrivals to a queueing system or being processed by a server is modeled as a continuous-time flow (Figure 6.1). Thus,

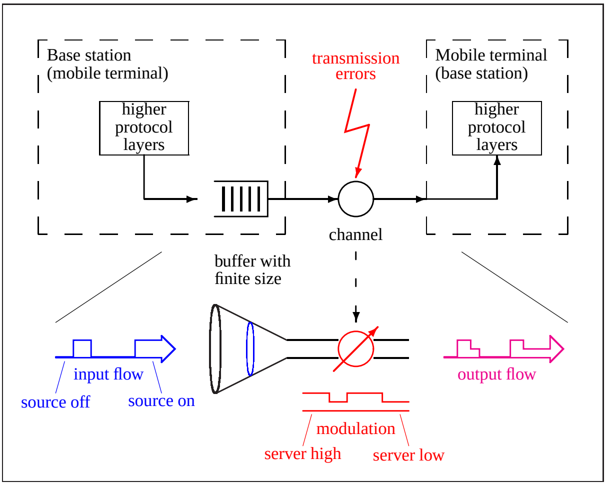

Figure 6.1: Application of the fluid flow model with variable capacity.

the fluid flow model is able to capture all kinds of effects that are due to time-variable source and server rates, i.e. that happen on burst or higher levels, while it ignores effects that happen on data unit level. This disadvantage is compensated by the ability to treat quite large state spaces on burst level – provided that the numerical calculations are carried out in a careful way [82, 83] or that the contribution of the buffer to QoS is quite small [77] – and by the fact that sources and servers might be treated in a similar manner [81, 80, 78]. The fact that the fluid flow model can be used both at the edge and inside the network as well as for wired and wireless communications [81, 80, 78] underlines its importance as a unifying framework for performance evaluation throughout the whole network.

Source and server models and buffer content behavior

In general, the processes that are driving the rate processes both at the entrance and at the exit of the buffer are modeled as (superposition of) Markov chains. However, [3] deals with Gaussian input, while [197] uses results from the M/G/1-PS queue. Sources and servers are merely distinguished by the sign of the rates with which they contribute to the buffer content process [81, 80, 78]. There belongs a drift value to each state, which is the sum of the corresponding (positive) source and (negative) server rates in that state. If an over-load state with positive drift occurs, the buffer fills up, while it empties once that an under-load state with negative drift occurs. Equilibrium states with vanishing drift freeze the buffer content temporarily.

Analysis

The content of a fluid flow buffer is described by a set of differential equations. Provided that the driving process is time-reversible, the vector of the joint probability density functions of the buffer content x for all states q of the governing Markov chain is given as

$$\vec{F}(x) = \sum_{\forall\, q} a_q(K)\, \vec{\varphi}_q\, \exp(z_q\, x). \tag{6.3}$$

First, the sets of eigenvalues $\{z_q\}$ and eigenvectors $\{\vec{\varphi}_q\}$ are obtained. Then, the adaptation of the solution to the boundary conditions that the buffer is never full in under-load states and never empty in over-load states happens via the determination of the set of coefficients $a_q(K)$ by solving a linear system of equations. An overview of the analysis that covers Variable Bit Rate (VBR) 2-state sources and servers, finite buffer sizes and vanishing drift values is given in [81, 78].

Numerical stabilization

The numerical side of the fluid flow model has a bad reputation. However, a pretty good numerical stabilization might be achieved by ensuring a high numerical quality of eigenvalues and -vectors and selecting a solution method that is able to cope with the large, ill-conditioned system of equations that already contains numerical errors from the eigensystem. The most versatile method for the latter step is the *Gaussian elimination with complete pivoting* [82, 83].

The buffer-less fluid flow model

By skipping the buffer, the buffer-less fluid flow model allows for very fast and numerically stable performance evaluations of large systems: merely the input (and server) rate distributions have to be calculated by convolution in case of independent sources/servers. The speed of the calculations might be raised by applying a suitable state space truncation [77].

