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# Scalable Network Architectures Supporting Quality of Service

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To my parents, who gave me life,
and to my wife, who gives it a meaning

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[illegible]

# ABSTRACT

The realization of the advantages of packet switching networks over their circuit-switching counterparts (efficiency, resilience, flexibility) and the economical benefits of providing multiple services over a single network infrastructure has spawned great interest in the introduction of new services in the Internet. However, many of these services have stringent Quality of Service requirements that the Internet was not designed to meet. The satisfactory support of these services over the Internet is, therefore, conditioned by its ability to provide good end-to-end QoS to the network flows, but there are technical issues in doing it at the Internet scale. This thesis deals with the problems associated with providing end-to-end QoS to individual flows in a scalable way.

The thesis is divided into two parts. The first part concerns distributed models for scalable QoS provisioning in the Internet, where network routers perform both data plane and control plane tasks without recourse to centralized, off-path control entities. We evaluate the RSVP Reservation Aggregation model, an IETF proposed standard for scalable provisioning of end-to-end QoS to individual flows, and define a resource management policy for use with this model. We also propose a new approach, the Scalable Reservation-Based QoS (SRBQ) architecture, based on flow aggregation on the data plane and on a scalable model of per-flow signaling on the control plane. An evaluation of SRBQ shows that it provides good QoS metrics with higher network utilization than RSVP Aggregation.

The second part of the thesis proposes an architecture for the QoS subsystem of a next-generation, IP-based mobile telecommunications system, based on centralized control entities, designated QoS brokers. Our proposal uses a layered resource management model, tackling the aspects of QoS control in the access, in the core, and in the inter-domain path segments. We propose three scenarios for the interaction of the different entities with the QoS brokers and three corresponding strategies for coordination between application signaling and network resource reservation signaling. These different strategies allow for the customization of service deployment in different use cases. We also propose several optimizations to the joint use of the Session Initiation Protocol (SIP) and Mobile IPv6, improving the efficiency of pre-session mobility. Finally, we propose a method for inter-domain QoS routing based on virtual trunks. We define an extension to the Border Gateway Protocol (BGP) for the transport of QoS metrics and their use in the path selection process. Resorting to simulation, our proposal is evaluated and compared to a competing proposal and to the optimal solution centrally obtained using Integer Linear Programming.

# RESUMO

A percepção das vantagens das redes de comutação de pacotes em relação às de comutação de circuitos (eficiência, robustez, flexibilidade) e das vantagens económicas do fornecimento de múltiplos serviços sobre uma infra-estrutura de rede única tem gerado um grande interesse na introdução de novos serviços na Internet. Contudo, muitos destes serviços têm requisitos rígidos de Qualidade de Serviço (QoS) que a Internet não foi concebida para satisfazer. O suporte de forma satisfatória destes serviços está, portanto, condicionado pela capacidade de garantir uma boa QoS aos fluxos na rede, mas há problemas técnicos em fazê-lo à escala da Internet. Esta tese aborda os problemas associados ao suporte de QoS extremo a extremo em fluxos individuais de forma escalável.

A tese divide-se em duas partes. A primeira parte concerne modelos distribuídos para o fornecimento de QoS na Internet, em que os *routers* da rede realizam tarefas tanto do plano de dados como do plano de controlo sem recurso a entidades centrais fora do percurso dos dados. É avaliado o modelo de Agregação de Reservas RSVP, uma norma proposta pelo organismo IETF para o fornecimento escalável de QoS extremo a extremo a fluxos individuais, e é definida uma política de gestão de recursos para usar neste modelo. Também é proposta uma nova aproximação, a arquitectura *Scalable Reservation-Based QoS* (SRBQ), baseada na agregação de fluxos no plano de dados e num modelo escalável de sinalização por fluxo no plano de controlo. A avaliação do SRBQ mostra que este modelo fornece bons valores de QoS, com uma utilização mais elevada dos recursos de rede que a Agregação RSVP.

A segunda parte da tese propõe uma arquitectura para o subsistema de QoS de um sistema de telecomunicações móveis da próxima geração baseado em IP, que se baseia em entidades de controlo centralizadas, designadas mediadores de QoS. A proposta usa um modelo por camadas para a gestão de recursos, solucionando os aspectos de controlo de QoS nos segmentos de acesso, de *core*, e inter-domínio do caminho. São propostos três cenários para a interacção das diferentes entidades com os mediadores de QoS e três estratégias correspondentes para a coordenação entre a sinalização de aplicação e a de reserva de recursos de rede. Estas diferentes estratégias permitem a adaptação do fornecimento dos serviços a diferentes casos de uso. Também se propõe um conjunto de optimizações no uso conjunto do *Session Initiation Protocol* (SIP) e do Mobile IPv6 que melhoram a eficiência da mobilidade pré-sessão. Por fim, propõe-se um método para o encaminhamento inter-domínio baseado em QoS usando *virtual trunks*. É definida uma extensão ao *Border Gateway Protocol* (BGP) para o transporte de medidas de QoS e para o seu uso no processo de selecção de caminhos. Recorrendo à simulação, avalia-se a proposta e comparam-se os resultados com os de uma proposta concorrente e com os valores óptimos obtidos de forma centralizada recorrendo a Programação Linear Inteira.

# RÉSUMÉ

La prise de conscience des avantages des réseaux de commutation de paquets comparés aux réseaux de commutation de circuits (plus efficaces, robustes et flexibles) ainsi que les avantages économiques des multiples services fournis sur une infrastructure de réseau unique, suscite un intérêt pour l'introduction de nouveaux services sur Internet. Néanmoins, beaucoup de ces services ont des exigences très fortes de Qualité de Service (QoS) auxquelles Internet n'était pas préparé à répondre. Ainsi, la fourniture satisfaisante de ces services sur Internet est conditionnée par sa capacité à garantir une bonne qualité de service de bout en bout aux flux du réseau, mais il existe des problèmes techniques pour le faire à l'échelle d'Internet. Cette thèse aborde les problèmes associés à la fourniture d'une QoS de bout en bout en flux distincts de manière scalable.

La thèse se divise en 2 parties. La première partie concerne les modèles distribués permettant de fournir une QoS scalable sur Internet, où les routeurs du réseau réalisent des tâches sur le plan des données et du contrôle sans recourir à des entités centrales de contrôle hors parcours des données. Le modèle évalué est celui de l'Agrégation de Réserve RSVP, une norme proposée par l'organisme IETF pour la fourniture scalable de QoS de bout en bout aux flux distincts, et une politique de gestion des ressources a été définie pour utiliser ce modèle. Une nouvelle approche est également présentée, l'architecture *Scalable Reservation-Based QoS* (SRBQ), basée sur l'agrégation de flux au niveau des données et sur un modèle scalable de signalisation par flot au niveau du contrôle. L'évaluation du SRBQ montre que ce modèle fournit de bons résultats de QoS, avec une utilisation plus élevée des ressources du réseau que l'Agrégation RSVP.

La deuxième partie de cette thèse propose une architecture pour les sous-systèmes de QoS d'un système de télécommunication mobile de nouvelle génération basée sur IP, qui utilise l'idée des entités de contrôles centralisés, appelées médiateurs de QoS. Cette

proposition utilise un modèle de gestion de ressources par couches, traitant les aspects de contrôle de QoS au niveau des segments d'accès, *core* et inter-domaine du chemin. Nous proposons trois scénarios pour l'interaction des différentes entités avec les médiateurs de QoS et trois stratégies correspondantes pour la coordination entre la signalisation de l'application et de la réserve des ressources réseau. Ces différentes stratégies permettent l'adaptation du déploiement des services à différents cas d'utilisation. Nous présentons aussi plusieurs optimisations pour l'utilisation conjointe du *Session Initiation Protocol* (SIP) et du Mobile IPv6, qui améliore l'efficacité de la mobilité de la pré-session. Enfin, nous proposons une méthode de routage de QoS inter-domaine en utilisant *virtual trunks*. Nous avons défini une extension au *Border Gateway Protocol* (BGP) pour le transport des mesures de QoS et pour leur utilisation dans le processus de sélection de chemin. En utilisant des processus de simulation, la solution est évaluée et comparée avec une proposition concurrente et avec la solution optimale centralement obtenue en utilisant la Programmation Linéaire Entière.

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# ACRONYMS AND SYMBOLS

| | |
|---|---|
| 3GPP | Third Generation Partnership Project |
| 3G | Third Generation (mobile telecommunication systems) |
| 4G | Fourth Generation (also referred to as B3G or NGN) |
| A4C | Authentication, Authorization, Accounting, Auditing and Charging |
| AAAC | Authentication, Authorization, Accounting and Charging |
| AAL | ATM Adaptation Layer |
| ACM | Association for Computing Machinery |
| AD | Administrative Domain |
| AF | Assured Forwarding |
| AN | Access Network |
| AoR | Address of Record |
| API | Application Programming Interface |
| AQM | Active Queue Management |
| AR | Access Router |
| ARM | Advanced Router Mechanisms |
| ARPANET | Advanced Research Projects Agency Network |
| AS | Autonomous System |
| ATM | Asynchronous Transfer Mode |
| B3G | Beyond 3G |
| BA | Behavior Aggregate |
| BB | Bandwidth Broker |
| BGP | Border Gateway Protocol |
| BGRP | Border Gateway Reservation Protocol |
| BReq | Binding Request |

| | |
|---|---|
| BR | Binding Response |
| BRR | Binding Refresh Request |
| BQ | Binding Query |
| CJVC | Core Jitter Virtual Clock |
| CL | Controlled Load |
| CoA | Care-of Address |
| CoT | Care-of Test |
| CoTI | Care-of Test Init |
| CPU | Central Processing Unit |
| CS | Circuit-Switched |
| CSCF | Call Session Control Function |
| CT | Context Transfer |
| DCCP | Datagram Congestion Control Protocol |
| DiffServ | Differentiated Services |
| DSCP | Differentiated Services Code Point |
| DNS | Domain Name Service |
| DoS | Denial of Service |
| DPS | Dynamic Packet State |
| DRR | Deficit Round-Robin |
| DS | Differentiated Services |
| DV | Distance Vector |
| DVB | Digital Video Broadcasting |
| DVB-H | Digital Video Broadcasting — Handheld |
| DVB-S | Digital Video Broadcasting — Satellite |
| DVB-T | Digital Video Broadcasting — Terrestrial |
| DWDM | Dense Wavelength Division Multiplexing |
| DWRR | Deficit Weighted Round-Robin (same as DRR) |
| EF | Expedited Forwarding |
| EoIP | Everything over IP |
| ER | Edge Router |
| FHO | Fast Handover |
| FIFO | First In, First Out |
| FTP | File Transfer Protocol |
| GGSN | Gateway GPRS Support Node |

| | |
|---|---|
| GIST | General Internet Signaling Transport |
| GLB | Greatest Lower Bound |
| GPRS | General Packet Radio Service |
| GS | Guaranteed Service |
| GSM | Global System for Mobile Communications |
| GTP | GPRS Tunneling Protocol |
| HA | Home Agent |
| HKT | Home Keygen Token |
| HoA | Home Address |
| HoT | Home Test |
| HoTI | Home Test Init |
| HSPA | High Speed Packet Access |
| HSPDA | High Speed Packet Downlink Access |
| HSPUA | High Speed Packet Uplink Access |
| HSS | Home Subscriber Server |
| HTTP | Hypertext Transfer Protocol |
| iBGP | Internal BGP |
| ICMP | Internet Control Message Protocol |
| I-CSCF | Interrogating Call Session Control Function |
| IEEE | Institute of Electrical and Electronics Engineers |
| IETF | Internet Engineering Task Force |
| IKE | Internet Key Exchange Protocol |
| IMS | IP Multimedia Subsystem |
| IntServ | Integrated Services |
| IP | Internet Protocol |
| ISP | Internet Service Provider |
| ITU-T | International Telecommunication Union — Telecommunication Standardization Sector |
| JVC | Jitter Virtual Clock |
| LAN | Local Area Network |
| l-QC | Local QoS Class |
| LS | Link State |
| LSP | Label-Switched Path |
| LUB | Least Upper Bound |

| | |
|---|---|
| MARQS | Mobility Management, AAA, Resource Management, QoS and Security |
| MBAC | Measurement-Based Admission Control |
| MGW | Media Gateway |
| MIME | Multipurpose Internet Mail Extensions |
| MMSP | Multimedia Service Proxy (also Multimedia Service Platform) |
| MN | Mobile Node |
| MPLS | Multiprotocol Label Switching |
| MQC | Meta QoS Class |
| MS | Measured Sum |
| MSC | Message Sequence Chart; also Mobile Switching Center (in UMTS) |
| MT | Mobile Terminal |
| MTU | Maximum Transmission Unit |
| NAI | Network Access Identifier |
| NAPTR | Naming Authority Pointer (DNS) |
| NAT | Network Address Translation |
| NGN | Next Generation Network |
| NSIS | Next Steps in Signaling |
| NSLP | NSIS Signaling Layer Protocol |
| NTLP | NSIS Transport Layer Protocol |
| NVUP | Network View of the User Profile |
| OSPF | Open Shortest Path First |
| PBAC | Parameter-Based Admission Control (also Probe-Based Admission Control) |
| PC | Paging Controller |
| P-CSCF | Proxy Call Session Control Function |
| PDP | Policy Decision Point |
| PEP | Policy Enforcement Point |
| PHB | Per-Hop Behavior |
| PRNG | Pseudo-Random Number Generator |
| PSTN | Public Switched Telephone Network |
| q-BGP | QoS-enhanced BGP |
| QoS | Quality of Service |
| RAM | Random Access Memory |
| RAN | Radio Access Network |
| RED | Random Early Detection |

| | |
|---|---|
| RIO | Random Early Detection with In/Out Bit |
| RRP | Return Routability Procedure |
| RSVP | Resource Reservation Protocol |
| RSVPRAgg | RSVP Reservation Aggregation |
| RTCP | Real-Time Control Protocol |
| RTP | Real-Time Protocol |
| RTT | Round Trip Time |
| SCORE | Stateless Core |
| S-CSCF | Serving Call Session Control Function |
| SCTP | Stream Control Transmission Protocol |
| SDP | Session Description Protocol |
| SDPng | Session Description Protocol — New Generation |
| SGSN | Serving GPRS Support Node |
| SIB | Seamless Integration of Broadcast |
| SIP | Session Initiation Protocol |
| SR | Subdomain Router |
| SRBQ | Scalable Reservation-Based QoS |
| SRV | Service Record (DNS) |
| SS7 | Signaling System #7 |
| SVUP | Service View of the User Profile |
| TCP | Transmission Control Protocol |
| TD-CDMA | Time Division — Code Division Multiple Access |
| TSC | Time Stamp Counter |
| UA | User Agent (SIP) |
| UDP | User Datagram Protocol |
| URI | Uniform Resource Identifier |
| URL | Uniform Resource Locator |
| USP | Ubiquitous and Seamless Pervasiveness |
| UTRAN | UMTS Terrestrial Radio Access Network |
| VC | Virtual Clock |
| VID | Virtual Identity (also Virtual Identifier) |
| VLL | Virtual Leased Line |
| VoIP | Voice over IP |
| VQ | Virtual Queues |

| | |
|---|---|
| WCDMA | Wideband Code Division Multiple Access |
| WFQ | Weighted Fair Queuing |
| WiFi | Wireless Fidelity (IEEE 802.11) |
| WiMax | Worldwide Interoperability for Microwave Access (IEEE 802.16) |
| WRR | Weighted Round-Robin |
| YESSIR | Yet Another Sender Session Internet Reservation Protocol |

# CHAPTER 1

# INTRODUCTION

*The Holy Grail of computer networking is to design a network that has the flexibility and low cost of the Internet, yet offers the end-to-end quality of service guarantees of the telephone network.*

*S. Keshav*

Having its roots in the military ARPANET, conceived as a data transport network with a focus on resilience, the Internet supports only a best effort service model, where all packets are treated the same way, therefore providing a single level of service. Now that the Internet is becoming the ubiquitous global communication infrastructure, new applications are emerging with more demanding and diversified requirements than data transport. Internet telephony, for example, has much stricter delay requirements than remote terminal, the most demanding of the original applications. The deployment of other service models providing better Quality of Service (QoS) than best effort is of great importance for the transport of these new applications.

Recommendation E.800 of the ITU-T [ITU-TE.800] has defined QoS as "the collective effect of service performance, which determines the degree of satisfaction of a user of the service." Notwithstanding the intrinsically subjective nature of the concept, QoS in packet switching networks may be ascertained through a number of objective network performance metrics. The most relevant metrics for QoS are the packet delay, delay variation (jitter), throughput and packet loss ratio.

For most practical purposes, the generous amounts of capacity made possible by recent technological advances (particularly in the optical link technologies used in the backbones) provide good QoS parameters to the applications as long as the utilization remains low. The main objective of QoS provisioning models is, therefore, to provide guarantees regarding those parameters, ensuring they remain adequate for the applications' requirements even when the network load increases, giving rise to congestion. Although adding more capacity to the network may alleviate the effects of congestion, in the long term the extra capacity will end up being used, making overprovisioning more of a band-aid than an actual solution to the QoS problem — a true solution must not only provide good QoS parameters, but also provide guarantees that make them predictable and dependable.

Aiming at the introduction of QoS support in the Internet, the IETF has proposed two major frameworks: Integrated Services (IntServ) and Differentiated Services (DiffServ). The IntServ architecture was the first one to emerge. Based on per-flow reservations subject to admission control, IntServ provides service classes with both strict and soft QoS guarantees. IntServ's strengths lie in its capability of providing both strict QoS guarantees and adequately efficient resource usage (particularly in the case of multicast flows). However, IntServ is fundamentally flawed regarding scalability: the need to implement complex packet scheduling algorithms and to classify every packet based on both layer 3 and layer 4 header information is unbearable to high-performance routers in core networks. Moreover, the use of RSVP signaling for reserving resources and maintaining per-flow state does not scale to a very large number of simultaneous flows. The DiffServ architecture emerged as a scalable alternative to IntServ that does not suffer from scalability problems: there is no per-flow resource reservation, flows are aggregated in classes according to specific characteristics, and service differentiation is performed based on the service class to which each packet belongs. The drawback of DiffServ is that without flow admission control mechanisms, traffic within a class may exceed the amount supported by the provisioned resources, degrading the QoS received by flows in that class due to cross-interaction either at the packet schedulers or at the edge traffic conditioners.

This thesis deals with the problems associated with providing end-to-end QoS to individual flows in a scalable way while maintaining good usage of the network resources, conciliating the benefits of the IntServ and DiffServ architectures. The text is divided into two parts. The first part deals with distributed approaches, where the network routers cooperate among themselves to perform not only the data plane functions (e.g., packet scheduling) necessary for providing QoS guarantees, but also the control plane functions (e.g., resource reservation signaling). We evaluate an existing model, based on the aggregation of RSVP reservations inside a network domain, and propose a new one, designated Scalable Reservation-Based QoS (SRBQ), based on flow aggregation on the data plane and on a scalable model of per-flow signaling on the control plane. In the second part of the thesis we propose an architecture for the QoS subsystem of a next generation IP-based mobile telecommunications system, supporting not only SIP-based multimedia applications, but also all kinds of IP-based applications, including legacy ones. In this architecture, QoS control is segmented, and control plane functions are centralized at entities designated QoS Brokers. We address several issues, mostly related to signaling, and propose a model for inter-domain QoS routing. The work described in this second part of the thesis was developed in the scope of the DAIDALOS Integrated Project (IST-2002-506997) in the Thematic Priority "Information Society Technologies" of EU Framework Programme 6 for Research and Development.

## 1.1 Main Contributions

The main contributions of this thesis to the state of the art are the following:

- An analysis of the RSVP Reservation Aggregation model proposed in [RFC3175] and the proposal of a bandwidth management policy and of a workaround to an identified problem with the loss of *ResvConf* in the standard. The analysis includes the identification of the strengths and weaknesses of the model and a performance evaluation regarding QoS parameters and scalability.
- Proposal and definition of a new QoS architecture based on the aggregation of flows on the data plane and on a scalable model of per-flow signaling on the data plane; the proposal of algorithms for achieving scalability with a per-flow signaling model; the evaluation of the architecture regarding QoS and scalability and its comparison to RSVP Aggregation.
- Contributions to the proposal of the QoS subsystem of a new architecture for next generation IP-based mobile networks.
- Proposal of several scenarios for the integration of application signaling and resource reservation signaling in the DAIDALOS architecture. Evaluation of the proposed scenarios.

- Identification of inefficiency problems with the joint use of SIP, Mobile IPv6 and end-to-end resource reservation, and the proposal of a number of optimizations for the elimination of those inefficiencies. Evaluation of the proposed optimizations.
- Proposal of a solution to the problem of inter-domain QoS routing based on virtual trunks, including the formulation of the problem in Integer Linear Programming for obtaining the optimal solution, and a practical solution based on a QoS extension of the Border Gateway Protocol (BGP). Validation of the practical solution and performance comparison with standard BGP and with another recently proposed inter-domain QoS routing protocol, as well as the optimal solutions.

Some other noteworthy, albeit less important, contributions of the work herein presented are ns-2 implementations of the RSVP Reservation Aggregation model, SRBQ and SIP, available from [PriorNS].

## 1.2 Publications

The work performed in the scope of this thesis gave rise to a number of publications: one technical report, 2 papers published in national conferences with referees, 14 in international conferences with referees (4 of which were published in the Lecture Notes in Computer Science book series), one journal paper and one chapter in an encyclopedia. Another journal paper has been submitted and is pending acceptance. Most of the work developed in the scope of the DAIDALOS project was also published in several project deliverables. These publications are enumerated below, categorized by type and in reverse chronological order.

### 1.2.1 Journals, Book Series and Books

- R. Prior and S. Sargento, "Inter-Domain QoS Routing — Optimal and Practical Study." In *IEICE Transactions on Communications*, vol. E90-B, no. 3, pp. 549–558, IEICE, March 2007.

- R. Prior and S. Sargento, "Scalable Reservation-Based QoS Architecture — SRBQ." In *Encyclopedia of Internet Technologies and Applications*, Idea Group Publishing. (to appear)

- R. Prior, S. Sargento, P. Brandão and S. Crisóstomo, "SRBQ and RSVPRAgg: A Comparative Study." In *Telecommunications and Networking — ICT 2004, 11th International Conference on Telecommunications*, Lecture Notes in Computer Science vol. 3124, pp. 1210–1217, Springer-Verlag, August 2004.

- R. Prior, S. Sargento, P. Brandão and S. Crisóstomo, “Performance Evaluation of the RSVP Reservation Aggregation Model.” In *High-Speed Networks and Multimedia Communications*, Lecture Notes in Computer Science vol. 3079, pp. 167–178, Springer-Verlag, June 2004.
- R. Prior, S. Sargento, P. Brandão and S. Crisóstomo, “Comparative Evaluation of Two Scalable QoS Architectures.” In *Networking-2004*, Lecture Notes in Computer Science vol. 3042, pp. 1452–1457, Springer-Verlag, May 2004.
- R. Prior, S. Sargento, P. Brandão and S. Crisóstomo, “Efficient Reservation-Based QoS Architecture.” In *Interactive Multimedia on Next Generation Networks*, Lecture Notes in Computer Science vol. 2899, pp. 168–181, Springer-Verlag, November 2003.

### 1.2.2 International Proceedings with Independent Reviewing

- R. Prior and S. Sargento, “SIP and MIPv6: Cross-Layer Mobility.” In *Proceedings of the 12th IEEE Symposium on Computers and Communications* (ISCC’2007). (to appear)
- R. Prior and S. Sargento, “Inter-Domain QoS Routing with Virtual Trunks.” In *Proceedings of the IEEE International Conference on Communications* (ICC’2007). (to appear)
- R. Prior and S. Sargento, “Virtual Trunk Based Inter-Domain QoS Routing.” In *Proceedings of the 6th Conference on Telecommunications* (ConfTele’2007). (to appear)
- R. Prior and S. Sargento, “Towards Inter-Domain QoS Control.” In *Proceedings of the 11th IEEE Symposium on Computers and Communications* (ISCC’2006), Cagliari, Sardinia, Italy, June 2006.
- R. Prior and S. Sargento, “QoS and Session Signaling in a 4G Network.” In *Proceedings of the IEEE International Conference on Networks* (ICON’2005), Kuala Lumpur, Malaysia, November 2005.
- R. Prior, S. Sargento, J. Gozdecki and R. Aguiar, “Providing End-to-End QoS in 4G Networks.” In *Proceedings of the 3rd IASTED International Conference on Communications and Computer Networks* (CCN’2005), Marina del Rey, CA, USA, October 2005.

- S. Sargento, R. Prior, F. Sousa, P. Gonçalves, J. Gozdecki, D. Gomes, E. Guainella, A. Cuevas, W. Dziunikowski and F. Fontes, “End-to-end QoS Architecture for 4G Scenarios.” In *Proceedings of the 14th Wireless and Mobile Communications Summit*, Dresden, Germany, June 2005.
- R. Prior, S. Sargento, D. Gomes and R. Aguiar, “Heterogeneous Signaling Framework for End-to-end QoS support in Next Generation networks.” In *Proceedings of the 38th Hawaii International Conference on System Sciences* (HICSS 38), Hawaii, January 2005.
- R. Prior, S. Sargento, P. Brandão and S. Crisóstomo, “Evaluation of a Scalable Reservation-Based QoS Architecture.” In *Proceedings of the 9th IEEE International Symposium on Computers and Communications* (ISCC’2004), Alexandria, Egypt, June 2004.
- R. Prior, S. Sargento, S. Crisóstomo and P. Brandão, “End-to-End QoS with Scalable Reservations.” In *Proceedings of the 11th International Conference on Telecommunication Systems, Modeling and Analysis* (ICTSM11), Monterey, CA, USA, October 2003.

### 1.2.3 National Proceedings with Independent Reviewing

- R. Prior and S. Sargento, “Arquitectura Escalável para o Suporte de QoS em Redes IP.” In *Actas da 7a Conferência sobre Redes de Computadores* (CRC’2004), Leiria, Portugal, September 2004.
- D. Gomes, R. Prior, S. Sargento and R. Aguiar, “Integration of Application-level and Network-level Signalling: From UMTS toAll-IP.” In *Actas da 7a Conferência sobre Redes de Computadores* (CRC’2004), Leiria, Portugal, September 2004.

### 1.2.4 Pending

- R. Prior and S. Sargento, “Cross-Layer Mobility with SIP and MIPv6.” Submitted to the *Journal of Internet Technology*, Special issue on IPv6-based Mobile/Multimedia Applications, Taiwan Academic Network.

### 1.2.5 Technical Reports

- R. Prior, S. Sargento, S. Crisóstomo and P. Brandão, “End-to-End QoS with Scalable Reservations.” Technical report DCC-2003-01, DCC-FC & LIACC, Universidade do Porto, April 2003.

### 1.2.6 Project Deliverables

- S. Sargento (ed.) et al., "QoS System Implementation Report." Daidalos (IST-2002-506997) consortium deliverable D322, October 2005.
- D. Bijwaard (ed.) et al., "Network Architecture Design and Sub-Systems Interoperation Specification." Daidalos (IST-2002-506997) consortium deliverable D312, February 2005 (updated January 2006).
- S. Sargento and D. Gomes (eds.) et al., "QoS Architecture and Protocol Design Specification." Daidalos (IST-2002-506997) consortium deliverable D321, August 2004.
- D. Bijwaard (ed.) et al., "Initial Network Architecture Design and sub-Systems Interoperation." Daidalos (IST-2002-506997) consortium deliverable D311, May 2004.

## *1.3 Organization*

The rest of this thesis is organized as follows. Chapter 2 gives an overview of the more relevant work within the different subjects dealt with in this thesis. It begins with a description of the different building blocks employed by QoS provisioning models, identifying some design choices and tradeoffs. Then it describes the two most prominent frameworks standardized by the Internet Engineering Task Force (IETF) for QoS support on the Internet, Integrated Services (IntServ) and Differentiated Services (DiffServ). These models represent extreme opposites in their characteristics and design options: IntServ excels in flexibility and fine per-flow QoS control, whereas DiffServ provides only coarse-grained, per-aggregate QoS control, but is extremely scalable and appropriate for use in high-speed core networks. Several other QoS frameworks are also presented, grouped according to their most important design characteristics: alternative signaling models, the use of flow aggregation, the elimination of state maintenance in core nodes, and the removal of QoS control plane functions from routers and their placement in central entities, designated Bandwidth Brokers. We also describe some models for addressing two aspects of (aggregate) inter-domain QoS: inter-domain resource reservation and inter-domain QoS routing. Next, we give an introduction to IP-based mobile telecommunication systems, including a retrospective of the convergence of UMTS towards an All-IP architecture centered on the IP Multimedia Subsystem (IMS), and architectural proposals for next generation networks with QoS support over heterogeneous access technologies. Finally, we introduce the Session Initiation Protocol (SIP), central to the IMS, discussing its integration with resource reservation signaling and with mobility management. The large amount of work in different areas described in chapter 2 stems from the broad spectrum of subjects covered by the work performed in this thesis.

Chapters 3 to 8 describe the original work performed in the scope of this thesis, and is broadly divided into two parts. The first part consists on chapters 3 and 4, and concerns distributed approaches to resource reservation and QoS provisioning in the Internet, where network elements along the data path perform both data plane and control plane functions.

Chapter 3 contains an analysis of the RSVP Reservation Aggregation model [RFC3175]. We identify some problems with the specification and clarify some areas where it is lacking, and define a bandwidth management policy for the aggregates, considered out of scope of [RFC3175]. Based on simulation results obtained using our implementation of the model in ns-2 [PriorNS], we evaluate the performance of this model regarding the standard QoS parameters (delay, jitter and packet loss ratio) and other relevant performance and scalability parameters, such as network utilization and the signaling load at core nodes; these results are compared to those obtained with the RSVP/IntServ model. We also analyze the influence of some tunable parameters in the behavior of the model and provide some guidelines for setting their values.

In chapter 4 we propose a new architecture for scalable QoS provisioning, named Scalable Reservation-Based QoS (SRBQ). This architecture combines aggregate data plane processing (packet classification, scheduling, policing and shaping) with a scalable model of per-flow signaling on the control plane. Signaling scalability is achieved through the minimization of the computational complexity of the different tasks involved, and some new mechanisms and algorithms are employed to this end. Resorting to ns-2, we perform an extensive evaluation of the SRBQ architecture, analyzing QoS and scalability aspects with several different experiments involving both synthetic test flows and real multimedia streams. We also compare SRBQ to the RSVP Reservation Aggregation architecture, both qualitatively and quantitatively.

The second part of the thesis, consisting on chapters 5–8, describes our contributions to the next generation IP-based mobile network architecture developed in the scope of phase 1 of the IST DAIDALOS Integrated Project [Daidalos], mostly centered on the QoS subsystem and on signaling aspects.

In chapter 5 we introduce the project, presenting its major goals, key concepts and development guidelines, its scenario-based development model, and its division into work packages. We give an overview of the network architecture, with a natural focus on the QoS subsystem, highlighting the differences from the UMTS/IMS architecture. We describe the interactions between the different components for providing QoS across the different segments of the end-to-end path (access, core and inter-domain), using a layered model of resource management that allows the architecture to scale well in the core and yet provide per-flow QoS.

Chapter 6 discusses resource management in the access. We propose several scenarios for the coordination of application-level signaling and network-level resource reservation signaling, centered on different entities — mobile terminal, access router or multimedia proxy. The different signaling scenarios provide support for virtually any type of application (examples are provided for SIP-based multimedia conferences and for legacy data applications) and allow for the optimization of different exploitation cases. We also describe how session renegotiation is coordinated with handover signaling in order to fully explore the resources available in the different network access technologies. A comparative analysis of the different scenarios is performed, based on their use cases and on efficiency and scalability results obtained from their simulation under ns-2.

In chapter 7 we identify some inefficiency problems arising from the joint use of SIP and Mobile IPv6, particularly when end-to-end resource reservations must be performed for the media, and propose an optimized initiation sequence using some cross-layer interactions that removes those inefficiencies. The benefits of our proposal are ascertained through a delay analysis of both standard and optimized sequences and through simulation results.

In chapter 8 we address the problem of inter-domain QoS routing, an integrating part of our proposed solution to the problem of end-to-end QoS and resource management in the DAIDALOS architecture. Our proposal is based on Service-Level Agreements (SLAs) for data transport between peering domains, using virtual-trunk type aggregates. We formally state the problem and formulate it in Integer Linear Programming (ILP), proving that routes obtained through the optimization process are free of cycles. Using a Mixed Integer Programming (MIP) code, the ILP formulation allows us to obtain the optimal set of routes for a given network configuration and traffic matrix; however, this solution cannot be used in practice, since the problem is NP-hard. Therefore, we propose a practical solution based on a QoS extension to the Border Gateway Protocol (BGP) based on two static metrics (assigned bandwidth and light load delay) and one coarse-grained dynamic metric (congestion alarm). Based on simulation results we validate our proposal and evaluate its performance compared with standard BGP, with the QoS_NLRI BGP extension [Cristallo04] and with the optimal route set provided by the ILP optimization.

Finally, chapter 9 presents the main conclusions of the work described in this thesis, along with some directions for further work.

# CHAPTER 2

# RELATED WORK

The introduction of Quality of Service (QoS) mechanisms in the Internet has been a heavily researched topic in networking for more than a decade. Numerous proposals of architectures, technologies and mechanisms have been made with the goal of enriching the Internet with QoS guarantees that the current best effort model cannot provide. This chapter introduces some of the more relevant research work that has been carried out in this field.

When discussing QoS, it is inevitable to mention Asynchronous Transfer Mode (ATM), due to its rich set of built-in QoS features. However, ATM is a layer 2, connection-oriented technology. Since our work concerns QoS in connectionless packet switching networks (even though QoS "connections" may be established as an overlay), we will not discuss ATM.

This chapter is organized as follows. Section 2.1 describes the general building blocks of QoS provisioning frameworks. Section 2.2 describes the two major QoS architectures standardized by the IETF, IntServ and DiffServ. Section 2.3 describes other proposed QoS models, categorized into models based on alternative signaling (sec. 2.3.1), models based on the aggregation of flows (sec. 2.3.2), models based on the elimination of state at the core (sec. 2.3.3), and models based on centralized management entities, generally designated Bandwidth Brokers (sec. 2.3.4). Section 2.4 describes different proposals for two orthogonal aspects of inter-domain QoS provisioning: inter-domain resource reservation and inter-domain QoS routing. Descending from a very different type of networks (circuit-switched cellular networks), mobile telecommunication systems are progressively converging towards an All-IP architecture that will ultimately be merged with the Internet, becoming the universal and ubiquitous communication infrastructure. Section 2.5 describes the origins and current state of the UMTS architecture, as well as some developments towards the next generation IP-based mobile networks. Finally, section 2.6 introduces SIP, the call signaling protocol used in UMTS' IP Multimedia Subsystem (IMS), and that will probably also be used for managing multimedia sessions in next generation networks. We give an overview of the protocol and describe its interaction with network resource reservation for QoS support and with mobility management.

## 2.1 Building Blocks for a QoS Framework

Before we start presenting existing QoS frameworks (both standardized and proposed), we will present the different blocks used to build them. Note that not all of these building blocks (or functions) are used in each QoS framework, in accordance to its scope or characteristics — for example, a framework based on static allocation does not require signaling. The efficiency of all these functions must be taken into consideration when designing a scalable QoS architecture.

### 2.1.1 Packet Classification

Contrary to the standard best-effort service, where all packets are treated equal, QoS frameworks usually involve some form of differentiation in packet handling. In order to perform this differentiation, a packet classification function is necessary for associating each packet to the service it will receive. Packet classification may be based on a single or multiple fields from the packet header.

### 2.1.2 Queuing/Scheduling

Packets are queued according to the results of the classification function. The scheduling function decides which, among the queued packets, is to be transmitted next. It is, thus, responsible for sharing the available capacity among the different flows, as well as controlling the queuing delay experienced by packets. Scheduling algorithms may be broadly classified into two types: work-conserving and non-work-conserving. With work-conserving schedulers, the link is never idle as long as there is at least one queued packet — they merely control the order in which packets get transmitted. Non-work-conserving schedulers, on the other hand, can delay the transmission of a packet until it becomes eligible; therefore, the link may be idle even when there are queued packets if none of those packets is eligible. Even though non-working-conserving schedulers waste transmission capacity whenever only non-eligible packets are queued, they are useful in controlling jitter.

The scheduling algorithm is probably the most determinant one in terms of scalability. On one hand, it is run for every packet traversing the router, and packets must be processed at line rate. On the other hand, a scheduler that performs differentiation involves sorting, and even the best available sorting algorithms have worst case complexity of O(log(N)) for the insertion or removal of each of the N elements. If N is the number of flows, that may be $10^6$ or higher in a core router, $\log_2(10^6) = 20$ elemental steps must be performed for queuing and/or dequeuing a packet; given that each elemental step involves a number of instructions, including conditionals, which are generally expensive, it is impractical to use an algorithm that implies such operation being performed for every packet at line rate.

### 2.1.3 Metering

The metering function is responsible for the measurement of temporal properties, the most common one being the rate, of a traffic stream (as determined by a classifier). The results of metering may be used to affect the marking, shaping or policing functions, and also for accounting purposes.

### 2.1.4 Traffic Shaping

The shaping function is responsible for delaying out-of-profile packets for the necessary amount of time for them to become in-profile. It is frequently used to restore the envelope of a traffic stream distorted by cross traffic, and is useful for reducing jitter within the stream.

### 2.1.5 Traffic Policing

Traffic policing is an alternative function for handling out-of-profile packets on a traffic stream. Unlike the shaping function, traffic policing does not attempt to force out-of-profile packets into conformance; instead, it “punishes” out-of-profile packets by dropping them or demoting them to a lesser class (usually, the best effort class). By doing so, it ensures that only in-profile packets receive the contracted treatment. Policing is preferred to shaping whenever the existence of some amount of packet loss is a lesser evil than the introduction of additional delay.

### 2.1.6 Packet Marking

The marking function is responsible for setting the value of some header field in the packet header; this information is used for classification (or other purposes) in downstream nodes. It may be used to provide a simpler means of classification in the core nodes of a network by marking a single field in the packet header with the results of multi-field classification performed at the ingress node. Packet marking may be performed based on the results of other functions, notably packet classification and traffic policing (e.g., for identifying out-of-profile packets).

### 2.1.7 Admission Control

Admission control is the function that decides whether a new flow should be accepted or rejected in the network. The decision is usually performed taking into account the announced profile of the flow, the current network or class load, and network policies. There are two main classes of admission control algorithms: Parameter-Based Admission Control (PBAC) and Measurement-Based Admission Control (MBAC); these classes are described in appendix A. The rejection of flows for which there would be insufficient resources prevents QoS degradation for the active (previously admitted) flows.

### 2.1.8 Signaling

The QoS signaling function is used to establish, maintain and remove reservation states for traffic streams in the network nodes. It usually invokes the admission control function, and frequently triggers changes in other modules (notably scheduling and classification). QoS signaling may be classified in two broad categories: single-tier and two-tier [Vali04]. Single-tier signaling assumes an end-to-end homogeneous QoS architecture for the Internet, with all routers along the path supporting the same mechanisms; QoS state is established and maintained in every intermediate router, and the same path is followed by signaling and data packets alike. Two-tier signaling recognizes that the Internet is a collection of interconnected Autonomous Systems (ASs) that are technologically and administratively independent; therefore, resource management is split into two categories — intra-domain, performed within each AS, and inter-domain, performed across the different ASs — and different signaling procedures are used in each category. The provision of end-to-end QoS requires appropriate interworking between intra- and inter-domain signaling. QoS signaling schemes may be further classified according to other aspects:

- In-band or out-of-band

- Per-flow or per-aggregate
- Sender-initiated or receiver-initiated
- Hard-state or soft-state (which must be periodically refreshed)
- Centralized or distributed
- Triggered by the end hosts, edge routers or other entities

## *2.2 Main IETF Frameworks for QoS*

This section describes the two major frameworks — Integrated Services (IntServ) and Differentiated Services (DiffServ) — standardized by the Internet Engineering Task Force (IETF) for QoS support on the Internet. Their radically different design stems from the very different premises that have driven their conception: IntServ was designed for providing tight and mathematically proven per-flow QoS on a "flat" Internet, whereas DiffServ was designed as a scalable framework for providing QoS to aggregates of flows on a hierarchical Internet, where different operators independently manage resources in their own domains using the mechanisms they see fit and establish Service Level Agreements (SLAs) for transport service provisioning with the other operators they connect to.

### 2.2.1 IntServ

Aiming at the introduction of services with QoS requirements that cannot be satisfied by the standard best effort delivery, the IETF Integrated Services (IntServ) working group specified an architecture for providing elevated services [RFC1633]. This architecture is an overlay to the standard IP routing infrastructure, conciliating the datagram model of the Internet with per-flow QoS guarantees obtained through the reservation of resources. The type of QoS guarantees and the quantification of the resources to be reserved are specified by the application, usually resorting to the Resource Reservation Protocol (RSVP), described below. IntServ supports two service classes, the Controlled Load service, providing soft QoS guarantees, and the Guaranteed Service, providing hard QoS guarantees in terms of packet delivery and bounded delay.

The resource reservation process is performed in several steps. The application identifies the flow — using the 5-tuple (*source IP address*, *destination IP address*, *transport protocol*, *source port*, *destination port*) in IPv4 — and characterizes its traffic envelope and QoS requirements. It then requests the network to reserve resources along the flow's path as defined by standard IP routing. At each router along the path, the request is subject to admission control; if the flow is admitted, the router installs reservation state for classifying and scheduling its packets. It is worth noting that IntServ is a one-tier model, where all routers along the path support the same mechanisms and procedures.

After successfully finishing the resource reservation along the entire path, the application may expect the negotiated QoS from the network, provided the path does not change and the flow respects the reserved profile. In the event of path changes, the service will be disrupted, as resources will not be reserved along the entire new path; there are, however, mechanisms to recover from this situation. Policing and shaping mechanisms ensure that the flow profile is respected; out-of-profile packets are treated as best effort or eliminated altogether.

The IntServ architecture is not limited to unicast flows. In fact, since the beginning it was conceived having support for multicast applications (notably audio- and video-conference) in mind, and provides mechanisms for merging and splitting of flows (see [RFC2211] and [RFC2212] for details). Even though multicast support is a key feature of the IntServ model, the complexity required to support it is one of its “Achilles’ heels”.

#### *2.2.1.1 Controlled Load Service*

The Controlled Load (CL) service [RFC2211] emulates the behavior of a Best Effort (BE) service provided under unloaded conditions. The big difference between CL and BE is that CL flows do not experience noticeable service degradation even when network is overloaded — a CL flow will consistently have a packet loss ratio comparable to that introduced by the error rate of the transmission medium, as well as a transit delay not greatly exceeding the minimum delay experienced by a successfully transmitted packet.