VBR 2-state sources and servers

VBR 2-state sources and servers with exponentially distributed “high” and “low” duration, which include the subset of on-/off sources and servers, play a special role in fluid flow analysis. The superposition of n homogeneous sources or servers again leads to a Markov chain with merely $n+1$ states, thus making it possible to study quite large systems [82, 83]. The eigensystem of each homogeneous subsystem containing 2-state sources or servers is obtained via an eigenvector-generating function approach that exploits the special structure of the modulating Markov chain. Heterogeneous systems of independent sources are composed out of homogeneous subsystems by using Kronecker algebra, thus taking advantage of their numerical quality.

6.3.4 Gaussian storages

The Gaussian traffic model was considered in subsection 3.2.2, and we use the same notation here. Non-exact but useful estimates of queueing performance can be easily obtained for general Gaussian input traffic. Details can be found for simple queues in [3] and for priority queues in [194]. The same technique was applied to study busy period lengths in the fractional Brownian motion (fBm) case [193]. Another type of queueing result is the Girsanov approach to fractional Brownian storage in [192].

Assume that the cumulative input process to a storage with unbounded capacity is described by a Gaussian process $(A_t : t \in \mathbb{R})$ with stationary increments. The stationary storage process is then defined as

$$V_t = \sup_{s \leq t}(A_t - A_s - c(t-s)), \tag{6.4}$$

where c is the service rate. Our basic approximation is

$$\begin{aligned} \mathrm{P}(V_0 > x) \quad &\geq \quad \sup_{t\geq 0} \mathrm{P}(A_t - ct > x) = \mathrm{P}(A_{t^*} - ct^* > x) \\ &\approx \quad \exp\left(-\frac{(x+t^*)^2}{2v(t^*)}\right), \end{aligned} \tag{6.5}$$

where t^*, the "typical time to wait for seeing storage level x", minimizes $(x + (c-m)t)^2/v(t)$ in the domain $t \geq 0$. The first estimate is a trivial strict lower bound, whereas the second seems to be an upper bound (exceptions may exist, since no proof is known).

A "bonus" feature of the simple Gaussian queue is that the most probable path (in maximum likelihood sense) to reach a given storage level x at time 0 can be readily written in terms of the covariance function. Its expression is

$$f_{x,t^*}(s) = -\frac{x + (c-m)t^*}{v(t^*)}\Gamma(-t^*, s). \tag{6.6}$$

The same technique extends in a straightforward way to priority queues, as long as the most probable path to reach total storage level x is such that only lowest priority traffic needs to queue. As an example, consider the case that there are two priority classes, the higher class is modelled by a Gaussian process with similar covariances as in the Anick-Mitra-Sondhi fluid flow model (a short-range dependent process), and the lower class is modelled by a fractional

Brownian motion with self-similarity parameter $H = 0.8$ (a long-range dependent process, used as a simplified model for Internet traffic). Figure 6.2 shows the traffic rates in the most probable way that the queue starts to accumulate at time 0 and reaches a maximum level 2. The queue is a result of the combined "effort" of both classes. Note also how long in advance fractional Brownian traffic "plans" its hump, compared to the short range dependent process.

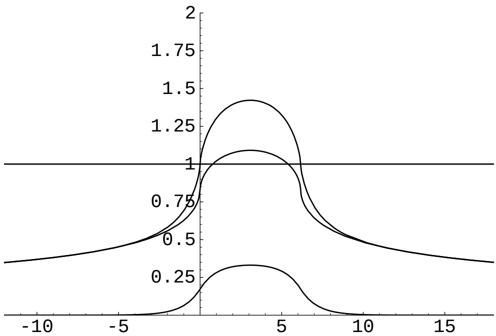

Figure 6.2: Traffic rates of most probable paths that yield a class 2 queue of size 2 (from bottom: class 1, class 2, and sum of both). Mean rate is normalized to 0 and output rate to 1.

6.4 Loss networks

Loss networks are used as models for connection-oriented networks, such as telephone networks or ATM networks, where the sharing of the available resources (typically, link capacities) is based on reservations made in the connection establishment phase. Calls (connection requests) arrive randomly according to some specified stochastic process. Which specific resources an arriving call needs depends on the class of the call. If these resources are not available at the arrival time, the call is blocked and lost. Otherwise the required resources are reserved and the connection is established.

The goal is, typically, to calculate the call blocking probability under the specified traffic conditions and system configuration, which is a relevant measure for the grade of service in these kinds of systems. The blocking probability is defined by means of the stationary distribution of the state of the system. In many cases this stationary distribution as a nice closed form representation, which is just a truncated form of the corresponding distribution in a system with infinite link capacities. The real problem when calculating the blocking probability comes from the so called state space explosion.