In order to provide such guarantees, all network elements along the path must make sure that enough resources are available to support all CL flows. This is achieved by means of Admission Control: clients requesting the CL service provide an estimated envelope (*TSpec*) for the data traffic they will generate; if there are enough resources to handle traffic conforming to the specified envelope, the flow is accepted; otherwise it is rejected. The *TSpec* used by the CL service is the TOKEN_BUCKET_TSPEC defined in [RFC2215], containing the following information:

- A token rate, $r$, which is an upper bound for the long term rate of the flow
- A bucket depth, $b$, limiting the burstiness of the flow
- A peak rate, $p$

- A maximum packet size, *M* — CL flows with a value of *M* larger than the MTU of a link must be rejected; otherwise, the queue could be jammed by an oversize packet that could never be transmitted
- A minimum policed unit, *m* — packets smaller than *m* are counted against the token bucket as being of size *m*

This specification, basically a double token bucket characterized by ((*r*,*b*),(*p*,*M*)) — refer to fig. 2.1 — is used for compatibility with other QoS services; however, the peak rate, *p*, has little significance for the CL service, and is usually ignored by the admission control (end hosts may set it to infinity). Traffic falling outside the requested *TSpec* is usually treated as BE traffic, though it is not explicitly required by [RFC2211].

The CL service is appropriate for a broad class of applications that can tolerate an occasional lost or delayed packet but are highly sensitive to network overload. Important members of this class are the so-called "adaptive real-time applications", such as IP telephony or videoconference, which work well in unloaded best-effort networks but degrade quickly under overload conditions.

#### *2.2.1.2 Guaranteed Service*

Some applications, notably circuit emulation and applications depending on it, have stricter QoS requirements than those provided by the CL class. The Guaranteed Service (GS) [RFC2212] provides these hard real-time flows with assured levels of bandwidth that result in firm, mathematically provable bounds on end-to-end packet delays and a complete absence of packet dropping due to queue overload for all conforming packets. This is achieved by reserving, in each router along the path, given amounts of transmission capacity and buffer space for each GS flow. To this end, in addition to the specification of the traffic envelope (*TSpec*), the GS service uses a specification of the reservation (*RSpec*), containing a reserved

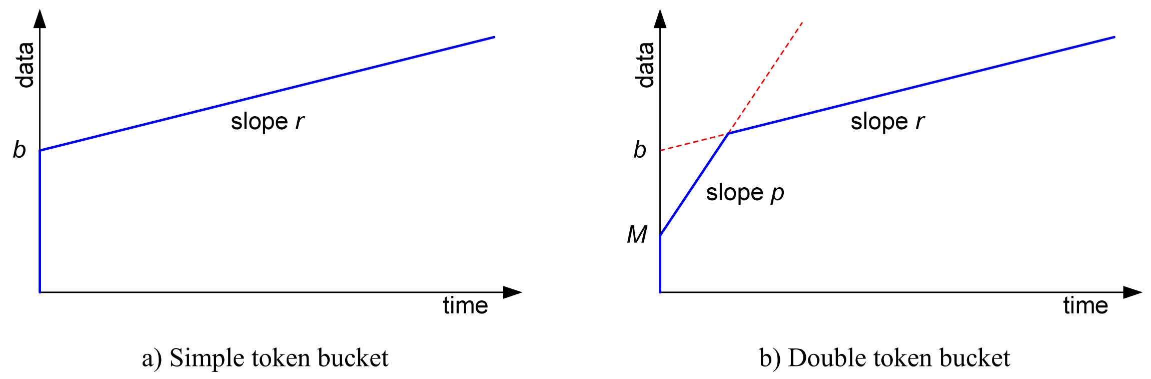

**Figure 2.1: Token bucket traffic specification**

rate $R$ and a slack parameter $S$ (the amount of buffer space is not included in *RSpec* because it may be derived by the router from the *TSpec*, *RSpec* and other received parameters).

In a fluid flow model, a flow bounded by a token bucket of rate $r$ and depth $b$, for which there is a reserved capacity $R$, is delayed by no more than $b/R$, provided that $R \geq r$. However, the GS is only an approximation of the fluid model (there is not a leased line, only an amount of reserved capacity on a line where this flow is multiplexed with others); therefore, two error terms are introduced to characterize the maximum deviation from the ideal model: $C$ (dependent on $R$) and $D$ (independent of $R$). Using these error terms, the delay bound becomes $b/R + C/R + D$. Since traffic envelope of GS flows is bounded by a double token bucket (the *TSpec*) that limits the peak rate to $p$, a tighter bound on the end-to-end queuing delay may be derived [Parekh93, Parekh94]; this bound is shown in eq. (2.1), where $C_{\text{tot}}$ and $D_{\text{tot}}$ represent the summed values of the $C$ and $D$ parameters for all routers along the path.

$$QDelay \leq \begin{cases} \dfrac{(b-M)(p-R)}{R(p-r)} + \dfrac{M + C_{\text{tot}}}{R} + D_{\text{tot}} & \Leftarrow p > R \geq r \\ \\ \dfrac{M + C_{\text{tot}}}{R} + D_{\text{tot}} & \Leftarrow R \geq p \geq r \end{cases} \tag{2.1}$$

Note the absence of the first term of the delay in the second case, where the peak rate is lower than the reserved rate — since the reserved capacity exceeds the peak rate, there is no need to drain out the bucket.

The slack term $S$ is used to add some flexibility to the reservation process, increasing the chances of a reservation being accepted. The receiver requests a larger amount of bandwidth than strictly required to meet the required delay bound, thus obtaining a lower value for the bound, and places the difference in $S$. Routers using deadline-based schedulers (which decouple rate and delay guarantees) that would not be able to meet the required delay bound but could accept a larger bound, may still accept the reservation provided the slack is at least as large as the difference between the bounds (the difference is subtracted from the value of $S$ sent upstream). On rate-based schedulers, a lower rate (than both the requested rate and the minimum rate required to meet the end-to-end delay bound) may be reserved at a router with insufficient capacity (and upstream) by consuming some slack, provided that eq. (2.2) holds. In eq. (2.2), $C_{\text{tot}i}$ is the sum of the $C$ terms upstream of the current node, $i$, including itself, ($R_{\text{in}}$, $S_{\text{in}}$) is the received *RSpec* and ($R_{\text{out}}$, $S_{\text{out}}$) the modified *RSpec* sent upstream.

$$S_{\text{out}} + \frac{b}{R_{\text{out}}} + \frac{C_{\text{tot}}}{R_{\text{out}}} \leq S_{\text{in}} + \frac{b}{R_{\text{in}}} + \frac{C_{\text{tot}i}}{R_{\text{in}}}, \quad r \leq R_{\text{out}} \leq R_{\text{in}} \tag{2.2}$$

It is worth noting that the reserved capacity that is not used by a GS flow is not wasted, it is used to temporarily benefit other flows (or classes).

Guaranteed Service flows must be policed at the network access or ingress points for ensuring conformance to the *TSpec*. Moreover, traffic reshaping is required at reservation merging or splitting points. Even though reshaping adds delay to some packets, it does not affect the delay bounds. Traffic is reshaped using a combination of a token bucket with a peak rate controller. Whenever the bucket is full, excess traffic is treated as best effort — the reshaping mechanism performs some policing itself.

#### 2.2.1.3 RSVP

Even though the IntServ model is independent from the resource reservation protocol (reservations may even be established by manual configuration or a management protocol [RFC2212]), its deployment is usually based on RSVP [RFC2205]. RSVP is a signaling protocol operating over unicast or multicast routes previously established by a routing protocol, of which it is independent. Some characteristics of RSVP are:

- Unidirectional (simplex) reservations
- Receiver-initiated reservations
- Aggregation of multicast reservations along the multicast trees
- Soft-state reservations
- Dynamic modification of established reservations

RSVP reservations are unidirectional for two reasons: (1) RSVP was conceived having multicast applications in mind, where there may be a large number of passive receivers, and unidirectional reservations make sense for such applications; (2) data paths in the two directions may be asymmetric. Applications requiring bidirectional resources may request two reservations, one in each direction. Multicast support is also the reason for receiver-initiated reservations — in a multicast session with a large number of receivers, it becomes impractical for the sender to keep track of all of them. RSVP provides aggregation of reservations on the multicast tree, even if the reservations are heterogeneous, preventing unnecessary waste of resources. The reservation upstream of the merging point is the least upper bound (LUB) of the downstream reservations. Moreover, different reservation styles are provided: wildcard (valid for any sender), fixed filter (valid for a single sender or a fixed set of senders) and dynamic filter (valid for a dynamically changeable set of senders).

In order to keep the robustness typical of datagram networks, where flows are rerouted in order to restore the service whenever a node or link fails, RSVP uses soft state reservations, which are torn down if not refreshed periodically. This means that should the path of a flow with an established reservation change, the stale reservation on the old path will be removed, freeing up resources for other flows. This, however, comes at the cost of increased overhead, introduced by the refresh messages.

RSVP allows for dynamic changes in the reservations. However, the changes must be subject to admission control, which may fail if the amount of reserved resources is being increased.

#### *2.2.1.3.1 Operation*

An IntServ reservation is established using RSVP as follows [RFC2210] (refer also to fig. 2.2). The sender begins by sending the *Path* message, used to install path state (as the name implies); this message follows the same path as the data packets, and contains the following information:

- Identification of the session, consisting on the destination IP address (which may be multicast), the layer 4 protocol and, optionally, the destination port
- Identification of the previous hop (initialized with the sender's IP address and updated at every router along the path)
- The traffic envelope (*TSpec*) of the flow
- Optionally, an ADSPEC object; it is used to gather information on some path characteristics (hop count, bottleneck bandwidth, minimum latency, MTU) and, in GS reservations, the error terms (*C* and *D*)

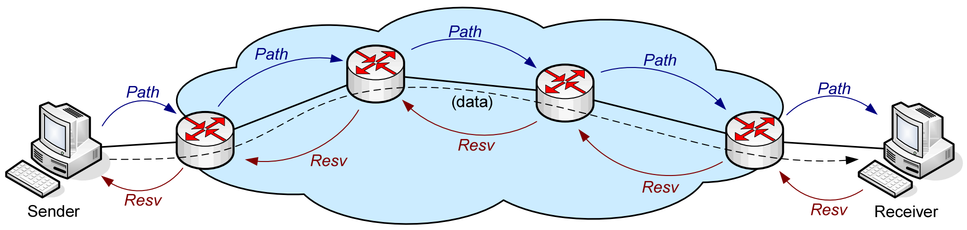

**Figure 2.2: RSVP messages**

After receiving the *Path* message, the receiver determines the amount of resources to be reserved and sends a *Resv* message upstream to the sender. Thanks to the *Path* state stored at the routers, the *Resv* message can follow exactly the reverse path of the *Path* message even if the routes are asymmetric. On receiving the *Resv* message, each router submits the request to admission control; if accepted, the message is forwarded upstream; otherwise, a *ResvErr* message is sent to the receiver reporting the fact.

Since RSVP is a soft state protocol, *Path* and *Resv* messages must be periodically refreshed, otherwise the reservation would timeout and be removed. Nevertheless, RSVP provides *PathTear* and *ResvTear* messages for a faster removal of path and reservation state, respectively, thus avoiding the waste of resources until the timeout.

#### 2.2.1.4 Issues with IntServ

Even though the IntServ architecture, supported by the RSVP protocol, is able to deliver the QoS guarantees necessary for soft and hard real-time applications on top of the datagram-based Internet, it has not gained widespread acceptance as once expected. The most frequently pointed out limitations of RSVP/IntServ concern its scalability to high-speed backbone networks.

The first issue is the necessity for maintenance of per-flow state at every intermediate node in the network, including core routers supporting a very large number of simultaneous flows. While this is not as big a problem as it may seem at first (refer to chapter 4), it is still a limitation, particularly given the complexity of RSVP processing. This complexity stems in large part from the multicast-oriented design of the protocol; however, multicast has not gained the importance once expected, and it is questionable if there is enough interest in multicast to justify the extra complexity.

Another, more severe, issue with RSVP/IntServ is the necessity for computationally complex packet scheduling algorithms and for packet classification based on the 5-tuples identifying the flows. These operations must be performed for every packet, but their complexity implies they cannot be performed at line rate in high speed backbone routers.

As a result of these scalability issues, RSVP/IntServ has been deployed only in small scale in internal networks, where they are mitigated by the much smaller number of simultaneous flows.

### 2.2.2 DiffServ

The scalability issues that plagued IntServ led the IETF to the development of a radically different approach to QoS — the Differentiated Services (DiffServ) architecture [RFC2475]. Contrary to IntServ, a fine-grained, flow-based mechanism, DiffServ is a coarse-grained, class-based mechanism for traffic management. DiffServ scales very well since core nodes do not perform most of the functions described in section 2.1 — they only perform packet scheduling according to a small number of traffic classes, selected according to a single field of the IP packet header. The concept of service differentiation according to a field in the packet header was not new: back in 1981, [RFC791] specified a Type of Service (ToS) byte, containing a field specifying a precedence level and another one specifying the requested type of service (the latter was primarily intended to be used for routing, though it could also affect other aspects of datagram handling)[1]. The ToS model, however, was never used in large scale.

A central concept in DiffServ is that of Per-Hop Behavior (PHB). A PHB is the externally observable forwarding behavior applied at each node to a traffic aggregate. Frequently, the description of a PHB is made with reference to other traffic, for example by guaranteeing a PHB a certain fraction of the link capacity. Resource management is performed according to the PHBs. When there are interdependencies in the specification of a set of PHBs, the set is defined as PHB group with a unified specification[2]. The implementation of PHBs is essentially based on queue management and packet scheduling mechanisms. However, the specification of a PHB is based on the behavioral characteristics relevant for the service, and frequently a given PHB may be implemented with different mechanisms. Two PHB groups have been standardized by the IETF: the Assured Forwarding (AF), providing high probability of forwarding to conformant packets, and the Expedited Forwarding (EF), used to build a low loss, low latency, low jitter, assured capacity, end-to-end transport service through DiffServ domains (“Virtual Leased Line” or “Premium” service). These PHB groups will be described in sections 2.2.2.1 and 2.2.2.2.

In DiffServ, aggregates are identified by a 6 bit field in the packet header, the Differentiated Services Code Point (DSCP). The DSCP corresponds to the leftmost 6 bits of the DS field (a redefinition of the ToS octet in IPv4 or the Traffic Class octet in IPv6). The collection of packets sharing the same DSCP and crossing a given link in a particular direction is designated Behavior Aggregate (BA).

Two other very important concepts in the DiffServ architecture are those of Service-Level Agreement (SLA) and Traffic Conditioning Agreement (TCA). An SLA is a contract between a network operator and a customer (or between peering operators) containing a specification of the network service to be provided — the Service Level Specification (SLS) — including traffic treatment and performance metrics. The SLA may also contain a set of traffic conditioning rules, designated TCA. The TCA specifies rules for packet classification,

[1] For a complete history of the ToS byte, please refer to sec. 22 of [RFC3168].

[2] A single stand-alone PHB is considered a special case of a PHB group.

and traffic profiles and the accompanying rules for metering, marking, and packet dropping and/or shaping.

The DiffServ architecture makes a clear distinction between the edge nodes and the core nodes of a DS domain (defined as a set of contiguous DiffServ nodes providing a common service policy and set of implemented PHBs). For efficiency reasons, core nodes implement only BA classification, based on the DSCP, and the forwarding behavior of the supported PHBs. Edge nodes, providing interconnection to other domains (which may or may not support DiffServ), contain additional classification and traffic conditioning functions (fig. 2.3); these functions are required for ensuring TCA conformance of ingress traffic and conditioning egress traffic. The multi-field (MF) classifiers may use different fields of the packet header (source and destination addresses, transport protocol, source and destination ports, etc.) for assigning packets to PHBs supported in the domain; the packets are marked with the corresponding DSCP for efficient classification in the core. The traffic conditioning functions correspond to those described in section 2.1.

The above described concepts are the building blocks for providing differentiated QoS and defining the forwarding treatment at individual nodes. However, providing QoS to a flow implies providing an end-to-end service with well-defined metrics. The first step towards this goal is the support for intra-domain QoS between the ingress and egress nodes of a single network. A concept for describing the overall treatment that a traffic aggregate will receive from edge-to-edge of a DS domain and how to configure the elements for providing that treatment, thus, became necessary. The Per-Domain Behavior (PDB) [RFC3086] provides such description. A PDB is characterized by specific metrics that quantify the treatment a set of packets with a particular DSCP (or set of DSCPs) will receive as it crosses a DS domain. A particular PHB group and traffic conditioning requirements are associated with each PDB. The measurable parameters of a PDB are suitable for use in SLSs at the network edges.

A Virtual Wire PDB has been proposed [Nichols04], defining an edge-to-edge transport service for providing circuit emulation as an overlay on top of an IP network, mimicking the behavior of a hard-wired circuit of some fixed capacity from the point of view

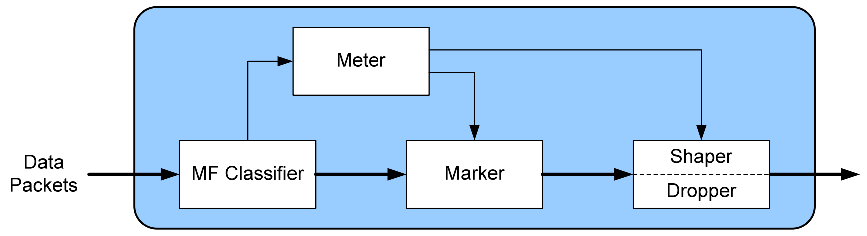

**Figure 2.3: Packet classification and traffic conditioning functions of a DiffServ edge node**

of the originating and terminating nodes. An alternative Virtual Wire PDB has been proposed in [Walter03] where the minimum jitter requirements are relaxed in order to obtain lower delay values. The only PDB that has been published in the RFC series, however, is the Lower Effort (LE) PDB [RFC3662], providing a background transport service for bulk applications that can be starved by Best Effort traffic.

#### 2.2.2.1 Assured Forwarding

The AF PHB group[3] provides delivery of IP packets in four independently forwarded AF classes. Each AF class is assigned a minimum amount of capacity and buffer space at every node. Within each AF class, a packet can be assigned one of three different levels of drop precedence. In case of congestion, the drop precedence of a packet determines the relative importance of the packet within the AF class. A congested DS node tries to protect packets with a lower drop precedence value from being lost by preferably discarding packets with a higher drop precedence value. An important property of AF is that a DiffServ node does not reorder packets of the same microflow if they belong to the same AF class, even though they may have different drop precedence levels.

In nodes supporting AF, short periods of congestion should be absorbed by buffering. Congestion in larger temporal scales needs to be controlled through packet dropping. However, packet dropping should be gradual rather than abrupt, requiring the use of Active Queue Management (AQM) techniques. Within an AF class, a DS node must not forward a packet with smaller probability if it contains a drop precedence value $p$ than if it contains a drop precedence value $q$ when $p < q$. Multi-level extensions of the Random Early Detection (RED) [Floyd93] mechanism with cumulative queue lengths such as RED with In and Out bit (RIO) [Clark98] or Generalized RED (GRED) [Almesberger99] are suitable for the implementation of the drop precedence levels within each AF class.

#### 2.2.2.2 Expedited Forwarding

The Expedited Forwarding PHB, first defined in [RFC2598], provides very low loss, queuing delay and jitter to conformant IP packets; this is achieved by always guaranteeing a minimum capacity for EF traffic that the arrival rate must not exceed, independently of traffic of other PHBs. The EF PHB is, therefore, appropriate for circuit emulation services.

[3] Strictly speaking, AF is a type of PHB group, since the operation of each AF class is entirely independent of the others; each AF class is an instance of the AF PHB group type [RFC3260].

The original definition lacked mathematical precision and introduced unnecessary limitations on the schedulers, and was obsoleted by a new, more formal definition in [RFC3246]. This new definition introduces packet-scale rate guarantees, in addition to the aggregate guarantees previously specified; it also clarifies the behavior of EF routers with multiple inputs and/or complex scheduling. Similarly to AF, EF packets belonging to the same microflow cannot be reordered.

The implementation of the EF PHB requires a scheduling mechanism that can guarantee a minimum rate at the packet scale. A simple implementation may be based on a strict priority scheduler, with EF traffic having the highest priority. Ensuring that the arrival rate does not exceed the minimum capacity requires traffic conditioning (policing/shaping) mechanisms to be performed at the network boundaries; this also ensures that other classes do not starve due to excess EF traffic.

#### *2.2.2.3 Issues with DiffServ*

Based on simple and efficient mechanisms, DiffServ provides a very scalable foundation for deploying QoS on high-speed IP networks. At the core nodes, only a minimal amount of state is maintained, corresponding to a small number of traffic classes, and packet classification and scheduling are efficient. However, DiffServ cannot provide QoS to end-user flows by itself — there is no per-flow reservation of resources or admission control. More than a limitation, it is a characteristic of DiffServ — it is a tool for network operators, not end-users, and provides QoS for aggregates, not individual flows. End-to-end QoS-enabled packet delivery services may be built on top of the DiffServ foundation using additional layers that provide the missing features; notable examples are the Bandwidth-Broker-based architectures described in section 2.3.4.

## *2.3 Other QoS Models*

In spite of its scalability problems, IntServ provides hard per-flow QoS guarantees, which cannot be achieved with DiffServ. A lot of research has undergone towards the goal of providing the per-flow guarantees of the IntServ model (or, at least, some of those guarantees) with improved scalability. This section describes some of the resulting proposals.

### 2.3.1 Alternative Signaling

Since some of the issues of RSVP/IntServ concern the signaling protocol itself, several proposals have been made for QoS signaling protocols with more desirable properties.

#### 2.3.1.1 Simplified Signaling

Based on the premise that a significant portion of the applications requiring QoS were multimedia oriented, the Yet Another Sender Session Internet Reservation (YESSIR) protocol was proposed [Pan99] as an extension of the Real-Time Control Protocol (RTCP), the companion to Real-Time Protocol (RTP) used by many such applications. YESSIR is a sender-initiated, soft-state protocol, and provides support for partial reservations that may be improved over time, as resources become available. YESSIR has faster processing and smaller overhead than RSVP, and yet retains most of its functions, notably multicast (though supporting only individual and shared reservation styles). With YESSIR, per-flow state is still maintained at the routers, and since it concerns only signaling, no improvement is made with respect to RSVP regarding packet classification and scheduling; therefore, it suffers from similar scalability limitations in these respects.

Another proposal for simplified QoS signaling is Boomerang [Fehér99], a duplex, soft-state reservation protocol. Boomerang was implemented as an extension of ICMP Echo function, and requires no special functionality on the far-end node. It is possible to use per-flow reservations, but measurement-based admission control may also be used; in the last case, there is no need to maintain per-flow state at the routers, but only soft QoS may be provided (a similar limitation exists in probing-based admission control, described below). Test results indicate that Boomerang is much lighter-weight than RSVP, both in terms of CPU power and memory space [Fehér99, Fehér02]; however, it has reduced functionality, and when used with per-flow QoS, packet classification and scheduling procedures are still scalability-limiting.

#### 2.3.1.2 Next Steps in Signaling (NSIS)

The Next Steps in Signaling (NSIS) Working Group has proposed a two-layer extensible signaling architecture [RFC4080] that addresses many limitations of RSVP, having QoS signaling as one of the first applications[4] [Fu05]. One interesting feature of NSIS is the separation between signaling message transport and signaling applications, provided by two different layers. The lower layer, designated NSIS Transport Layer Protocol (NTLP), provides a generic transport service for different signaling applications that reside in the upper layer, the NSIS Signaling Layer Protocols (NSLPs). This layered approach simplifies the design of new signaling applications.

[4] Other applications are Network Address Translation (NAT) hole punching / Firewall control and metering.

The main part of the NTLP is the General Internet Signaling Transport (GIST) protocol [Schulzrinne06]. It runs on top of standard transport protocols — User Datagram Protocol (UDP), Transmission Control Protocol (TCP), Stream Control Transmission Protocol (SCTP) and Datagram Congestion Control Protocol (DCCP) — and reuses existing Transport Layer Security (TLS) and IP layer security (IPSec/IKEv2). The use of a cryptographically random session identifier for signaling sessions, independent of the flow identifier, is useful in mobility scenarios, where the flow identifier may change. In addition to signaling transport, GIST provides peer discovery and capability querying services, and supports advanced features such as the negotiation of transport and security protocols, recovery from route changes and interaction with NAT and IP Tunneling.

The proposed NSLP for QoS [Manner06], together with GIST, provides functionality extending that of RSVP: it is also a soft-state protocol, and supports sender-initiated, receiver-initiated and bidirectional reservations, as well as reservations between arbitrary nodes (end-to-end, edge-to-edge, end-to-edge, etc.). The QoS NSLP is independent of the underlying QoS architecture, and provides support for different reservation models, including models based on flow aggregation (described in section 2.3.2). A separate document [Bader06] specifies the use of NSIS to implement the Resource Management in DiffServ (RMD) model, described below (section 2.3.2.3). Unlike RSVP, there is no support for multicast, thus reducing the complexity for the majority of the applications, which are unicast.

NSIS concerns signaling only, and was designed to support any QoS model. As a result, its characteristics in terms of QoS, resource utilization and scalability are largely dependent on the underlying QoS model.

### 2.3.2 Aggregation-based schemes

The aggregation of individual flows, the basic idea behind DiffServ, allows not only for a substantial reduction in the amount of state that routers are required to maintain, but also, more importantly, for more computationally efficient packet classification and scheduling mechanisms. Under certain circumstances, aggregation may also allow for a large decrease in signaling overhead. This section describes some QoS models using flow aggregation to attain better scalability.

#### *2.3.2.1 IntServ over DiffServ*

With the goal of simultaneously reaping the benefits of RSVP/IntServ (per-flow QoS) and DiffServ (scalability in the core), a framework was proposed for supporting IntServ over DiffServ networks [RFC2998]. End-to-end, quantitative QoS is provided by applying the

IntServ model end-to-end across a network containing one or more DiffServ regions. Access networks, where the number of simultaneous flows is relatively low, use MF classifiers and per-flow traffic control; resource reservations are requested by the end-hosts, usually resorting to RSVP. In core regions, DiffServ aggregation of flows is performed, and scalability is achieved by using BA classifiers and per-class traffic control. From the perspective of IntServ, the DiffServ regions are treated as virtual links connecting IntServ-capable nodes (fig. 2.4).

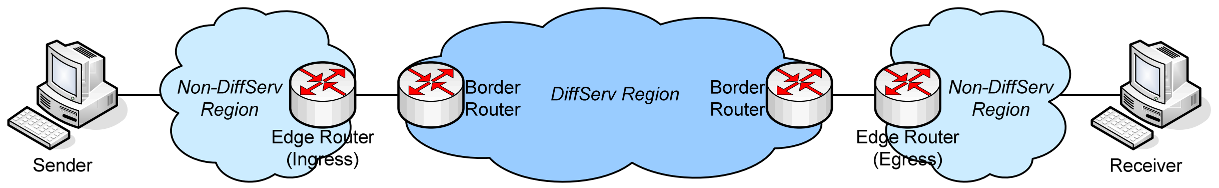

**Figure 2.4: IntServ over DiffServ — sample network configuration**

Requests for IntServ services must be mapped onto the underlying capabilities of the DiffServ network region; this mapping involves:

- Selecting an appropriate PHB (or PHB group) for the requested service
- Exporting IntServ parameters from the DiffServ region for ADSPEC updating (please refer to sec. 2.2.1.3.1)
- Performing admission control on the IntServ requests that takes into account the resource availability in the DiffServ region
- Performing appropriate policing (and, eventually, shaping or remarking) at the edges of the DiffServ region

Inside the DiffServ region, resource management may be performed in a number of different ways, including statically provisioned resources, resources dynamically provisioned by RSVP, and resources dynamically provisioned using Bandwidth Brokers. In the first case, resources are provisioned according to an SLA, of which the ADSPEC parameters are derived from. RSVP messages are transparently carried across the DiffServ region. Though scalable, this solution is inflexible. In the second case, nodes inside the DiffServ region are also RSVP speakers — the data plane is DiffServ, but the control plane is RSVP. Due to aggregate classification and scheduling, it is more scalable than pure RSVP/IntServ, but the use of per-flow RSVP in the core is still limiting. A more scalable alternative, which will be described later, is the use of RSVP for aggregates. The third alternative uses centralized entities designated Bandwidth Brokers to perform resource management and control plane functions; such approaches are further discussed in section 2.3.4.

The IntServ over DiffServ framework requires the DiffServ regions of the network to provide support for the standard IntServ services between the border routers. While such support is relatively easy to implement for the Controlled Load service, particularly in DiffServ networks with Bandwidth-Broker-based control of resources, it is hard to provide the strict delay bounds of the Guaranteed Service in aggregation-based networks. Such bounds can be achieved by priority schedulers, but only at the cost of very low link utilization for the traffic class supporting the GS traffic [Charny00].

The IntServ over DiffServ framework does not provide a complete solution that can be readily deployed — [RFC2998] clearly states that more work is required in several areas (service mapping, functionality for using RSVP signaling with aggregate traffic control, resource management mechanisms) before coming up with such solution.

#### 2.3.2.2 RSVP Reservation Aggregation

The RSVP Reservation Aggregation model, standardized in [RFC3175], defines an extension to RSVP by which individual end-to-end flows may be aggregated into a single reservation inside a DiffServ region, where they share common ingress and egress nodes. The establishment of a smaller number of aggregate reservations on behalf of a larger number of end-to-end reservations yields the corresponding reduction in the amount of state maintained at the routers. Hierarchical aggregation may be achieved by applying the method recursively.

Aggregate reservations are dynamically established between the ingress nodes (aggregators) and the egress nodes (deaggregators), and are updated in bulk quantities much larger than the individual rates of the flows in order to reduce the signaling overhead. Whenever a flow requests admission in an aggregate region, the edge routers of the region check if there is enough bandwidth to accept the flow on the aggregate. If resources are available, the flow will be accepted without any need for signaling the core routers. Otherwise, the core routers will be signaled in an attempt to increase the aggregate's bandwidth. If this attempt succeeds, the flow is admitted; otherwise, it is rejected.

By using DiffServ mechanisms for packet classification and scheduling (rather than performing them per aggregate reservation), the amount of classification and scheduling state and the complexity of these procedures in the aggregation region is even further reduced — it is independent not only of the number of end-to-end reservations, but also of the number of aggregate reservations in the aggregation region.

The main disadvantage of this model is the underutilization of network resources. Since the bandwidth of each aggregate is updated in bulk quantities, each aggregate's bandwidth is almost never fully utilized. The unused bandwidth of all aggregates traversing a link adds up, leading to a significant amount of wasted link capacity. The RSVP Reservation Aggregation model is described and analyzed in more detail in the next chapter.

#### *2.3.2.3 Resource Management in DiffServ (RMD)*

Based on similar principles to RSVPRAgg, the Resource Management in DiffServ (RMD) [Westberg02] was introduced as a method for dynamic reservation of resources within a DiffServ domain. It provides admission control for flows entering the domain and a congestion handling algorithm that is able to terminate flows in case of congestion due to a sudden failure (e.g., link, router) within the domain.

The RMD framework defines two types of protocols: the Per-Domain Reservation (PDR) protocol and the Per-Hop Reservation (PHR) protocol. The PDR is triggered at the edge nodes, and is used for resource management in the entire domain. Though the PDR could be an existing protocol, such as RSVP, a newly defined one was used in [Westberg02]. The PHR is a complement to the DiffServ PHB, providing reservation of resources per traffic class at every node within the domain. PDR messages are usually carried encapsulated in PHR messages. A PHR protocol named RMD On-Demand (RODA) protocol was defined; it is reservation-based, unicast, edge-to-edge protocol designed for a single DiffServ domain, aiming at simplicity, low implementation cost and scalability.

Similarly to RSVP Reservation Aggregation, scalability in RMD is achieved by separating a fine-grained reservation mechanism used in the edge nodes of the DiffServ domain from a much simpler reservation mechanism used in the interior nodes. The limited functionality supported by the interior nodes allows for fast processing of signaling messages. As previously stated, the RMD model is also supported by the NSIS signaling stack.

### 2.3.3 Elimination of State in the Core

Since the necessity for maintaining state in core routers is usually regarded as one of the major factors limiting the scalability of the RSVP/IntServ model, a significant amount of work has undergone in the development of models based on stateless core routers. This section describes the most prominent models in this class.

#### *2.3.3.1 Probing-Based Admission Control*

In accordance with the "end-to-end principle" [Saltzer84], one of the architectural guidelines of the Internet [RFC3439], several schemes have been proposed where the complexity is moved to the endpoints and (some degree of) QoS is provided with minimal or no intervention of the network routers [Bianchi00, Breslau00b, Elek00, Gibbens99, Kelly00, Sargento01, Key03]. The probing approach may be regarded as an extreme case of aggregation: all flows are aggregated into a single queue per output port of the routers[5], and QoS provisioning is based on admission control performed by the terminals themselves. The admission control decision is based on the network congestion level as measured by the endpoints.

The probing technique is split into two phases: the probing phase and the data phase. In the probing phase, a packet sequence is sent with similar characteristics to the flow being admitted for a certain time. At the end of this phase, the receiver sends information on the QoS of the received stream to the sender; based on this information, the sender decides if the flow is admitted or rejected. If the flow is admitted, the actual flow data may be transmitted; this is called the data phase. Some QoS parameters commonly used for the admission control are packet delay or delay variation and packet loss or marking ratio.

There are some differences among the proposed probing mechanisms. In [Elek00] and [Bianchi00], probe packets are sent with lower priority than data packets. If the loss ratio after the probing period is acceptable, the flow is admitted — odds are that the loss ratio for the data packets will be less than that of the probes, which have lower priority. In [Gibbens99], [Kelly01] and [Key03], probe and data packets are treated equally, and admission is decided based on the ratio of packets marked with Explicit Congestion Notification (ECN) [RFC3168], instead of the packet loss. It is expected that routers start marking packets long before the congestion level leads to packet loss. Since the number of marked packets is much larger than the number of dropped packets the duration of the probing phase may be significantly reduced [Kelly01]. In order to avoid a problem known as thrashing (a large number of flows simultaneously try admission and fail, even though the network has enough resources to admit some of them), [Breslau00] proposes two techniques. The Slow Start Probing technique consists on splitting the probing phase into intervals and sending probe packets at increasing rates; in the last interval, probes are sent at the same rate of the flow. If, at the end of an interval, the loss rate exceeds a given threshold, the flow is rejected and probes stop being sent. The Early Reject technique is similar, but probes are sent at the final rate in all intervals. Another identified problem was resource stealing: since there is no resource reservation, a new flow may reduce the QoS received by previously admitted flows. The ε-probing technique [Sargento01] mitigates this problem in multi-class networks: in

[5] While, strictly speaking, this is not necessarily true (there is no incompatibility between probing and differentiation), the use of different queues for service differentiation is orthogonal to the probing concept.

addition to the probe itself, low rate ε-probes are sent on the remaining classes, and the flow is admitted only if both the probe and the ε-probes have QoS levels better than given thresholds.

Given their nature, probing-based schemes have excellent scalability properties, and they are more appealing in high-speed networks, where the large number of multiplexed flows allows for better estimation of the received QoS. However, since there is no resource reservation, only soft QoS may be provided, and they are of limited use if there is no differentiation between flows that use probing and flows that do not use it.

#### 2.3.3.2 Scalable Reservation Protocol (SRP)

The Scalable Reservation Protocol (SRP) [Almesberger98] is based on a similar idea as the probing schemes, but requires more support from the network. Packet scheduling at the routers is performed in two classes, one for traffic with reservations and another one for best effort traffic (fig. 2.5). An in-band protocol is used for gaining access to the reserved service class: flows with requirements for improved QoS start by marking the packets as *request* packets. When a *request* packet is received by a router, an estimator checks whether accepting the packet would exceed the available resources. If the packet can be accepted, it is forwarded in the reserved class, and the router commits to accept further *reserved* packets at the same rate; otherwise the packet is re-marked as *best effort* and forwarded in that class.

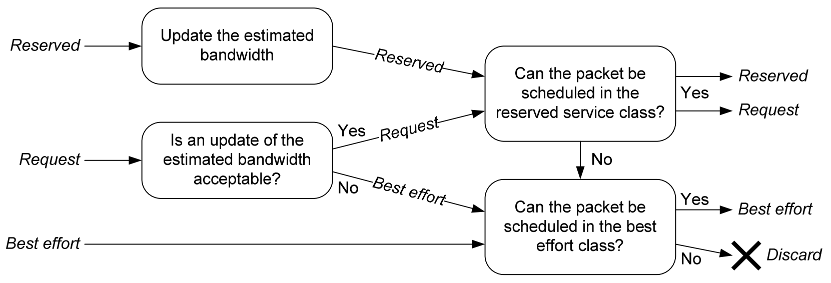
Reserved
Update the estimated bandwidth
Reserved
Can the packet be scheduled in the reserved service class?
Yes
Reserved
Request
Request
Is an update of the estimated bandwidth acceptable?
Yes
Request
No
Best effort
No
Can the packet be scheduled in the best effort class?
Yes
Best effort
No
Discard
Best effort

**Figure 2.5: SRP packet processing by routers**

Periodically, the receiver sends feedback to the sender using a different protocol (which may be implemented on top of RTCP); this protocol is not interpreted by the routers, only by the sender. The feedback information concerns the receiver's estimate of the reserved rate, the summed rate of received *request* and *reserved* packets[6]. The sender maintains an independent estimate of the reserved rate, and the maximum rate at which *reserved* packets may be sent is max(*feedback*, *src_estimate*) — the remaining packets are sent as *request* packets. Routers are mostly stateless: only an estimate of the aggregate reserved rate is maintained per output port.

[6] Actually, the maximum value of this sum over a time window.

With minimal support from the routers, the SRP model provides a differentiated reserved class, which supports dynamically changeable and partial reservations, and dispenses with a probing phase. However, similarly to probing schemes, it provides only soft QoS.

#### 2.3.3.3 Egress Admission Control

The Egress Admission Control proposal [Cetinkaya01] holds some similarities to the RSVP Reservation Aggregation proposal: end-to-end reservation requests are hidden inside the network, and the flow admission decision is taken by the egress router (deaggregator in RSVPRAgg). However, in this case no state is maintained in the interior nodes: there are no reservations for edge-to-edge aggregates, and flow admission decisions are taken based on a "black box" model of the network, characterized by measured arrival and service envelopes of the aggregate traffic flowing between the ingress and the egress routers. The arrival and service envelopes are measurement-based statistical counterparts of the deterministic arrival curve and service curve concepts. They account for interfering cross traffic without explicitly measuring or controlling it.

Even though only statistical guarantees (soft QoS) can be provided, this framework supports different traffic classes with varying degrees of QoS guarantees. Such differentiation is obtained not only through the envelopes of the different traffic classes, but also by the use of a level of confidence in the flow admission process, expressing the confidence that the system will actually deliver the announced QoS. As a result, Egress Admission Control is able to provide better guarantees than the previously mentioned core-stateless schemes.

#### 2.3.3.4 SCORE

At the high-end of the core-stateless architectures in terms of QoS guarantees is the Stateless Core (SCORE) [Stoica99, Stoica00]. Based on the concept of Dynamic Packet State (DPS), where each packet carries state information initialized by the ingress router and updated at every hop, the SCORE architecture is able to provide IntServ end-to-end per-flow rate and delay guarantees without recourse to state maintenance at core nodes.

A SCORE network closely approximates the behavior of a per-flow stateful network, specifically a network where every node implements the Jitter Virtual Clock (JVC) scheduling algorithm. It has been shown that, as long as a flow's arrival rate does not exceed its reserved rate, such network is able to provide the same delay bounds as a network implementing the Weighted Fair Queuing (WFQ) scheduling algorithm [Demers89], commonly used in the deployment of the RSVP/IntServ model.

JVC is a non-work-conserving scheduling algorithm combining a Virtual Clock (VC) scheduler [Zhang90] with a rate controller. Upon arrival, a packet is assigned an eligible time and a transmission deadline. The packet is held in the rate controller until it becomes eligible, and the scheduler orders the transmission of eligible packets according to their transmission deadlines. The eligible time $e_{i,j}^k$ and deadline $d_{i,j}^k$ of the $k$th packet of flow $i$ at the $j$th node are computed according to the following formulae:

$$e_{i,j}^k = \begin{cases} a_{i,j}^1 & \Leftarrow k = 1 \\ \max\left(a_{i,j}^k + g_{i,j-1}^k, d_{i,j}^{k-1}\right) & \Leftarrow k > 1 \end{cases} , \quad i, j \geq 1 \tag{2.3}$$

$$d_{i,j}^k = e_{i,j}^k + \frac{l_i^k}{r_i}, \quad i, j, k \geq 1 \tag{2.4}$$

where $r_i$ is the reserved rate of the flow, $l_i^k$ the length of the packet and $a_{i,j}^k$ its arrival time at the $j$th node, and $g_{i,j}^k$ is the amount of time the packet was transmitted earlier than its deadline at the previous node, stamped in the packet header.

While JVC requires the maintenance of per-flow state at each node, more precisely the maintenance of the deadline of the last transmitted packet, $d_{i,j}^{k-1}$, this value is only used in a max operation in eq. (2.3). The Core Jitter Virtual Clock (CJVC) scheduling algorithm eliminates the need for state maintenance by adding a slack variable $\delta_i^k$ to the other term in the max operation such that it is always larger than $d_{i,j}^{k-1}$. It has been shown that using the formula derived in [Stoica99] for computing $\delta_i^k$, the deadline of a packet at the egress node of a CJVC network is equal to its deadline at the same node on a corresponding JVC network; therefore, CJVC can provide the same delay bounds of a network based on WFQ and, thus, supports the requirements of IntServ's guaranteed service.

A lightweight protocol is used between the ingress and egress nodes for requesting resource reservations. Admission control is performed at every node in order to ensure that the sum of the reserved rates of flows traversing a link does not exceed its capacity. An estimated upper bound on the total reserved capacity, $R_{\text{bound}}$, is maintained per output port; admission control merely involves checking that $R_{bound} + r \leq C$, where $r$ is the requested rate for the new flow and $C$ is the capacity of the output link. The algorithms for ensuring that $R_{\text{bound}}$ is (1) always an upper bound and (2) a close upper bound are detailed in [Stoica99].

The SCORE architecture succeeds in eliminating the need for state maintenance in the core. In doing so, the need for packet classification at core nodes is also eliminated, which is important for scalability, as complex MF classification is computationally expensive. However, the scheduling mechanism is still complex, as packets need to be sorted according to their deadlines; moreover, packets in the rate controller need also be sorted according to their eligible times.

### 2.3.4 Bandwidth Brokers

A number of proposals have been made where resource management, signaling and admission control are performed by centralized entities, commonly designated Bandwidth Brokers (BBs) [RFC2638, Terzis99, Duan04], but also known under different names (Agents [Schelén98], Oracles [RFC2998], Clearing Houses [Chuah00], QoS Brokers [Marques03] or Domain Resource Managers [Hillebrand04]). Models based on BBs decouple the QoS control plane from the data plane. Since many control plane functions are performed per flow, scalability can be greatly enhanced by offloading these responsibilities from the core nodes.