In addition to ordinary point-to-point connections, loss models can be utilized in the case of *point-to-multipoint* (or *multicast*) *connections*. By a multicast connection we mean a one-way tree-like connection from a source to a number of receivers receiving the same information stream simultaneously.

In general, the analysis of loss models is nowadays relatively mature. There are already monographs such as [232] devoted to this topic. Much research effort was also put on this topic in the COST-242 project (see [228, Chap. 18]). In the COST-257 project, the following topics related to loss models were studied:

- Efficient simulation of multirate loss systems for the approximation of blocking probabilities [141, 140, 143, 144]
- Calculation of blocking probabilities in multicast networks based on multicast connections with dynamic membership [121, 198, 1]
- Transient analysis of Erlang's single link loss model [282]
- Loss systems with stationary and periodic input [285]

Concerning the first item, the results achieved were remarkable: new methods were found that could accelerate the simulation by several orders of magnitude. The research of the second item started almost from the scratch: so the results achieved are quite fundamental. As regards the third item, an alternative approach to calculate transient blocking probabilities was developed. Regarding the fourth item a comprehensive study of loss systems with stationary and periodic input and their mathematical treatment by marked point processes is provided. Below a short summary of the results achieved in the project is given.

6.4.1 Multirate loss systems

In this study the problem of estimating call blocking probabilities in a multirate loss system was considered. The exact solution to this can be given an analytic expression. However, it can be used for only very small networks as the state space of the system grows exponentially. The focus has been on devising efficient simulation methods for estimating the blocking probabilities. The results reported here represent progressive improvements in the speed-up factor.

Under certain assumptions the steady state distribution for the number of calls in progress in each class has a product form. The blocking probability of class-k calls, B_k, is the probability of the set of blocking states. As the form of the distribution is known, the static Monte Carlo method can be used. To increase the efficiency of the simulation, the conditional expectations (CE) method and importance sampling (IS) have been applied (see e.g. [234]).

The first results on applying the CE method were reported in [142]. The idea is based on expressing B_k as a conditional expectation, where the state space is partitioned into subsets in a suitable way such that the contribution to the blocking probability from a particular subset can be computed analytically. In [140] the CE method was generalized to the case, where the subsets were not restricted to be entirely inside the allowed state space. Also, results were reported on the combination of the CE method with the IS heuristics proposed by Ross [232, Chap. 6]). The same partitioning was used to generate the samples with the so called Gibbs sampler in [141], where the CE method is also applied. In summary, it can be said that the methods discussed above give a substantial variance reduction but even better results were obtained by applying IS.

In general, the idea of IS is to generate the samples with a distribution that makes the interesting samples more probable. The bias in the samples is then corrected by weighting them with the so called likelihood ratio. In [143] exponentially twisted distributions (ETDs) of the original distribution were used. An IS distribution was derived which attempted to make all the blocking states more probable. This was accomplished by using a composite form distribution consisting of a weighted sum of ETDs, one for each link the traffic class k uses. The parameters of the ETDs were chosen such that the main probability mass of the IS distribution was centered around the most probable blocking states. Heuristics were also presented to fix the weights of the composite distribution. This method represents a major step forward in the problem of efficiently estimating B_k.

When using just ETDs sample generation is easy. However, in [144] it was shown how a much better approximation of the ideal IS distribution can be obtained by accepting a slight increase in the computational effort of generating samples. To this end, it was first observed that the estimation of B_k can be broken into separate sub-problems each corresponding to the blocking probability contribution from one of the links the traffic class k uses. Then two effective IS methods to solve each sub-problem were given. In these methods the earlier used ETDs are replaced with a more accurate approximation of the ideal IS distribution. In both methods the idea is to generate samples directly into the set of blocking states of a given link, assuming solely that link to have a finite capacity. The first method, based on using an inverse convolution, achieves this objective exactly. The second one is an approximation of the first method, where a Gaussian approximation is used. The trade-off in the two methods is between the better performance of the first method and the lower memory consumption of the second method. The two methods drastically improve the performance of the IS sampling.