BB-based models are often used in conjunction with DiffServ, since the two technologies are complementary: DiffServ provides a scalable model for data plane QoS functions, such as edge traffic conditioning and packet classification and scheduling, and BBs perform QoS signaling, flow admission control and resource management, control plane functions that are missing in DiffServ.

One important advantage of centralizing QoS control in BBs is the possibility of using sophisticated admission control and QoS provisioning algorithms that allow for a network-wide optimization of the resources, something that is difficult to achieve with distributed schemes. This optimization can easily incorporate policy aspects. Moreover, BBs can also support additional functions, such as support for mobility or inter-domain resource reservation; these functions and QoS control may be performed in an integrated fashion. Another advantage of a centralized approach is that QoS state consistency issues are avoided — in distributed approaches, these issues are partially solved by using soft states; however, the need for periodical refreshment of soft states increases the signaling overhead.

Schelén and Pink [Schelén98] proposed a model where a BB gains knowledge of the network topology by passively listening to a link state routing protocol (e.g., OSPF) and retrieves detailed information, such as link bandwidth, using a network management protocol (e.g., SNMP). Parameter-based admission control for priority traffic is performed based on information maintained at the BB regarding reserved resources at each link. In addition to intra-domain resource management, BBs in different domains use a protocol for establishing inter-domain aggregate reservations, designated funnels, towards a given subnetwork, identified by its address prefix.

Originally published as an Internet Draft in 1997, [RFC2638] proposes an architecture supporting two elevated services — a Premium service with firm QoS guarantees that may be used to emulate virtual leased lines, and an Assured service providing the soft guarantee of a very low probability of packet dropping. On the data plane, differentiation is based on a two-bit field of the packet header, a mechanism that came to be the basis for the DiffServ model. On the control plane, each domain has a BB which keeps track of reservations, that can be static or dynamic, manages resources, and configures the traffic conditioning mechanisms at the border routers (aggregate) and access routers (per flow). Additionally, BBs in different domains exchange messages in order to establish end-to-end reservations that are aggregated according to the destination.

A BB-based two-tier model for resource management was proposed in [Terzis99]. Different resource management models are used for access (leaf) and transit domains. Intra-domain resource management is mostly done using RSVP, either per-flow (access domains) or aggregate (transit domains), and BBs are mostly used to manage inter-domain reservations. At access domains, the sender uses RSVP, but the *Path* message is intercepted by the first hop router, which sends a reservation request to the BB[7]. If enough bandwidth is available at the egress router towards the downstream domain, the BB tells the first hop router to forward the *Path* message, and the egress router to start sending *Resv* messages towards the sender, thus performing an RSVP/IntServ reservation between the sender and the egress router. Since RSVP is terminated at the egress of the sender's domain, receivers need an out-of-band mechanism to know the traffic profile of the source (it is suggested that this is performed at the application layer). The receiver sends a request to the BB, and the BB tells the ingress router to perform an RSVP/IntServ reservation to the receiver.

In transit domains, QoS at the data plane is provided by DiffServ. Ingress routers measure aggregate traffic towards each egress router (which they know since they are assumed to participate in inter-domain routing using the Border Gateway Protocol [RFC4271]). Using this information, they build a Tspec that is sent in an aggregate RSVP *Path* message towards the egress node; this node responds with an aggregate *Resv* message.

The BBs of adjacent domains communicate among themselves to establish dynamic traffic conditioning aggreements. Inter-domain reservations take into consideration only the

[7] This implicitly limits the architecture to a controlled load service, since in guaranteed service the amount of resources to reserve can only be known from the Resv message.

downstream domain, not the entire path. Edge routers measure the rate of the outgoing aggregate using a time window algorithm (see appendix A), and use a watermark-based algorithm to request an increase or decrease in the reservation. If more than a fraction $w$ of the reserved capacity is in use, the BB is informed to increase the reserved rate to the downstram domain; in response, the BB of the downstream domain tells the ingress router to increase the reservation by a value $\delta$, which is proportionally distributed among the aggregates originated at that node. If less than a fraction $l$ (with $l < w$) is in use, the BB is informed to decrease the reserved rate, using a hysteresis mechanism to avoid Ripple effect.

An advantage of this model is a simple, albeit imprecise, method for managing inter-domain resources. Some disadvantages are a mixed approach to resource reservation at the access domains that is cumbersome, and the support for controlled load services only.

A hierachical BB model is proposed in [Chuah00]. Basic routing domains managed by a Local Clearing House (LCH) are aggregated into a hierarchy of logical domains, associated with Global Clearing Houses (GCH) that manage inter-domain reservations. For performance reasons, resources are reserved in advance using a Gaussian predictor based on measured aggregate traffic (the possibility of on-demand reservations is mentioned, but not specified in the proposal). Two other techniques are used to improve the responsiveness of the reservation mechanism: RxW scheduling of reservation requests and caching of intra- and inter-domain computed paths of previous reservations. An interesting feature of the architecture is the possibility for secure real-time billing.

An entirely core-stateless approach supporting the IntServ per-flow Guaranteed Service, as well as a class-based guaranteed delay service with flow aggregation, was proposed in [Duan04]. The data plane is based on an improved version of the SCORE architecture, which may use a combination of core-stateless rate-based schedulers — such as the Core-Stateless Virtual Clock (C$VC) [Zhang00b], a work-conserving version of the CJVC described in section 2.3.3.4 — and delay-based schedulers — such as the Virtual Time Earliest Deadline First (VT-EDF) [Zhang00b]. On the control plane, a BB with detailed knowledge of the network topology keeps track of the reservations and performs admission control, ensuring that the worst case edge-to-edge delay requirements of the flows, defined by a dual token bucket *TSpec*, can be met. Flow admission requests are issued by the edge routers in response to external stimuli, such as the arrival of an RSVP reservation request. In an attempt at reducing the admission control delay, the process is split into two phases: admission test and bookkeeping; the latter can be performed after the reservation response. Nevertheless, while this splitting ensures O(1) complexity for the admission test, it requires the bookkeeping phase to update not only the state of all routers along the edge-to-edge path, but also for all paths traversing any of those routers. In practical terms, it means that admission control is very responsive for a single request but slow when a number of requests is performed in a short period.

The greatest concern with BBs is that by concentrating the intelligence for resource management, they become single points of failure, and may easily become bottlenecks in the control plane. However, these problems can be mitigated through the use of standard techniques for server redundancy and load sharing. Moreover, they allow the optimization of the control plane to be handled orthogonally to the optimization of the data plane.

One important aspect of BB-based architectures for QoS provisioning is the ease of integration of mobility management in the resource management model. This aspect is of major importance for providing QoS to mobile terminals with wireless access technologies, since such integration is necessary for minimizing the network service disruption as the terminals move across different cells. Additionally, other operational aspects of the network, such as policy enforcement, accounting and billing may easily be incorporated into the BB, making BB-based models very appealing for next generation IP-based mobile networks. Section 2.5.2 presents a proposed BB-based architecture for next generation networks.

## 2.4 Inter-Domain QoS

The Internet is not a flat collection of interconnected routers; it is organized in a two-level hierarchy. At the higher level, the Internet consists on a large number of interconnected Autonomous Systems (ASs). An AS is a set of routers under a single technical administration, having a single coherent interior routing plan and presenting a consistent picture of the destinations that are reachable through it. ASs are connected via gateways (the border routers), which run an inter-domain routing protocol to exchange routing information about which hosts are reachable by each AS. As a result, each gateway constructs a routing table that maps each IP address to a neighbor AS that knows a path to that IP address. The ASs can be broadly classified into two types: stub ASs, which only carry traffic generated at or destined to internal addresses (even though they may be multi-homed), and transit ASs, which have multiple connections to other ASs and carry traffic originated at and destined to external addresses. Transit ASs vary widely in dimension and geographical presence, and may be accordingly classified in tiers. An interesting analysis of the structure of the Internet, based on the inter-domain routing tables observed at several vantage points, is presented in [Subramanian02].

There are two different aspects involved in providing inter-domain QoS: the first one is finding a route capable of satisfying the QoS requirements of the application flows; the second one is performing reservations for ensuring that enough resources are available for the traffic demand. These aspects are complementary, but orthogonal: resource reservations may be performed over a QoS-optimized path or over a non-QoS-optimized path provided by BGP; conversely, inter-domain QoS routing is useful even without resource reservations. The next sections discuss these two aspects.

### 2.4.1 Inter-Domain Resource Reservation

The greatest technical problem with inter-domain resource reservation is scalability: the large number of ASs in the Internet[8] makes even aggregate reservations per (*source AS*, *destination AS*) challenging. The next paragraphs describe some of the more relevant work in this field.

The Border Gateway Reservation Protocol (BGRP) [Pan00] operates end-to-end, but only between border routers. It is a soft-state protocol based on the aggregation of reservations along the sink trees created by BGP, rooted on the destination domain (fig. 2.6). BGRP uses five control message types: *Probe* and *Graft*, used to establish a reservation, *Refresh* to keep the reservations active and update them, *Tear* for quicker removal of reservations, and *Error* to report errors during probing or grafting; these messages are sent reliably. BGRP signaling holds some similarities to RSVP signaling — *Probe* and *Graft* messages (fig. 2.6), for example, work quite similarly to RSVP's *Path* and *Resv* messages (fig. 2.2). There are, however, some important differences. (1) BGRP runs only between border routers. (2) BGRP uses stateless probing: no state is stored in the routers on processing *Probe* messages; instead, the router's address is added to a route record in the message itself, which is used to source route the corresponding *Graft* message along the reverse path. (3) BGRP does not work with individual reservations, but with aggregates; moreover, reservations from different upstream domains to the same sink are aggregated into a single downstream reservation (sum of the upstream reservations), ensuring that the amount of stored state is O(N), where N is the number of different domains (possible sinks). (4) Soft-state refreshments are bundled, in order to reduce the signaling overhead. (5) BGRP reservations are sender-initiated — *Probe* messages contain the reservation request, and admission control is performed when they are processed.

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[8] As of December 2006, there are nearly 24000 different advertised ASs in the Internet [CIDRRep].

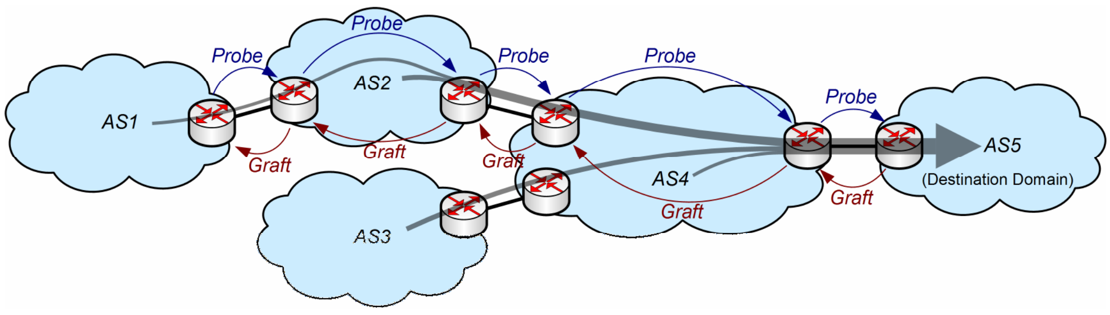


**Figure 2.6: BGRP signaling and sink tree aggregation**

Developed in the scope of the Premium IP Cluster project AQUILA [Aquila], the BGRP+ protocol [Salsano03, Sampatakos04] improves the BGRP protocol by adding a quiet grafting feature[9] that allows for an appreciable reduction in signaling overhead. The quiet grafting mechanism is based on the existence of a pre-reserved resource cushion for the sink tree at a given border router, so that when a new request arrives at that router, it can guarantee resource availability without interacting with the downstream routers. The mechanism used in AQUILA for providing such resource cushion is the delayed release of resources: when an upstream reservation is reduced or release, downstream resources are not immediately released in the hopes that if a new request arrives in the meantime, resources are already available.

In the Shared-segment Inter-domain Control Aggregation Protocol (SICAP) [Sofia03], aggregation is based on path segments that different reservations may share. Reservations may be merged into aggregates that do not necessarily extend all the way to the destination; instead, Intermediate De-aggregation Locations (IDLs) are established (preferably in ASs with large degree). This approach increases the probability of accommodating different requests in the same aggregates, minimizing the number of aggregates and reducing the amount of state required. In order to further reduce this amount of state, SICAP maintains a list of destination prefixes advertised by the AS where the reservation is terminated, merging the reservations to any of those prefixes. SICAP signaling works similarly to BGRP, and the signaling load of both approaches is comparable (both protocols exchange messages per individual reservation).

The Internet2 QBone Signaling Design Team [QBone] has developed a signaling protocol that runs between the BBs of different domains for performing resource reservations. Although the Simple Inter-domain Bandwidth Broker Signaling (SIBBS) protocol

[9] The possibility of quiet grafting was already mentioned in [Pan00], but was fully specified and implemented only for BGRP+.

[Teitelbaum00, Chimento02] assumes an application-to-application reservation model, it can also be used to establish core tunnels extending from an origin to a destination domain. The amount of stored state is, therefore $O(N^2)$, where N is the number of different domains. Even though the aggregation of core tunnels according to destination domain has been proposed, which would reduce the amount of state to O(N) (similarly to BGRP), the specification of SIBBS does not include such mechanism.

### 2.4.2 Inter-Domain QoS Routing

Routing in the Internet is performed in two layers, in accordance with the two-level hierarchy. While the protocol for intra-domain routing (usually referred to as the Interior Gateway Protocol — IGP) may be chosen by network owner at will, inter-domain routing in the Internet is performed using the *de facto* standard Border Gateway Protocol (BGP), currently in version 4 [RFC4271]. BGP is a path vector protocol for exchanging reachability information between connected ASs. Routes selected by BGP are propagated to the intra-domain routing protocol used within the AS (usually referred to as the Interior Gateway Protocol — IGP) by the border routers. The reachability information is conveyed in *UPDATE* messages, each containing an advertisement of a new or changed route to a given network destination, specified by its network prefix, and/or a set of withdrawals of routes to destinations that may no longer be reached via the AS originating the *UPDATE*. A network prefix represents a block of contiguous addresses, and is specified by a base address and an associated mask represented by the number of left-aligned 1 bits (for example, 192.168.0.0/24 represents the block of contiguous addresses ranging from 192.168.0.0 to 192.168.0.255).

Besides the destination prefix, route announcements include attributes specifying the IP address of the downstream router that must be used to reach the destination (*Next Hop*), and a list of the ASs that will be traversed en route to the destination (*AS Path*), used to check for routing loops. The length of the *AS Path* attribute is also used as a metric for route selection. Other attributes may also be present: in fact, BGP can be easily extended through the addition of optional attributes. Optional path attributes may be further classified into transitive or intransitive: transitive attributes are transparently passed to upstream nodes by a BGP node not supporting the attribute; intransitive attributes are dropped.

The reception of an *UPDATE* message triggers a three step decision process: (1) a degree of preference is assigned to the new route (if any) based on a set of policies; (2) one of the available routes to the destination is selected and propagated to the IGP; (3) if the route is different from the previously installed one, it is propagated to the peering ASs (unless otherwise specified by the policy-based route exporting rules). One important aspect that is worth stressing is that by announcing a route to a given destination to a neighbor AS, an AS is committing to forward traffic to that destination coming from that neighbor; due to the commercial nature of connections between ASs, this is frequently undesirable, thus the importance of route exporting rules.

There are two variants of BGP: external BGP (eBGP) and internal BGP (iBGP). The above described process of announcing routes to neighboring ASs is performed by eBGP, while iBGP is used to distribute the best learned routes from neighboring ASs among the other border routers of the AS.

The introduction of inter-domain QoS routing in the Internet is a complex issue. The numerous ASs are managed by independent entities motivated by business self-interests that lead to different (and frequently conflicting) goals. More importantly, inter-domain routing is the glue that holds the Internet together — without it, the Internet would break apart into a series of isolated network islands. Since a problem in inter-domain routing could seriously harm Internet connectivity, any evolution, including the addition of QoS parameters, has to be performed in small, well-tested and proven steps, and simultaneously ensuring full backward compatibility with plain BGP. The next paragraphs describe several proposals for the introduction of inter-domain QoS routing in the Internet.

A series of techniques for achieving basic inter-domain traffic engineering and/or QoS-based routing using plain BGP were described in [Quoitin03]. The selected paths for outgoing traffic may easily be controlled through the use of the *Local Pref* attribute; this attribute is used to rank the (multiple) received routes to a given destination. Manipulation of the *Local Pref* attribute based on passive or active measurements can be used for selecting the best routes, QoS-wise, for outgoing traffic[10]. Some degree of control is also possible for incoming traffic. An AS multi-connected to another AS may used the *Multiple Exit Discriminator (MED)* attribute to select the incoming link for traffic destined to a given prefix. An AS connecting to multiple ASs may advertise a given destination only to one (or a subset of) these ASs, forcing incoming traffic to that destination to enter the AS only from the peer(s) to which the route was advertised; this technique can be combined with the use of more specific routes, since BGP gives them preference over less specific ones. Finally, since one of the BGP path selection rules is the shortest *AS Path*, an AS may artificially increase the *AS Path* length by inserting itself several times in routes announced to some peers.

[10] Several service providers offer commercial route controllers that, based on measurements, select the best paths for Internet traffic at companies that are connected to more than one ISP [Bartlett02, Borthick02].

Crawley et al. [RFC2386] defined a framework for QoS-based routing in the Internet, adopting the traditional separation between intra- and inter-domain routing. They discussed the goals of inter-domain QoS routing and the associated issues that must be addressed, and provided general guidelines that should be followed by any viable solution to QoS routing in the Internet. However, they do not specify the set of QoS metrics to be transported or the algorithms for using such metrics in the choice of inter-domain routes.

A series of statistical metrics for QoS information advertisement and routing, tailored for inter-domain QoS routing (though also applicable to intra-domain routing) were defined by Xiao et al. [Xiao04], along with algorithms to compute them along the path. These metrics, the Available Bandwidth Index (ABI), the Delay Index (DI), the Available Bandwidth Histogram (ABH) and the Delay Histogram (DH), convey information expressed in terms of one or more probabilistic intervals. Simulation results show that by using these metrics, selected routes are closer to optimality than when using static metrics; moreover, the overhead is lower and the stability higher than when using the corresponding instantaneous (purely dynamic) metrics. However, these approaches consider only a single QoS parameter, making it difficult to simultaneously satisfy different requirements. When optimizing by bandwidth, paths with large delay may be chosen while others with less, yet sufficient, available bandwidth and much lower delay may be available. Conversely, when optimizing by delay, a route with low available bandwidth may be selected; switching to this route may cause congestion, increasing the delay. When the delay information is updated, the previous route might be selected again, and so on, causing route flapping (though on longer time scales than with dynamic metrics).

Cristallo and Jacquenet proposed an extension to the BGP with a new optional and transitive attribute, QoS_NLRI, for the transport of several types of QoS information [Cristallo04]. An important feature of this extension is that QoS improvement is observed even if only a fraction of the ASs supports it, making an incremental deployment possible. This work is focused on the specification of the attribute, including the formats for transporting the different parameters, such as reserved data rate or minimum one-way delay, and does not specify how the information is to be used in path selection. Some simulation results demonstrating its use with (static) information on one-way packet delay are provided, though.

The MESCAL project [Mescal], devoted to the development of solutions for the deployment and delivery of inter-domain QoS across the Internet, proposed the use of QoS routing based on Meta QoS Classes (MQCs) [Levis05]. An MQC is a standardized set of qualitative QoS transfer capabilities, corresponding to a set of common application requirements. Each domain supporting a given MQC must map it into a Local QoS Class (l-QC) supported by its infrastructure[11]. The set of ASs supporting each MQC and their adjacencies form a virtual overlay topology designated MQC plane. Inter-domain routing is performed with a QoS-enhanced BGP (q-BGP), based on the above mentioned QoS_NLRI extension, that selects, for each destination, one path per MQC plane (from an abstract view, it works as if a different instance of BGP ran on each MQC plane). Without the use of dynamic QoS information, q-BGP does not react to congestion — the networks must be provisioned so that congestion does not occur in the MQC planes where it is relevant.

## 2.5 QoS in IP-Based Mobile Telecommunication Systems

The market for information and communications technology is currently undergoing a structural change. The traditional boundaries between broadcasting networks, fixed telephony, mobile telephony and data networks are being progressively diluted, as we transition from a vertical to a horizontal model for the integration of services. In vertical network structures, services (e.g. telephony, television) can only be received with suitable networks and end devices. With a horizontal approach, users will be given the possibility of using the desired services with a single end device, regardless of the platform and network access technology. IP plays a central role in this horizontalization process, since it is available globally and, at least in principle, can be used to support virtually all the services and applications in all the networks. While the transition to an Everything over IP (EoIP) paradigm is already taking place — Triple Play services (television + telephony + Internet access) are being provided over cable and twisted pair, and the 3G mobile telephony is moving towards an All-IP model —, Next Generation Networks (NGNs) are expected to take the concept even further, providing not only uniform access to the services using different network access technologies, but also freedom of motion across those technologies without service disruption.

### 2.5.1 UMTS

The Universal Mobile Telecommunications System (UMTS), defined by the Third Generation Partnership Project (3GPP) is the most prominent of the third generation (3G) mobile phone technologies. Initially focused on backward compatibility with Global System for Mobile Communications (GSM), with voice calls performed in the circuit-switched (CS)

[11] An l-QC corresponds to a DiffServ PHB, and is identified by the DSCP.

domain and a packet-switched (PS) domain providing only basic IP connectivity, UMTS has been evolving towards an All-IP architecture, release after release.

Release 99 is strongly focused on a smooth evolution from GSM to UMTS networks. The UMTS network must be backward compatible with GSM networks and be able to interoperate with GSM. Compared to GSM, the most important enhancement is a new radio interface: the UMTS Terrestrial Radio Access Network (UTRAN), introduced by R99, uses the more efficient (with better spectral efficiency) Wideband Code Division Multiple Access (WCDMA) radio access method. Voice calls use the CS domain, and the PS domain provides only basic IP connectivity. Asynchronous Transfer Mode (ATM) transport is used in both CS and PS domains.

The UMTS Release 4 emphasizes the separation between the bearer and the control functions in the CS domain by splitting the Media Switching Center (MSC) into MSC Servers and Media Gateways. Media Gateways are responsible for connection maintenance and switching, while the MSC Server is responsible for the control of the connections. Due to these new elements and functionalities, the CS domain is able to scale freely: if more switching capacity is required, Media Gateways (MGWs) are added; when more control capacity is needed, an MSC Server can be added[12]. The MGWs can packetize the voice connections, allowing the operators to benefit from the efficiency of Voice over IP (VoIP) by moving to a single packet-switched core, shared by voice and data. Packetized voice, however, could not yet be used end-to-end, since in the access voice bearers and their control are still provided in the CS domain. Other innovations introduced with R4 were broadcast services and network-assisted location services.

The greatest step towards an All-IP network was taken in Release 5, with the introduction of the IP Multimedia Subsystem (IMS). The IMS provides control of voice and multimedia sessions (including all related functions such as accounting and charging) in a standard way, based on the Session Initiation Protocol (SIP) [RFC3261]. Voice (and multimedia) calls may now be performed entirely in the PS-domain. Release 6 further improved the IMS. Release 7 is expected to introduce Voice Call Continuity (VCC), allowing for handover of voice calls between the PS and CS domains, as well as interworking with different access networks, notably WiFi.

Figure 2.7 shows a simplified view of the PS domain of the UMTS architecture with the IMS (therefore, corresponding to Release 5 or later); the CS domain is omitted since this thesis deals with Quality of Service on packet switching networks only.

[12] A single MSC Server can control several Media Gateways.

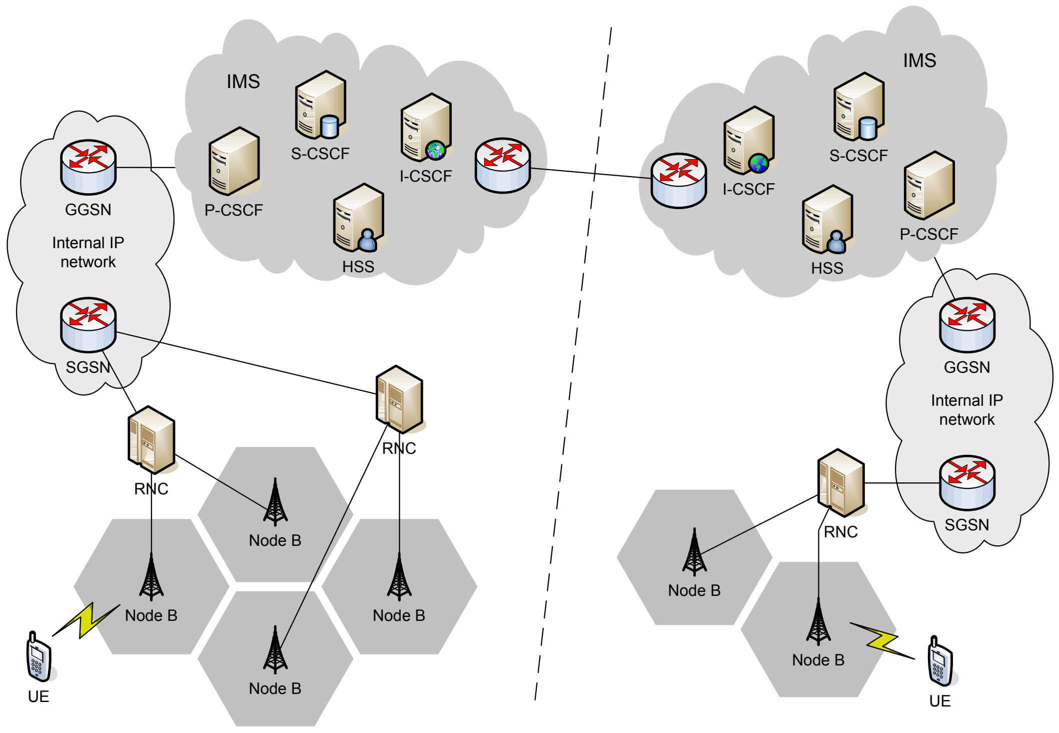


**Figure 2.7: Packet-switched domain of the UMTS architecture (with IMS)**

The Node Bs are the Base Stations, which communicate with the User Equipments (UEs) using WCDMA. The Radio Network Controllers (RNCs) perform radio resource management functions, including the control of handovers between the Node Bs under their responsibility. The Serving GPRS Support Node (SGSN) performs functions such as mobility management among different RNCs and billing user data; together with the Gateway GPRS Support Node (GGSN), it is responsible for connecting the radio access network to the IP network and mapping QoS at the IP layer to QoS at the radio layer. The protocol stack from the GGSN downwards to the UE is pretty complex, as may be seen in fig. 2.8, and the multi-level encapsulation generates appreciable overhead.

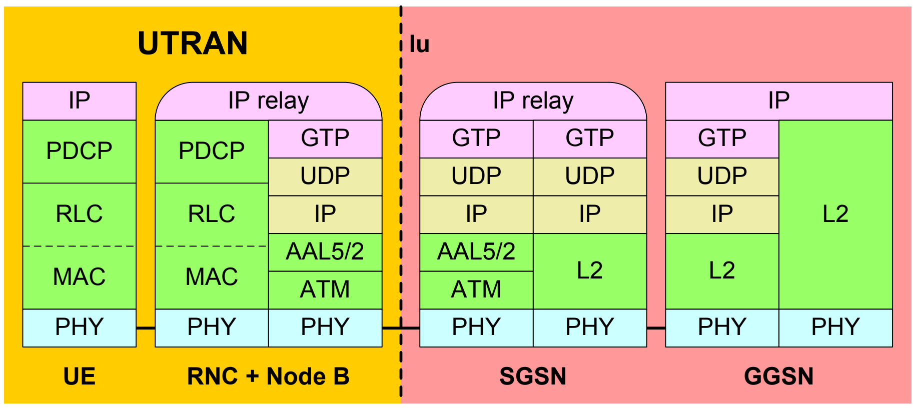


**Figure 2.8: UMTS protocol stack**

The IMS incorporates the Home Subscriber Server (HSS) and three Call Session Control Functions (CSCFs): the Proxy-CSCF (P-CSCF), the Serving-CSCF (S-CSCF) and the Interrogating-CSCF (I-CSCF). The HSS is the master user database; it contains subscription-related information (user profiles), and aids the call control servers in completing the routing/roaming procedures by solving authentication, authorization, name/address resolution, and user location issues. The P-CSCF provides coordination between events in the application layer and resource management in the IP bearer layer, acting as a Policy Decision Point (PDP) for the GGSN. The I-CSCF is mainly the contact point, within an operator's network, for all IMS connections destined to a subscriber of that network operator or to a roaming subscriber currently located in that operator's service area. The I-CSCF's IP address is published in the operator's Domain Name Service (DNS) so that remote servers can find it. The S-CSCF is the central node of the signaling plane: it handles the session states in the network, managing ongoing sessions, and providing accounting mechanisms. As previously stated, signaling in the IMS is based on SIP; communication between the P-CSCF in the IMS and the GGSN is performed using the Common Open Policy Service (COPS) protocol [RFC2748]. In the Radio network, signaling to the UE is based on GPRS mechanisms.

Since the IMS belongs to the IP Backbone, QoS is supported by DiffServ mechanisms. The GGSN is the entity responsible for providing DiffServ edge functionality. Four different QoS classes with different QoS guarantees are provided between the GGSN and the users: conversational (the most demanding), streaming, interactive and background (no QoS guarantees). These classes, in an operator-driven scheme, can be mapped into the DSCP field of the IP header depending on the bandwidth and resource provisioning.

The UMTS network architecture was developed considering not only future networks' QoS requirements, but also the support for legacy GSM/GPRS technologies that have coarser support for QoS. Nevertheless, the mechanisms deployed for UMTS provides the means to achieve full end-to-end QoS. However, the most commonly deployed scenario is based on an overprovisioned core, with per-flow resource reservation being performed only for the radio access.

### 2.5.2 Next Generation IP-Based Mobile Networks

The excessive complexity of the UMTS stack and the need for supporting seamless movement across heterogeneous access networks, along with the requirement for fast development and deployment of new telecommunication services, led researchers to work on simpler and more flexible All-IP architectures for Next Generation Networks (NGNs).

Envisioning an evolution of the 3G mobile and wireless infrastructure towards the Internet, the Moby Dick project [MobyDick] has defined, implemented, and evaluated an IPv6-based, mobility-enabled QoS architecture for heterogeneous networks, based on IETF's QoS models, Mobile IPv6, and Authentication, Authorization, Accounting and Charging (AAAC) framework. The proposed architecture [Marques03] is centered on QoS Brokers and their cooperation with an AAAC system. The architecture uses IPv6 as a convergence layer, providing a common ground for the use of different access technologies. Mobility management is performed at layer 3, using Mobile IPv6 with Fast Handover (FHO) extensions, and is controlled by the QoS Brokers. The FHO support uses a "make before break" approach, where resources are reserved on the new cell before disconnecting from the previous one; moreover, packets are bicast to both cells until the handover is finished. The use of QoS-aware Fast Handovers, combined with the use of context transfer, allows for seamless mobility across different access technologies.

QoS on the data plane is provided by DiffServ. On the control plane, QoS support is based on the concept of services. Users subscribe to SLAs consisting on sets of services with high-level descriptions (e.g., telephony); these services are mapped by the operators into network services (consisting on a traffic class, identified by the DSCP, and associated rate limits). The set of subscribed network services constitutes the Network View of the User Profile (NVUP). QoS Brokers use the NVUP for admission control (combined with information on available resources) and for configuring the Access Routers (ARs). QoS signaling is implicit and performed in-band. In order to request a service, the user starts sending packets marked with the corresponding DSCP. The first packet is intercepted at the AR, triggering an admission control request to the QoS Broker; if the request is accepted, the AR is instructed to let the packets marked with that DSCP flow to/from the requesting Mobile Terminal. For termination, every network service has an associated timeout: if no packets with the corresponding DSCP are sent or received during this timeout period, the network elements will automatically free the resources and stop accounting usage of the service. Resource management in the access is, thus, performed per (Terminal, Service). In the core, resources are managed by aggregate, independently of the individual flows, using an overprovisioning approach.

The Moby Dick architecture provides an integrated solution addressing most relevant issues for NGN operators (AAAC, QoS, mobility, security). However, QoS is relatively coarse-grained (it allows only one instance of a network service per terminal at any time), and relies on overprovisioning outside the access link. The network part of the DAIDALOS architecture introduced in chapter 5 is an evolution of the Moby Dick architecture.

## 2.6 SIP — Session Initiation Protocol

The Session Initiation Protocol (SIP) [RFC3261] is an application-layer signaling protocol for establishing, modifying and terminating sessions between two or more participants over IP networks. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences. SIP can be used to invite participants to already existing sessions, such as multicast conferences, and also to add or remove media to/from an existing session.

Originally developed around 1996 from the merging of two proposed protocols for the multimedia conference control (the Session Invitation Protocol and the Simple Conference Invitation Protocol), SIP has been very actively developed due to the enormous interest it generated, resulting from the great promises of IP telephony. SIP was first standardized in [RFC2543], and is now specified by [RFC3261] along with a large number of extensions. The adoption of SIP by the 3GPP as a mandatory protocol for signaling in the IMS was not only a landmark for establishing it as the most prominent protocol in IP telephony, but also a major drive for standardizing the extensions required for mobile telephony.

### 2.6.1 Protocol Overview

SIP is a protocol for controlling sessions consisting of one or more media streams (audio, video, or any other IP-based communication mechanism). It is a text-based request/ response protocol, heavily drawing from the Hypertext Transfer Protocol (HTTP) [RFC2616]: SIP messages consist on a request or status line, header fields and an optional Multipurpose Internet Mail Extensions (MIME) body. A notable difference is that SIP can use different transport protocols, including UDP, using retransmissions for ensuring request reliability over unreliable protocols.

SIP endpoints are addressed by Uniform Resource Identifiers (URIs), usually *sip* or *sips*, similar to email addresses (e.g., *sip:alice@wonderland.org*). SIP requests contain a source address (*From* header) and two destination addresses, one with the original, logical destination (*To* header) and the other with the current destination (URI in the request line), which may change as a result of the user location (call routing) process. Moreover, the *Contact* header specifies the address where future requests should be sent.

SIP defines four logical entities (user agents, registrars, redirect servers and proxies), and an abstract location service; it also requires DNS support for translating names to IP addresses and for finding servers in remote domains. User agents (UAs) originate and terminate requests, and are the only elements where SIP signaling and the media converge; soft phones and PSTN gateways are examples of UAs. Registrars handle registration requests, placing the received information (including UA location) in the location service. Redirect servers receive requests and, using the location service, respond with an indication of one or more URIs where the requestor should send the request to. Proxies are intermediaries used mainly for routing requests towards the destination. SIP proxies can be stateless or stateful. A stateless proxy simply forwards incoming requests to another server without ensuring reliability. A stateful proxy maintains state for transactions (consisting on a request and responses to that request), therefore, it may ensure reliability. A stateful proxy may also iterate or fork requests, attempting to reach the destination at different locations (sequentially or simultaneously). Stateful proxies may further be classified into transaction stateful, which maintain state only for the duration of the transactions, or call-stateful, which control calls up for their entire duration (call-stateful proxies insert their address into a *Record-Route* header in the first request in order to force subsequent requests to traverse them as well). It is worth noting that SIP does not define how these logical entities are implemented or deployed, and multiple logical entities can be implemented in a single box. Frequently, a proxy, a redirect server and a registrar are integrated into a single SIP server.

The base specification of SIP defines six request methods. The most important is the *INVITE* method, used to establish new sessions or modify existing ones. The *ACK* confirms the establishment of a session, completing a three-way handshake that adds reliability to the *INVITE* transaction. The *BYE* method is used to terminate an established session, and the *CANCEL* method to terminate an ongoing request for a session that is not yet completely established. The *OPTIONS* request is used to query for capabilities and negotiate options prior to establishing a session. The *REGISTER* method is used by the UA to, periodically, send location information to the Registrar. Extensions to the base protocol may add new request methods to this set: for example, [RFC3311] defines the *UPDATE* method that allows a client to update parameters of a session (such as the set of media streams and their codecs) without impacting the state of the dialog. SIP responses are identified by a three digit status code. Provisional responses (status code 1xx) indicate that the request was received and is being processed, and also provide information on the progress in contacting the called user (e.g., phone is ringing). Provisional responses are not sent reliably in the base protocol; however, there is a standard extension [RFC3262] for sending them reliably, when necessary, by requesting that their delivery be confirmed by means of a *PRACK (PRovisinal ACKnowledge)* request. Final responses (status codes 2xx-6xx) indicate a resolution of the request; they are divided into success (2xx), redirects (3xx) and different types of failures (4xx-6xx).

Session negotiation is not performed by SIP itself: details such as the type of media, codec or sampling rate are not described and negotiated using SIP, but rather by a protocol such as the Session Description Protocol (SDP) [RFC2327], using an offer/answer model [RFC3264]. The messages of the negotiation protocol are transported, MIME-encoded, in the body of SIP messages.

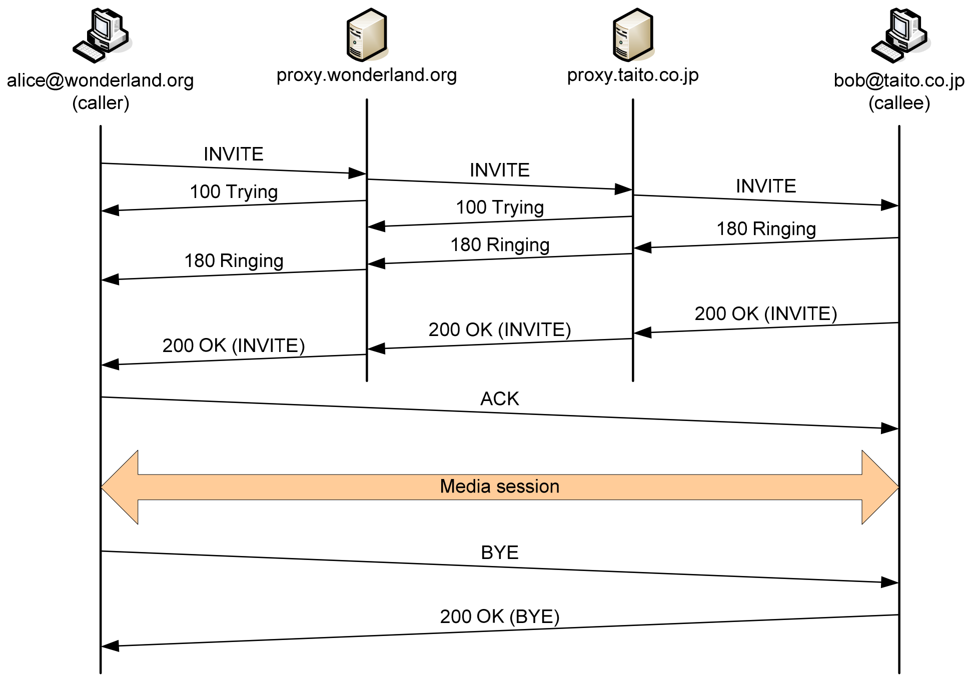


**Figure 2.9: Session initiation with SIP**

Figure 2.9 shows an example of session initiation with SIP using transaction-stateful proxies. The caller (Alice) sends the *INVITE* through her outbound proxy, which replies with a *100 Trying* message informing her that it is working to forward the request. This message quenches retransmissions of the *INVITE*. Using DNS, Alice's outbound proxy discovers the proxy responsible for Bob's domain, to which it forwards the request. After receiving the *INVITE*, Bob's soft phone starts ringing, and sends the corresponding response to Alice (180 Ringing). All responses to a request follow the exact inverse path of the request. When Bob answers the call, a *200 OK* message is sent, indicating that the call is accepted; this message is acknowledged with an *ACK* request. Since the proxies have not added themselves to a *Record-Route* header in the *INVITE*, further requests (and, consequently, their replies) are directly sent between the UAs. The media is always sent directly between the UAs. When Alice hangs up, a *BYE* request is sent to Bob to terminate the call (had it been Bob to hang up first, the *BYE* would have been sent in the opposite direction).

## 2.6.2 Integration with QoS Signaling — Preconditions

Multimedia conferencing in general and IP telephony in particular are applications which depend heavily on the QoS provided by the underlying networks and, consequently, benefit greatly from the reservation of resources. Coordination between resource reservation signaling and SIP is achieved through the use of preconditions [RFC3312, RFC4032]. A precondition is a set of constraints about the session which are introduced in the offer. The recipient of the offer generates an answer, but does not alert the user or otherwise proceed with session establishment, namely alerting the user, until the preconditions are met. This can be known through a local event (such as a confirmation of a resource reservation), or through a new offer sent by the caller.

Figure 2.10 shows an example of the use of preconditions for coordinating SIP and resource reservation (end-to-end in this case). After the first offer/answer round, in possession of the IP all required information (IP address and port of the other party, codec), both endpoints perform resource reservation (each one for its sending direction). After performing the reservation specified by the callee for confirmation in SDP2 (*conf* line), the caller sends an *UPDATE* request with a new offer reflecting the change. Meanwhile, the callee had already performed its part of the reservation, so it sends an answer (SDP4) in the response to the *UPDATE* and immediately starts ringing, resuming the establishment of the session.

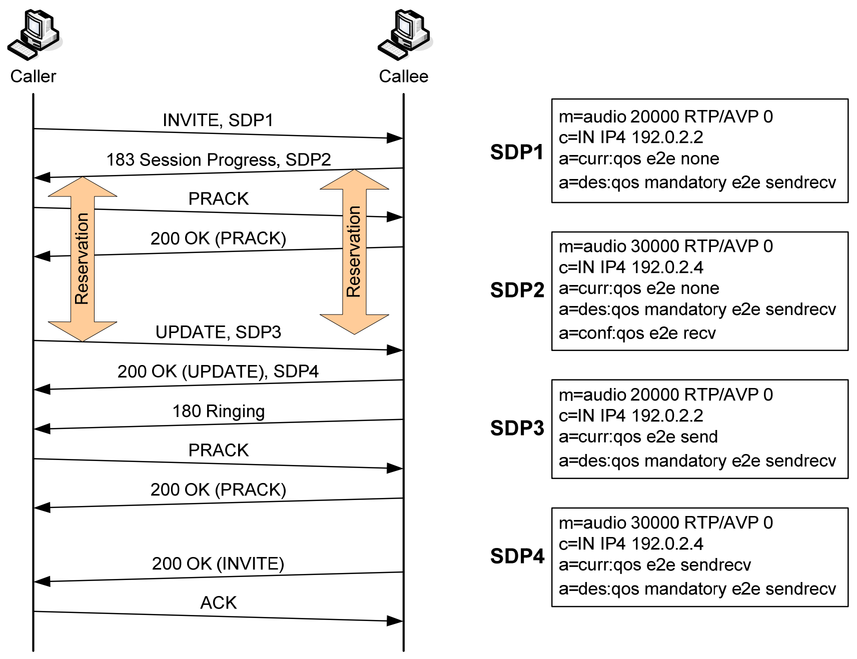


**Figure 2.10: Integration of SIP and resource reservation using preconditions (example)**

### 2.6.3 SIP and Mobility

SIP has built-in support for application layer terminal mobility management [Wedlund99, Schulzrinne00]. Pre-session mobility is trivially provided by the location service: the terminal simply re-registers with its "home" registrar every time it obtains a new IP address. Mid-session mobility is supported by sending a *re-INVITE* to the other party, informing it of the new IP address. This *re-INVITE* is routed much faster than the original *INVITE*, since the moving terminal already has the direct contact of the other party, and only proxies that added themselves to the *Record-Route* header will be traversed.