6.4.2 Blocking in multicast networks

In [121, 198, 1], the following model of a multicast network is considered. There is a single server node connected with a collection of user sites by a transport network with tree topology. A predefined set of multicast connections with dynamic membership use this transport network, with the server acting as the source and the receivers located at the user sites. As an example, one can think of the delivery of TV or radio channels by a multicast capable telecommunication network, each channel requiring its own multicast connection. Requests from the receivers to join the related multicast connections arrive randomly, according to a Poisson process. If, upon the arrival of such a request, there are not enough resources available to establish a new leg from the receiver to the nearest node that already carries this connection, the request is blocked and lost. Otherwise the new leg is established. Whenever a receiver disconnects, the branch of the connection tree dedicated to this specific receiver is dropped.

In [121], it is shown how to calculate the call blocking probability for any of the multicast connections in a specific link with finite capacity assuming that all the other links have infinite capacity. The idea is to transform this problem to that of calculating the call blocking probability in a so called generalized Engset system.

The general case with any number of finite capacity links is treated in [198]. The purpose is now to calculate the end-to-end blocking probability for any receiver of any multicast connection. The blocking probability can be defined by means of the stationary distribution of the state of the system. In this case, the state is described by a collection of vectors, each of them consisting of two-state elements. Element i of vector u tells whether any of the receivers located at user site u is connected to multicast connection i, or not. It is observed in [198] that the stationary distribution of the state of the system is just a truncated form of the corresponding distribution in a system with infinite link capacities. However, calculating blocking probabilities in the resulting closed form expression is practically possible only for very small networks with few multicast connections. The main result of [198] is therefore the development of a faster algorithm for the calculation of blocking probabilities. The complexity of the algorithm depends only linearly on the size of the network. Unfortunately, neither this algorithm is applicable to networks with many multicast connections (due to the state space explosion).

In [1], it is, however, shown that, by making certain restrictive assumptions, the calculation of blocking probabilities becomes feasible even for networks with many multicast connections. The idea is to restrict oneself to the cases where connections are symmetrical.

6.4.3 Transient analysis of Erlang's loss model

In [282], the M/M/n/n loss system is considered. The evolution of the state of this system is described by a Markov jump process. Thus, the transient state probabilities $P_{j|i}(t)$ (to be in state j at time t given that the initial state is i) are determined by the system of Kolmogorov's differential equations. The transient analysis of Erlang's loss model is motivated by a dynamic VP bandwidth management scheme in an ATM network, where the bandwidth allocation for a VP is adjusted at regular time intervals. At the beginning of an interval the system occupancy i is observed and new VP capacity allocation is done in such a way that the expected time average of the blocking probability in the interval will be less than a predefined limit. The general solution methods require that the evolution of all the probabilities $P_{j|i}(t)$ be calculated, although the only interesting probability (in this application) is $P_{n|i}(t)$. In [282], a novel method is presented which allows the calculation of $P_{n|i}(t)$ separately without the need to consider the evolution of the other state probabilities. The idea is to transform the original system of n differential equations to a single integral equation of Volterra type.

6.4.4 Loss systems with stationary and periodic input streams

In [285] Willie studies some basic problems concerning loss systems in queueing theory. The problems resulted from practical questions in the design of modern telecommunication networks and they are treated using the theory of random marked point processes.

In the first part basic questions relating to loss systems with a superposition of inputs are investigated. The input of a loss system consisting of a finite number of fully available, identical servers is assumed to be a superposition of a finite number of partial traffic streams, which need not be independent and are represented by a random marked point process. First, existence, uniqueness and ergodic statements for the steady state of the system at several observation points, i.e. at an arbitrary point in time and at the arrival instants of calls belonging to a fixed partial stream are derived. Practicable sufficient conditions, formulated in terms of the partial inputs, are given. Then a method for calculating the individual call blocking probabilities is presented when two independent traffic streams, a Poissonian stream and a stream induced by a semi-Markov point process, are offered together to a finite number of service stations. Using this procedure and the possibility of approximating every stationary point process by a sequence of stationary semi-Markov point processes and a continuity argument, one can calculate the individual call loss probabilities when a general stationary traffic stream and a Poissonian stream are offered simultaneously to N servers. Explicit formulas are derived for the time congestion, the total call blocking probability and the individual call blocking probabilities in a single server loss system whose input consists of a finite superposition of independent general stationary traffic streams with exponentially distributed service times. The results are used for studying to what extent two arrival processes with coinciding customer-stationary state distributions are similar or even identical, and whether such an arrival process with time-stationary state distribution is of the Poisson type. Moreover, a weakened insensitivity property can be found in this model.