The use of SIP for mobility management, however, trades generality for ease of deployment. SIP mobility is not suitable for TCP connections, as they depend on the IP addresses of both endpoints. This, and the fact that different procedures must be used for SIP and non-SIP sessions, constitutes a disadvantage of SIP-based mobility compared to a general layer 3 solution like Mobile IP (MIP) [RFC3344] or Mobile IPv6 (MIPv6) [RFC3775]. However, the duplication of mobility management functions in the network and application layers leads to inefficiency issues. The following paragraphs describe some proposals for different degrees of integration of SIP and MIP (v4 and v6), aimed at providing improved mobility management capabilities.

Jung et al. [Jung03] proposed the use of integrated mobility agents for SIP and MIP(v4). Some of the MIP functions (like binding refreshments) are transposed to SIP, and mobility is communicated to the correspondent nodes (CNs) via *re-INVITE* requests. This approach imposes different handover procedures for SIP and non-SIP sessions (UDP or TCP), and the security issues of establishing bindings with CNs via SIP were not addressed.

Politis et al. [Politis03] proposed a hybrid SIP/MIP(v4) scheme for inter-domain mobility. Their approach avoids the IP-in-IP MIP encapsulation for SIP sessions, but not for non-SIP ones, for which the handovers are managed by MIP. Their work mostly concerns mid-session mobility which, in our case, is handled by a modified MIPv6 with Fast Handover extensions. Moreover, the encapsulation problem is mitigated in MIPv6 by the use of routing optimization.

Wang et al. [Wang03, Wang04] proposed an integrated SIP-MIP mobility management architecture, where MIP and SIP agents are broken down into functional blocks and then integrated, without duplication, into unified Home and Foreign Mobility Servers (HMS/FMS), thus avoiding redundancy. Their proposal mostly intends to solve the problems associated with different types of mid-session mobility that in our case are addressed at the layer 3. Moreover, their architecture is different from ours in that it requires one FMS per (access) network.

An interesting problem that none of these proposals has addressed concerns the inefficiency issues resulting from the integration of end-to-end resource reservation with session signaling in mobile scenarios.

## 2.7 Summary

This chapter provided some background for the work described in this thesis, with an overview of the most relevant related work. We described the different functions that can be used as building blocks for QoS provisioning. We introduced the two major QoS frameworks proposed by the IETF (IntServ and DiffServ) and gave an overview of the most important proposals for QoS models aimed at the conciliation of the benefits of IntServ (per-flow guarantees) and DiffServ (scalability); the proposals were grouped according to the central concept employed — alternative signaling model, flow aggregation, elimination of state in the core, or centralization of control plane functions in Bandwidth Brokers. We also described proposed solutions for the two orthogonal problems of inter-domain QoS — inter-domain resource reservation and inter-domain QoS routing. With respect to QoS in IP-based mobile networks, we gave an overview of UMTS, focusing on the packet-switched domain, and pointed out directions for the next generation networks. Finally, we introduced the Session Initiation Protocol, central to UMTS' IP Multimedia Subsystem and the strongest contender in the field of multimedia session establishment and control protocols for the Internet; we gave an overview of the protocol, of its coordination with network resource reservation signaling, and its relation with mobility.

# CHAPTER 3

# EVALUATION OF RSVP RESERVATION AGGREGATION

Even though the RSVP Reservation Aggregation (RSVPRAgg) [RFC3175] model was generally considered the best candidate for a scalable replacement of RSVP/IntServ [RFC2205, RFC1633] in high-speed core networks, no performance study had been carried out on a packet network simulator, though numerical simulations for aggregation in general did exist [Sargento02]. In fact, no implementation of the model which could be used to perform such study was available for any packet network simulator. This state of affairs, along with the need for such implementation to perform the comparison with the Scalable Reservation-Based QoS architecture presented in the next chapter, led us to the development of an RSVPRAgg extension module for the ns-2 simulator [NS2], publicly available for download from [PriorNS], and to the simulation study presented in this chapter and originally published in [Prior04c].

In this chapter we present an overview of the RSVPRAgg QoS model, its implementation and performance evaluation. We begin with an overview of the model and its principles in the next section. Section 3.2 describes the implementation in more detail, regarding message processing and the bandwidth management policy for aggregates, which is considered out of the scope of [RFC3175]. Some particularities of our implementation and limitations of the model are also discussed in this section. A performance evaluation, based on simulation results, is presented in section 3.3. We analyze the standard QoS parameters (delay, jitter and packet loss ratio), as well as other parameters relevant to the performance and scalability of the architecture, such as network resource utilization and the number of signaling messages processed at core nodes, and compare them to those obtained with the standard RSVP/IntServ architecture in similar conditions. The results show that, while RSVPRAgg is able to meet the QoS requirements of a controlled load class in a scalable way, it suffers from underutilization of network resources. With these simulations we also evaluated the influence of some tunable parameters: the *bulk* size and the hysteresis time. Based on the results we derive some guidelines for setting these parameters. Finally, section 3.4 presents the conclusions of the evaluation herein performed.

## 3.1 Overview of RSVP Reservation Aggregation

The RSVP Reservation Aggregation model [RFC3175] preconizes the use of a single RSVP reservation to aggregate different RSVP reservations inside an aggregation region, provided they share the ingress and egress nodes (aggregator and deaggregator) and belong to the same traffic class. Each aggregate reservation has a token bucket specification whose rate and bucket depth are larger or equal to the summed values of the individual end-to-end RSVP reservations using it. Furthermore, aggregate reservations are updated in quantities much larger than average individual flows' token bucket specifications, termed *bulks*. The establishment of a smaller number of aggregate reservations on behalf of a larger number of end-to-end ones, combined with an update rate much lower than the reservation setup/termination/modification rate of the end-to-end flows, allows for a considerable reduction in the amount of state stored and the number of signaling messages processed at the interior nodes of the aggregation region. Edge nodes, on the other hand, must keep state information and process signaling messages for both end-to-end and aggregate reservations. Therefore, the amount of state and message processing overhead at the edge nodes is somewhat higher than in regular RSVP.

In order to hide end-to-end RSVP messages from RSVP-capable routers inside the aggregation region, the aggregator changes the IP protocol number field of some RSVP messages (namely *Path*, *PathTear* and *ResvConf*) to RSVP-E2E-IGNORE. These messages are forwarded as normal IP datagrams inside the aggregation region, meaning they are not processed and no corresponding state is stored. At the deaggregator, the IP protocol number field is restored to RSVP. The previous hop perceived (and stored) by the deaggregator for these messages is the aggregator. This implies that the protocol number for other end-to-end RSVP messages needs not be changed, as they are unicast from RSVP hop to RSVP hop[1].

[1] That is, are directly sent from one hop to the following, using their addresses as source and destination IP.

The use of DiffServ (Differentiated Services) [RFC2475] mechanisms for the classification and scheduling of traffic supported by aggregate reservations makes the amount of classification and scheduling state stored inside the aggregation region independent not only of the number of end-to-end reservations but also of the number of aggregate reservations. The process of classifying and scheduling packets inside the aggregation region is, therefore, comparatively light in terms of required processing power.

Traffic covered by aggregate reservations is identified by one or more Differentiated Services Code Points (DSCPs), corresponding to one or more Per-Hop Behaviors (PHBs) that provide the required forwarding treatment. Different traffic classes are supported by having more than one aggregate between each pair of ingress/egress routers. In this case, a different DSCP (or DSCP set) and a different PHB (or PHB group) are assigned to each traffic class (more than one DSCP set and PHB group may be assigned to a single traffic class in order to achieve differentiation within that class).

Aggregate reservations are performed in a similar way to regular RSVP reservations, only using a different session specification which carries the IP address of the deaggregator and the (main) DSCP of the aggregate.

The main advantage of aggregating end-to-end reservations using the model briefly described in the preceding paragraphs is a much enhanced scalability than is the case with regular RSVP. This is due to (1) the much lighter packet classification and scheduling procedures, (2) the reduced amount of state stored at the interior nodes and (3) the lower number of signaling messages processed at these nodes. Another significant advantage of this architecture is transparency: terminals and access domains do not need to be modified or even know that core domains are using aggregation. At access domains, where scalability is not such an important issue as it is in core domains, standard RSVP/IntServ is used.

The main disadvantage of the aggregation model is the underutilization of network resources. In fact, there must be at least one aggregate (more if more than one service class is supported) for each (*aggregator*, *deaggregator*) pair and, in order to reduce signaling processing and profit from the resulting scalability enhancement, the bandwidth of each aggregate must be updated in bulk quantities much larger than the average flow rate. The bandwidth of the aggregates is, therefore, almost never fully utilized. The unused bandwidth of all aggregates traversing a link adds up, meaning that there will be a significant amount of wasted link capacity. The *bulk* size must, therefore, be chosen weighting the higher signaling reduction from large *bulk* sizes against the increased network resource underutilization.

Another disadvantage of the model is the loss of isolation between end-to-end flows, implying that one flow may suffer delay from the bursts of others. This, however, may not be a big problem, since there is evidence [Clark92] suggesting that aggregating flows does not increase their mean delay and may, in fact, reduce the tail of the delay distribution curve. A further disadvantage is the above mentioned need for edge routers to maintain state information and process signaling messages for both end-to-end and aggregate reservations, meaning that it is even worse than regular RSVP/IntServ in this respect.

## 3.2 Implemented Solution

The RSVPRAgg model was implemented in the ns-2.26 network simulator [NS2] as an extension to an existing implementation of the RSVP protocol by Marc Greis [Greis98]. The code is available in [PriorNS]. This section describes the implemented solution, regarding message processing and the aggregate bandwidth management policy. It also describes some implementation particularities and some limitations of the aggregation model and its specification.

### 3.2.1 Message Processing

The first event at the aggregation region is the arrival of an end-to-end *Path* message. The edge node that receives the message becomes the aggregator for an eventual future reservation. It stores the path information and forwards the message by the interior link using an IP protocol field value of RSVP-E2E-IGNORE (marked with the * symbol in figs. 3.1, 3.2 and 3.3). The message is forwarded by the core routers as a normal IP data packet (that is, without processing it). We have chosen to use a DSCP for signaling messages that provides them preferential treatment over other packets. When an edge node receives this hidden end-to-end *Path* message and finds that the outgoing interface is exterior, it processes the message, becoming the next hop to the aggregator for this end-to-end path. The next few paragraphs describe the message sequences for three different scenarios: in the first one, the

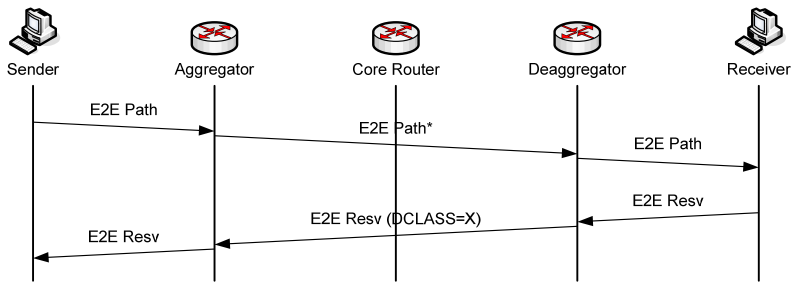


**Figure 3.1: Simplest case — the aggregate may accommodate the new flow.**

aggregate to which the new flow is assigned already exists and has enough bandwidth to accommodate it (fig. 3.1); in the second one, the aggregate exists but has insufficient bandwidth (fig. 3.2); in the third one, the aggregate does not yet exist and must, therefore, be created (fig. 3.3).

If path information for the aggregate to which the new flow is assigned already exists, with or without available bandwidth to accommodate the new flow, the *Path* message is forwarded towards the receiver, with the IP protocol field value restored to RSVP (figs. 3.1 and 3.2). In this case, when an end-to-end *Resv* message arrives at the deaggregator, it first checks if there is enough bandwidth allocated to the aggregate to support the new flow. If enough bandwidth is available (fig. 3.1), the reservation information is stored and the end-to-end *Resv* message is sent to the perceived Previous HOP (PHOP) — the aggregator. A DCLASS object is added to this message in order to tell the aggregator which aggregate the flow will belong to — this is necessary for supporting multiple aggregates having the same (*aggregator*, *deaggregator*) pair but different service classes. Otherwise (fig. 3.2), the reservation is held as pending and a bandwidth increase (in multiples of the *bulk* size) for the aggregate is requested. The procedure to modify the bandwidth of an existent aggregate will be explained in section 3.2.2.

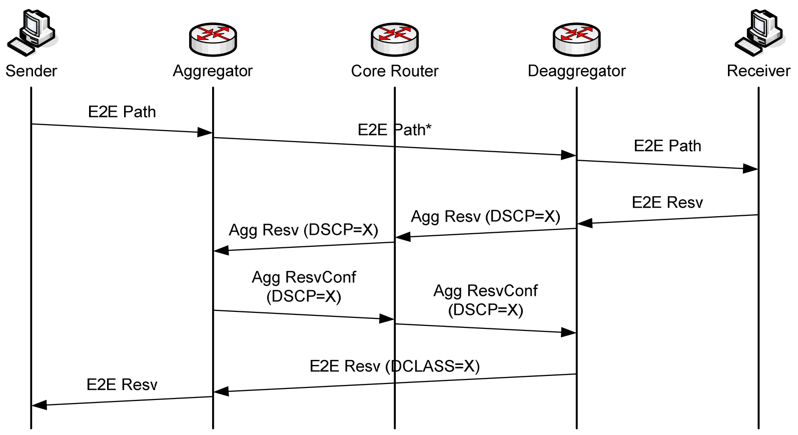


**Figure 3.2: Aggregate exists, but has insufficient bandwidth**

If no path information exists for the aggregate to which the new flow is assigned (fig. 3.3), a pending session is created, holding the end-to-end *Path* information, and an end-to-end *PathError* message is sent to the aggregator signaling the request for a new aggregate path. In this case, the aggregator responds with an aggregate *Path* message, using the DSCP from the DCLASS object sent with the end-to-end *PathError* message. When the aggregate *Path* message arrives at the deaggregator, the end-to-end path information may be moved from the pending session to a regular one, and the end-to-end *Path* message is forwarded towards the receiver. It also tries to allocate some bandwidth (one *bulk*) for this aggregate, since odds are that soon a corresponding end-to-end *Resv* message will arrive.

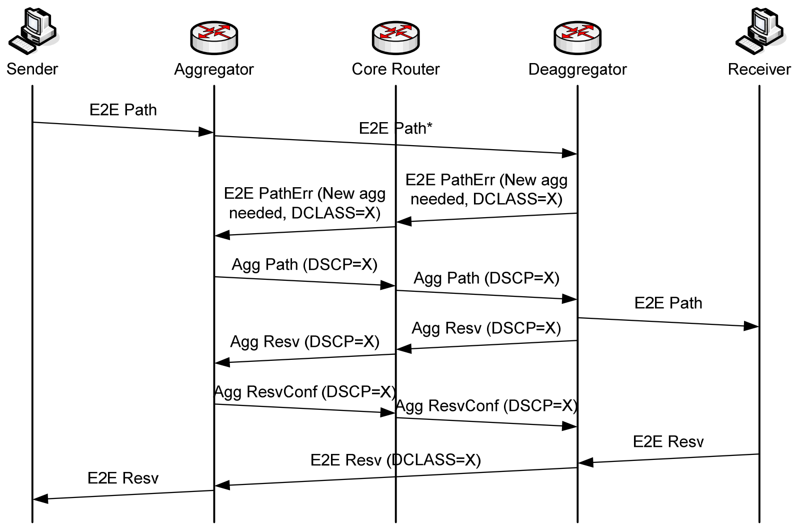


**Figure 3.3: The aggregate does not exist**

The procedure for requesting a new aggregate reservation or modifying an existing one consists on sending an aggregate *Resv* message with the requested *flowspec* towards the aggregator. Notice that it is essential for the deaggregator to be signaled that the aggregate reservation was successfully modified. One method to do this, proposed in [RFC3175] and used in our implementation, is the confirmation of changes to reservations by means of *ResvConf* messages. If there is enough available bandwidth along the path to accommodate the requested aggregate bandwidth up to the aggregator, a *ResvConf* message will be sent to the deaggregator; it may then try admission control for the pending reservations belonging to this aggregate. If not enough bandwidth is available, the deaggregator will receive an aggregate *ResvErr* message, which will trigger the emission of end-to-end *ResvErr* messages for all pending reservations belonging to the aggregate. Since the rejection of the (modification of the) aggregate may occur in any node from the deaggregator up to the aggregator, when the former receives the aggregate *ResvErr* message, the bandwidth reserved for the aggregate may be the larger requested one up to some interior node, though not up to the latter. The deaggregator is, therefore, responsible for the removal of the excess bandwidth, which will not be used, by sending a new aggregate *Resv* message with the last confirmed *flowspec*. Notice that double admission control is performed at the deaggregator: (1) for the output link and (2) for the aggregate. In the second case, the aggregate *flowspec* used for admission control is the Greatest Lower Bound (GLB) of the last requested and confirmed aggregate *flowspecs*.

In order to avoid repeated unsuccessful attempts at increasing the bandwidth for an aggregate, a (configurable) hold time is also imposed after the first unsuccessful attempt. During this period, no end-to-end flow will be admitted in the aggregate, unless some bandwidth is freed by other flows sharing that aggregate.

End-to-end reservations may be terminated in one of three ways: (1) by a *ResvTear* message sent upstream by the receiver, (2) by not refreshing the reservation, or (3) by a *PathTear* message sent downstream by the sender. Whenever one of these events is processed by the deaggregator, the flow's bandwidth is released and made available for other flows sharing the aggregate which may request it.

### 3.2.2 Aggregate Bandwidth Policy

Although [RFC3175] defines no actual policy for aggregate bandwidth management, since it is considered out of the scope of the document, it provides general guidelines for such policy. In particular, it states that the bandwidth of the aggregates should be modified infrequently, and that some hysteresis should be used in order to avoid oscillations under stable operating conditions. Figure 3.4 illustrates the aggregate bandwidth management policy used in our implementation and described in the next paragraphs. The aggregate bandwidth is plotted along with the sum of the reservations belonging to the aggregate.

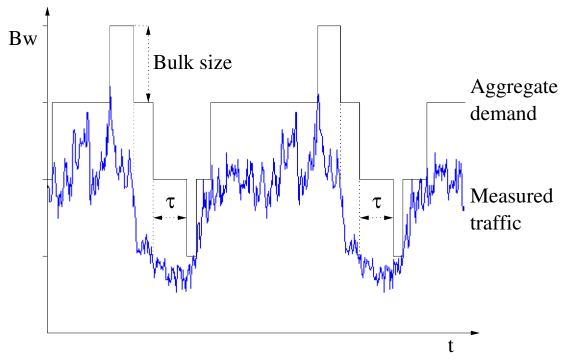


**Figure 3.4: Aggregate bandwidth management**

Bandwidth updates for aggregates are always performed in multiples of a *bulk*. The *bulk* size is configurable, and should be set to a value much larger than the individual flows' rates. Bandwidth increase for aggregates is performed on demand, i.e., when a new end-to-end reservation request arrives at the deaggregator, it is assigned to a certain aggregate and there is no bandwidth to accommodate the new flow on that aggregate. Since we are dealing with simulation, the definition of rules to predictively estimate traffic patterns and perform bandwidth management accordingly would not be as meaningful as in the case of real networks with actual customer traffic over large time spans. Though it may lead to an increased reservation setup delay, this reactive policy tends to increase network utilization, since it leaves more bandwidth available for other aggregates which will hold actual traffic. The sole exception to the reactive policy rule happens at the creation time of a new aggregate. Since it is triggered by the reception of a *Path* message, odds are that a request for a reservation assigned to the new aggregate will soon be received. By predictively allocating some bandwidth (one *bulk*) to the aggregate, it is possible to reduce the setup time for that end-to-end reservation.

Bandwidth reduction for aggregates is not performed immediately when it ceases to be needed. Instead, it is delayed until the excess *bulk* has not been in use for a certain, configurable, time period $\tau$. This hysteresis mechanism is intended to avoid unnecessary message processing at the interior nodes by successively increasing and decreasing the bandwidth in a stable operating point around a multiple of the *bulk* size. If, at a certain instant, the wasted bandwidth of an aggregate exceeds two *bulks*, though, bandwidth reduction is performed immediately, leaving only one excess *bulk* and restarting the hysteresis timer.

In order to avoid repeatedly trying to increase an aggregate's bandwidth without success, leading to unnecessary message processing at the core (interior) nodes, a configurable hold time was also implemented during which no aggregate bandwidth increase will be tried in response to the arrival of a new end-to-end reservation request assigned to that particular aggregate. During that time period, new end-to-end reservation requests will either be accepted immediately, which may happen if other flows belonging to the same aggregate were terminated leaving some bandwidth available, or they will be rejected.

### 3.2.3 Particularities and Limitations

Our ns-2 implementation of the aggregation model has some particularities and limitations which will be discussed in the next paragraphs. Of these, some are due to simplifications in our extension to RSVP/ns, while others are limitations of the specification of the RSVP reservation aggregation model itself.

One limitation of the aggregation model is related to the Guaranteed Service (GS). With aggregation, the rate-delay coupling assumption is not valid — all flows aggregated into a single queue share the same delay bounds. Due to the disparity of these bounds, GS flows sharing the same aggregator and deaggregator nodes cannot be assigned to a single aggregate; instead, they must be partitioned into a set of aggregates, each corresponding to a different delay bound [Schmitt99]. Dynamically partitioning the flows into aggregates would be too complex and generate too much signaling to be scalable. One must, therefore, resort to static partitioning, leading to sub-optimal results. This sub-optimal partitioning inevitably leads to even more underutilization of network resources in a model which already suffers significantly from this problem. In any case, partitioning leads to a larger number of aggregates and, therefore, to more signaling and underutilization.

One limitation of the model specification, stated in section 1.4.8 of [RFC3175], is the present lack of multicast support. Several factors contribute to this, namely (1) the difficulty in constructing a multicast tree that assures that aggregate *Path* messages follow the same path as data packets and (2) the amount of heterogeneity that may exist in an aggregate multicast reservation. Even if (1) is solved, addressing (2) would probably lead to a set of procedures which provide no substantial reduction in the amount of state stored and messages processed by interior nodes. The proposed solution is a hybrid one, where the reservations are setup using end-to-end signaling, making use of aggregation only for packet classification and scheduling. Partly due to this limitation, partly due to the fact that multicast evaluation was not one among our main goals, there is currently no support for multicast in our implementation.

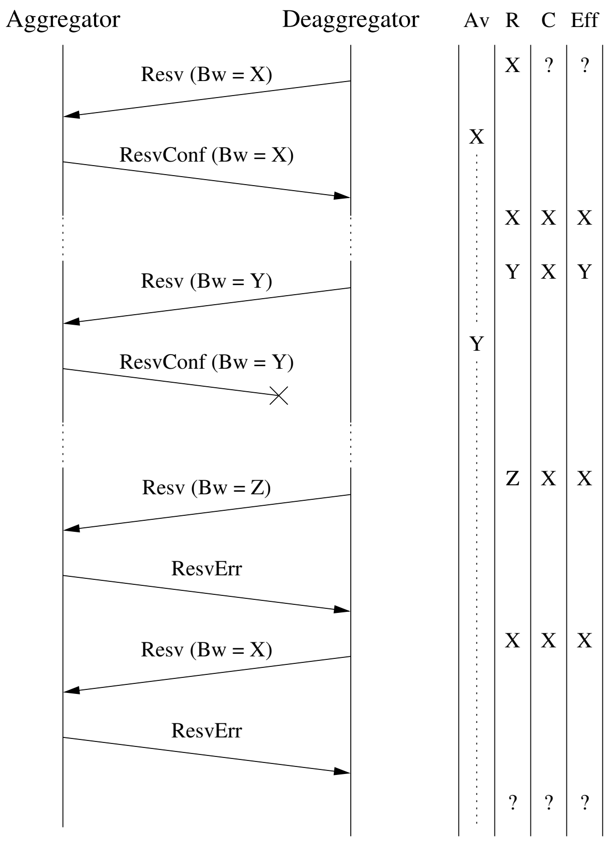


**Figure 3.5: Loss of *ResvConf* problem**

Another potential problem of the model not addressed by [RFC3175] is as follows. As previously stated, the *flowspec* value effectively used for admission control in the aggregate must be the GLB of the last requested and confirmed *flowspecs*, since it is the value guaranteed to be available all the way up to the aggregator. The way the aggregation model is specified, it may lead to inconsistencies if a single *ResvConf* message is lost. Figure 3.5 illustrates this problem. *Av* is the minimum *flowspec* installed in all the links from the deaggregator up to the aggregator (the available *flowspec*); *R* and *C* are, respectively, the last requested and confirmed *flowspecs* (at the deaggregator); *Eff* is the *flowspec* used for admission control to the aggregate (also at the deaggregator). Suppose a reservation with a bandwidth value of *X* was successfully installed and confirmed. Then, at some instant, the deaggregator decides to release some unused bandwidth, setting up a reservation with a bandwidth value of $Y<X$. If the confirmation for this reservation was lost in transit, the last confirmed bandwidth value remains *X*. Now suppose a new modification is attempted, increasing the bandwidth to a value of $Z>Y$ (and, in the illustrated case, also $Z>X$). At that time, the GLB of the two *flowspecs* becomes *X*, when effectively only *Y* bandwidth is reserved. Worse, if this modification fails, triggering a *ResvErr* message, the deaggregator will try to restore the *flowspec X* in order to avoid bandwidth wastage. However, since $X>Y$, this request may also fail, totally confusing the deaggregator. This problem may be solved by adding a rule stating that the effective bandwidth used for admission control can only be increased in response to the arrival of a *ResvConf* message.

There was also a problem with the original RSVP/ns implementation: as soon as one reservation from a session was changed, a refresh for the whole reservation state of the session would be performed. This is in violation of what is stated in section 3.3 of [RFC2205], and is fixed in our implementation so that only the changed reservation is refreshed, delaying the others until the scheduled periodic refresh instant.

Dynamic routing is not yet fully supported by our implementation. In order for dynamic routing to work properly, some mechanism would be needed to try to restore aggregate reservations when routes change with a bandwidth as close as possible to that of the previously established aggregate, and then to send *ResvErr* messages for the previously accepted end-to-end reservations which would be in excess with the new bandwidth value. While this would not be hard to implement, some policy would need to be defined to select the excess end-to-end flows to remove, which falls out of the scope of this performance evaluation.

## 3.3 Performance Analysis

In this section we evaluate the performance of the RSVP Reservation Aggregation architecture based on results from several different sets of simulations. The obtained results are compared against those of simulations performed using the standard RSVP/IntServ architecture with the same (dumbbell) topology. We analyze the standard QoS parameters (delay, jitter and packet loss ratio), the network resource utilization at the core link, and the reservation setup time. The number of signaling packets processed at the core is also analyzed in order to ascertain the scalability of the architecture and the improvement over standard RSVP.

Although admission control for aggregates must be parameter-based (PBAC), admission control for flows inside aggregates may be either parameter- or measurement-based (MBAC)[2]. We performed simulations with PBAC and MBAC. In the RSVP/IntServ simulations we used PBAC.

Figure 3.6 shows the topology used in these simulations. It consists of a transit (core) domain, TD, and 6 access domains, AD1–AD6. Each terminal in the access domains simulates a set of terminals. The bandwidth of the links in the transit domain and the interconnections between the transit and the access domains is 10 Mbps. The propagation delay is 2 ms in the transit domain and 1 ms in the interconnections between domains. There are up to 9 different aggregates in the link between C1 and C2, since there are 3 edge routers connected to C1 and other 3 connected to C2. The bandwidth assigned to the Controlled Load (CL) class is 7 Mbps. The bandwidth assigned to signaling traffic is 1 Mbps. Notice that, although this seems very high, it is only an upper limit. The unused remaining 2 Mbps, as

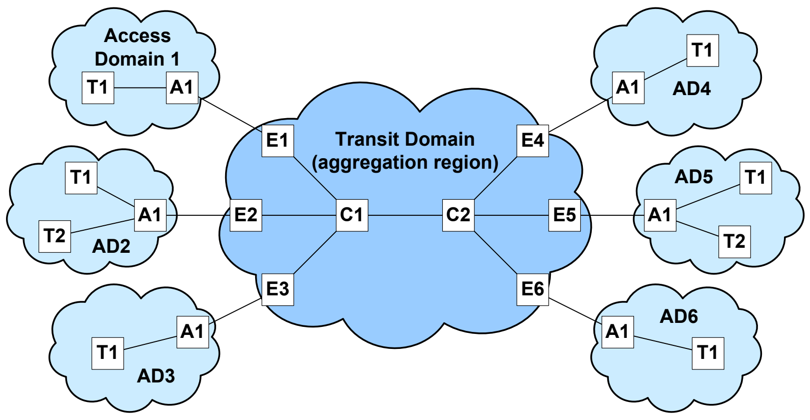


**Figure 3.6: Simulation topology for the evaluation of RSVP Reservation Aggregation**

[2] The admission control mechanisms are described Appendix A.

well as the unused bandwidth from the CL and signaling classes, is used for best-effort (BE) traffic. The GS class was not used in these simulations due to the model's limitations in GS support explained above in section 3.2.3.

The RSVPRAgg implementation has some tunable parameters. Except where otherwise noted, we used a *bulk* size of 500 kbps and a hysteresis time (unused *bulk* removal delay) of 15 s.

Each terminal of the access domains on the left side generates a set of flows belonging to the CL class, as well as filler traffic for the BE class. Each source may generate traffic to all destinations (terminals on the access domains of the right side), and the destination of each flow is randomly chosen. Filler BE traffic is composed of Pareto on-off and FTP flows. The traffic in the CL class is composed of a mixture of different types of flows, both synthetic — Constant Bit-Rate (CBR) and Exponential on-off (Exp.) — and real world multimedia streams — packet traces from H.263 videos, available from [VidTraces]. We used several different video traces for each bit-rate, starting each flow from a random point in the trace in order to avoid unwanted correlations between flows. The characteristics of the set of flows used are summarized in table 3.1. These flows are initiated according to a Poisson process with a certain mean time between calls (MTBC), and each flow has a duration which is distributed exponentially (synthetic flows) or according to a Pareto distribution (video traces), with the average value shown in the table (Avg. dur.). BE flows are active for all the duration of the simulations.

| Type | Avg. rate (kbps) | Peak rate (kbps) | On time (ms) | Off time (ms) | Resv. rate (kbps) | Resv. burst (bytes) | MTBC (s) | Avg. dur. (s) | MOL (kbps) | ROL (kbps) |
|---|---|---|---|---|---|---|---|---|---|---|
| CBR | 48 | - | - | - | 48 | 1500 | 13.8 | 120 | 1670 | 1670 |
| Exp. | 48 | 96 | 200 | 200 | 64 | 2500 | 6.8 | 120 | 3388 | 4518 |
| Video | 16 | - | - | - | 17 | 4000 | 17.1 | 180 | 674 | 716 |
| Video | 64 | - | - | - | 68 | 8000 | 17.1 | 180 | 2695 | 2863 |

**Table 3.1: Flow characteristics**

The largest mean offered load (MOL) in the CL class is, in terms of average traffic rates, about 20% higher than the bandwidth allocated to that class, which translates in an excess of about 40% in terms of reserved rates (ROL — Reserved Offered Load).

All simulations presented in this chapter are run for 5400 simulation seconds, and data for the first 1800 seconds is discarded. All values presented are an average of, at least, 5 simulation runs with different random seeds. The next sub-sections discuss the results of these experiments.

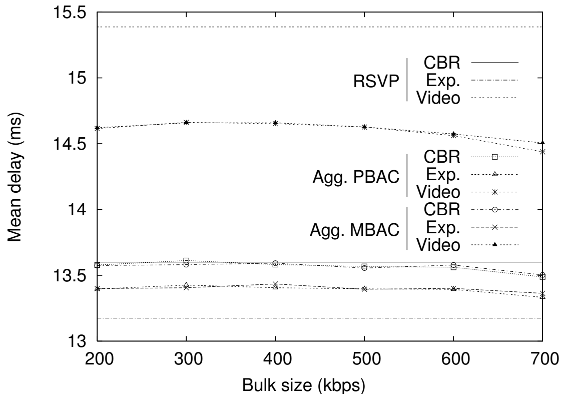
### 3.3.1 Variable *Bulk* Size

An important parameter in the RSVPRAgg architecture is the *bulk* size, which has implications on both the network resource usage and the signaling scalability. In the first experiment we vary the *bulk* size from 200 kbps to 700 kbps and use the maximum offered load (corresponding to the values presented in table 3.1). The results from this experiment are presented in fig. 3.7, using either PBAC or MBAC in the CL class. Reference values obtained with standard RSVP/IntServ are also provided.

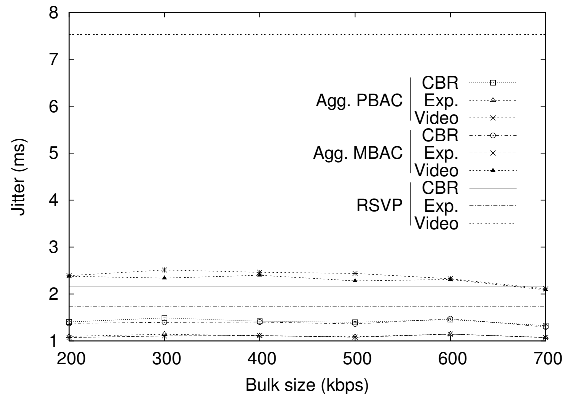
As we may see from fig. 3.7.a the mean delay does not vary much with the *bulk* size. It is about the same as in RSVP for CBR flows, slightly higher for Exponential flows, and somewhat lower for the video streams. Jitter (fig. 3.7.b) is always lower in RSVPRAgg, particularly in the case of video streams. This indicates that, indeed, the upper tail of the delay distribution is reduced by aggregating flows. Packet losses (fig. 3.7.c) for video streams are slightly higher in RSVPRAgg than in standard RSVP. Both in RSVPRAgg and in RSVP there are no packet losses in CBR flows. Contrary to RSVP, there is a small amount of loss (<0.005%) in exponential flows in RSVPRAgg. This amount of loss is, however, acceptable in a controlled load class. The admission control method for flows in aggregates does not seem to have a significant impact on the QoS parameters.

Regarding the utilization of the CL class (fig. 3.7.d), we may see that it is noticeably lower in RSVPRAgg than in standard RSVP. This is due to the fact that sometimes bandwidth is needed in an aggregate when it is not available, though there is spare bandwidth in other aggregates. As expected, utilization is even lower when using PBAC than when using MBAC since fewer flows are admitted in each aggregate. It is interesting to notice that there are local maxima in network resource utilization for *bulk* sizes of 500 kbps and 700 kbps, which are submultiples of the bandwidth available for the CL class (7 Mbps). This shows that it is good practice to choose a *bulk* size that is submultiple of the bandwidth allocated to the service class.

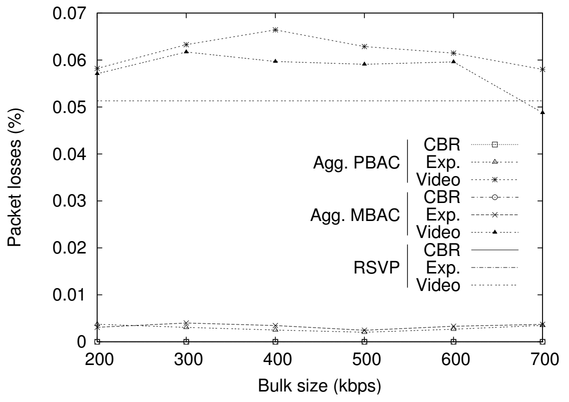
An important parameter in the evaluation of the signaling scalability is the number of signaling packets processed at core nodes. Figure 3.7.e shows the number of signaling packets processed at node C1. As may be seen, the number of messages processed at the core is reduced more than tenfold from RSVP to RSVPRAgg (from about 23000 to about 1800). This represents, indeed, a very significant increase in signaling scalability. Though not easily seen in the figure, there are local minima in the number of messages processed at the core with *bulk* sizes of 500 kbps and 700 kbps, which is another reason to choose a submultiple of the assigned bandwidth as the *bulk* size.



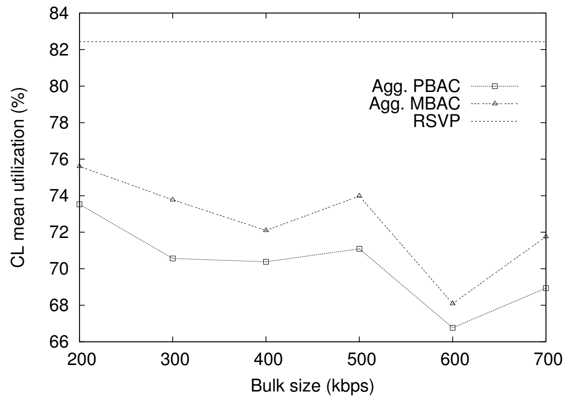
a) Mean delay



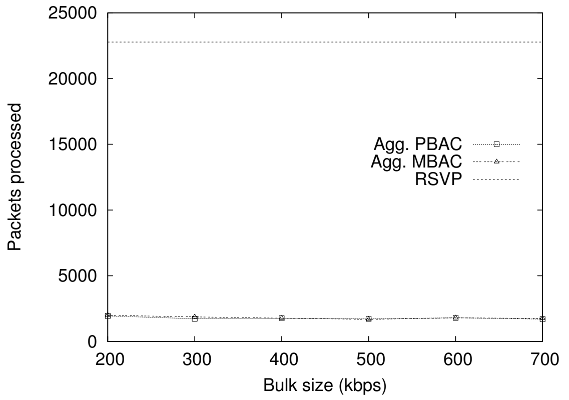
b) Jitter



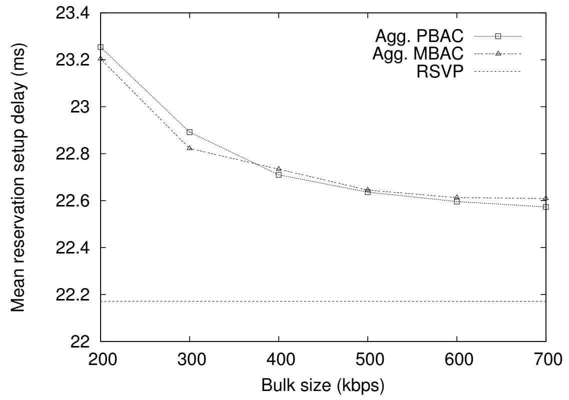
c) Packet losses



d) CL class utilization



e) Processed signaling packets at core node 1



f) Reservation setup delay

**Figure 3.7: Simulation results with variable bulk size**

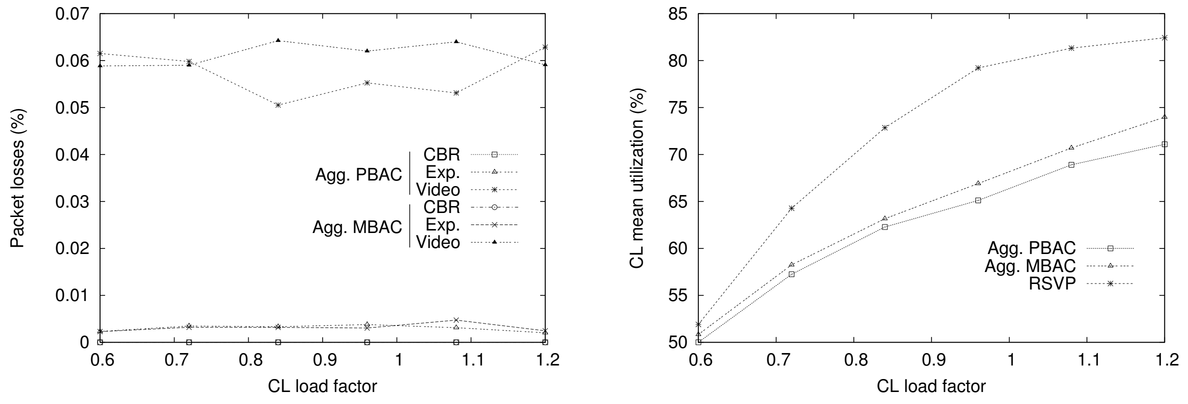
Figure 3.7.f shows the reservation setup delay. It is very important to notice that the curves relate only to the delays imposed by signaling message exchange and do not include processing time, since the ns-2 simulator is not suitable for the measurement of processing delays. The reservation setup delay decreases with increasing values of the *bulk* size. This behavior is expected since with larger *bulk* sizes more reservations are accepted without the need for increasing the aggregate bandwidth, which requires additional signaling. In the simplest case (appendix 2 in [RFC3175]), it basically consists on a round-trip time, the same as standard RSVP. In the more complex cases (appendices 1 and 3 in [RFC3175]), one or two round-trip times for the aggregation region are added. With the inclusion of processing times, the setup delay would be much lower for RSVPRAgg than for the scalability-impaired RSVP.

The results presented above indicate that the RSVPRAgg architecture is able to meet the QoS requirements of a controlled load class, being able to replace the standard RSVP/IntServ architecture with substantial gains in scalability. The drawback is a lower usage of network resources.

## 3.3.2 Variable Offered Load

In the second experiment we evaluate the behavior of the RSVPRAgg architecture with varying offered load. The flows are the ones shown in table 3.1, but the mean time between calls (MTBC) is adjusted to vary the offered load from 60% (load factor of 0.6) to 120% (load factor of 1.2) of the bandwidth assigned to the CL class. The MTBC values presented in the table correspond to a load factor of 1.2. Figure 3.8 shows some results from this experiment.

All QoS parameters are essentially constant, not depending on the offered load factor. Admission and traffic control are, therefore, being effective in emulating the behavior of a lightly loaded best-effort network, characteristic of the controlled load class. Regarding CL class utilization, for low values of offered load it is almost the same in RSVPRAgg and in standard RSVP, but it grows much faster in RSVP as the load factor approaches 1. At this point, the utilization curve for RSVP saturates, while those of RSVPRAgg continue to grow, exhibiting no visible saturation. The utilization with MBAC is slightly higher than with PBAC, since more flows are accepted. The largest difference in utilization between RSVP and



a) Packet losses b) CL class utilization

**Figure 3.8: Simulation results with variable offered load**

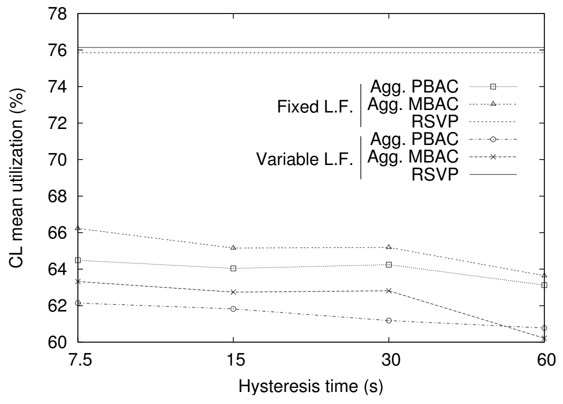
RSVPRAgg is about 15% of the bandwidth allocated to the CL class (about 1 Mbps difference).

### 3.3.3 Variable Hysteresis Time

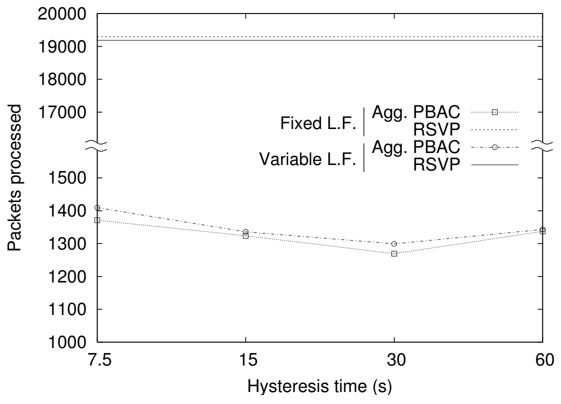
With this experiment we evaluate the influence of the hysteresis time in the utilization of the CL class and in the number of signaling packets processed at the core. Hysteresis is needed in order to avoid oscillation in the reserved rate of an aggregate when operating in stable conditions, with the sum of reservations for the aggregate around a multiple of the *bulk* size. In these simulations, only one terminal in each access domain is transmitting. The offered load is 90% of the bandwidth allocated to the class in terms of traffic and 105% in terms of reserved rates. We performed two different sets of simulations, one using the same average amount of offered load in all transmitting terminals at all times (Fixed LF — Load Factor), and another one affecting the offered load in each terminal by a multiplicative factor of 0.5, 1 or 1.5[3] (Variable LF), keeping the same total offered load. These factors are rotated between the transmitting terminals every 400 simulation seconds. This rotation has the effect of forcing bandwidth to be released from some aggregates and requested in different ones.

Figure 3.9 shows the results with the variation of the hysteresis time (i.e., the delay for the removal of an unused *bulk*) from 7.5 s to 60 s. As expected, the utilization decreases when the hysteresis time increases. This is due to the fact that unused bandwidth *bulks* are being held in aggregates for longer periods of time before being released and made available to other aggregates which may need it. The largest difference in utilization is obtained in the variable (rotating) load factor simulations when using PBAC; in this case, the difference is larger than 3% of the bandwidth allocated to the class. The number of signaling packets processed at the core also depends on the hysteresis time, although the variation is not very large, particularly if compared with the gains of using RSVPRAgg instead of the standard RSVP. It is interesting to notice that there is a minimum in the number of packets processed for a hysteresis time of 30 s. This behavior is due to the prevalence of one of two factors. For low values of hysteresis time, increasing this value means an increased probability that a flow will be admitted into the aggregate without need for increasing its bandwidth, since spare bandwidth is being held for longer periods. For higher values of hysteresis time another factor becomes dominant: the higher number of failed attempts to increase the bandwidth in some aggregates while spare bandwidth is being held in others. Although there is a minimum hold

[3] The total offered load, therefore, remains the same.



a) CL class utilization



b) Processed signaling packets at core node 1

**Figure 3.9: Simulation results with variable hysteresis time**

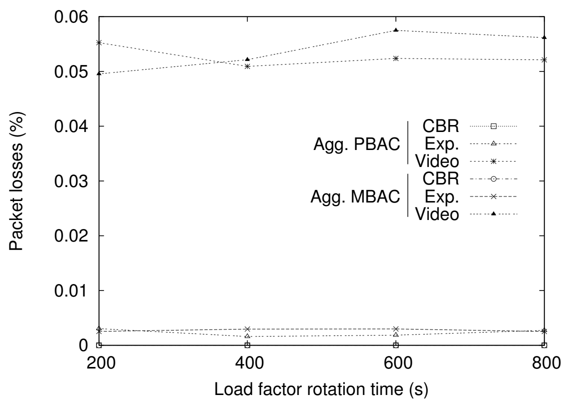
period between attempts to increase an aggregate's bandwidth, it was fixed at 5 s in these simulations. In face of these results, large values of hysteresis time are not recommended.

We performed a similar experiment keeping the hysteresis time constant at 15 s and varying the offered load rotation time between 200 s and 800 s. The results show that this variation does not noticeably affect the CL class utilization.

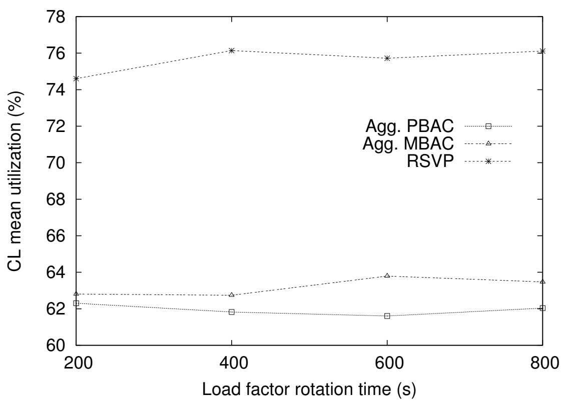
## 3.3.4 Variable Offered Load Rotation Time

The last experiment is meant to evaluate the effect of changing conditions in the offered load from different access domains. Similarly to the previous experiment, we use 3 transmitting terminals, one from each access domain, affected by rotating load factors of 0.5, 1 and 1.5; however, this time we vary the delay between rotations of the load factors. The global offered load is the same as in the previous experiment.

Figure 3.10 shows the packet losses and the CL class utilization when the rotation



a) Packet losses



b) CL class utilization

**Figure 3.10: Simulation results with variable offered load rotation time**

delay varies between 200 s and 800 s; both parameters are essentially unaffected by this variation. This is probably due to the fact that the hysteresis time (15 s) is much shorter than the minimum rotation delay. On the other hand, shorter rotation delays would not be effective due to the average duration of the flows. This means that by using a hysteresis time which is small when compared to the average duration of the flows, the model is made insensitive to variations in the distribution of the offered load coming from different domains.

## 3.4 Conclusions

In this chapter we performed an evaluation of the RSVP Reservation Aggregation architecture, proposed by the IETF as a scalable alternative to the standard RSVP/IntServ architecture for use in high-speed core domains. We gave an overview of the architecture, pointing out its main strengths and weaknesses. We described our implementation of RSVPRAgg in the ns-2 simulator and discussed some particularities of the implementation and limitations of the architecture and its definition. Policies which are considered out of the scope of [RFC3175] were defined, namely the aggregate bandwidth management policy. The tunable parameters of our implementation were also presented.

The simulation results indicated that the RSVPRAgg architecture is able to meet the QoS requirements of the controlled load IntServ class. This is achieved with much lighter classification, forwarding and signaling procedures than those of RSVP/IntServ. A comparison of the number of signaling packets processed at the core in RSVPRAgg and standard RSVP/IntServ shows that signaling is much lighter and more scalable in the former. Combined with the fact that packet classification and scheduling are performed per class (therefore independent not only of the number of simultaneous flows, but also of the number of aggregates), this makes RSVPRAgg appropriate for deployment in high-speed core networks. The drawback, as demonstrated, is a lower utilization of network resources. Based on the analysis of the simulation results, we also provide some guidelines for setting the tunable parameters, namely the *bulk* size and the hysteresis time.

# CHAPTER 4

# SCALABLE RESERVATION-BASED QOS

We have seen in the previous chapter that while the RSVPRAgg model is a scalable alternative to RSVP/IntServ, it suffers from network underutilization stemming from the aggregation of state information and bulk updates to that information. If it were possible to combine the scalability properties of the DiffServ model for packet classification, policing, shaping and scheduling underneath RSVPRAgg, with a scalable model of per-flow reservation signaling, we would be able to simultaneously benefit from the scalability properties and avoid the underutilization limitation.

Despite the widespread belief that per-flow signaling and maintenance of state is not scalable to high bandwidth core routers, no evidence has ever been produced that this statement is universally true. This belief has its roots in the lack of scalability of the RSVP/IntServ model, but does the same lack of scalability necessarily apply to any conceivable application of per-flow state? We have strong reasons to believe it is possible to develop a scalable per-flow signaling approach. Signaling traffic represents only a very small fraction of the data traffic, and their proportion is similar at the access and the core. Storage of per-flow information requires routers with sufficient memory, but this requirement is already satisfied in today's routers. Consider, for example, a router handling a maximum of a hundred thousand simultaneous flows. If 100 bytes are required to store the state of each flow, the total amount of memory necessary for storing per-flow information is 10 Mbytes, an amount that is small by today's standards, and expected to be even more so in the future due to "Moore's Law." The Achilles' heel of per-flow state maintenance is the burden of processing the signaling messages, which is intimately tied to the algorithmic complexity of the tasks such processing involves. If, however, the computational complexity of every task associated with processing of signaling messages could be made independent on the number of simultaneous flows, then the overall processing power required for per-flow reservation signaling would grow no more than linearly with this number, making it scalable. Based on these principles, we developed the Scalable Reservation-Based QoS (SRBQ) architecture, combining aggregate packet processing (classification, scheduling, policing and shaping) with a scalable model for per-flow, signaling-based resource reservation.

In this chapter we describe the SRBQ architecture, detailing its building blocks, evaluate the architecture regarding different aspects, like QoS and scalability, and perform a comparative analysis of SRBQ against RSVPRAgg, discussed in the previous chapter. Different parts of this work were published in [Prior03a, Prior03b, Prior03c, Prior04a, Prior04b, Prior04d, Prior07a]. We begin with a high level overview of SRBQ in section 4.1. Section 4.2 details the different aspects of the architecture: the packet scheduling, classification, shaping and policing mechanisms (4.2.1), the label mechanism (4.2.2), the signaling protocol for performing resource reservations (4.2.3), the (4.2.4) packet processing sequence, and the expiration timers for soft-state reservations (4.2.5). Section 4.3 presents an extensive evaluation of SRBQ, using both synthetic and real multimedia flows, analyzing its performance QoS- and scalability-wise. Section 4.4 compares the SRBQ and RSVPRAgg architectures. Section 4.5 closes the chapter with a summary of its main conclusions.

## 4.1 SRBQ Architecture Overview

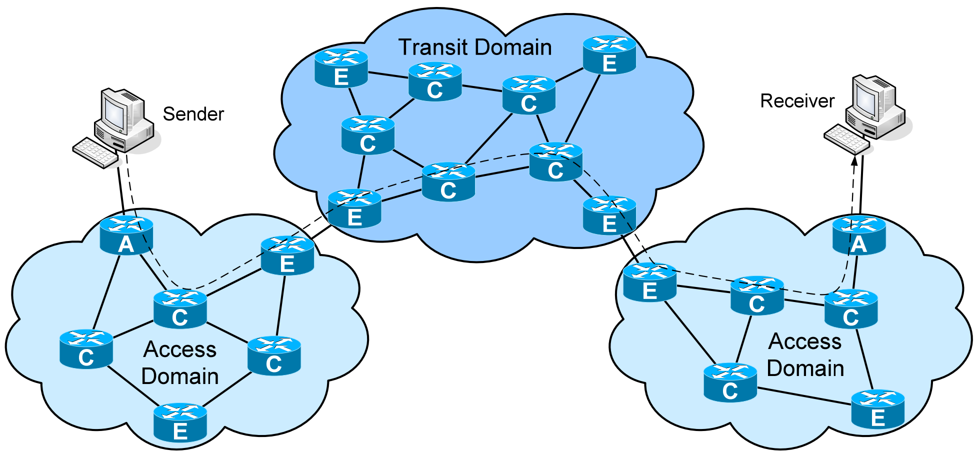


**Figure 4.1: Architecture overview**

The SRBQ architecture (fig. 4.1) combines the strict end-to-end QoS guarantees of a signaling-based approach with per-flow reservations subject to admission control (both in terms of bounded delay and minimal loss) with the efficiency and scalability provided by flow aggregation and by several mechanisms and algorithms we developed (described later in this chapter).

The underlying architecture for our model is strongly based on the DiffServ architecture [RFC2475]. In fact, not only is the underlying infrastructure very DiffServ-like, but the two models may peacefully coexist in the same domain. Resource allocation for the two models can be administratively configured. On top of the DiffServ-like infrastructure, we added signaling-based reservations subject to admission control.

The network is partitioned in domains, consisting of core (C) and edge (E) nodes. Access domains have also access routers (A), to which individual terminals are connected. Individual flows are aggregated according to the service class. Service classes are mapped to DiffServ-compatible PHBs (Per-Hop Behaviors), and aggregate classification is performed based on the DS (Differentiated Services) Field of the packet header [RFC2474]. Edge nodes perform policing of the incoming aggregates in order to ensure conformance to the aggregate reservation (the sum of the individual reservations of all flows in the aggregate). Since traffic is policed in all edge nodes, core nodes do not have policing functionalities.

The reservations are unidirectional, sender-initiated and soft state. There are several advantages to this approach: (1) it is more directly mapped to existing business models, in which a provider sells a service (e.g., streaming) at a price which includes not only the content but also the delivery with the appropriate quality; (2) the sender has more information on the flow characteristics and, therefore, is able to parameterize the reservation in a way that assures appropriate quality without waste; (3) unidirectional reservations inherently support path asymmetry. Reservations are setup by means of a signaling protocol.

Every node along the end-to-end path must perform admission control for every flow and must, therefore, run the signaling protocol. This protocol was implemented as an extension to RSVP [RFC2205], providing the functionality needed for our model. Although signaling is performed per-flow, several techniques and algorithms have been developed aiming at the minimization of the computational complexity and, therefore, the improvement of signaling scalability. More specifically, a label mechanism was developed with the goal of reducing the signaling message processing time at each router. This mechanism provides each router with direct access to the flow reservation structures, using a label in the signaling messages that is directly mapped to the memory address where the reservation state for the flow is stored. The processing load of the routers in the end-to-end path is further decreased by the use of an algorithm we developed for the efficient implementation of expiration timers used in the soft reservations. This algorithm has low computational complexity compared to currently available algorithms, and provides enough functionality.

## 4.2 SRBQ Architecture Details

This section describes SRBQ in detail, tackling the different aspects and components of the architecture.

### 4.2.1 Service Differentiation and Traffic Control

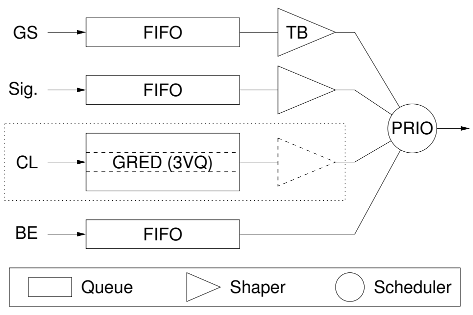
We begin by presenting the service classes available in our model, their mapping within the DiffServ architecture, and the resource reservation and control techniques applied to the traffic flows, such as the admission control, traffic shaping and policing.

#### 4.2.1.1 Service Classes and Mapping Strategies

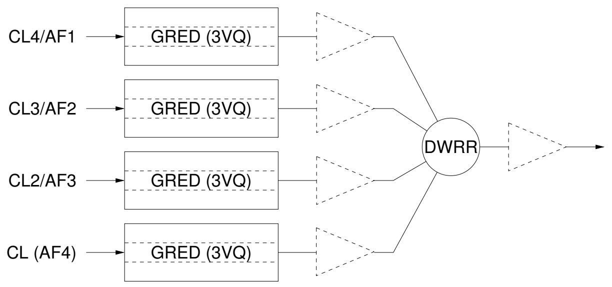
Besides best effort, our model provides two additional service classes: the GS (Guaranteed Service) class that is characterized by hard QoS assurance in terms of both delivery guarantee and maximum delay; and the CL (Controlled Load) classes that emulate the behavior of lightly loaded best-effort networks. Reservations for traffic flows using the GS class are characterized by a token bucket. Reservations for traffic flows using CL classes are characterized by three average rate watermarks: packets exceeding the first two watermarks will receive a degraded service in terms of drop probability; packets exceeding the third watermark will be dropped. The GS class is especially appropriate for delay- and loss-intolerant flows having a well-defined traffic envelope, while the CL class is more appropriate for flows with looser QoS requirements, flows for which a traffic envelope is not easily derived, and elastic TCP flows. The admission control algorithms for the GS and CL classes are different, as will be seen in subsection 4.2.1.2.

As previously mentioned, the SRBQ model is based on DiffServ. The service class is conveyed in the Differentiated Services Code Point (DSCP) field of the packet header. Packet classification at the routers is performed based on both the DSCP field and the input interface at which the packet arrived. The GS class is based on the same principles as the Expedited Forwarding (EF) PHB in DiffServ: low delay and very high delivery assurance, provided by a service rate which must be greater than or equal to the maximum arrival rate. Similarly, the CL class is based on the Assured Forwarding (AF) PHB, providing a better than best effort service with three different drop precedence levels. Packets from a flow using the controlled load service are marked for one of the drop precedence levels according to rate levels established by the signaling protocol. It is worth noting that DSCP-based packet classification can be implemented with perfect hashing (with a mere 64 hash buckets) and is, therefore, extremely efficient, having worst-case O(1) complexity.

The simplest queuing model for the routers is depicted in fig. 4.2.a. The main scheduler is priority-based, with at least four different queues. The highest priority queue is used for the highly delay sensitive traffic of the GS class. The service discipline inside this class is classical First In, First Out (FIFO). With a priority immediately below that of the GS there is a queue for other critical traffic with lower delay sensitivity, such as signaling and routing protocols (Sig.), internally served in a FIFO fashion as well. Obviously, this class is not subject to admission control. With lower priority, there is a queue for the CL traffic class, containing three virtual queues (VQ), one for each drop precedence level. The internal service discipline may be any one of the Generalized Random Early Detection (GRED) algorithms [Almesberger99], e.g., RED with In/Out bit (RIO) [Clark98]. In the case of multiple (independent) CL classes, the CL queuing block is replaced by the one shown in fig. 4.2.b. A Deficit Weighted Round Robin (DWRR) discipline [Shreedhar95] is used to dequeue from individual CL / DiffServ AF classes. At the lowest priority there is a queue for best effort traffic served in a FIFO, RED, or other appropriate service discipline.



a) Single CL class



b) Multiple CL classes

**Figure 4.2: Queuing model**

All of the algorithms used for packet classification and scheduling in SRBQ have a computational complexity that is independent on the number of simultaneous flows, meaning that the packet classification, queuing and scheduling model of SRBQ scales very well to a huge number of simultaneous flows.

The recommended DSCP for the GS class is binary 101100. This value is different from the one used in the EF class, 101110, enabling coexistence of our model with DiffServ by protecting GS flows — subject to admission control, policing and shaping — from the EF traffic. The recommended DSCPs for the CL service class are binary 100010, 100100 and 100110 which are, in fact, the DSCPs of the DiffServ class AF4 (therefore, when both our model and the DiffServ model work together, the latter must not use the AF4 PHB). Our model may also contain different CL service classes using DSCPs from other AF classes, provided these are not used by DiffServ.

#### 4.2.1.2 Resource Reservation and Control

Admission control is performed at every node along the flow's path. As we have already mentioned, the admission control algorithms are different for the GS and the CL Classes. In the GS class, parameter-based admission control must be performed in order to strictly guarantee that enough bandwidth and buffer space are available at every router, even in the worst case. QoS requirements are much looser in the CL class, allowing for the use of measurement-based admission control in order to improve the utilization of network resources (though parameter-based algorithms may likewise be used).

A GS flow $i$ is characterized by a token bucket $(r_i, b_i)$. In order to perform admission control for a new GS flow, the router computes (incrementally) the summed token bucket for all admitted flows sharing the same service class and output interface, using the formulas $R_{sum} = \sum_i r_i$ and $B_{sum} = \sum_i b_i$, where $r_i$ and $b_i$ are, respectively, the token bucket rate and depth of the individual flows. A new flow $j$ is accepted iff $R_{sum} + r_j \leq R_{GS\max}$ and $B_{sum} + b_j \leq B_{GS\max}$, where $R_{GS\max}$ and $B_{GS\max}$ are the configured limits of GS rate and bucket depth for the output interface.

CL flows are characterized by three average rate watermarks, two for degrading service and a third one for packet dropping. The three watermarks $R_{sum,w}$ that characterize the aggregate are obtained by summing the corresponding watermarks $r_{i,w}$ from the individual flows, $R_{sum,w} = \sum_i r_{i,w}$, and a new flow $j$ is admitted iff $R_{sum,w} + r_{j,w} \leq R_{CL\max,w}, \forall w$, where $R_{CL\max,w}$ is the configured maximum rate for the $w^{\text{th}}$ CL watermark for the output interface. Since some amount of packet loss is tolerated, admission control is not as critical as in the case of GS. This allows us to somewhat improve network resource utilization, either by using Measurement-Based Admission Control (MBAC) algorithms, replacing $R_{sum,w}$ with an estimate, or by using Parameter-Based Admission Control (PBAC) with overbooking, multiplying the configured maximum rates by an overbooking factor $\beta$. Please refer to Appendix A for more information on PBAC and MBAC algorithms.

#### *4.2.1.2.1 Traffic Shaping*

As can be seen in fig. 4.2.a, the highest priority queues, corresponding to the GS traffic class and signaling/routing traffic, must be subject to a traffic shaper. The traffic shaper for the GS class is of token bucket type, and is parameterized using the summed values $R_{sum}$ and $B_{sum}$. This shaper is necessary in order to avoid packet discarding by the policer at the input interface of the following node (or, more precisely, at the following edge node, since no policing is performed at the core), and also helps in reducing multiplexing jitter. Since the traffic is shaped to the summed token bucket of the class, which is an upper bound to the actual traffic in the aggregate, this shaper does not degrade the quality of service guarantees of the GS flows because "greedy shapers come for free" [Boudec01].

A different kind of shaper is also used in our model. Not all terminal nodes are able to perform precise token bucket scheduling for GS flows, and even if they do, jitter is inevitably introduced by multiplexing several GS flows, or a GS flow and traffic from other classes. Since this jitter could lead to packets being dropped by the policing module at the access router even for applications generating conformant traffic, GS flows are ingress-shaped, instead of policed, at the access router. This shaping is performed on a per-flow basis. Since all GS flows are shaped into conformance, the GS aggregate is also guaranteed to be conformant to the aggregate reservation.

The signaling/routing traffic, though not subject to admission control, must also be shaped in order to prevent starvation of the bandwidth assigned to the CL class. Since this traffic will not be subject to policing at the input interface of the following node, the shaper may be simplified to a work-conserving rate limiter (i.e., rate is only limited when lower priority classes have queued packets waiting for transmission).

The CL class may also be shaped; however this is only needed if the network administrator wants to make sure that some amount of bandwidth always remains for best effort traffic. Since traffic from the CL class tolerates some amount of packet dropping, the shaper may be a simple work-conserving rate limiter.

#### *4.2.1.2.2 Traffic Policing*

There are four different types of nodes in our model: end nodes, access routers (A), edge routers (E) and core routers (C). All of them perform signaling and support the queuing model described above, in subsection 4.2.1.1. The access nodes perform per-flow policing, ensuring that traffic from each individual flow does not exceed its reservation parameters. As previously stated, GS flows are shaped at these nodes, instead of policed. Edge nodes perform aggregate policing only, with an aggregate consisting on a DSCP value (or a set of DSCP values corresponding to a PHB group) per input interface. There is no need for edge routers to perform per-flow policing, since this has already been done at the access routers. Edge nodes may also perform re-marking to a higher drop probability within the CL class. Since the re-marking is to be performed at aggregate level, it must be *color-aware*[1] in order to avoid unfair service degradation to "well behaved" flows. Core routers perform no policing, as they assume that all traffic has already been policed in their domain, either by an edge router (for traffic coming from different domains) or by an access router (for eventual traffic generated inside their own domain) and is, therefore, conformant to the reservations.

Contrary to traffic shaping (performed at the output interface) which is always performed on traffic class aggregates, policing (at the input interface) is performed in a per-flow basis at the access routers. This is required for ensuring that flows are conformant to the reservations, and also to perform drop precedence (re-)marking for CL flows. Forcing every flow to conform to its reservation ensures that no service degradation is inflicted by a greedy flow on the remaining flows sharing the same service class and output interface.

In order to avoid packet dropping in the GS class even in presence of some clock drift between routers (or timer imprecision at the previous router), the bucket of the aggregate GS policer may be increased by a small amount (e.g., an MTU).

### 4.2.2 Label Mechanism

Usually, the scalability limitation is imposed neither by the signaling traffic nor by the maintenance of the flow information — signaling traffic represents only a very small fraction of the data traffic, and the amount of memory required for storage of per-flow information is available on today's routers. The scalability limitation stems from the computational complexity of the algorithms used in processing the signaling messages.

One potentially scalability-limiting task for the core routers, particularly in terms of worst case, is the lookup of the stored flow information, based on the 5-tuple parameters that specify the flow. Usually, this lookup is implemented using hash tables, but perfect hashing for the 5-tuple is infeasible. In order to efficiently access the reservation structures we developed a label mechanism which allows direct access to these structures without any need for hash lookups. These labels are 32-bit values whose meaning is externally opaque, but

[1] *Color-aware* re-marking means that a packet may be demoted to a higher drop probability, but not promoted to a lower one (the re-marker is aware of the packet's previous "color").

internally are the memory address of the reservation structure (most routers will use 32-bit processors; in routers with more than 32-bits of addressing space, the label may be an offset).

Once a signaling message is received, its LABEL object will be used to directly access the reservation flow state. This procedure significantly reduces the message processing time at the router and the signaling complexity. As will be shown, only the initial messages sent to perform resource reservation are excluded from this efficient access, but since the reservation does not yet exist, they would not need it anyway. Allocating a new structure is trivial, consisting in taking the first element from a linked list. The installation of the labels in the flow reservation structures will be detailed in subsection 4.2.3.2.

The label mechanism is also significantly advantageous for all per-flow processing. For example, per-flow policing performed at the access routers needs to access the flow state each time a packet is received. Although the number of flows at the access network is much smaller than that of core networks, the use of a mechanism that provides direct access to the flow information allows for a significant reduction in its processing load. Finally, the route change detection (in case of rerouting) can also make use of the labels. More specifically, the router compares the next hop address obtained from the routing table with the next hop address stored in the reservation structure of the flow. A mismatch between these values indicates that the route has changed, and the router will react accordingly.

In order to benefit from these advantages on per-flow processing and fast and efficient route change detection, however, data packets need to carry the label information in the header. Notice that in the GS class it is strictly required that no packet is admitted at a router unless there is a valid reservation in place for the corresponding flow. CL classes are more tolerant to packet losses, and therefore, do not require the label in the packet header.

Whenever a GS packet arrives at a node, it is necessary to check the reservation structure before accepting it. Checking the reservation implies the existence of the label information in the GS packet headers in order to be performed in an efficient way. In IPv4 we could use the *Identification* and *Fragment Offset* fields of the packet header for this purpose, since these fields are only used when fragmentation occurs (although it was mentioned in [Stoica00] that only 0.13% of the packets in the Internet are fragmented). Since it is possible to avoid fragmentation by setting the *Don't Fragment* flag, these fields can be used without affecting the system operation. The GS service class excludes fragmentation anyway, since the additional headers would increase the effective rate of the flow, eventually leading to packet losses (the RSVP/IntServ Controlled Load and Guaranteed services also preclude fragmentation due to the inaccessibility of port information to build the 5-tuple flow identifier

[RFC2211, RFC2212]). Since the sum of the available bits is only 29 (less than the 32 bits required for the Label object), the three last bits of the labels must be zero, implying that all memory structure addresses are aligned to 8-byte (i.e., 64-bit) boundaries. As an alternative, the label could be conveyed by means of an IPv4 header option, though this would increase the overhead.

In IPv6 we could only make use of the *Flow Label* header field to carry the labels. However, the length of this field is just 20 bits, which is not enough to carry a memory address. In order to avoid the overhead of using an IPv6 extension header to convey the 32-bit label, an alternative labeling scheme is used — in this case, the label is used as an index to a table of reservations. Access to the reservation structure is still O(1), as its memory address is given by the simple formula $ResvAddr = TableAddr + Index \times StructSize$. This use of the *Flow Label* field, however, is not compliant to [RFC3697], which states that a *Flow Label* value is valid only within the context of the 3-tuple (*Source Address*, *Destination Address*, *Flow Label*) and that it must be delivered unchanged to the destination node.

It is worth noting that, in spite of all the advantages, the labels are not used for packet classification (except perhaps at the access routers), since it is performed on an aggregate basis, using just the DS field of the IP header.

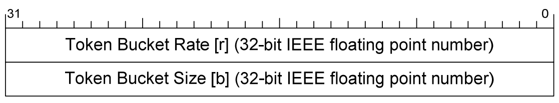
## 4.2.3 Signaling Protocol

As we have already mentioned, SRBQ signaling works on a hop-by-hop basis, providing unidirectional, soft state, sender-initiated reservations. Instead of creating a whole new protocol from the ground up, we have chosen to implement it as an extension to the RSVP protocol for several reasons: (1) RSVP is a widely known and supported protocol; (2) it also works on a hop-by-hop basis; (3) it also uses routes established by an underlying routing protocol; and (4) RSVP already implements some of the required functionality for our model. Although our signaling protocol is an extension of RSVP, it is much more scalable, since (1) the access to the flow information is simple with O(1) complexity, (2) expiration timers for the soft reservations are implemented in a very effective, low complexity way, and (3) it uses simple identification of the reservations in order to decrease the length of the refresh and explicit tear down messages. These topics will be described in more detail in the following subsections.

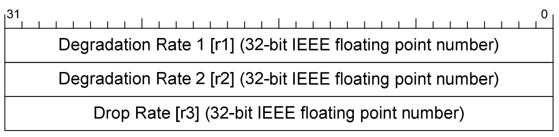
### *4.2.3.1 Signaling Messages*

Since RSVP is meant to perform receiver-initiated reservations, we had to extend it by adding three new message types: *SResv* (Sender Reservation), *SResvStat* (Sender Reservation

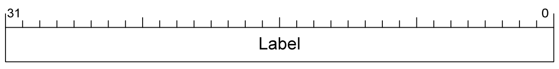
Status) and *SResvTear* (Sender Reservation Tear Down). The first one is used to establish new sender-initiated reservations and to modify and refresh existing ones; the second one is used to report status (success or error conditions); the last one is used to explicitly terminate an existing reservation.


31
0
Token Bucket Rate [r] (32-bit IEEE floating point number)
Token Bucket Size [b] (32-bit IEEE floating point number)

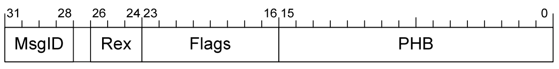
a) FLOWSPEC (GS)


31
0
Degradation Rate 1 [r1] (32-bit IEEE floating point number)
Degradation Rate 2 [r2] (32-bit IEEE floating point number)
Drop Rate [r3] (32-bit IEEE floating point number)

b) FLOWSPEC (CL)



c) LABEL / LABEL_SETUP



d) SRESV_PARMS

**Figure 4.3: Newly defined objects**

*SResv* messages are usually originated at the sender node, but may also be originated at any other node along the path for local repair purposes in error conditions, particularly in the case of route changes. Full *SResv* messages include an identification of the flow, consisting of both a SESSION and a FILTER_SPEC object, an identifier of the service class, a reservation timeout value and a FLOWSPEC class object parameterizing the reservation. The service class is specified by the PHB for that class, encoded as specified in section 2 of [RFC3140], and is included in the SRESV_PARMS object. The service class implies a particular type of FLOWSPEC class object (RSVP object C-Type). Two different FLOWSPEC types have been newly defined for this purpose, which are shown in fig. 4.3 (a and b). GS class reservations are parameterized by a token bucket, whereas CL class reservations are parameterized by three rate thresholds, two for drop precedence degradation and the third one for dropping. Though the rate parameters are specified in IEEE-754 floating point format, the nodes must perform all the calculations for aggregates using fixed point math, in order to avoid the accumulation of floating point rounding errors over time (eventually storing these fixed point values in the memory structure holding the reservation parameters). Optionally, the *SResv* message may contain an ADSPEC object containing the accumulated service degradation along the path, updated at every node.

```
<SResv Message (initial)> ::= <Common Header> [ <INTEGRITY> ] <SESSION>
                              <FILTER_SPEC> [ <LABEL> ]<LABEL_SETUP> <RSVP_HOP>
                              <SRESV_PARMS> <FLOWSPEC> [ <ADSPEC> ]
                              [ <POLICY_DATA> ]
```

The LABEL_SETUP and LABEL objects, whose format is shown in fig. 4.3.c, include a 32-bit value, the label, which is directly mapped to the memory address containing the flow reservation information. These objects will be used, respectively, to install the label at reservation setup time and to access the memory address when further messages are processed. The process of label installation is detailed in subsection 4.2.3.2.

The SRESV_PARMS object is shown in fig. 4.3.d. In addition to the PHB specifying the service class, it includes a 3-bit value (Rex) that represents the reservation expiration time. This value is in a 2-logarithmic scale and will be used with the developed algorithm to efficiently implement expiration timers. This algorithm will be detailed in subsection 4.2.5. It also includes a message identification number (MsgID) which allows for an association between the *SResv* message and an eventual *SResvStat* message in response to a *SResv* message.

*SResv* messages used for refreshing reservations without changing parameters (with the possible exception of reservation expiration timeout) are simpler than the full *SResv* messages used for the initial reservation setup. The SESSION and FILTER_SPEC objects are absent, as is the FLOWSPEC class object. The LABEL_SETUP object is also absent from refresh-only *SResv* messages.

```
<SResv Message (refresh)> ::= <Common Header> [ <INTEGRITY> ] <LABEL> <RSVP_HOP>
                              <SRESV_PARMS> [ <POLICY_DATA> ]
```

*SResv* messages may include a POLICY_DATA object that contains policy information applicable to the requested reservation. This feature may be used by the service providers to account and charge for an increased service quality and/or to enforce an acceptance policy for reservations. A POLICY_DATA object may be added to an *SResv* message, when entering the domain, by the ingress router. The information therein contained may be retrieved from an AAAC (Authentication, Authorization, Accounting and Charging) server. The policy information is used by routers inside the domain, which will perform admission control decisions based not only in available network resources, but also on the acceptance policy; it is removed at the egress router. The definition of the POLICY_DATA object format and processing may be domain-specific, and is out of the scope of this work. Basic security in signaling is achieved by means of INTEGRITY objects, as in the standard RSVP protocol.

*SResvStat* messages are used for both reservation confirmation and error reporting. The first kind is generated at the receiver node, triggered by the reception of an *SResv* message, and is propagated up to the sender. The second kind may be generated by any other node for reporting an error condition to the previous node which, depending on the error type, may be propagated up to the sender, or trigger a local repair procedure. The ERROR_SPEC object is used to report a successful reservation or to describe an error condition. *SResvStat* messages sent in response to an *SResv* message also include a copy of the SRESV_PARMS object from that *SResv* message.

```
<SResvStat> ::= <Common Header> [ <INTEGRITY> ] <LABEL> [ <LABEL_SETUP> ]
                <RSVP_HOP> [ <SRESV_PARMS> ] [ <ADSPEC> ] <ERROR_SPEC>
```

*SResvTear* messages are generated either by the sender node to explicitly terminate a reservation, or by any other node along the path which detects a route change, in order to remove the stale reservation from the old path. Among other objects, they include the LABEL object, so the nodes can directly access to the flow information in order to remove the reservation.

```
<SResvTear Message> ::= <Common Header> [ <INTEGRITY> ] <LABEL> <RSVP_HOP>
```

If there is no explicit teardown and no refresh messages, the timer associated to the specific flow expires. The flow reservation is then terminated, and the corresponding information is removed from the reservation table.

Since in SRBQ signaling must be performed at every node (including core routers), much care has been taken in reducing the processing load imposed by the signaling protocol, making use of computationally efficient algorithms and techniques only. The next subsections will address the details of the signaling protocol that increase its scalability.

#### 4.2.3.2 Signaling Dynamics

As we have already mentioned, although the base signaling protocol is RSVP, the proposed protocol is sender-initiated. This characteristic introduces a large difference between the proposed protocol and standard RSVP. Being sender-initiated means that upon receiving the first message (*SResv*), the nodes along the path run the admission control algorithm (based on the token bucket parameters or on network measurements), and perform resource reservation for the new flow. Notice that, in this model, performing resource reservation for a new flow is equivalent to increasing the bandwidth allocation for the class to which the new flow belongs, since our model is based on DiffServ.

When a router receives an initial *SResv* message requesting a new reservation and the request is accepted, the *SResv* message is forwarded to the following router and the updated traffic specification for the class is committed to the admission control module. This traffic specification, though, is not committed to the policing and queuing modules until the corresponding *SResvStat*, confirming a successful reservation up to the receiver, arrives. This ensures that whenever a router updates its resource allocation for a class, all routers towards the receiver terminal already have performed their updates. Otherwise, packet dropping could be inflicted on the class by the policing module of the following node in the time interval between resource allocation updates of the two routers. The *SResvStat* message will not only inform the sender about the success or failure of the admission request, but will also trigger the commitment of the resource reservation to both the policing and the queuing modules at the routers if the reservation succeeded, or remove it from the admission control module if it failed.

Each reservation structure stores, among other parameters, the flow specification and three label fields: *B*, *T* and *F*. These label fields are, respectively, the label to be used in signaling messages (or data packets) sent backwards (towards the sender), the label for the router itself (which may be implicit), and the label to be used in messages sent forwards (towards the receiver). The obvious exceptions are the end nodes: the sender has no *B* label and the receiver has no *F* label. Upon receiving a refresh *SResv* message, a node checks the label field in the message, directly accesses the state information of the flow, updates its expiration timer, and copies the *F* label field stored in the reservation structure to the label field of the *SResv* message to be forwarded. This label will be used by the next node to directly access the reservation state of the flow.

Labels are installed at reservation setup time, as shown in fig. 4.4, using LABEL_SETUP objects in the *SResv* and *SResvStat* messages.

The SRBQ signaling protocol works as follows (fig. 4.4). The initial *SResv* message, originated at the sender (1), contains, among other objects, the flow reservation specification and the label $T_S$. This label, conveyed by the LABEL_SETUP object, will be used in messages sent by the $R_1$ router to the sender. $R_1$ performs admission control, based on the flow reservation parameters. If the flow can be accepted, the router updates the resource reservation of the flow's class[2], allocates a reservation structure for the flow, stores the label at the B field for this reservation, and forwards the *SResv* message to $R_2$ (2) after changing the LABEL_SETUP to $T_1$. If the flow cannot be accepted, a *SResvStat* message is sent towards

[2] As has been previously stated, this reservation is only committed to the admission control module at this point.

the sender reporting the error. Each router along the path that receives this message removes resource reservation information for this flow and releases the reserved bandwidth. If the flow is accepted, the router forwards the *SResv* message to the next hop and the procedure is repeated at every router along the forward path until the *SResv* message finally arrives at the receiver (3) with $T_2$ in the LABEL_SETUP. At this point, all routers along the path have reserved resources for the new flow and all labels required for backward message processing are installed in the reservation state.

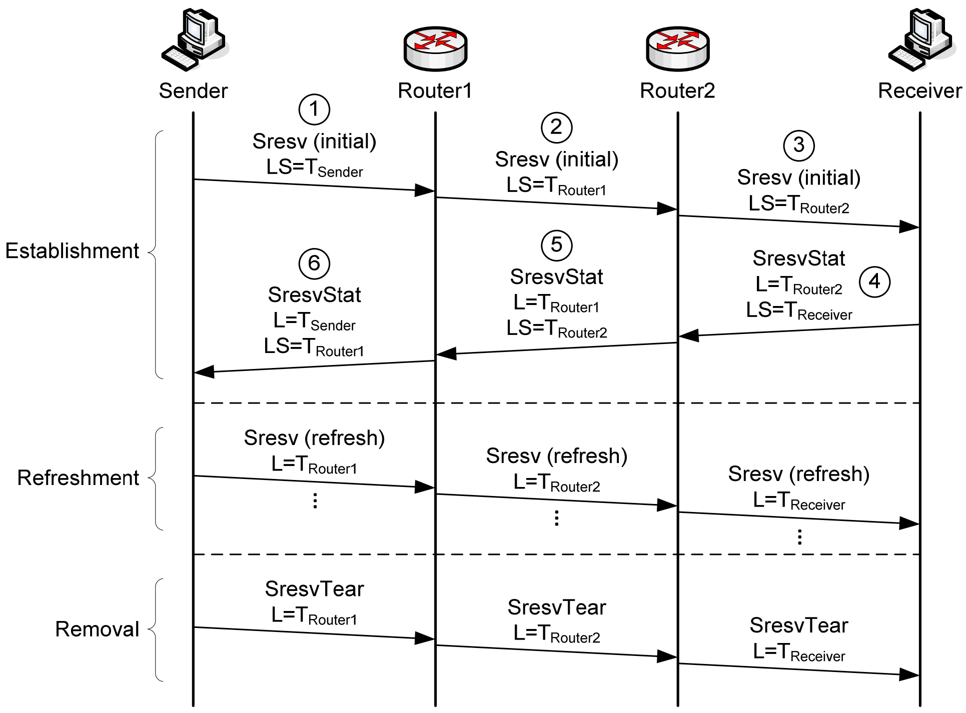


**Figure 4.4: Message flow**

The receiver will acknowledge the successful reservation by sending a *SResvStat* message towards the sender reporting success. Since the labels required for backward message processing are already installed, the *SResvStat* message will make use of the labels. Note that even an *SResvStat* message reporting a failure reservation originated by a node in the middle of the path can make use of the labels already installed upstream. The *SResvStat* message reporting successful reservation (4) contains a LABEL object with the label $T_2$ and a LABEL_SETUP object with the label. $R_2$ uses the label $T_2$ in the LABEL object to access the memory structure for this reservation and commits the updates to the policing and queuing modules. The label $T_R$ is stored at the $F$ field in $R_2$. It changes the LABEL to the value in the B field, $T_1$, and forwards the message to $R_1$, containing $T_2$ in LABEL_SETUP (5). When the *SResvStat* message reaches the sender (6) with $T_1$ in LABEL_SETUP, all labels are installed, the reservation is acknowledged, and no further LABEL_SETUP and flow reservation objects need to be sent, except in the case of route changes. Notice that the path of the *SResvStat* message is the symmetric path of the *SResv* message. Since the signaling protocol is hop-by-hop, the *SResvStat* message has information on the previous hop of the *SResv* message. However, these two messages are used in a one way reservation. Therefore, the reservation in the opposite direction can have a completely different path.