The second part addresses a basic problem in the foundations of queueing theory – the existence of a unique periodic steady-state for loss systems and convergence to it – which is of great theoretical and practical interest. The input of a multi-server loss system is assumed to be a periodic random marked point process which has, with probability one, infinitely many construction points. It is shown that, independently of the initial distribution, there exists a unique periodic process modeling the periodic steady-state behaviour of the loss system. In addition, practicable sufficient conditions for the existence of enough construction points are derived. Finally, the concept of Palm distributions for periodic random marked point processes is introduced and investigated. Employing this concept in the framework of the former multi-server loss system, it is shown that, independently of the initial distribution, there exists a unique stationary stochastic sequence modeling the behaviour of the system at the arrival instants of customers in periodic steady state.

6.5 Applications

This section deals with the work of COST-257 on some specific network elements, such as switches, resequencing units, traffic shapers and rate adapters, and some applications, for the study of which elements of queueing theory were used.

The performance evaluation of an optical transparent packet *switch* that uses optical delay lines to solve contention is addressed in [37]. In [167], an accurate novel approximation technique for the performance analysis of cell-based switch architectures is presented. The technique is based on the separate study of two parts of the switch: the internal switching fabric and the buffers associated with the output line interfaces. A blocking switch with both input and output FIFO HOL queueing is investigated in [242]. The switch consists of N input queues, N output queues and an internal switch server. It is assumed that the maximum bit rates of input and output links are equal, while the switch server speed is k times higher. The system is considered under the assumption that the input queues are saturated. In addition, the probability of a cell being addressed to any given output equals $1/N$. Under these assumptions, the cell arrival rate to any output queue is studied, when the system is in the stationary state. In the case that N tends to infinity, analytical results are obtained based on the analysis of a discrete-time M/D[k]/1 queueing system, where service occurs at discrete times, times at which at most k cells leave the system. For finite values of N, results are obtained by simulation. In [284], the performance of a prioritized $N \times N$ ATM output-queueing switch with Bernoulli arrivals is evaluated.

In [57], a discrete-time model for a so-called common buffer *resequencing unit* is proposed and analyzed. This common buffer consists of a resequencing buffer (in which cells are artificially delayed until they have experienced a constant predetermined delay in the switch) and a number of output buffers of the switch. The model used for the common buffer is a discrete-time GI-D$_L$-∞ queue followed by m GI-D-1 queues, where D and D$_L$ stand for deterministic service times of 1 and L slots, respectively. The total buffer occupancy in the common buffer is investigated by means of a generating-functions approach.

In [81, 80], the fluid flow model with variable server rate is used for studying the influence of varying channel capacity due to transmission errors on a wireless link on QoS (see Figure 6.1). The results reveal a great sensitivity to the ratio of channel and source time constants and may be used for buffer dimensioning purposes.

In [2], a continuous-time M/M/1 queueing system is analyzed in which the server can serve at two different speeds. The actual speed of the server depends on the state (empty or nonempty) of a fluid buffer. Fluid flows continuously into the fluid buffer at a constant rate, but is released from the buffer only during busy periods of the server. Hence, the contents of the fluid buffer is in turn determined by the queueing system. The queueing model serves as a mathematical model for a two-level *traffic shaper* at the edge of an ATM network. The stationary joint distribution of the number of customers in the system and the contents of the fluid buffer is investigated. From this distribution, various performance measures such as the steady-state sojourn time distribution of a customer can be obtained.

In [104], an analytical queueing analysis is presented of a *rate adaptation buffer* in case the cell arrival stream on the incoming link is modeled as an interrupted Bernoulli process and the ratio of the arrival rate versus the transmission rate can take any rational value. In particular, the means and tail distributions of buffer contents and cell delay and an accurate closed-form approximation for the cell loss ratio are obtained.