Each established reservation has an associated timer, which is proportional to the expected flow duration. The algorithm devised to provide efficient timers' implementation will be detailed in subsection 4.2.5. A reservation is explicitly released upon the reception of a *SResvTear* message, or implicitly released (soft reservation) upon the expiration of the timer for that reservation. In order to refresh the reservation and update the expiration timer (so the reservation is not released), the sender originates a simplified (refresh) *SResv* message towards the receiver. Processing of this message will make use of the labels for direct access to the flow reservation information, and each router along the path updates the expiration timer. Refresh-only *SResv* messages do not require a confirmation from the receiver.

When a sender wants to explicitly terminate a flow session, it sends a *SResvTear* message towards the receiver, also making use of the labels. Upon reception of this message, each router along the path removes the reservation information for that flow and releases its bandwidth for future flows. Before removing the reservation, though, the router must wait for the queue to be flushed (or the last enqueued packet belonging to the flow to be transmitted). Only then the reservation parameters may be subtracted from the aggregate reservation and the *SResvTear* message forwarded to the next hop.

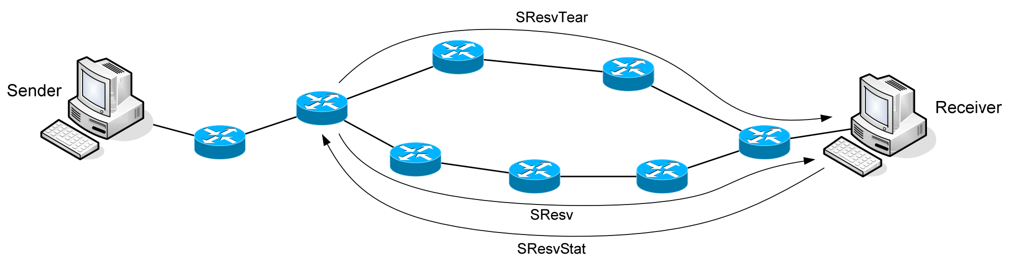


**Figure 4.5: Route change**

The *SResv* messages are also required when a route changes (fig. 4.5). In this case, the router that detects a route change towards the receiver originates a full *SResv* message along the new path in order to reserve resources and install the labels required for scalable signaling processing. Each router along the new path processes this message and forwards it to the next hop. The receiver, upon receiving the message, sends back a *SResvStat* message acknowledging the new reservation. The expiration time for this *SResv* message must be greater or equal to the remaining time for reservation expiration at the router detecting the route change. A *SResvTear* message should also be sent, when possible, to the old next hop in order to remove the stale reservation from the old path.

If a timer expires for lack of refreshment, the reservation parameters are immediately removed from the policing module. However, only after the queue of the traffic class to which the flow belongs has been flushed, or the last enqueued packet from that flow has been transmitted, the reservation may be removed from the admission control and traffic control modules. This ensures that, under every circumstance, there is enough bandwidth allocated to the class to send those packets without negatively affecting other flows in the same class.

### 4.2.4 Packet Processing

Whenever a data packet arrives at a router, it must be processed according to specific rules which, due to different requirements, are different for the GS and CL classes. In particular, no GS packet may be admitted if there is not an up-to-date reservation in place, like in the case of expiration of the reservation or an occurrence of a route change.

The first action performed by the router when a data packet arrives is to look at the DSCP and find out if the packet belongs to a service class with reservations. If the packet belongs to the GS class, the following sequence of actions is then performed:

1. Get the label from the packet header;
2. Check out if there is a valid reservation in place corresponding to the label; if not, the packet is dropped (or re-marked to best-effort);
3. If there is a valid reservation, the packet is passed to the policing module (except at the core routers);
4. The next hop given by the routing table is then compared to the one stored in the reservation structure; if they are different, the route has changed — the packet is dropped (or re-marked to best-effort) and a procedure to reestablish the reservation through the new route (and, if possible, to remove the stale reservation from the old route) is triggered;
5. If there is a valid reservation in place, the policing module accepts the packet and no route change has occurred, the packet is finally enqueued in the GS queue of the output interface.

If, on the other hand, the packet belongs to (one of) the CL class(es), the sequence is simpler. Since these flows are tolerant to small amounts of packet loss and flow interference, immediate route change detection and reservation checking for every packet are not required. This means that it is possible to drop the requirement for the data packets to carry the label and the prohibition of datagram fragmentation. In this case, a route change could go undetected until the next *SResv* (refresh) arrives. CL packets, then, are immediately passed to the policer (if applicable) and then enqueued at the output interface given by the routing table.

Best-effort and signaling packets are processed as usual.

### 4.2.5 Soft Reservations and Efficient Timer Implementation

In SRBQ, reservations are soft state: if no *SResvTear* message is received and the reservation is not refreshed, the associated timer expires and it is removed. Soft state reservations have the obvious advantage of providing adaptability to changing network conditions; however, they require the implementation of expiration timers. The basic implementation concept for timers is a sorted event queue: the processor waits until the first timer value in the list expires, dequeues it, performs the appropriate processing, and then goes on waiting for the next timer value to expire. While dequeuing an event is trivial, inserting an event with a random expiration time is an expensive operation, highly dependent on the total number of events queued. Some algorithms have been proposed [Varghese97, Aron00] for the efficient implementation of timers, but while general enough for implementing any kind of timer, these algorithms are still overly complex for our purpose. Contrasting to the complexity of generic timers, fixed delay timers are very simple and efficient to implement: in this case, the event queue is treated in FIFO fashion, providing both trivial event queuing and dequeuing. Fixed delay termination timers, however, are undesirable in the case of reservations which may have very dissimilar life spans.

Trying to achieve a balance between the simplicity of fixed delay timers and the flexibility of generic timers, we have created an algorithm which has trivial timer queuing and a low and constant cost timer dequeuing, providing eight possible timer delays in a base-2 logarithmic scale. These values map to a timer delay range of 1:128, which is enough for our purposes. The implementation is based on eight different queues, each of which has an associated fixed delay. Internally, therefore, these queues are served using a FIFO discipline. Flow timeouts are chosen from one of the eight possible values using a three bit field in a SRESV_PARMS object included in *SResv* messages. Queuing an event is a simple matter of adding it to the tail of the corresponding queue, which is trivial. Dequeuing an event means choosing one of the eight possible queues (the one whose timers expires first) and taking the first event from that queue. Anyway, timers should not expire very frequently, as most of the time reservations will either be refreshed or explicitly terminated by means of a *SResvTear* message, accessing the timer using a reference stored in the flow structure.

Having a good range of reservation expiration timer values means that short-lived flows will not remain stale for long times whenever something unusual occurs (such as an application lockup or premature termination, or an undetected route change), but longer-lived flows will not generate too much signaling traffic just to refresh the reservation. Figure 4.6 shows the relative weight of the refresh *SResv* messages in the total signaling traffic for flows with life spans varying from 15 s to 240 s using the eight possible different reservation timer values. The base timer is 4 s, and refresh messages are sent at a rate that is 4 times larger than the expiration timer rate to ensure that the reservation is correctly refreshed even in the presence of some signaling traffic losses. Notice that in order to tolerate the loss of $k$ consecutive refresh messages, refreshes must be sent at least every $T/(k+1+\Delta)$ seconds, where $T$ is the expiration timer value and $0<\Delta<1$ is a tolerance factor. As can be seen, the weight of the refresh messages in the overall signaling traffic may vary from 0 to 98.6%, increases with the lifespan of the flows and decreases with the timer duration.

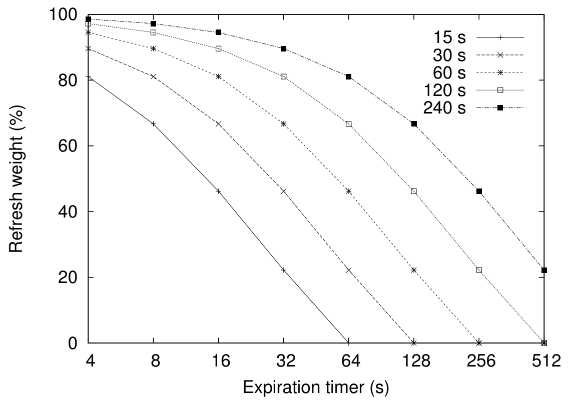


**Figure 4.6: Relative weight of refresh messages**

Applications should use timer values representing a good tradeoff between signaling traffic and fast recovery from faults for the expected flow lifespan. When the lifespan cannot be estimated *a priori*, the application may use a short timer at first and increase it using the SRESV_PARMS objects in refresh messages.

## 4.3 Performance Evaluation

The SRBQ architecture has been implemented using the ns-2 simulator [NS2], and some extensions to the Nortel DiffServ implementation and Marc Greis' RSVP implementation [Greis98]. The DiffServ extensions are publicly available in [PriorNS]. In this section we discuss the performance results obtained by simulation, assessing the performance of SRBQ regarding QoS guarantees and scalability.

The simulated scenario is depicted in fig. 4.7. It includes 1 transit and 5 access domains. Each terminal in the access domains represents a set of terminals. The reason for having more than one access domain connected to the edge node of the transit domain is to check that correct aggregate policing is performed at the entry of that domain. The bandwidth of the connections in the transit domain and in the interconnections between the transit and the access domains is 10 Mbps. The propagation delay is 2 ms in the transit domain connections and 1 ms in the interconnections between the access and the transit domain.

In this scenario we consider the coexistence of GS, CL and BE classes. At each referred connection, the bandwidth assigned to the signaling traffic is 1 Mbps. Note that, although this seems very high, the excess bandwidth can be used for BE traffic. The bandwidth assigned to the GS class is 3 Mbps, while for CL it is 4 Mbps. The remaining bandwidth is used for BE traffic. The bandwidth reserved for the GS and CL classes and left unused is also used for BE.

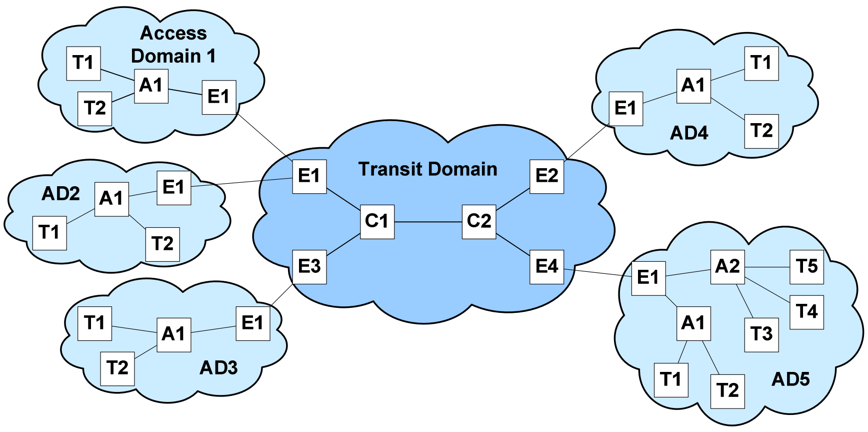


**Figure 4.7: Topology used for SRBQ evaluation**

Each terminal on the access domains on the left side generates a set of flows belonging to the GS, CL and BE classes. The destination of each flow is randomly chosen from the set of the terminals on the right side access domains; each source may generate traffic to all destinations. The traffic in each class is composed of a mixture of different types of flows.

In the GS class admission control is parameter-based. In the CL class we compare the performance results with both PBAC and MBAC algorithms. The MBAC algorithm used is an adaptation of Measured Sum (MS) [Jamin97] for 3 drop probability levels, where the estimated traffic for each level is added to the estimated traffic of the lower drop probability ones. The overall target utilization factor is 95%.

All simulations presented in this chapter are run for 5400 simulation seconds, and data for the first 1800 seconds is discarded. All values presented are an average of at least 5 simulation runs with different random seeds. The next subsections present the results of these experiments.

## 4.3.1 End-to-End QoS Guarantees

In this subsection we evaluate the end-to-end QoS guarantees of both GS and CL classes in different experiments, first varying the amount of offered load, and second varying the requested rate of flows. The largest Mean Offered Load (MOL) in the GS and CL classes is, in terms of average rates, about 20% higher than the one assigned to those classes. Due to different mixtures of flow types, this translates in excess figures of 26% (GS) and 42% (CL) in terms of requested reserved rates (ROL — Requested Offered Load). In these experiments the set of flows is distributed in the following way: (1) traffic in the GS class is composed by Constant Bit Rate (CBR) flows (Voice and Video256) and on-off exponential (Exp1gs) flows; (2) traffic in the CL class is composed by on-off exponential (Exp1cl) and Pareto (Pareto1cl) flows; and (3) traffic in the BE class is composed by on-off Pareto (Pareto1be) and FTP (Ftpbe) flows. Flows belonging to the BE class are active for the overall duration of the simulations (there are 3 FTP and 2 Pareto flows per source), while flows in the other classes are initiated according to a Poisson process with a certain mean time interval between calls (MTBC), having an average duration (Avg dur.) exponentially distributed. The characteristics of these flows are summarized in table 4.1.

| Class | Type | Peak rate (kbps) | On time (ms) | Off time (ms) | Avg. rate (kbps) | Pkt size (bytes) | Resv rate (kbps) | Resv burst (bytes) | Low RR (kbps) | High RR (kbps) | MTBC (s) | Avg dur. (s) | MOL (kbps) | ROL (kbps) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| GS | Voice | 48 | - | - | 48 | 80 | 48.048 | 81 | - | - | 45 | 120 | 768 | 769 |
|  | Video256 | 256 | - | - | 256 | 1000 | 256.256 | 1050 | - | - | 180 | 240 | 2048 | 2050 |
|  | Exp1gs | 256 | 200 | 200 | 128 | 1000 | 160 | 5000 | - | - | 90 | 90 | 768 | 960 |
| CL | Pareto1cl | 256 | 200 | 200 | 128 | 1000 | 150 | - | 64 | 256 | 38 | 120 | 2425 | 2842 |
|  | Exp1cl | 256 | 200 | 200 | 128 | 1000 | 150 | - | 64 | 256 | 38 | 120 | 2425 | 2842 |
|  |  |  |  |  |  |  |  |  |  |  | Simult. Flows |  |  |  |
| BE | Ftpbe | - | - | - | - | 1040 | - | - | - | - | 3 per src terminal |  | var. | N.A. |
|  | Pareto1be | 256 | 200 | 200 | 128 | 1000 | - | - | - | - | 2 per src terminal |  | 2304 | N.A. |

**Table 4.1: Traffic flows for end-to-end QoS tests**

For GS flows, the reservation rate (Resv rate) represents the rate of the token bucket and the reservation burst (Resv burst) represents its depth. The reservation parameters provide a small amount of slack to compensate for numerical errors in floating point calculations. For CL flows, Low RR (Reservation Rate), Resv rate and High RR represent the three rate watermarks used for drop precedence selection and packet dropping at the policer. In these simulations, both parameter-based and measurement-based admission control were used for the CL class, with the utilization limits for the three rate watermarks set to 0.7, 1.0 and 1.7 times the bandwidth assigned to this class. The sum of the rates in each watermark for all flows in the class must not exceed the respective utilization limits. Notice that only admission control is performed per-flow; both scheduling and policing are performed on a per-class basis (except at the access routers).

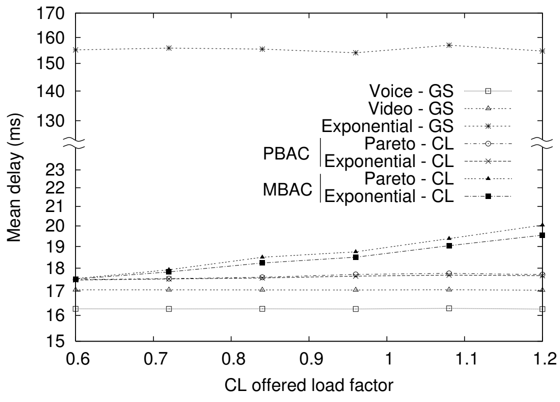
In the following subsections we present the comparison between the performance results obtained with a variation in some crucial parameters.

#### *4.3.1.1 Offered Load*

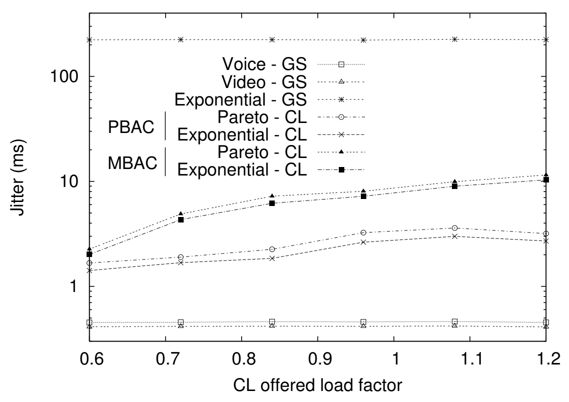
In the first experiment we varied the offered load factor for the CL class from 0.6 to 1.2, keeping a constant offered load factor for the GS class of 1.2. A load factor of 1.2 corresponds to the values of MTBC and Average duration presented in table 4.1. Lower amounts of offered traffic are obtained by increasing the mean time between flow generation events in the inverse proportion of the offered load factor. Figure 4.8 presents the delay, jitter, packet losses and core link utilization results, combined from simulations with PBAC and MBAC in the CL class. The GS class always uses PBAC, and since the results are the same in the two types of simulations, a single curve per flow type is shown.

With PBAC, the average delay remains very low and almost constant for all types of flows, except for the GS exponential flows. For all flows except the GS exponential ones, the delay is mostly the sum of transmission and propagation delays. GS exponential flows suffer an additional, and potentially large, delay at the ingress shaper of the access router when they send at a rate larger than what they requested for long periods of time. It is the applications' fault, though, for transmitting non-conformant traffic. The fact that the delay for the other GS flows remains very low shows that they are not adversely affected. The delay for CL flows remains almost constant, independently of the offered traffic. When using MBAC, the delay for the CL class increases with the offered load. This is due to a higher utilization of the class. Jitter values exhibit a similar behavior for GS flows. Jitter for CL flows increases with the offered CL load, which is expected due to the increased multiplexing. This increase is much more noticeable when using MBAC. Regarding packet losses, they are null for voice and video GS flows. Losses for exponential GS flows are higher, though small (<0.14%), and are due to buffer space limitation at the ingress shapers (access routers). In CL flows packet loss increases with the offered load, but remains nevertheless very low (less than 0.03%) when using PBAC. This means we should probably be more aggressive by reducing the requested rate watermarks for these flows, which will be evaluated in subsection 4.3.1.2. With MBAC, packet losses are much higher, going up to 0.25% for the heavy-tailed Pareto flows with an offered load factor of 1.2.

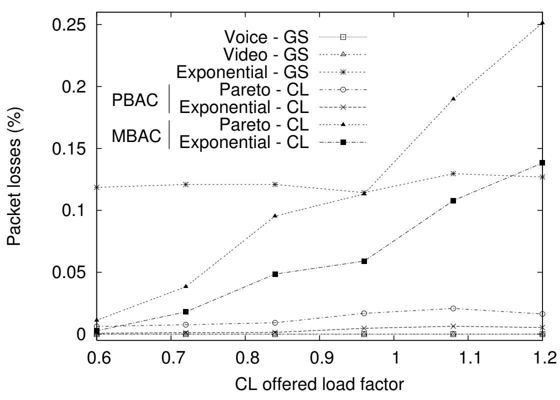
The utilization of the CL class increases from about 2.4 Mbps (60%) to about 3.1 Mbps (78%) with PBAC. Keep in mind that we are reserving 150 kbps for flows with an average rate of 128 kbps, which imposes an upper limit in the mean utilization of the CL class of about 3.4 Mbps (85%) when using PBAC. With MBAC the reserved values are not so important, and the utilization goes up to 3.3 Mbps (83%). This higher utilization is responsible for the worse QoS results (delay, jitter and losses) in the CL class, since a perfect prediction of the class occupancy cannot be performed. In the GS class the utilization is almost constant with a value of 2.5 Mbps (83%). Though not shown in the chart (since the curve would not be visible), signaling traffic goes up from about 2.9 kbps to about 3.3 kbps with the increasing CL offered load. This means that signaling traffic is less than 1/1000 of the data traffic with reservations subject to admission control (GS and CL). The BE class uses the remaining bandwidth.



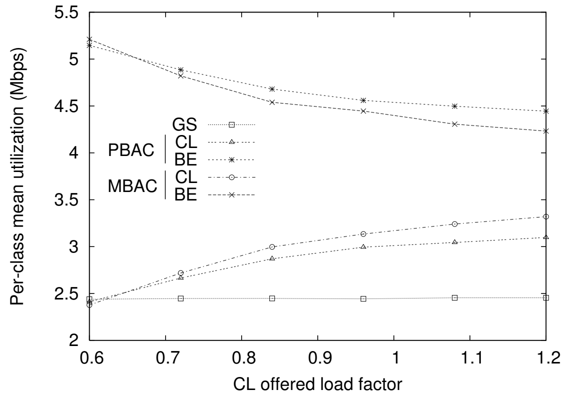
a) Mean delay



b) Jitter



c) Packet loss



d) Utilization

**Figure 4.8: QoS and per-class utilization results with varying offered CL load**

A second experiment was performed where the offered load for the CL class was kept constant at 1.2, while that for the GS class was varied from 0.6 to 1.2. The curves for all QoS results (fig. 4.9) are essentially constant, meaning that no noticeable degradation is introduced by increasing the GS load. These results were expected, since the average CL offered load is fixed and the degradation inflicted by the per-flow ingress shaping of the GS flows is independent of the number of accepted flows — it depends only on the degree of conformance of the flow to the reserved token bucket.

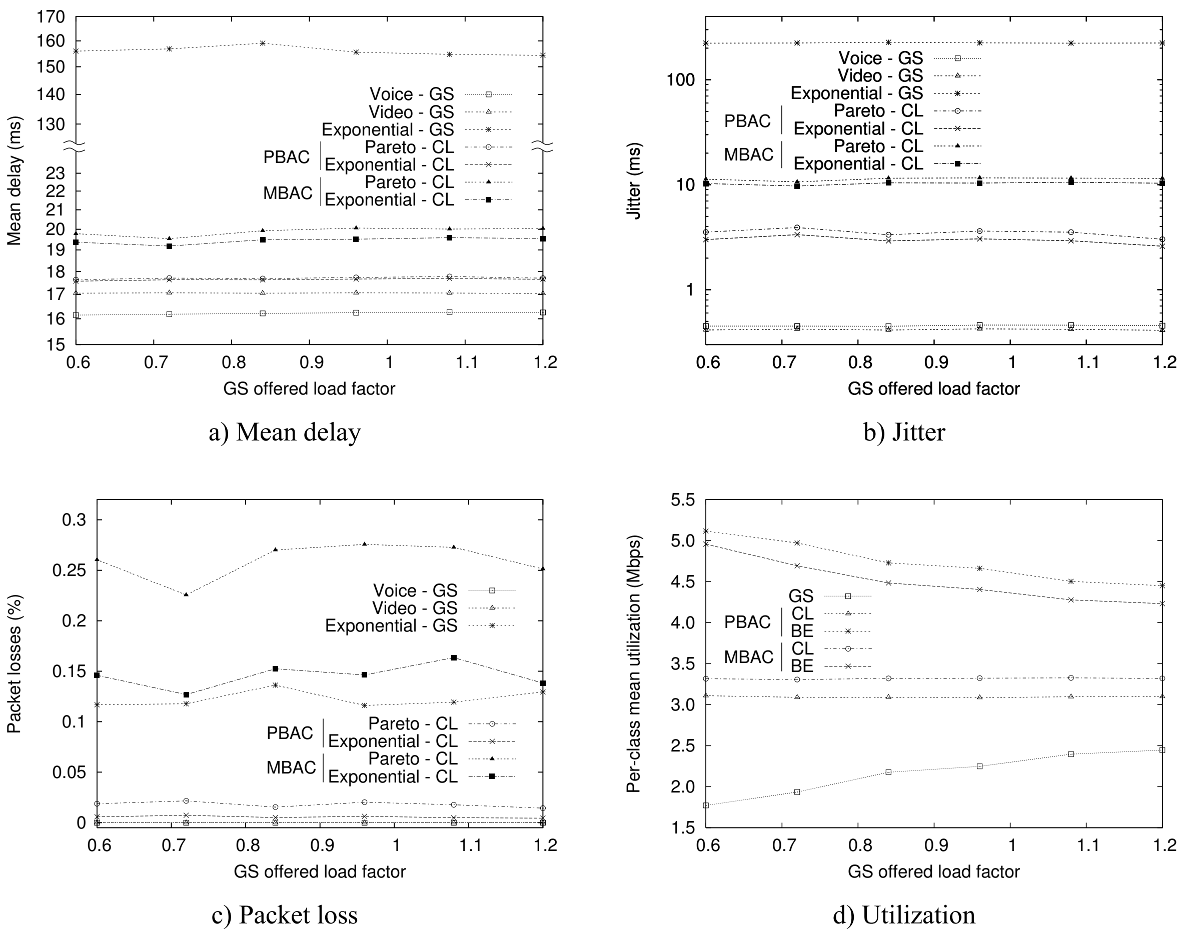
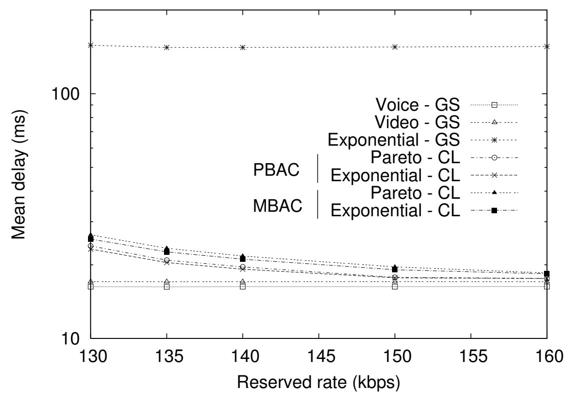


**Figure 4.9: QoS and per-class utilization results with varying offered GS load**

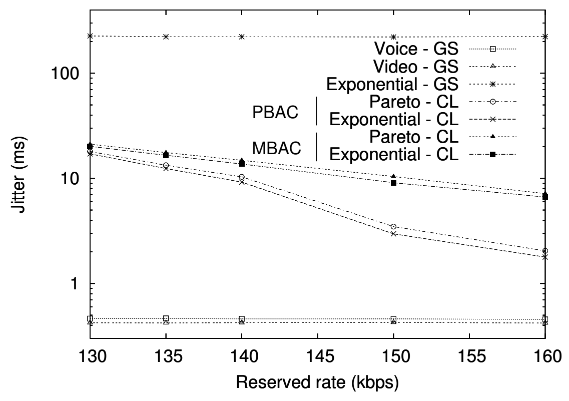
### *4.3.1.2 Requested Rate*

As was previously mentioned, we could be more aggressive on the requested rate of the CL traffic flows. With the next experiment we analyze the effect, in terms of QoS and link utilization of both GS and CL classes, of decreasing the requested rate. Figure 4.10 shows the variation of delay, jitter, packet loss, and utilization values with varying requested rates for CL flows using both PBAC and MBAC for the CL class. Here we have set the flow acceptance utilization limits of the three rate watermarks to 0.7, 1.0 and 2.0 times the bandwidth assigned to CL in order to ensure that flow admission would be performed based on the second rate watermark, the varying factor in these experiments. Since the average rate for both types of CL flows used in this experiment is 128 kbps, we varied the requested rate from 130 kbps to 160 kbps, a little higher than the 150 kbps used in the previous experiments.

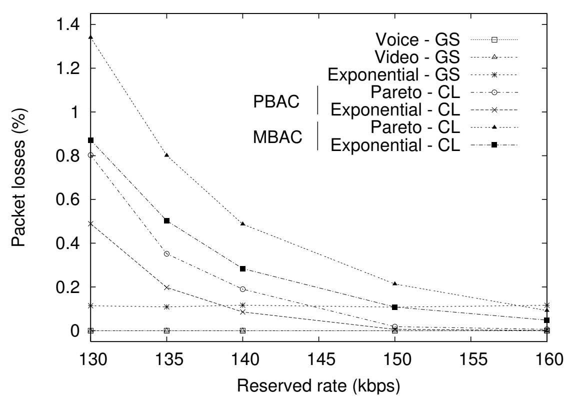
The QoS values for GS flows are unaffected by the admission control method used for CL flows, which is normal since GS traffic has higher priority and always uses PBAC. As expected, delay, jitter and losses for CL flows decrease with the increasing requested rate, since the number of accepted flows is lower.



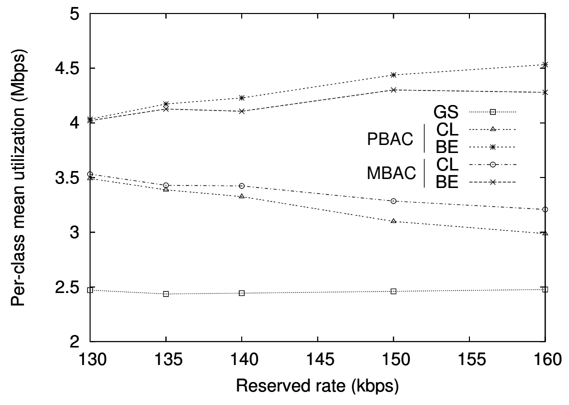
a) Mean delay



b) Jitter



c) Packet loss



d) Utilization

**Figure 4.10: QoS and per-class utilization results with varying reserved rates for CL flows**

Delay, jitter and packet loss values for the CL class are higher when using MBAC than when using PBAC, as can be seen in fig. 4.10. The higher values obtained are again due to the better resource usage with MBAC and to an underestimation in some time intervals of the bandwidth occupancy in the next intervals[3]. Regarding link utilization, it is higher and decreases more slowly with the increasing reservation values with MBAC than with PBAC. The slower decrease is due to the smaller influence in MBAC of the reserved rate, since the reserved rates of the already admitted flows are not taken into consideration, only the actual measured traffic.

Even with a requested rate of 130 kbps, which is only 1.6% higher than the average transmission rate, packet losses are lower than 0.8% and 1.4%, respectively, in the PBAC and MBAC cases.

[3] PBAC always “estimates” a usage equal to the sum of the full reservation values.

The previous sets of experiments show that our model, though being aggregation-based, is able to support both strict and soft QoS guarantees. They also show that aggressive CL flows (Pareto) are more penalized in terms of loss probability than those with friendlier traffic envelopes (Exponential). In contrast, the more aggressive GS flows are essentially penalized in terms of delay due to ingress shaping, but only if they exceed the reserved traffic specification.

### 4.3.2 Independence between Flows

In this subsection we evaluate the performance of the architecture in the presence of misbehaved flows, that is, flows that send at a rate much higher than the one they requested for considerable periods of time. Moreover, we also analyze the influence of misbehaved flows on well behaved ones. In order to protect the network from the former flows, the access router performs per-flow ingress shaping for the GS class. This shaper absorbs multiplexing jitter from the terminal and ensures that the traffic injected into the network does not exceed the reserved parameters by absorbing application bursts above the requested bucket (of 5 packets in this case), thus protecting the other GS flows. CL flows, on the other hand, are tolerant to small amounts of packet losses, meaning that the CL class does not need this degree of protection. CL flows are policed, instead of shaped, at the access router, meaning that a single misbehaved CL flow will be penalized in terms of packet losses but will not be significantly affected in terms of delay. Since the CL policer is not a strict token bucket, but rather based on the average rate measured over a period of time, it is not noticeably affected by multiplexing jitter. Performing per-flow policing, instead of shaping, at the access router has the advantages of lighter processing and avoidance of the introduction of unnecessary delays in bursty flows.

In this experiment, measurement-based admission control is used in the CL class. The offered load for the GS and CL classes is, respectively 43% and 39% larger, in terms of reserved rates (ROL), than their assigned bandwidth. In terms of actual traffic (MOL), this translates in an excess of 23% over the assigned rate of both CL and GS. In each class, there are three types of flows, as shown in table 4.2: (1) a CBR flow simulating a video stream at 64 kbps that is considered a well behaved flow; (2) an on-off exponential flow with a fixed average duty cycle of 50% (Exp1) that is considered a nearly well behaved flow, since it sends at a rate a little bit higher than it is requesting; and (3) an on-off exponential flow (Exp2) that is considered a misbehaved flow, since it can send at a rate much larger than the one it is requesting for considerable periods of time. Its average duty cycle is variable, from 50% (equal average busy and idle times) to 12.5% (busy time is 12.5% of the cycle, on average). The sum of the average busy and idle times remains constant at 400 ms. In order to keep the average rate constant, the peak rate varies between 256 kbps (average busy and idle times of 200 ms) and 1024 kbps (average busy and idle times of 50 ms and 350 ms, respectively) — lower values of duty cycle correspond to increased burstiness[4]. The peak rate value of 1024 kbps for a duty cycle of 12.5% introduces a large mismatch between the requested and transmitted rates, turning this type of flow into a misbehaved one.

| Class | Type | Peak rate (kbps) | On time (ms) | Off time (ms) | Avg. rate (kbps) | Pkt size (bytes) | Resv rate (kbps) | Resv burst (bytes) | Low RR (kbps) | High RR (kbps) | MTBC (s) | Avg dur. (s) | MOL (kbps) | ROL (kbps) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| GS | Video64gs | 64 | - | - | 64 | 500 | 64.064 | 501 | - | - | 75 | 240 | 1229 | 1230 |
| | Exp1gs | 256 | 200 | 200 | 128 | 1000 | 160 | 5000 | - | - | 75 | 120 | 1229 | 1536 |
| | Exp2gs | var. | var. | var. | 128 | 1000 | 160 | 5000 | - | - | 75 | 120 | 1229 | 1536 |
| CL | Video64cl | 64 | - | - | 64 | 500 | 65 | - | 64 | 66 | 75 | 240 | 1229 | 1248 |
| | Exp1cl | 256 | 200 | 200 | 128 | 1000 | 150 | - | 64 | 256 | 50 | 120 | 1843 | 2160 |
| | Exp2cl | var. | var. | var. | 128 | 1000 | 150 | - | 64 | 256 | 50 | 120 | 1843 | 2160 |

**Table 4.2: Traffic flows for the isolation test**

Figure 4.11 shows the mean delay and the packet loss ratio for all three types of flows on both classes with increasing duty cycle values of the misbehaved (Exp2) flows. We may observe that the delay of the well behaved flows, in both GS and CL, is very low, consisting mainly on the sum of transmission and propagation delays. Misbehaved CL flows are not penalized in terms of delay; the delay difference between them and the CBR flows is due to larger transmission delays, since their packet size is larger. GS flows, on the other hand, are shaped at the access router, so the non-conformance to the reservation specification is translated in large delays. In fact, with a duty cycle of 12.5%, the average delay for misbehaved GS flows is more than 400 ms. Notice that since all GS flows are aggregated and use the same queue, internally served in a FIFO fashion, the queuing delay is shared by all GS flows. Therefore the large delays and packet losses for nearly well behaved and misbehaved flows are inflicted at the ingress shaper only.

Contrary to what happens regarding delay, both GS and CL misbehaved flows are penalized in terms of packet losses, as may be seen in fig. 4.11.b. With a duty cycle of 12.5%, packet losses for misbehaved flows reach 7.1% in GS and 4.7% in CL. The other flows have very small losses, showing that they are not adversely affected by the burstiness of the other flows. While not null, losses in nearly well behaved GS flows are low and constant, therefore unaffected by the burstiness of the other flows. These losses are not inflicted by network congestion, but rather by the ingress shaper. Notice that, while not a single packet was lost in well behaved GS flows, both well behaved and nearly well behaved CL flows are slightly influenced by misbehaved flows, having losses that increase when the duty cycle of the

[4] In fact, using the common definition of burstiness as the ratio of peak to average rate of the flow, the burstiness is the inverse of the duty cycle.

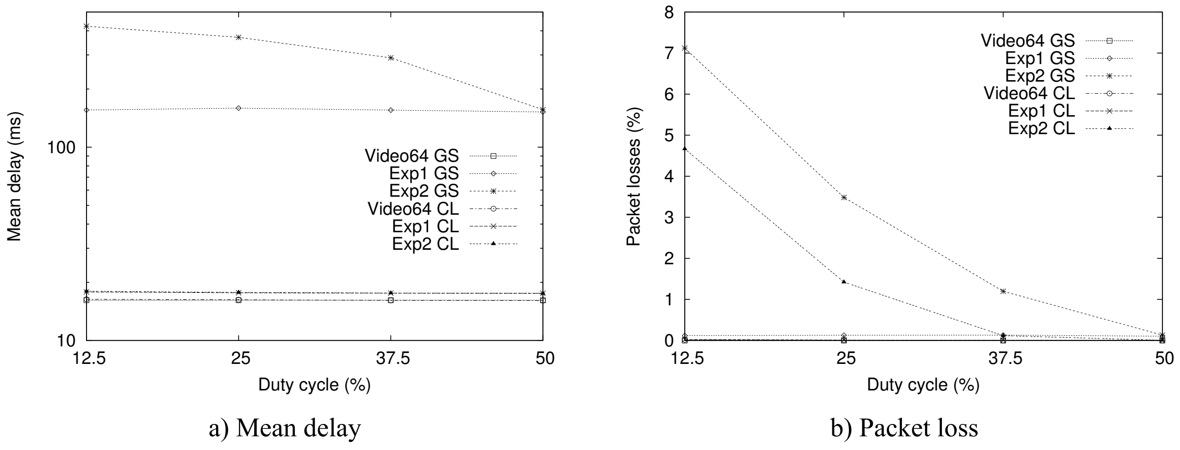


**Figure 4.11: Delay and packet losses with varying burstiness of misbehaved flows**

misbehaved flows goes from 50% to 12.5%. While their maximum values are a mere 0.015% and 0.025%, respectively, for the well behaved and nearly well behaved CL flows, they are not null as is the case in GS.

In both classes there is a clear reaction against misbehaved flows, which is meant to protect the other flows.

From the previous results, we may conclude that the CL class is appropriate for bursty traffic which tolerates some packet loss. In fact, being based on average rates, this class is much more tolerant to bursts, and since there is no shaping, no additional delays are inflicted to bursty flows. The GS class, on the other hand, is ideal for flows intolerant to packet losses and having well defined traffic envelopes which they do not exceed, but heavily penalizing for non-conformant flows. The strict guarantees, which may not be provided by the CL class, justify the additional complexity of the GS service.

This experiment shows that the system reacts accordingly in the presence of misbehaved flows. These results are not unusual, and are to be expected in guaranteed service type classes. The main achievement of our model is to provide these guarantees in a scalable, aggregation-based architecture.

### 4.3.3 Real Multimedia Streams

All the previous experiments were based on synthetic flows with specific characteristics meant to evaluate particular aspects of the performance of the SRBQ architecture. In order to evaluate its performance under normal working conditions we performed a set of simulations using packet traces from real multimedia flows. The trace files we used correspond to H.263 video streams with average bit rates ranging from 16 kbps to 256 kbps, and are available from [VidTraces].

In this experiment, we assigned 2 Mbps to the GS class and 5 Mbps to the CL class. The rate watermarks for the CL class were adjusted to 0.8, 1.0 and 2.0 times the bandwidth assigned to it. Table 4.3 summarizes the parameters of the flows. Traces from several different video streams are used for each bit rate, and the starting point in the stream for each flow is randomly chosen. Flows are initiated according to a Poisson process and have a duration following a Pareto distribution. The highest mean offered load for both the GS and the CL classes is 20% higher than their assigned bandwidth (load factor of 1.2).

| Class | Type | Avg rate | Pkt size | Resv rate | Resv burst | Low RR | High RR | MTBC | Avg dur. | MOL | ROL |
|---|---|---|---|---|---|---|---|---|---|---|---|
| | | (kbps) | (bytes) | (kbps) | (bytes) | (kbps) | (kbps) | (s) | (s) | (kbps) | (kbps) |
| GS | video | 16 | var. | 17 | 4000 | - | - | 151 | 180 | 114 | 122 |
| | video | 64 | var. | 68 | 5000 | - | - | 151 | 180 | 458 | 486 |
| | video | 256 | var. | 272 | 8000 | - | - | 151 | 180 | 1831 | 1945 |
| CL | video | 16 | var. | 16.5 | - | 12 | 40 | 60 | 180 | 288 | 297 |
| | video | 64 | var. | 66 | - | 48 | 96 | 60 | 180 | 1152 | 1188 |
| | video | 256 | var. | 264 | - | 192 | 352 | 60 | 180 | 4608 | 4752 |

**Table 4.3: Characteristics of the real-world multimedia streams**

Figure 4.12 shows the performance and utilization results for varying offered load factors in both classes, combining results from the simulations using PBAC and MBAC in the CL class; PBAC is always used in the GS class. Delay in the GS class does not seem to be affected by the offered load factor, while that of the CL class exhibits a slight growth trend, more evident when using PBAC. Notice that the delay is always smaller than 20 ms. Jitter figures have a similar behavior. The higher jitter values in the GS class are due to ingress shaping at the access router, performed in order to force the flows into conformance with the reservations.

There are no packet losses in the GS class: burstiness above the reserved rate is absorbed by the ingress shaper at the access router (within certain bounds), and translates into increased delay and jitter rather than losses. Packet loss curves for the CL class exhibit a seemingly contradictory behavior: they are higher for lower values of offered load. There is, however, an explanation for this fact, which stems from a combination of several factors. (1) Lower bit rate H.263 flows are burstier and have smaller packets than higher bit rate ones. (2) The second rate watermark (for which the class has a target utilization factor of 1) used in the reservations for all these flows is 3% higher than their target bit rate. Therefore, the absolute difference between the reserved rate and the target bit rate is larger in higher bit rate flows. The absolute difference between the third watermark and the target bit rate is also larger in higher bit rate flows. (3) Packet re-marking and policing is performed on an aggregate basis at the edge routers. These three factors combined mean that when there are more high bit rate flows in the network, low bit rate ones take advantage of the bandwidth excess from the

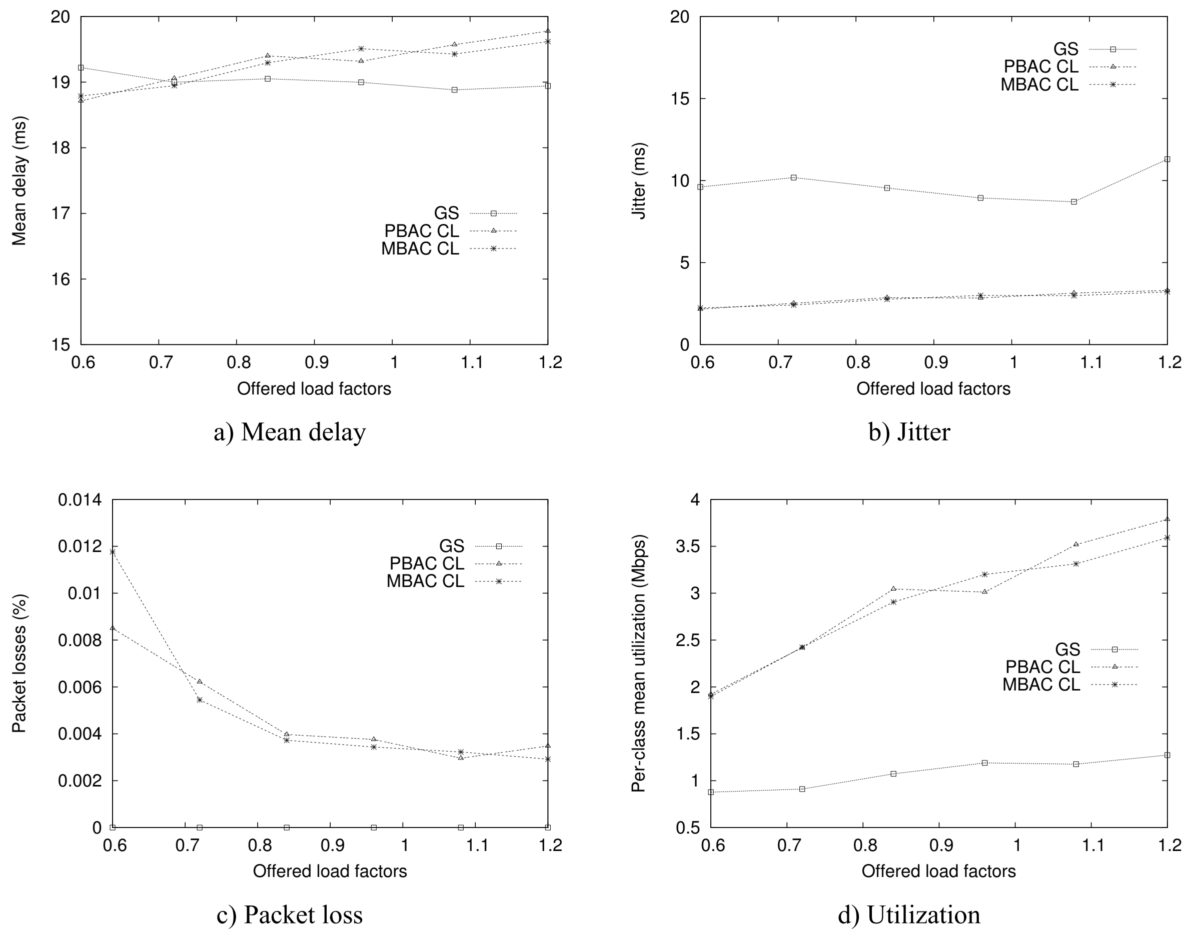


**Figure 4.12: QoS and utilization results with varying offered loads using real video flows**

higher bit rate ones, therefore decreasing their loss ratio. This fact has a much higher weight in the overall packet loss ratio than the (minimal) amount of network congestion.

The utilization of each class, as expected, grows with the offered load. With a load factor of 1.2, the mean utilization of the GS class is 1.3 Mbps (65%), and that of the CL class is 3.8 Mbps (75%) when using PBAC and 3.6 Mbps (72%) when using MBAC. The lower utilization with MBAC than with PBAC has a simple explanation. The reserved rate (second watermark) for these flows is very close to their target bit rate (only 3% higher). The MBAC target utilization factor is 95%; this means that most of the time the estimated bandwidth will be higher than 95% of the sum of the reservations, leading to lower utilization figures when using MBAC.

The results presented in the previous paragraphs show that the SRBQ architecture is able to meet the QoS requirements of the supported service classes when using real-world multimedia flows.

### 4.3.4 Scalability

SRBQ combines DiffServ-like aggregate packet processing (classification, scheduling, policing and shaping), whose scalability properties are well known, with a model of per-flow signaling-based resource reservation carefully designed to have message processing times that are low and algorithmically independent on the number of simultaneous flows. This independence property is fundamental to the scalability of the signaling protocol, as it implies that the overall necessary processing power in face of the number of simultaneous flows can be upper bounded by a linear curve. However, an evaluation of SRBQ would not be complete without a measurement of processing delay of the signaling messages, necessary to ascertain the feasibility of the protocol as of today.

Usually, such an evaluation would be undertaken in a physical testbed using a prototype implementation. An event-driven network simulator, such as ns-2, in not generally appropriate for such measurements, as the simulated clock progresses independently of the wall clock, and all processing appears to be instantaneous. However, if the implementation of a piece of code in the simulator is close enough to what a real implementation would be, a meaningful measure of the processing delay associated to that piece of code may be obtained by ignoring the simulation clock and using the system clock instead. We decided to take this approach in order to avoid the burden of re-implementing SRBQ in a real operating system and setting up the testbed.

In this experiment we used the same set of flows of section 4.3.1 and varied the load factor between 0.033 and 33.3 in order to evaluate the processing delay with different loads. However, since we wanted to maintain the flow acceptance ratio, the upper bounds on the aggregate values were varied in the same proportion. Using the TSC (Time Stamp Counter) register of the processor, which counts the number of clock cycles since boot-up, we measured the processing delay of *SResv* messages, including both initial and refreshments of accepted flows. The processing delay was obtained by dividing the number of elapsed cycles by the Central Processing Unit's (CPU) clock frequency.

Figure 4.13 shows the processing delay of *SResv* messages for different values of the load factor. These delays were measured in an AMD Athlon XP running at 2095 MHz (a low-end desktop processor by today's standards), and include the generation of the messages to be forwarded to the next hop. There were roughly 7 refresh messages for each initial *SResv*, on average. About 1800 messages were processed per simulated second at intermediate nodes with load factor 33.3. The results exhibited an unexpected behavior: even though no piece of code inside the *SResv* message processing routine depends on the number of simultaneous

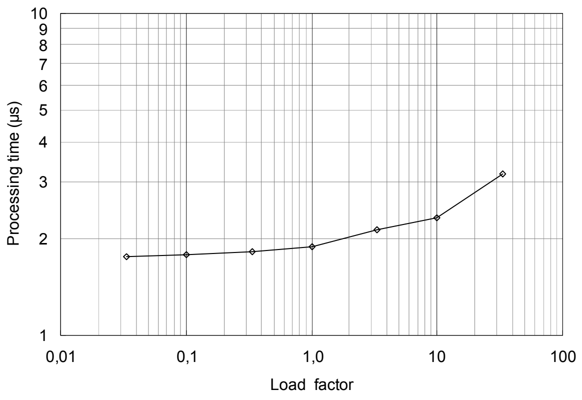


**Figure 4.13: Average processing delay of *SResv* messages**

flows, the measured processing delay has increased with the load factor. The routine was double-checked for such code (which would be an implementation error), but none was found. Therefore, we attribute this behavior to an increased cache miss ratio due to the loss of locality of reference. Notice that even though our routine is independent on the number of simultaneous reservations or the number of packets "in flight," other code in ns-2 is temporally and spatially dependent on those (and other) factors; this problem is further aggravated by the fact that more than 30 nodes are being simulated inside a single processor. It is worth noting that, contrary to an algorithmic dependence on the number of flows, performance decrease due to increased cache miss ratio will have a limited growth — in the limit, it would stop growing at the point where the cache miss ratio is 100%, and all data is retrieved from main RAM.

From the message processing delay results, it is easy to compute the CPU load associated to processing *SResv* messages. If there are $10^5$ simultaneous flows and a refresh *SResv* is sent for each flow every 10 s, about $10^4$ *SResv* messages will be processed per second, provided the average duration of the flows is substantially larger than the refresh period. The approximate CPU occupation for processing *SResv* messages is $T_{SResv}/10^4$, where $T_{SResv}$ is the average processing delay for an *SResv* message. If $T_{SResv} = 3$ µs, then the CPU will spend 3% of the time processing *SResv* messages. However, we believe that in a real implementation the delay would be lower: the evaluated code was a simulation prototype written in C++ and compiled without any optimization, and more than 30 nodes were simulated on a single processor, which translates into an increased cache miss ratio. Moreover, while the simulator process was run with soft real-time priority for the evaluation, it could still be preempted by interrupt handlers, adding to the delay. The profiling code itself also takes some cycles, though this factor has a negligible effect. In a real implementation using hand-tuned C code at kernel level, we believe the processing times would be significantly lower without any modification to the signaling protocol. Further improvements could still be obtained by a simplification of the protocol using fixed format messages, which are much simpler to parse, instead of the object-oriented RSVP extension approach.

## *4.4 Comparison between SRBQ and RSVPRAgg*

This section presents a comparison between SRBQ and RSVPRAgg, the IETF proposal for a scalable aggregation-based QoS architecture with reservations. We analyze the QoS guarantees, resource utilization and scalability of both models in order to evaluate their relative merits and shortcomings, as well as their suitability to replace the reference RSVP/IntServ architecture, which suffers from scalability problems that disallow its usage in high traffic core networks. We perform both a qualitative comparison, based on the nature of the architectures, and a quantitative comparison, based on simulation results.

A mapping between the QoS architectures needs to be performed in order to obtain a meaningful comparison. The aggregation regions of RSVPRAgg and the non-aggregated RSVP regions correspond to the core and access domains of SRBQ, respectively; the aggregators and deaggregators in RSVPRAgg correspond to edge routers in SRBQ. The network topology used in the simulations is the same given this mapping.

### 4.4.1 Qualitative Comparison

Both the RSVPRAgg and the SRBQ model aim at providing QoS levels comparable to those of the RSVP/IntServ model but, unlike this one, in a scalable manner. Resorting only to the nature of the architectures, we may state that both are able to support strict and soft QoS guarantees, since no flow (not belonging to the best effort class) is admitted in the network when the available end-to-end network resources are not sufficient. The admission control algorithm for the CL classes is token bucket based in RSVPRAgg and average rate watermarks based in SRBQ. Both of these architectures make use of aggregation of flows in order to achieve high efficiency in packet classification and scheduling, and both of them use the DSCP field of the IP header to this end. The approach to reducing the signaling processing load is radically different, though.

The core routers in the RSVPRAgg architecture only need to store the state of each aggregate. Contrasting to this low amount of state, the one stored in the SRBQ architecture is per-flow. However, considering the available routers nowadays, this does not seem to be a limiting factor of the SRBQ architecture, as previously demonstrated. A more scalability-limiting factor at the core routers would be the lookup of the stored flow information, based on the 5-tuple parameters that specify the flow, when the number of flows is very high. Since the SRBQ architecture overcomes this problem by using the labels, the existence of per-flow reservation structures is not a limitation. The scalability of the classification and scheduling procedures at the core nodes is similar in both architectures, since they are performed on a per-aggregate basis, according to the DSCP of the flows. At the edge routers, classification is much lighter in the SRBQ model, since no per-flow mapping needs to be performed. As scheduling on both architectures is performed per-class, using simple queuing disciplines in the underlying DiffServ-like infrastructure, the efficiency is comparable. At the access routers, packet classification is performed per-flow in both models. The processing load involved for classification is, therefore, similar; however, it may be highly reduced in SRBQ if labels are used in data packets (which is always true in the GS class). Packet scheduling is usually more efficient in SRBQ, since it is based on the DSCP, whereas in RSVPRAgg, at the access routers (therefore outside the aggregation regions), usually Weighted Fair Queuing (WFQ) or a similar discipline is used.

One disadvantage of the SRBQ model is the mandatory shaping of the GS class at every router, and the ingress shaping of this class at the edge routers. Notice, however, that since the whole class is treated as a single flow, shaping may be performed very efficiently, and may trivially be implemented in dedicated hardware.

Both models have a soft-state approach to reservations, meaning that a reservation that is neither explicitly terminated nor refreshed for a certain period will timeout and be removed. This soft-state character provides them with adaptability to changing network conditions (e.g., route changes) and failures which inevitably occur in a dynamic Internet. In order to efficiently implement reservation expiration timers, a specific timer algorithm was developed for SRBQ having O(1) computational complexity. RSVPRAgg, like RSVP, makes use of generic timers, but has a much lower number of (aggregate) reservations to manage at core nodes. At edge nodes, however, reservations are per-flow, and generic timers impose a scalability limitation to RSVPRAgg.

The approach at making signaling processing scalable is very different in these two architectures. In SRBQ the end-to-end character of reservation signaling is retained, and scalability is achieved by the use of computationally efficient techniques and algorithms (like labels and efficient timers). In RSVPRAgg, end-to-end reservations are aggregated, reducing the signaling processing at the core to that needed to maintain and update aggregate reservations, whose updates are performed in bulk quantities in order to reduce the signaling load. This approach has two disadvantages: (1) the number of signaling messages processed at the edge nodes of the aggregation region, which may be a high-traffic transit domain, is even higher than in the case of regular RSVP, since both end-to-end and aggregate messages must be processed; worse, packet classification at the edge is per-flow; and (2) the reduction in signaling is highly dependent on the bulk size, but large bulk sizes lead to a very poor utilization of network resources.

In the SRBQ architecture the utilization of network resources is very high since the reserved bandwidth always matches the bandwidth of the admitted traffic. The aggregation efficiency in the RSVPRAgg architecture depends on the matching between the admitted traffic and the aggregate reservation. If the bulk bandwidth is too large, the signaling load is highly reduced but the network resources can be highly under-utilized. On the other hand, with a small bulk size, the aggregate reserved bandwidth is close to that of the admitted traffic (resources are not wasted), but the signaling load is very high, similar to that of per-flow reservations.

Concerning reservation setup delays, they may be divided in two components: (1) the delay related to message exchange between routers and (2) the processing time for these messages. In the SRBQ model, the first component of the delay is one end-to-end round trip time. In RSVPRAgg, it is equivalent to one end-to-end round trip time in the best case, when the corresponding aggregate already exists and has enough bandwidth to accommodate the end-to-end flow; when the aggregate exists but has not enough bandwidth, a round trip time for the aggregation region is added; when the aggregate does not yet exist, two round trip times for the aggregation region must be added. The second component of the delay is more difficult to evaluate. In the SRBQ model it is approximately fixed for a given topology. In RSVPRAgg it depends on several factors: (1) the need to modify the aggregate bandwidth; (2) the ratio of routers in the aggregation region to the total number of routers in the end-to-end path; (3) the number of end-to-end flows at the edge routers of the aggregation region and at the routers outside this region; (4) the efficiency of the hashing functions used, and other factors. In the end, we expect the high efficiency of signaling message processing in SRBQ model to compensate the higher number of signaling messages processed.

### 4.4.2 Quantitative Comparison

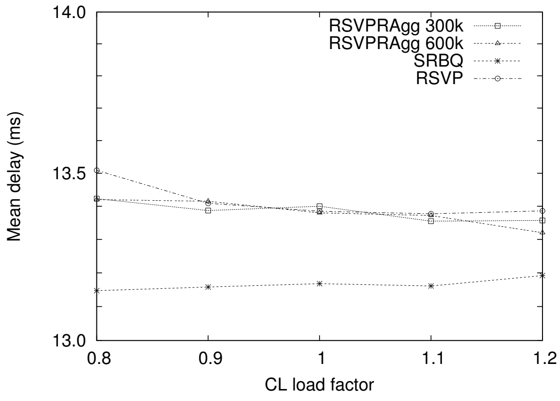
As has been previously mentioned, we have implemented the SRBQ and RSVPRAgg models in the ns-2 simulator. We will now present a quantitative comparison of these architectures based on a series of simulation experiments performed with these implementations. An existing implementation of RSVP for this simulator [Greis98] was also used to obtain reference results.

These models have some adjustable parameters. In RSVP and RSVPRAgg, the R parameter (average refresh period) used was the default of 30 s. The reservation expiration timer in SRBQ was chosen so that refreshes would be sent every 32 s, the closest value. The target utilization factors for the 3 watermarks in the SRBQ model were adjusted to 0.999, 1.0 and 3.0. This ensures that using the reservation parameters given below for each set of simulations, admission control would be performed based on the second rate watermark. In the RSVPRAgg model there are two tunable parameters related to aggregate bandwidth management: we used a value of 15 s for the bulk release delay timer, related to hysteresis, and a value of 5 s for the hold timer, which prevents repeated failed attempts at increasing the bandwidth of a given aggregate. Simulations with the RSVPRAgg model were performed with two different bulk sizes: 300 kbps and 600 kbps. The admission control used in all models is parameter-based (PBAC).

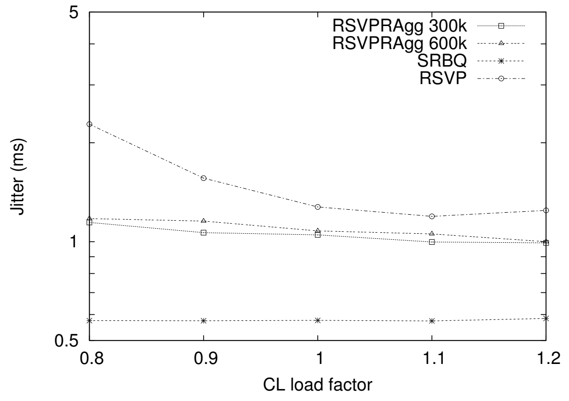
The topology used in these simulations is the same we used for the evaluation of RSVPRAgg, and is illustrated in fig. 3.6 and described in section 3.3 of the previous chapter (p. 91). Once again, it is worth noting that with 3 ingress and 3 egress routers in the aggregation region (left to right), the number of end-to-end aggregates in this domain is 9. Three traffic classes were simulated: signaling, CL and BE. We assigned 1 Mbps to the signaling class and 7 Mbps to the CL class. The remaining bandwidth, as well as unused CL and signaling bandwidth, is used for BE traffic. Traffic belonging to the CL class is a mixture of different types of flows: CBR, exponential on-off and Pareto on-off. These flows are initiated according to a Poisson process with a certain mean time interval between calls (MTBC), and flows' durations are exponentially distributed. Filler traffic in the BE class is composed by on-off Pareto and FTP flows. The simulation results are presented in the next subsections.

#### *4.4.2.1 Single Flow Type*

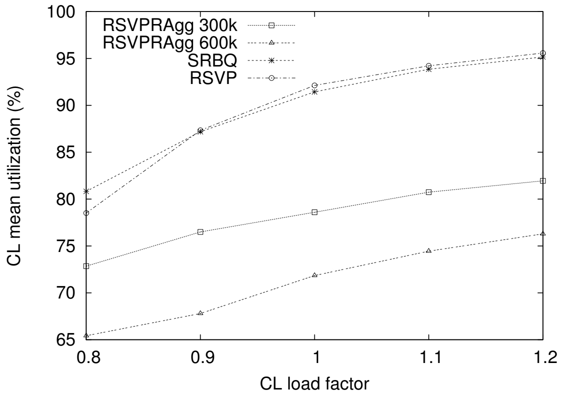
In the first set of comparative simulations we used, in the CL class, only 64 kbps constant bit-rate (CBR) flows with a packet size of 500 bytes. This is done to completely avoid the unfairness originated by disproportionate rejection rates between flows of different types in different models. The average flow duration is 120 s, and the mean time between calls is adjusted to vary the offered load factor between 0.8 and 1.2 times the bandwidth allocated to the CL class at the core link. The results from this set of simulations are presented in fig. 4.14. We consider two different sets of results in RSVPRAgg: with a bulk size of 300 Kbps (RSVPRAgg 300k) and a bulk size of 600 Kbps (RSVPRAgg 600k).



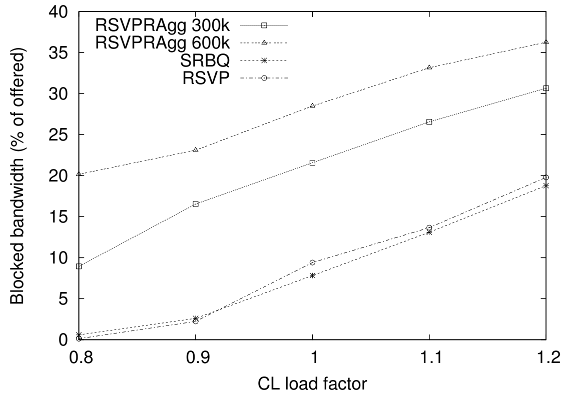
a) Mean delay



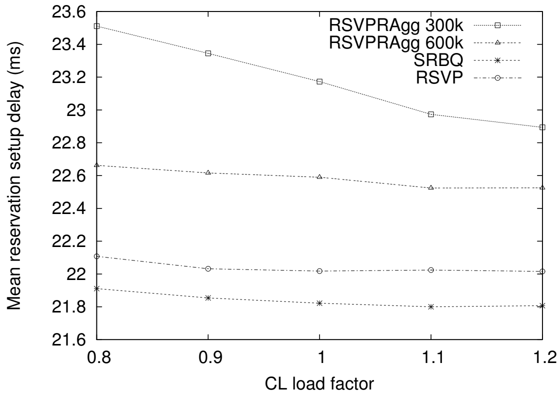
b) Jitter



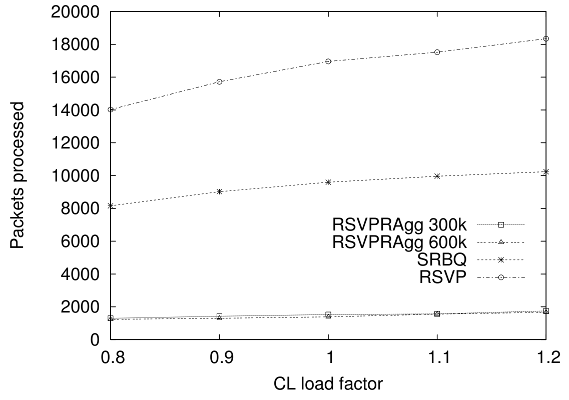
c) CL class utilization



d) Blocking probability



e) Reservation setup delay



f) Processed signaling packets at core node 1

**Figure 4.14: QoS and performance results of SRBQ, RSVPRAgg and RSVP with varying offered load factor, and the same type of traffic flows**

In all models, the mean delay is not much higher than the sum of transmission and propagation delays (12.08 ms), meaning that the time spent in queues is low. Nevertheless, it is lower in SRBQ. Jitter is also lower in the SRBQ model. These results are probably due to the use of WFQ with quite large per-queue limits in RSVP and in RSVPRAgg outside the aggregation domain. In all models presented there are no losses, since the reserved bandwidth of the CBR flows is equal to the maximum required bandwidth, being sufficient to accommodate the accepted flows. Regarding the utilization of bandwidth allocated to the CL class, it is much higher in SRBQ (similar to RSVP) than in RSVPRAgg. In the former, the curve shows saturation around the offered load factor of 1, whereas in the latter there is no visible saturation. In RSVPRAgg, the utilization is very noticeably lower with a bulk size of 600 kbps than with a bulk size of 300 kbps. Notice that a bulk size of 600 kbps is less than a factor of 10 higher than the flow rates. This suggests that the use of larger bulk sizes in order to increase the scalability would lead to very poor network resource utilization. Corresponding to the lower utilization figures, the blocked bandwidth in the RSVPRAgg model is higher than in SRBQ, and is higher when using larger bulk sizes. In SRBQ the blocked bandwidth figures are similar to regular RSVP, since in both end-to-end reservations are accepted up to the bandwidth reserved for the CL class, contrasting to the RSVPRAgg model in which end-to-end reservations are only accepted up to the reserved rate of the corresponding aggregate.

The reservation setup delay is another evaluated parameter. In SRBQ, it was measured from the instant the reservation procedure is triggered by sending an *SResv* message to the instant the corresponding *SResvStat* message is received at the sender, confirming the successful establishment of the reservation. In RSVPRAgg and regular RSVP it is measured from the instant the first *Path* message for the flow is emitted to the instant the corresponding *Resv* arrives at the sender, indicating the existence of the reservation along the whole path, given that the receiver sends a *Resv* message as soon as it receives the first *Path* message for a new flow. Since the ns-2 simulator does not measure message processing times, the values presented correspond only to the signaling packet exchange. The lowest setup times correspond to SRBQ and RSVP, since in both models they correspond to a round trip time for the messages, being larger in RSVP due to the larger message size. In RSVPRAgg, on the other hand, when an aggregate does not exist or its bandwidth needs to be updated, additional messages are exchanged inside the aggregation region, leading to higher end-to-end reservation setup delays, which increase with decreasing bulk sizes. The reduction in reservation setup time with increasing offered load factor and a bulk size of 300 kbps is explained by the fact that more flows are being accepted into aggregates without modifying their bandwidth, since a higher portion of those requiring bandwidth increase are rejected. Notice that if processing times were measured, the reservation setup delay of regular RSVP would be much higher than in the other two models.

The last parameter evaluated in this test is the number of signaling messages processed at core node 1 (refer to fig. 3.6 in the previous chapter). The average refresh interval used for both RSVP and RSVPRAgg is the default of 30 s; the SRBQ expiration timer was chosen so that refreshes are sent every 32 s, the closest value. By a wide margin, the number of messages processed at the core is much lower in RSVPRAgg (about 1500 packets on average during the 3600 useful simulation seconds) than in SRBQ or RSVP (respectively, about 9000 and 16000 packets processed under the same conditions). This is an obvious result due to the fact that at interior nodes only aggregate messages are processed. The almost twofold difference between SRBQ and RSVP is due to the fact that in RSVP both *Path* and *Resv* refreshes are needed.

From the number of processed messages only, the RSVPRAgg model would be the clear winner in terms of signaling processing scalability. Keep in mind, though, that one big strength of the SRBQ model is the use of low complexity, highly efficient algorithms (labels, timers, etc.), which translate in much less CPU time used to process each message.

#### 4.4.2.2 Multiple Flow Types

The second set of comparative simulations is meant to evaluate the behavior of the different models in presence of multiple flow types with different characteristics in the CL class. The set of flow types used is shown in table 4.4. There are two types of CBR flows with rates of 48 kbps and 64 kbps, one exponential on-off type and one Pareto on-off type, both with an average transmission rate of 48 kbps, average on and off times of 200 ms, and peak transmission rate of 96 kbps. The reservation parameters for RSVPRAgg and RSVP (token bucket) and SRBQ (3 watermarks) are shown in the table. The average flow duration for all types is 120 s, and the mean time between calls (MTBC) is adjusted so that all flow types have, on average, the same amount of reserved bandwidth. Also, the amount of offered load is varied from 0.8 to 1.2 (load factor) times the CL bandwidth at the core by changing the MTBC in all flows. The MTBC value presented in the table corresponds to a load factor of 1.2.

| Type | Avg. rate (kbps) | Pkt. size (Bytes) | On (ms) | Off (ms) | Pk. rate (kbps) | Token Bucket | | Watermarks (kbps) | | | MTBC (s) | Avg. dur. (s) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | | | | | | R (kbps) | B (Bytes) | 1 | 2 | 3 | | |
| cbr48cl | 48 | 500 | | | | 48 | 1500 | 48 | 48.048 | 56 | 11 | 120 |
| cbr64cl | 64 | 500 | | | | 64 | 1500 | 64 | 64.064 | 72 | 14.6 | 120 |
| exp1cl | 48 | 500 | 200 | 200 | 96 | 64 | 15000 | 32 | 64 | 96 | 14.6 | 120 |
| pareto1cl | 48 | 500 | 200 | 200 | 96 | 64 | 15000 | 32 | 64 | 96 | 14.6 | 120 |

**Table 4.4: Flow characteristics for the multiple flows test**

Figure 4.15 contains the most relevant results for this test. The mean delay is shown for all flow types in SRBQ and RSVPRAgg with a bulk size of 300 kbps. As can be seen, all flows in SRBQ suffer similar average delays, which is an obvious result since they share all the queues. In RSVPRAgg the average delay inflicted in different flow types is different, which is due to the use of WFQ outside the aggregation region; Pareto flows in this model have significantly higher queuing delays than the other flows. Regarding packet losses, we may observe in the figure that, contrary to all other types, Pareto flows have a very significant packet loss ratio of about 10% in the RSVPRAgg model. It is important to keep in mind that the Pareto distribution is a heavy-tailed one, with infinite variance. This implies that Pareto on-off flows are not well suited for token bucket characterization, since unless we use disproportionately large bucket sizes, packet losses will always be high. The three rate watermarks characterization used in SRBQ for the CL class is much more appropriate for this kind of flows: losses for Pareto flows in SRBQ are always less than 0.003%. Packet losses for exponential on-off flows are very low in RSVPRAgg (about 0.003%) and null in SRBQ. All CBR flows have no packet losses in both models.

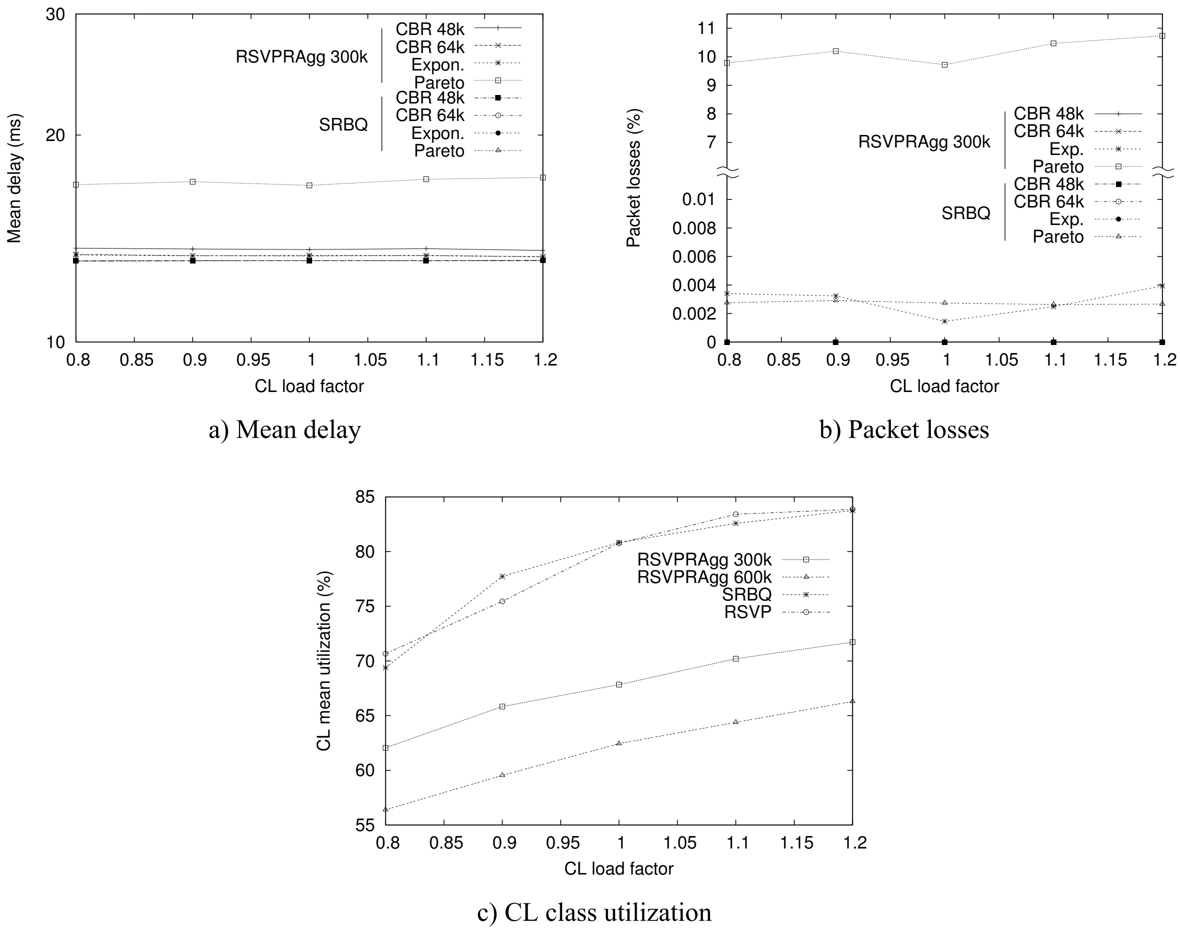


**Figure 4.15: QoS and utilization results of SRBQ and RSVPRAgg with varying CL load factor, and several types of traffic flows**

Link utilization in this test follows the same pattern as in the previous one, although with lower values. This is explained by the fact that two flow types, exponential and Pareto, perform reservations with a rate that is higher than their actual average transmission rate. Since we are using parameter-based admission control, the CL bandwidth available at the core link is an upper bound to the sum of reservations, not the actual traffic. SRBQ and RSVP have about the same utilization figures, showing link saturation around an offered load factor of 1; RSVPRAgg has lower utilization values which decrease with increasing bulk size, and does not exhibit saturation.

From the previous results we may conclude that both models provide adequate QoS, except for Pareto ones in RSVPRAgg, which are not well suited for the token bucket type reservations used in that model.

#### 4.4.2.3 Bursty Flows

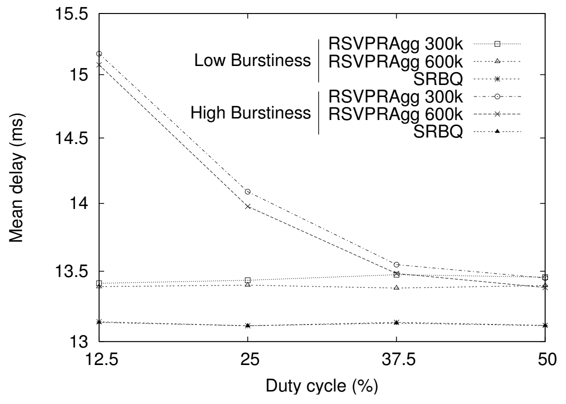
With this set of simulations we evaluate the behavior of the different models in the presence of misbehaved flows. We measure not only the quality of service received by these bursty flows but also the impact in the other flows. Table 4.5 shows the characteristics of the CL flows used in this experiment: cbr64cl (well behaved flows), exp1cl (nearly well-behaved flows) and exp2cl (misbehaved flows). The peak rate of the misbehaved flows varies between 96 kbps (duty cycle of 50%) and 384 kbps (duty cycle of 12.5%) in order to keep the average rate constant at 48 kbps.

| Type | Avg. rate (kbps) | Pkt. size (Bytes) | On (ms) | Off (ms) | Pk. rate (kbps) | Token Bucket | | Watermarks (kbps) | | | MTBC (s) | Avg. dur. (s) |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | | | | | | R (kbps) | B (Bytes) | 1 | 2 | 3 | | |
| cbr64cl | 64 | 500 | | | | 64 | 1500 | 64 | 64.064 | 72 | 11 | 120 |
| exp1cl | 48 | 500 | 200 | 200 | 96 | 64 | 15000 | 32 | 64 | 96 | 11 | 120 |
| exp2cl | 48 | 500 | var | var | var | 64 | 15000 | 32 | 64 | 96 | 11 | 120 |

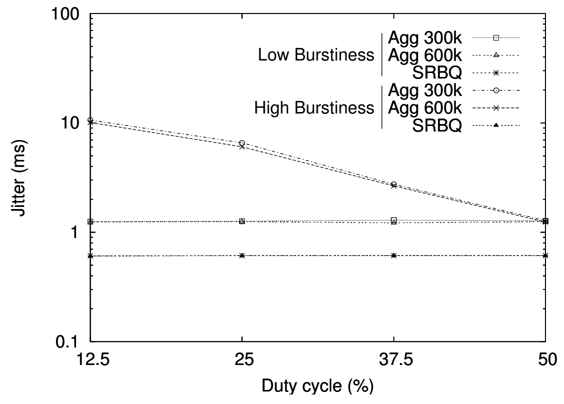
**Table 4.5: Flow characteristics for the burstiness test**

Each of the 4 transmitting terminals generates flows with these parameters, in the CL class. The total mean offered load at the core link is, therefore, 20% larger than the bandwidth allocated to CL, in terms of reserved rate.

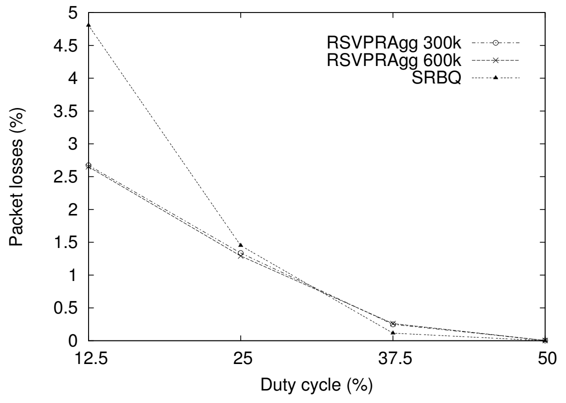
Figure 4.16 shows some results from these simulations. As observed, the mean delay for misbehaved flows is not affected by their burstiness in SRBQ (where it is mostly the sum of transmission and propagation delays), contrary to the other models. This is due to the fact that in the SRBQ model all CL data packets share the same queue. Notice, however, that in this model the traffic is policed before entering the domain. On the other hand, we may observe that in all models highly bursty flows have no noticeable impact in the delay of the low burstiness flows; the delay of CBR flows (not shown) is not affected either. Jitter results are similar to delay ones, though with proportionally larger variation in high burstiness flows in RSVPRAgg.



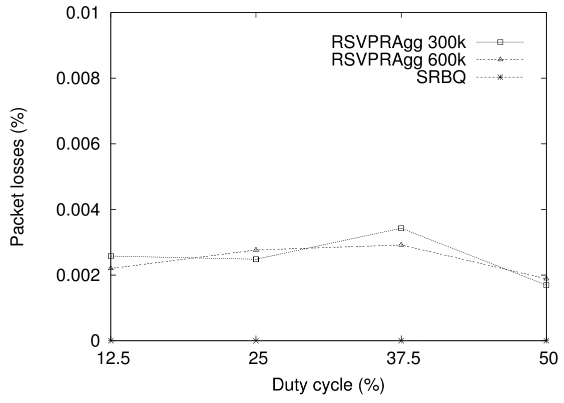
a) Mean delay



b) Jitter



c) Losses for high burstiness flows



d) Losses for low burstiness flows

**Figure 4.16: Delay and packet losses of SRBQ and RSVPRAgg with varying burstiness of misbehaved flows**

Misbehaved flows are heavily penalized in terms of packet losses. In SRBQ, packet losses for these flows almost reach 5% with a duty cycle of 12.5%, while they are about 2.6% in RSVPRAgg. The high loss values are due to the fact that these flows transmit at rates much higher than they reserved for considerable periods of time. A relatively large bucket size absorbs these bursts up to some level in RSVPRAgg, but in SRBQ the reservations for CL traffic have no bucket parameter, only 3 rate watermarks, of which even the third one is much lower (96 kbps) than the peak transmission rate (384 kbps). Clearly, these flows are violating their contracts, so measures must be taken against them to prevent QoS degradation for other flows. SRBQ is inflicting higher penalizations, better protecting other flows. Loss figures for other flows are not affected by the misbehaved ones. They are very low in RSVPRAgg and null in SRBQ for low burstiness exponential flows and are null in all models for CBR flows (not shown).

These results show that all models are able to provide adequate QoS levels to flows respecting their traffic contracts even in presence of misbehaved flows. The service of these flows is degraded in order to protect the well behaved flows: in SRBQ this degradation is only in terms of packet losses; in RSVPRAgg they are less penalized in terms of losses, but also have increased delay and jitter.

## 4.5 Conclusions

In this chapter we proposed and evaluated the Scalable Reservation-Based QoS architecture, a QoS model where scalability is achieved by using exclusively low computational complexity algorithms whose execution time is independent on the number of simultaneous flows. SRBQ combines aggregate packet processing (classification, scheduling, policing and shaping) with a scalable model of per-flow signaling-based resource reservation. The underlying architecture is based on DiffServ: individual flows are aggregated into service classes, mapped to DiffServ-compatible per-hop behaviors, and aggregate classification is performed based on the DS field of the packet header. Two service classes are provided: the Guaranteed Service class, with strict QoS guarantees, and a Controlled Load class, more tolerant to burstiness. Some techniques were developed in order to ensure that per-flow resource reservation signaling scales up to a very large number of simultaneous flows, namely a label mechanism that provides routers with direct access to resource reservation information of the flows, and an efficient implementation of soft state expiration timers.

Based on simulation results from our implementation of SRBQ in the ns-2 simulator, we have analyzed the standard QoS parameters (delay, jitter and packet loss ratio) and the per-class network resource utilization in different experiments, using both synthetic and real-world multimedia streams. The results show that in spite of being based on aggregation, SRBQ is able to support both strict (GS) and soft (CL) QoS guarantees, and provide adequate isolation of in-profile flows from out-of-profile ones, much stronger in the GS class. Out-of-profile CL flows are more penalized in terms of loss probability, while out-of-profile GS flows are essentially penalized in terms of delay due to ingress shaping. The CL class is appropriate for (eventually bursty) traffic which tolerates some packet loss. In fact, being based on average rates, this class is much more tolerant to bursts, and, since there is no shaping, no additional delays are inflicted to bursty flows. The GS class, on the other hand, is ideal for flows intolerant to packet losses and having well defined traffic envelopes which they do not exceed, but heavily penalizing for non-conformant flows.

The processing delay of reservation messages was also evaluated, in order to compute the expected processor load it imposes. Although the results, obtained by measuring the execution delay of the C++ method that processes *SResv* messages in the simulator code, were not entirely conclusive, we have enough reasons to believe that a real-world, optimized implementation would be scalable enough to cope with the huge number of simultaneous flows traversing a high-speed core router.

A comparative analysis of SRBQ against the RSVP Reservation Aggregation architecture, proposed by the IETF as a scalable alternative to RSVP/IntServ that retains the character of a signaling-based approach (as opposed to DiffServ), has shown that even though both models are able to provide adequate QoS in a scalable way, most QoS results are favorable to SRBQ and, more important, with SRBQ they are achieved with a much improved usage of network resources.

# CHAPTER 5

# DAIDALOS APPROACH TO QUALITY OF SERVICE

In the previous chapters we have analyzed an existing scalable architecture for providing QoS on IP internetworks and proposed a new one with improved characteristics. Although based on the aggregation of individual flows, at least partially, both of these architectures embody distributed models, where network elements along the data path (routers) share the responsibility for ensuring that enough resources are available to the flows for providing adequate QoS; there is no dedicated control element centralizing decisions or actions. This design principle is consistent with the traditional, decentralized view of the Internet, where a series of hierarchically flat network elements provide transport service for data packets, in this case with QoS support as added value. The Internet, nevertheless, is rapidly shifting away from this paradigm, and becoming the universal and unified telecommunications infrastructure, providing communication services that go far beyond data transport for which it was originally conceived. These services include those previously supported by separate and dedicated infrastructures — telephony, broadcasting — and new, previously inexistent ones — e.g., online gaming. Traditional telecom operators are migrating to this new, converged platform and, as they shift to the Everything over IP (EoIP) paradigm, they bring along some of their established practices, such as the use of out-of-band signaling and of centralized equipment for processing signaling messages[1] and keeping track of calls. Therefore, a QoS architecture based on Bandwidth Brokers (or, more generally, on QoS

[1] Quasi-associated signaling is the preferred mode in SS7 (Signaling System #7).

Brokers) is, understandably, more appealing to these new players. Centralized control entities are easier to manage and have a lower cost, as the intelligence does not have to be replicated on every router. Moreover, a Bandwidth Broker-based design provides an easier upgrade path, as functionality may be added by changing only a few control entities, not the whole routing infrastructure. A further advantage of the use of Bandwidth Brokers (BBs) is simplified design, as centralized algorithms are usually simpler than their distributed counterparts. The main drawback of BB-based architectures is the fact that BBs are single points of failure; however, the potential resilience problem posed by single points of failure may be overcome by other means, like the use of failover redundant BBs.