In [94], resource allocation in the case of heterogeneous bursts has been analyzed for small and large buffers with the M[k_1]+M[k_2]/D[k]/1/K discrete-time queueing model. This model represents a finite-capacity queue of size K, where bursts of size k_1 and k_2 cells arrive according to a Poisson process of respective rate λ_1 and λ_2. The service occurs at discrete time instants and up to k cells can be served during a time slot. The authors derive the distribution of the buffer occupancy and the cell loss probability by numerically solving a set of balance equations.

In [106], the author presents a good overview of existing analysis methods and algorithms for continuous-time queueing systems with a Poisson arrival process and constant service times. Both the case of a finite and of an infinite buffer capacity are considered. In [107], a simple algorithm, originally published by Badran ([31]), for evaluating GI/M/c loss systems with renewal arrival process, exponential service times and c servers is presented and explained.

In [89], the performance of the statistical multiplexing of packetized voice in Code Division Multiple Access (CDMA) systems is investigated. For the analysis of the resulting queueing system an iterative algorithm is developed. A discrete-time queueing model of the ATM Adaptation Layer 2 (AAL-2) multiplexing mechanism combined with a spacer is analyzed in [88] by means of numerical techniques.

An analytical model to evaluate the performance of Early Packet Discard (EPD) with per-VC queueing at a single node is developed in [249].

The Guaranteed Frame Rate (GFR) service allows users to send data in excess of their guaranteed service rate, but only guarantees that such traffic is transported on best effort. In order to distinguish traffic sent in line with the guaranteed service rate from traffic sent in excess, a frame-based version of the Generic Cell Rate Algorithm (GCRA) is employed. In [224], an analytical discrete-time approach is presented that forms the basis for dimensioning traffic descriptors related to the framed-based GCRA. In [275], a discrete-time analysis of the GFR service category is performed under the assumption of constant available bandwidth.

The impact of segmentation on the performance of a Go-back-N automatic-repeat-request (ARQ) protocol subject to correlated errors of the transported protocol data units is studied in [134].

In [148, 149, 226, 197], processor sharing models, M/G/1-PS or M/G/R-PS, are used as approximations for the elastic traffic.

Finally, within the COST-257 project, several queueing analyses were performed in the context of connection admission control (CAC) and dimensioning issues, see [32, 33, 56, 77, 79, 82, 149, 170, 263], the available bit rate (ABR) service category, see [15, 16, 21, 26, 27, 147, 148, 197, 223, 255, 266], internet issues, see [155, 173, 200, 268, 207], and wireless networks, see [258, 259] . For a discussion of these contributions, we refer to the chapters 5, 1 and 4.

Appendix A
COST-257 work related to Gaussian traffic models

The interest of COST-257 in Gaussian traffic models has its origin in VTT's early (since 1992) COST242 work on fractional Brownian motion as a model for LAN and Internet traffic [190, 95, 191].

A.1 Girsanov formula for fractional Brownian motion

Fractional Brownian motion is a self-similar Gaussian process with a distinctly non-Markovian character, therefore the analytical tools suitable for its handling were borrowed from the general theory of Gaussian processes instead of the renewal arguments that are familiar from queueing theory.

Since an explicit prediction formula for fBm had been found in [95], there was some hope to find also an explicit Cameron-Martin-Girsanov type formula that could be useful in some queueing problems. Such a formula was indeed found in 1996 [196].(Later it turned out that the same formula had already been discovered in the late 1960's by G. Molchan [177]!) The basic discovery was that if Z is a normalized fBm with Hurst parameter H, then the process

$$M_t = \int_0^t c_1 s^{\frac{1}{2}-H} (t-s)^{\frac{1}{2}-H} \, \mathrm{d}Z_s$$

has independent increments, and

$$\mathrm{E}M_t^2 = c_2^2 t^{2-2H}, \quad \mathrm{E}Z_s M_t = s \text{ for } 0 \leq s \leq t.$$

The constants c_1 and c_2 have explicit expressions in terms of Gamma functions.

In this context, a “handbook” [196] of fBm formulae was compiled. All formulae were checked using the ”Mathematica” software by Wolfram Research Inc.

The Girsanov formula was used in an interesting new expression for the stationary distribution function of the queue length in fractional Brownian storage in [192]. This expression, however, still contained unknown quantities which had to be replaced by approximations whose accuracy remained open.

A.2 Exact asymptotics of the queue length with fractional Brownian input

In 1996 L. Massoulie and A. Simonian at CNET found out how to apply techniques of the theory of extremes of Gaussian processes to obtain the exact tail behaviour of the distribution of fractional Brownian storage. Although the work was not reported as a COST-257 Temporary Document, their paper [171] is mentioned here since the authors were also participants of COST-257. The result is that

$$\mathrm{P}(V > x) \sim K x^{-\gamma} \exp\left(\frac{-x^{2-2H}}{2\kappa(H)^2}\right),$$

where $\kappa(H) = H^H(1-H)^{1-H}, \quad \gamma = (1-H)(2H-1)/H.$

A.3 Use of the theory of large deviations in path space

In fall 1997, it was understood at VTT how simple it is to obtain the large deviation asymptotics of fractional Brownian storage using the generalized Schilder’s theorem. In addition to the asymptotics themselves which had already been obtained by Duffield and O’Connell in 1994, this approach also gave the most probable paths along which the big queues typically develop. The technique was also successfully applied to busy period lengths, whose logarithmic asymptotics were identified with high accuracy [193].

In the following year the same technique was applied at VTT to obtain approximate queue length distributions with general Gaussian input traffic as described briefly in subsection 6.3.4. The paper [3] gives a detailed presentation of this approach. Special attention was given to the case when the cumulative arrival process has smooth paths, in contrast to the rugged paths of fBm. As an example of the effectiveness of the method, it was observed that it gives very easily the same most probable paths that were obtained in [243] as the asymptotic ones for an Anick-Mitra-Sondhi system, when the Gaussian process was given the covariance structure of the A-M-S input process.

Finally, the technique was extended to priority queues in [194]. An artificial example of the most probable paths in a system with two priorities is shown in Figure 6.2. Such paths and a corresponding probability estimate are obtained very easily as long as only lowest class traffic needs to queue when the process develops along the most probable path to a given total queue size. The general case remains essentially open.

A.4 Estimation of parameters of fractional Brownian traffic

A new method for estimating the parameters of fractional Brownian traffic was developed at Helsinki University of Technology [277] in 1999. The idea is that although a long observation time is necessary to estimate the self-similarity parameter H, it is not at all necessary to record sample values with regular small intervals whose number then becomes huge. Instead, one can use samples of the cumulative traffic process A_t at a geometric sequence of time points $t_n = t_0 + \delta b^n$, where $b > 1$. The results are excellent, outperforming the now popular wavelet estimators. The method does not, however, work for general Gaussian processes in its present form. The estimator software is available at http://keskus.hut.fi/tutkimus/com2/fbm.

A.5 A fast simulator for Gaussian traffic

A conditionalized Random Midpoint Displacement algorithm for generating samples of fractional Brownian noise was described in the COST-242 document [189] by VTT but programmed only in the simplest case where the accuracy is only satisfactory. The general form with arbitrary accuracy (with practically same speed) was realized in 1997 [195]. A version for on-the-fly simulation was developed and realized as well. The program is freely available at http://www.vtt.fi/tte/tte23/cost257/. Later it was extended to give samples of general Gaussian traffic with cumulative variance function as input, and, by Jorma Virtamo, to the case of a 2-dimensional parameter space.

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Impacts of New Services on the Architecture and Performance of Broadband Networks

COST-257 Final Report

COST-257 entitled “Impacts of new services on the architecture and performance of broadband networks” was inaugurated in November 1996. The project period of four years witnessed a dramatic change in network technology and emerging applications. At the beginning of the project, ATM was the dominating technology. During the project, IP-oriented network structures and applications gradually lessened the importance of ATM. In addition the emerging wireless networks in conjunction with their performance issues assumed also a large share of the projects core.

This booklet is intended to give a readily accessible and comprehensive overview of the results achieved during the COST-257 action. The results of COST-257 are structured in the following chapters: Network Control, Traffic Measurement, Traffic Characterization, Internet Performance, Wireless Networks and Queuing Theory.

The CD contains an extended version of the booklet text and provides multimedia information for the interested reader. While the text serves as a navigation aid to the results of COST-257, links to the references and to the technical documents of COST-257 provide a more detailed insight into the technical achievements of COST-257. Direct access to the TDs is enabled by either a searchable keyword or full text or author index.

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