Another aspect in which the new and converged infrastructure differs from the legacy Internet is the access. Traditional access through a wired Local Area Network (LAN) or via modem has no support for mobility — there is a fixed link with constant characteristics connecting terminal nodes to the network. However, the incorporation of a multitude of wireless access technologies is gradually providing users with ubiquitous access to the network, allowing them to move freely without disruption in the service, even though access conditions may vary widely. This change implies that network topology may no longer be considered flat, as the access links are clearly distinct in every aspect from the rest of the infrastructure.

The second part of this thesis, consisting on chapters 5–8, describes work we performed in the scope of phase 1 of the DAIDALOS project [Daidalos], mostly concerning the QoS subsystem of the proposed network architecture. Our contributions include the high-level design of this subsystem and the specification and analysis of some of its layers, including resource management and signaling in the access, as well as inter-domain QoS support. We have also worked on the optimization of session setup signaling in mobile scenarios. In this chapter we introduce the DAIDALOS approach to QoS provisioning. We begin with an overview of the project, its vision, goals and development model in section 5.1. The QoS-related subset of the network architecture is described in section 5.3. Section 5.4 provides an overview of how end-to-end QoS control is achieved within DAIDALOS, and section 5.5 introduces the different aspects of the network and problems that are dealt with in the next chapters.

## 5.1 The DAIDALOS Project

Designing Advanced network Interfaces for the Delivery and Administration of Location-independent, Optimized personal Services (DAIDALOS) is an EU-funded

Integrated Project dedicated to the design of advanced network infrastructures and access technologies for location-independent, personalized communication services. The DAIDALOS vision is to seamlessly integrate heterogeneous network technologies in order to achieve an open architecture based on a common IPv6 network protocol. Network operators and service providers will be enabled to provide intelligent access combined with dynamic service provisioning, supporting a wide range of voice, data and multimedia services. Several business models and business process interactions are supported from an architectural point of view. The new pervasive network and communication infrastructure developed by DAIDALOS enables every user to benefit from customized communication services, such as personalized newscasts tailored to their own individual preferences.

The design of the DAIDALOS architecture is based on five key concepts, which may be summarized as follows:

- Mobility Management, AAA, Resource Management, QoS and Security (MARQS), supporting functional integration for end-to-end services across heterogeneous technologies
- Virtual Identity (VID), which separates the user from a device, thereby enables flexibility as well as privacy and personalization
- Ubiquitous and Seamless Pervasiveness (USP), enabling pervasiveness across personal and embedded devices, and allowing adaptation to changing contexts, movement and user requests
- Seamless Integration of Broadcast (SIB), which integrates broadcast at both the technology level, such as DVB-S/T-H, and at the services level, such TV, carousels and datacast
- Federation, which allows network operators and service providers to offer and receive services, allowing players to enter and leave the field in a dynamic business environment

Besides these five key concepts, there are two different, competing and cooperating, development guidelines. The first one is user control: DAIDALOS acknowledges that the future of telecommunications will ultimately empower the user with the choice of service usage. Service delivery, however, is not a detail that should pertain to the user, and thus, the second guideline is a bias towards telecoms, reflecting their vision of potentially realistic scenarios for the telecommunications industry. Thus, most of the visions are associated with the existence of large telecom operators as business entities on the process of service creation and delivery.

A scenario-based design methodology has been chosen in the project — technical requirements are derived from user-centric scenarios (real life descriptions of communication-based activities). These scenarios are used in an iterative process of refining the requirements for system and architecture design. Two major scenarios are used in this process. The first one, "Mobile University", involves foreign students having access to their personal set of services and able to dynamically discover local services and devices. This scenario addresses several aspects such as: the organization of everyday life at the university (friends, appointments and reservations, classes, projects, exams, entertainment); locating people and devices, checking availability, discovering local services; searching for the best and/or cheapest available infrastructure; personal broadcasting (e.g., of classes and speeches). The second one, the "Automotive" scenario, involves mobility-supporting services in and around the vehicle, with aspects of personal multimedia, ad-hoc mobile networking and session mobility. This scenario addresses aspects such as: access to personal information and services inside and outside the vehicle; locating and detecting presence; service and content adaptation based on QoS across network and operator boundaries; session mobility between terminals (including vehicles), and across organizational and operational domains; broadcast services for entertainment, inter-vehicle safety, and regional traffic information services. Together, these scenarios are highly representative of a broad variety of education, entertainment and business situations in the real mobile world.

Since DAIDALOS aims at the development of a network and service architecture, the views of the telecom-operators currently deploying such architecture have to be considered. The architecture has a set of characteristics coming from this background: flexibility, a layered structure, optimization capabilities, and the possibility of being instantiated in different business scenarios. Services may be produced at any point of the network — service providers are not expected to be in specific locations in the network, though they are expected to have specific business relationships with the transport operator, identifying the objective contractual relationship expected between the two. However, DAIDALOS acknowledges that several typical major players will still exist in the future (e.g., mobile telephony operators, TV broadcasters), and has explicitly identified control elements for some of these players (notably mobile operators, the potential core business). Overall, the direction of the architecture led to the development of a universal, pervasive environment concept, with integrated personalization as a keynote, conciliating users' choices and operators' interests.

The project is divided in five work packages, following a horizontal approach to the system development. Three main work packages handle different aspects of the problem:

WP2 addresses the access network issues; WP3 addresses core network issues and SIP-based services; and WP4 designs the overall pervasive support intelligent system. The two remaining work packages address the design of the global architecture of the system at a higher level (WP1) and the integration of the work from the different work packages and the evaluation of the developed system (WP5). Our research and development work in DAIDALOS, described in this and the next three chapters, has been performed in the scope of WP3.

## 5.2 DAIDALOS Architecture

The DAIDALOS architecture relies on IP technologies, DiffServ- and multicast-aware, and includes fast mobility schemes, integrated with authentication and QoS. This provides for an economic transport layer across multiple technologies and a stable development layer for application developers. The architecture adheres to the vision of a future horizontalization of service provisioning, providing large sets of functionality which are independent of the physical layer technology. The support of broadcast technologies aims at seamlessly integrating traffic from two different and currently separate businesses, mobile communications and broadcasting, paving the way to a novel, user-oriented service platform.

Figure 5.1 shows a high level view of the DAIDALOS architecture and its main

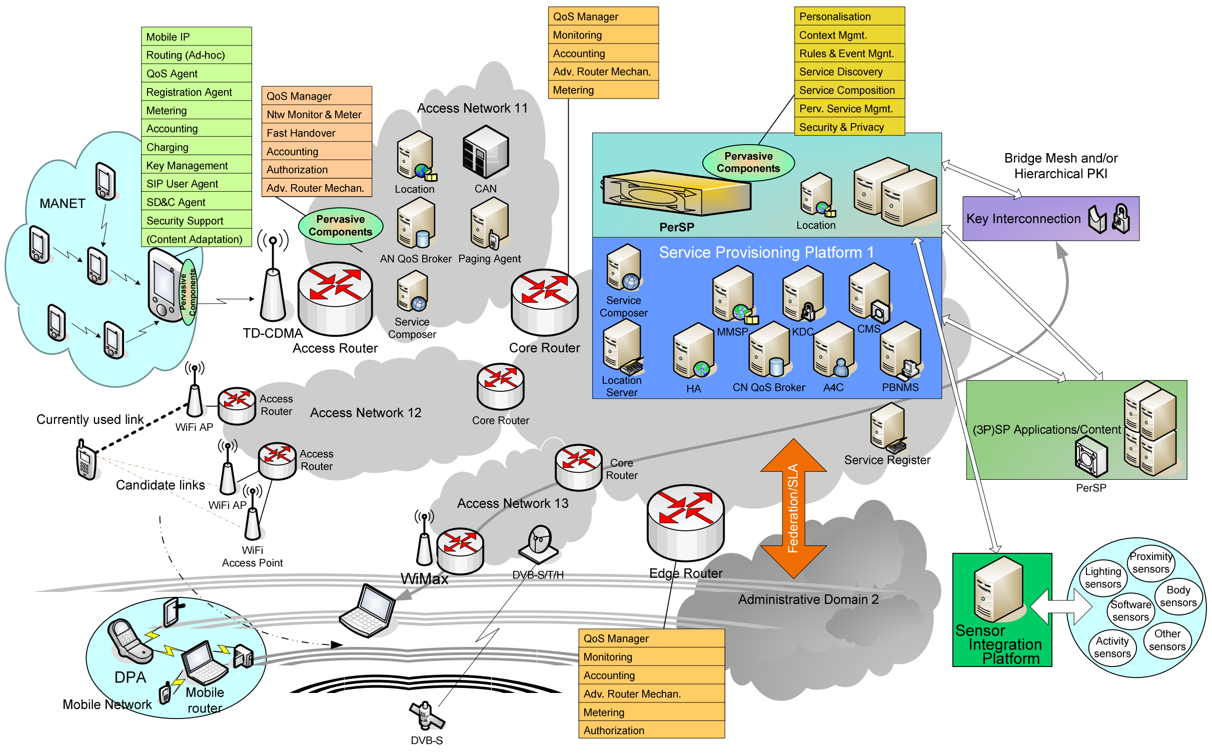


**Figure 5.1: Overall DAIDALOS architecture**

components. The architecture is quite complex, as it is expected to support virtually almost any economically sensible telecommunications scenario and provide for economic deployment of traditional communications scenarios. The figure mostly represents one administrative domain. This domain may be federated (or have some sort of SLA established) with other domains. The administrative domain is separated in core and access networks. Three different access networks with wireless access are represented. Supported technologies include WiFi, WiMax (even if no mobile standard exists, DAIDALOS is already integrating this technology in its portfolio), TD-CDMA and overall Access Cells with DVB (both Satellite and Terrestrial, though Daidalos has mostly focused on DVB-T).

Access Networks (ANs) are structured in cells. Each cell is controlled by an Access Point (or a Base Station in some technologies), and an Access Router (AR) controls a set of cells. DAIDALOS allows for the same AR to control several different APs for economic reasons (both equipment cost and wireless performance).

Several components may be deployed in the AN. The AN QoS Broker performs admission decisions and ensures Quality of Service. A Paging Controller allows power saving in wireless environments. A Content Adaptation Node (CAN) may also be present to perform multimedia codec adaptation whenever necessary.

Two other entities are deployed in the AN: a Service Composer, able to dynamically compose the user-requested services, and a Location Controller, the physical support of location — using diverse types of technologies, it provides the current location of the user to a central location server. This is an aspect where privacy must be considered, and complex rules may be in place regarding who can and who cannot access this information (part of the DAIDALOS pervasive environment support). Note that Service Composer may need location information for providing location-based services.

The architecture supports the use of Mobile Ad Hoc Networks (MANETs) for extending the network coverage. Typically, at least one of the devices in the MANET will also be connected to the access network, providing the others with access to the global Internet and other operator services. Mobile Networks, where a series of devices and a Mobile Router though which they are connected travel as a single entity[2], are also supported. Even though similarities exist between the MANET and Mobile Network concepts, there are differences that imply different technology support.

[2] Typically, this can be a car and the whole set of devices inside.

On the core network, interconnected by routers, the operator holds a Service Provisioning Platform. This platform provides a large set of functions required for the efficient provision of telecommunication services:

- A Location Server, central repository for user location
- A Global Service Composer, providing the tools for long term service provision
- A Home Agent, for handling network layer movement of terminals
- A Key Distribution Center (KDC), providing the crypto information required for the A4C operation; this entity will be interconnected to a global Public Key Infrastructure
- A CN QoS Broker to manage the core network transport infrastructure according to long term statistics (this entity is different in nature from the AN QoS Broker, which focused on admission control)
- A Multimedia Service Platform (MMSP), providing support for SIP-based services
- A Central Monitoring System (CMS) that collects information from probes in multiple entities and acts as a central query service for real time monitoring information
- An A4C platform, centralizing functions related to Authentication, Authorization, Auditing, Accounting and Charging
- A Policy-Based Network Management System (PBNMS)

The last five components are further described in the next section. Third party service providers can be interconnected to the Service Provisioning Platform, providing value-added services (notably content) on top of the base communication services.

Even though a global Pervasive Service Platform is represented in the SPP, it is not the only entity that handles pervasiveness. Pervasiveness, the capability of providing transparent service usage, is a complex function, achieved through the synergistic cooperation of multiple entities. Pervasive components are distributed along all entities, from the mobile terminal to the access router, into the SPP. These are all components that have degrees of personalization and privacy. Their control can be achieved by different degrees of intelligence. In DAIDALOS, the user has his service profile, containing rules that may be quite complex and change as time goes by. The Pervasive Service Platform server is simply the central rules engine in the network.

The Sensor Integration Platform collects information from sensors and provides that information as a service to any entity requiring it. It is used, for example, to provide pervasive and location-dependent services.

The architecture was defined in many aspects, even though explicit interface definition was not a concern, given the research nature of the project. The innovation concerns different aspects, including end-to-end QoS, overall security, VID-based authentication mechanisms, and content adaptation procedures. Another innovative aspect is the pervasiveness-oriented control overlay on top of the flexible telecommunication infrastructure (allowed for by the layered nature of the architecture with minimal overlaps) sustaining the integration capabilities sought after in a B3G (Beyond 3$^{rd}$ Generation) network.

## 5.3 QoS Subsystem Overview

This section provides an overview of the QoS-related components of the DAIDALOS WP3 architecture. As stated in the previous section, the architecture supports a wide range of services with seamless mobility of users across very heterogeneous networks. This heterogeneity stems not only from the diversity of network access technologies which must be supported in order to optimize the coverage/performance/cost factor under very different utilization scenarios, but also from the need to be inclusive of operators with quite different dimensions, characteristics and business cases. The development and fast deployment of advanced communication services in such a heterogeneous environment required the definition of a uniform architecture, capable of hiding the inherent complexity from those services. This uniformity is achieved by using IPv6 [RFC2460] as a convergence layer that hides the specificities of the different access technologies from the applications and services. The native support of mobility in IPv6 [RFC3775] is also of major importance. However, in order to provide completely seamless mobility with voice-graded handovers, an extension based on the support for fast handovers [RFC4068] is applied to IPv6.

Figure 5.2 shows the QoS-related subset of the proposed network architecture. Each Administrative Domain (AD) may contain several Access Networks (ANs), capable of supporting different access technologies, and a core subdomain providing interconnection between the access networks, via Subdomain Routers (SRs), and to other administrative domains, via Edge Routers (ERs). The architecture contains QoS elements in the AN, named AN QoS Brokers, that control the admission of new flows and the handovers. They are responsible for resource management in the access subdomains, including the access (wireless) links, possibly supporting different access technologies, and the interconnections below the SRs. The AN QoS Broker's decisions are enforced at the Access Routers (ARs), which they configure in a PDP-PEP (Policy Decision/Enforcement Point) relationship. Communication between the ARs and AN QoS Brokers is based on the Common Open Policy Service (COPS) protocol [RFC2748]. An important feature of the AN QoS Brokers is the ability to optimize the usage of operator resources by load balancing users and sessions

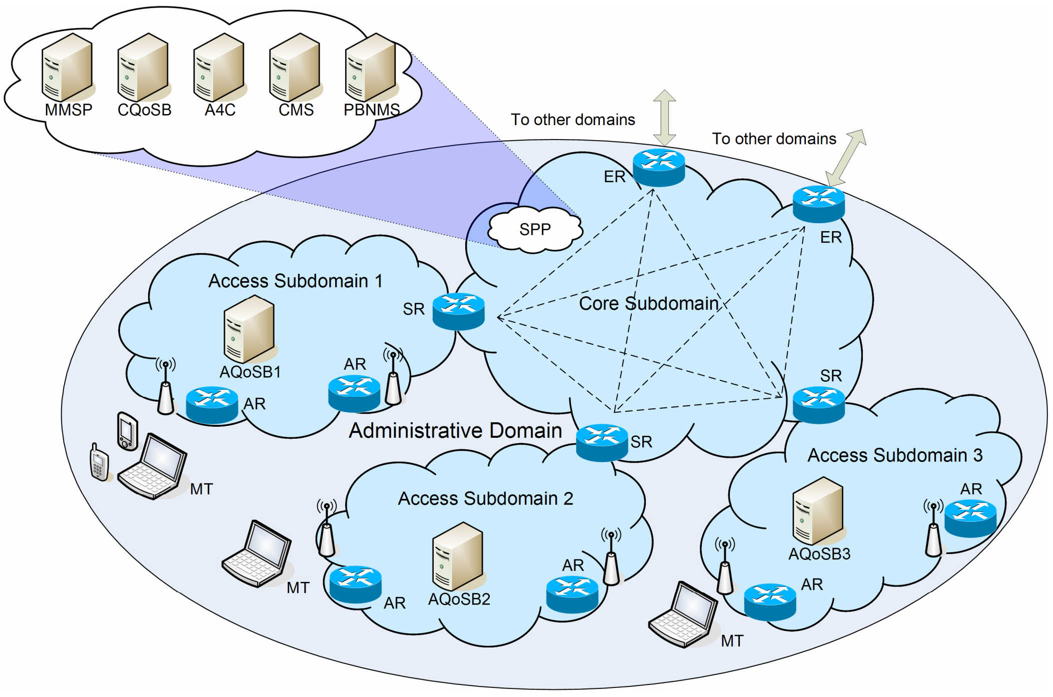


**Figure 5.2: Network architecture (QoS-related subset)**

among the available networks (possibly with different access technologies) through the use of network-initiated handovers. Another important feature is the possibility of downgrading or shutting down sessions of low priority users[3] when high priority users initiate a new session or handover an existing one to a cell with insufficient capacity.

In order to provide QoS to all kinds of services, including legacy IP applications, novel functionalities are added to the ARs to mark and recognize individual flows, and to translate other QoS reservation mechanisms, such as the IntServ [RFC1633] Resource Reservation Protocol (RSVP) [RFC2205] into DSCP markings and QoS Broker requests. The entity performing these functions is designated ARM, which stands for Advanced Router Mechanisms [Gomes04a]. A QoS client module in the terminals, able to mark application packets for a QoS service and to issue requests to the broker, may also perform the resource requests.

In the Core Network (CN) there is a Service Provisioning Platform (SPP) that provides the building blocks for creating services and applications on top of the network. The SPP contains a CN QoS Broker, which is responsible for resource management in the core, and deals with aggregates of flows traversing the core and inter-domain resources. Policies for resource management are defined by the Policy-Based Network Management System (PBNMS) and sent to the CN QoS Broker, where they are cached in a local repository for use.

[3] Priority is part of the user's profile.

The Central Monitoring System (CMS) collects statistics and network usage data from the Network Monitoring Entities (NMEs), and configures these entities to perform both passive and active probing. The information collected and processed by the CMS is fed to the PBNMS and the QoS Brokers, which use it for proper network (re)configuration and resource management. A Multimedia Service Platform (MMSP), consisting of a broker and proxy servers, is responsible for the provision and control of multimedia services. It is also capable of mapping application level QoS configurations to network resource requirements and of performing QoS requests for the flows. This architecture, thus, has a large degree of flexibility in QoS signaling, enabling the use of a diversity of QoS access signaling scenarios that fulfill the needs of the different applications and business cases of different operators. Unification of the scenarios is achieved by the centralization of admission and handover control at the AN QoS Brokers.

An Authentication, Authorization, Accounting, Auditing and Charging (A4C) server is also present in each domain. In order to improve the network efficiency and scalability, AN QoS Brokers retrieve from the A4C a subset of the user profile when a user registers in the network. This subset, termed Network View of the User Profile (NVUP), contains information on the set of network level services (classes of service, bandwidth parameters) that may be provided to the user, reflecting its contract with the operator. Similarly, a SVUP (Service View of the User Profile), containing information on the higher level services available to the user (e.g., voice calls, video telephony, and the respective codecs), is retrieved by the MMSP to control multimedia services. The advantages of the NVUP and the SVUP are twofold: on one hand, they speed up the resource reservation and service initiation processes, as decisions are local to the AN QoS Broker or the MMSP, without need for communication with the A4C; on the other hand, the load on the A4C servers, which are fundamental parts of operator networks, is reduced.

QoS support in the core is based on the DiffServ model, for scalability reasons; in the access, where (usually radio) resources are scarce, IntServ-like per-flow reservations are used for better control. Even though resource management is performed on an aggregate basis in the core and inter-domain segments of the path, information on the aggregates is propagated to the AN QoS Brokers, where it is used for admission control in order to achieve end-to-end QoS. This combination of per-flow and per-aggregate processing in a two-layer hierarchy allows our architecture to provide fine-grained QoS control while keeping the scalability properties of per-aggregate core resource management, decoupled from per-session signaling.

### 5.3.1 Comparison with UMTS

Compared to UMTS, the protocol stack in the DAIDALOS architecture is much simplified: the IP layer is connected to the physical layer only through layer 2, contrary to UMTS, which has a much more complex stack with multi-level encapsulation (please refer to fig. 2.8 in page 155). Even though UMTS has the advantage that handover control is mostly performed at the radio interface level, this strategy is not usable in heterogeneous networks[4]. Moreover, the simpler DAIDALOS stack translates in less overhead, leading to lower bandwidth consumption.

An important aspect in which the DAIDALOS architecture differs from UMTS is the decoupling of the PDP functions (AN QoS Broker) from the MMSP; in UMTS both functions are performed by the same element, the P-CSCF. Decoupling service level from network level functions provides the flexibility for simultaneously supporting any required application signaling protocol, freeing the architecture from being tied to a single protocol (SIP, in the case of UMTS). The ability to support different application signaling protocols is an issue of major importance, not only because there may be multiple application signaling protocols requiring QoS (not necessarily related to multimedia), but also in order to make the investment in infrastructure future-proof — the network infrastructure should be able to provide support for a new protocol providing functions not possible to implement in SIP with minimal changes. In the DAIDALOS architecture, supporting a new application signaling protocol essentially requires adding QoS-aware proxies supporting the protocol, keeping the rest of the network unchanged.

Probably the most important advantage of the DAIDALOS architecture over UMTS is the support for heterogeneous access networks. The use of IPv6 protocol as a convergence layer provides the network with independency on the underlying technologies and significantly simplifies the support for mobility and QoS across heterogeneous networks, "merely" requiring the mapping of IP QoS parameters into their layer 2 counterparts in order to take maximal advantage of the QoS features provided by each technology. In contrast, UMTS handovers are mostly performed at layer 2, without intervention of the IP entities. QoS control during handovers is provided both in inter-Radio Network handovers and in inter-SGSN handovers; in both cases, QoS levels are maintained, and no re-negotiation is performed by the IMS entities — previously negotiated profiles are simply transferred from the old path to the new one. This strategy cannot naturally be transposed to heterogeneous

[4] In the future, the emerging IEEE 802.21 standard, currently in draft state, could change this state of affairs.

network environments. The L2 orientation of UMTS hinders its usage across multiple access technologies, providing fewer possibilities for the optimization of the coverage/performance/cost factor under different usage scenarios than the DAIDALOS architecture. Interworking of the IMS with different access technologies is being worked upon, but no standard has yet been released.

## *5.4 End-to-End QoS Control*

The AN QoS Broker is the central element that performs admission control for new flows and controls the handovers. For this purpose, the AN QoS Broker has detailed knowledge on the topology and resource usage of the AN, and is aided by metering information collected by the CMS. Although core and inter-domain resources are managed on an aggregate basis, communication between CN and AN QoS Brokers provides the latter with the necessary information to build maps containing the available resources to the different access networks in the same domain and to other administrative domains. Three tables are maintained by the AN QoS Brokers: one with information on resources of the AN, another with the resources information of the paths between ANs and between the AN and the ER, and a third one with alarm levels corresponding to the availability of resources in the inter-domain route. These tables, along with information on the set of network QoS services available to the user, contained in the NVUP, are used by the AN QoS Brokers for admission control.

In order to establish a reservation for a flow with fully end-to-end QoS, requests must be performed to the AN QoS Brokers of both endpoints of the flow. The admission control process will be different according to the relative location of the endpoints. When a mobile terminal, MT1, initiates a session to another one, MT2, there are 3 possibilities for their relative location: (1) they are connected to the same AN, (2) they are connected to different ANs in the same domain, or (3) they are connected to different administrative domains. In the first case, a single AN QoS Broker is involved, and resource checking is performed for the AN only, since communication is local. In the second case, ANQoSB1 checks for resources in the first access network, AN1, and the core, and ANQoSB2 checks for resources in AN2. In the third case, each AN QoS Broker checks for resources in the respective AN and in the core of the domain where they belong, and for transmission resources in the inter-domain path segment, as illustrated in fig. 5.3.

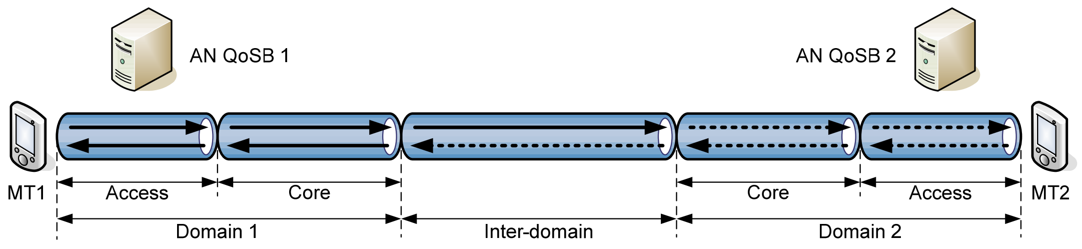


**Figure 5.3: Admission control in the inter-domain call scenario**

## 5.4.1 Per-Flow QoS — Session Setup

As was previously mentioned, the DAIDALOS network architecture is very flexible regarding the initiator of the QoS requests, which may be the MT, the ARM or the MMSP (or even an application server). The necessity for such a flexible resource request model stems from the need to support all the required applications, ranging from legacy, QoS-unaware data transfer to advanced multimedia communication services with stringent QoS requirements, as well as diverse deployment scenarios corresponding to different operator business cases. This flexibility regarding the QoS request initiator led to the development of different scenarios for the integration of application-level session setup signaling and network-level QoS signaling.

In the first scenario, suitable for SIP-based services, terminals perform only application layer signaling. Based on the contents of application-level negotiation messages (notably codec information) along with the user profile, the MMSP (or a SIP application server) infers network level resource requirements and issues QoS Broker requests accordingly. Communication between the MMSP and the AN QoS Broker is based on the COPS protocol.

In the second scenario, the terminals themselves issue QoS Broker requests. Applications use an Application Programming Interface (API) for requesting resources, which gives them a much higher degree of control of received QoS. Contrary to the previous scenario, this one supports non-SIP applications. However, both SIP and non-SIP applications must be coded to use the API, disallowing the use of *off-the-shelf* applications without modification. Since the terminals are not allowed to communicate directly with the QoS Broker, mostly for security reasons, a QoS Attendant at the AR works as an intermediary. Communication between the attendant and the QoS Broker is based on COPS, but a different protocol is used between the terminal and the attendant (currently an RSVP derivative).

In the last scenario, application signaling (when it exists) or data packets (when it does not) are intercepted by the ARM module at the AR. This module determines the amount of resources to reserve for the session based on the type of application, the contents of application layer signaling (if applicable) and the user's NVUP, and issues the QoS Broker requests. The main advantage of this scenario is the support of unmodified applications based on standard or otherwise well-know protocols (and, to a lesser degree, of most non-standard protocols); the drawbacks are less application control over QoS than in the MT scenario and less powerful SIP services than in the MMSP scenario (since the ARM is not, and cannot be, a fully-fledged SIP entity).

In spite of the differences, admission control and resource management is always performed by the AN QoS Broker, thus unifying the signaling scenarios. A thorough description and analysis of these scenarios is presented in the next chapter. The next two sections, respectively, give an overview of the intra-domain (core) and inter-domain resource management procedures that, combined with access resource management, provide end-to-end QoS control capabilities to the architecture.

### 5.4.2 Intra-Domain QoS Control

The intra-domain QoS control covers QoS resource management for an administrative domain from the user terminal to the edge router (ER). The main requirements for the intra-domain QoS architecture are: 1) scalability of the signaling within the administrative domain; 2) flexibility (easy to manage); 3) efficiency in the usage of network resources; 4) support for the mobility of users.

In this DiffServ environment, per-class aggregate resources are dynamically allocated, by the CN QoS Broker, based on actual network traffic, operator polices and other conditions. The monitoring subsystem plays an important role in this process, identifying aggregates to/from where resources should be reassigned. Resource management in the core is based on:

- Policies received from the PBNMS — information containing the description of the different transport services and the network topology. The CN QoS Broker has a bilateral interface with PBNMS: it requests for policies at start-up time and receives unsolicited policy definition when policies are changed in PBNMS. The CN QoS Broker generates alarms to the PBNMS reporting, e.g., continuous resource over usage or AN QoS Broker fault.
- Measurements supplied by the CMS — the CN QoS Broker detects if usage of a given link is above a certain threshold and can reallocate resources from less used links in order to increase the capacity of that link (upper part of fig. 5.4).
- Requests from the AN QoS Broker — AN QoS Broker can directly ask the CN QoS Broker to change the amount of resources of a given link (lower part of fig. 5.4).

Figure 5.4 depicts the resource management process in the core. The CN QoS Broker (CQoSB) reconfigures the bandwidth reserved for the aggregates on the basis of measurements and in response to requests sent by AN QoS Brokers. The CMS periodically sends the *Measurement Data* message with the monitoring results (the bandwidth occupied per class, the mean/maximum packet delay and loss in a class, etc). With this information, the CN QoS Broker has information on the congestion status of each class, and can reconfigure its routers (bandwidth per class, queue length, etc.) if required. In the case of core reconfiguration, the CN QoS Broker sends an *Agg Info* message to the AN QoS Brokers of the access networks affected by the reconfiguration to push an aggregate map update. Measurement information is usually used for long term reconfigurations, enforcing domain policies. Note that the CN QoS Broker can be provided with the measurement data on a periodic basis as well as on the requests sent to the CMS.

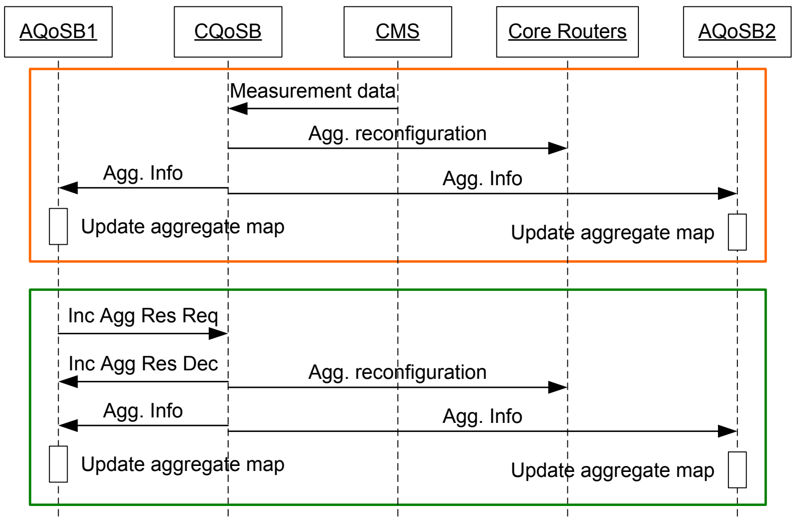


**Figure 5.4: Resource management in the core network**

Core reconfigurations may also be requested by an AN QoS Broker by sending an *Inc Agg Res Req* message to the CN QoS Broker when more bandwidth is required in a core aggregate to its access network. The CN QoS Broker answers this request and, if possible, reconfigures the routers and sends an *Agg Info* message to the AN QoS Brokers affected by the reconfiguration to update their aggregate maps.

The joint usage of these two mechanisms assures network flexibility while simultaneously minimizing the amount of signalling information exchanged in the connections between the CN QoS Broker and the CMS, and between the CN and AN QoS Brokers.

Considering that the core network is usually not the bottleneck in terms of bandwidth, core reconfigurations should be infrequent, and so should measurement information sent by the CMS. Note that each core reconfiguration may imply sending resource map updates to all the AN QoS Brokers that have to refresh the information related to this reconfiguration. Therefore, the use of partial, on demand reconfigurations decreases the signaling load and improves the network efficiency.

### 5.4.3 Inter-Domain QoS Control

Though much attention has been paid to intra-domain QoS, much less has been done in the scope of inter-domain QoS control. While the solutions of over provisioning or static DiffServ configurations are simple, they cannot provide any guarantees regarding end-to-end QoS. Additional mechanisms must, therefore, be used for inter-domain resource management. A solution for inter-domain QoS should be scalable and based on an evolution from the existing inter-domain routing, which the dynamic nature of QoS information should not compromise. Additionally, in order to gain acceptance, it should be simple and impose minimum requirements on intra-domain routing and QoS control.

Our approach to inter-domain QoS control is based on inter-domain routing with QoS metrics and constraints, without explicit resource requests. We have extended BGP (Border Gateway Protocol) [RFC4271] to convey and use three QoS metrics (expected path delay in light load conditions, bottleneck allocated bandwidth, a path congestion alarm) as described in chapter 8.

The information on inter-domain routes must be retrieved by CN QoS Broker in order to manage core resources (fig. 5.5); this task is performed by a BGP module installed in the CN QoS Brokers, which are, therefore, Internal BGP (iBGP) speakers. Conversely, bandwidth and alarm information on the aggregates between edge routers in the domain must be propagated to other domains. This information, stored at the BGP Policy Information Base (PIB) of the edge routers for use by the respective Decision Processes (the processes used to select a route to a destination AS among all the available ones), is configured and updated by the CN QoS Broker in a similar way to the other router parameters. If the route is selected, the

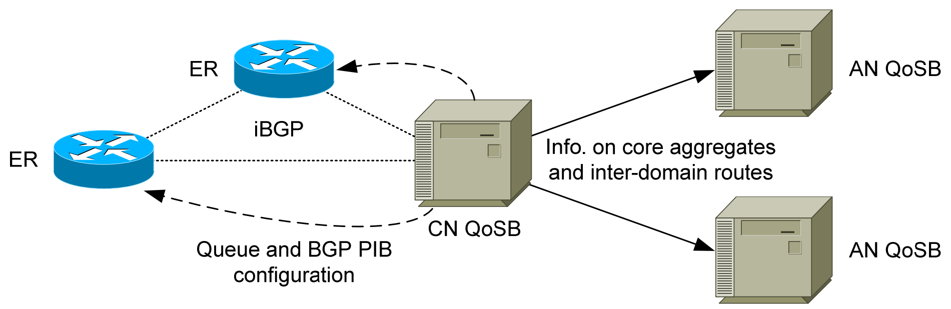

ER
ER
iBGP
CN QoSB
AN QoSB
AN QoSB
Info. on core aggregates and inter-domain routes
Queue and BGP PIB configuration

**Figure 5.5: Inter-domain QoS control — signaling**

edge routers propagate it to their peers in the upstream domain with the updated QoS metrics. As was previously mentioned, the CN QoS Broker propagates information on the inter-domain routes to the AN QoS Brokers, where it is used for admission control purposes.

## 5.5 Summary

In this chapter we have introduced the DAIDALOS project, in scope of which we have performed the work described in the second part of this thesis (chapters 5–8). We have mentioned the five key concepts (MARQS, VID, USP, SIB and Federation) and the two development guidelines (user control and telecom bias) that drive the project, as well as its scenario-based development model. We have described the major goals of the project, and the division of the research and development work into Work Packages that address the different aspects of the telecommunication system to be developed. We have given an overview of the QoS-related subset of the DAIDALOS WP3 architecture and how the relevant components interact for providing communication services with end-to-end QoS support across heterogeneous network access technologies. This overview will allow for a better understanding of the next chapters, as the QoS-related subset of the architecture is the frame of reference for the work therein presented. We have described the layered approach to resource management, performed per-flow in the access and in an aggregate basis in the core and inter-domain connections, and how the QoS information on the different layers and path segments is combined in order to provide end-to-end QoS to the application flows.

The next three chapters will describe some problems we addressed in the scope of DAIDALOS WP3 which, together with the collaboration in the definition of the architecture, constitute most of our research and development work in the project. Chapter 6 deals with the integration of application-level signaling and network-level resource reservation signaling, an aspect closely related to the per-flow resource management performed at the access. We describe our proposed signaling scenarios, which arise from the intrinsic flexibility of the network in terms of the initiator of resource reservation, and discuss the coordination between handover signaling and session QoS renegotiation signaling, of great importance in vertical handovers. A comparative analysis of the scenarios is performed, based on simulation results, addressing system behavior in normal operation and in overload conditions.

Chapter 7 addresses the inefficiency problems that arise from the joint use of SIP and MIPv6, particularly when end-to-end resource reservation must be performed. We propose an optimization based on cross-layer interactions and, again through simulation, demonstrate that our proposal greatly reduces the setup delay of multimedia sessions with end-to-end QoS support.

In chapter 8 we address the problem of inter-domain QoS routing, an integrating part of our proposed solution to the problem of end-to-end QoS and resource management. We propose a “black box” model based on virtual trunks, which may be regarded as traffic pipes traversing transit domains and that, concatenated, form the end-to-end path for data flow aggregates. The problem is formally stated and formulated in Integer Linear Programming (ILP), allowing us to obtain the optimal set of routes for a given network configuration and traffic matrix. A practical solution based on an extension of BGP using QoS metrics is proposed. Through simulation results, we compare the performance of our proposed extension with standard BGP, with the QoS_NLRI extension to BGP [Cristallo04] and with the optimal route set provided by the ILP optimization.

# CHAPTER 6

# SESSION AND RESOURCE RESERVATION SIGNALING

As mentioned in the previous chapter, in order to support all the required applications and operator business cases, the network architecture is very flexible regarding the initiator of the QoS requests, which may be the Mobile Terminal (MT), the Advanced Router Mechanisms (ARM), the Multimedia Service Proxy (MMSP), or even an application server. This flexibility led to the development of different scenarios for the integration of application setup and negotiation signaling and network QoS signaling, necessary for the establishment of sessions with end-to-end QoS. The scenarios are unified by the centralization of admission and handover control at the AN QoS Brokers.

In this chapter we present the different proposed signaling scenarios, illustrated by Message Sequence Charts (MSCs) representing the initiation of a multimedia session between two mobile terminals. Since the MT and ARM scenarios support legacy, non-QoS-aware data applications, MSCs representing the initiation of this type of application are also provided for these scenarios. We also discuss mobility issues, particularly the integration of session QoS renegotiation with the handover signaling. The signaling scenarios are comparatively analyzed, based on the results of several experiments performed in the ns-2 simulator, in which we evaluated the efficiency of session setup under normal load and the system behavior in overload conditions. This work was originally published in [Prior05a] and [Prior05c].

This chapter is organized as follows. The next section discusses session initiation in the different signaling scenarios. Section 6.2 discusses mobility and QoS issues. A comparative analysis of the different signaling scenarios is presented in section 6.3. Finally, section 6.4 contains the main conclusions of the work described in this chapter.

## 6.1 Session Initiation

In this section we illustrate the different scenarios for integration of session setup and negotiation signaling with resource reservation and QoS signaling based on MSCs for the initiation of a multimedia call between two terminals. Even though these examples are based on Session Initiation Protocol (SIP), other signaling protocols could be used in this architecture, leading to message exchange sequences not much different from these ones. In addition to multimedia sessions, the ARM and MT scenarios support more traditional data services that do not make use of an out-of-band signaling protocol. The initiation of such services is also illustrated in these scenarios.

### 6.1.1 MT Scenario

Figure 6.1 illustrates the MT scenario in a simplified example of a multimedia session initiation. The figure considers that the terminals are connected to different ANs (in the same or in different domains). Mobile terminal MT1, through the respective QoS Client, maps the application requirements to network services and QoS requirements, and sends a request to its serving QoS Broker, ANQoSB1, with this information. The request is sent indirectly, via a QoS attendant at AR1, as terminals are not allowed to connect directly to the QoS Brokers for

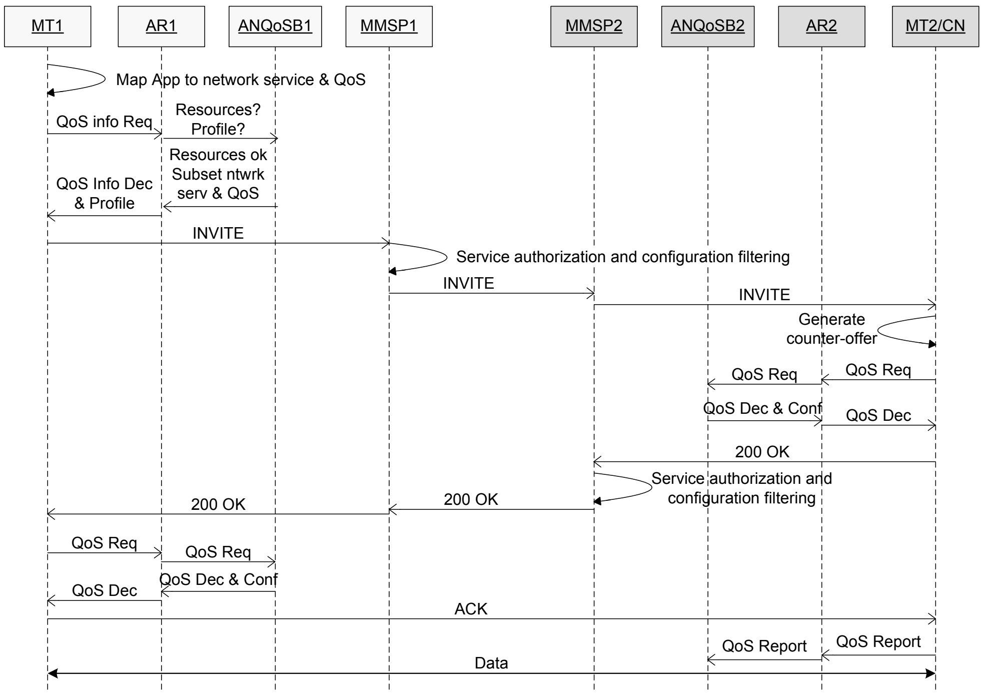


**Figure 6.1: SIP session initiation — MT scenario**

security reasons. QoS signaling between the QoS Client and the attendant is implemented as an extension to RSVP [RFC2205], which is local between the MT and the AR. ANQoSB1 answers with information on the network service levels available to the user, according to the user profile and the current network status. If allowed by ANQoSB1, MT1 sends an *INVITE* message with an initial offer of application level QoS configurations to MT2. When receiving the *INVITE*, MMSP1 performs service authorization, filtering out services not allowed by the SVUP. If the service is authorized, the *INVITE* is forwarded to the MT2. MT2 matches the QoS configurations in the *INVITE* to the set supported by itself, requests network resources to ANQoSB2, and generates a counter-offer according to the response; this counter-offer is included in the *200 OK* message (the *180 Ringing* message, not relevant for QoS, is omitted in the figure). On receiving this message, MMSP2 authorizes the service and filters unauthorized configurations from the counter-offer. When MT1 receives this message, it chooses the configuration to use, and updates the resource request to ANQoSB1, which reconfigures the policing and queuing modules at AR1 according to the amount of requested resources for the new flow. MT1 sends an *ACK* containing the final QoS configuration that will be used to MT2. If the final configuration requires less resources than were previously reserved, a QoS report is sent to ANQoSB2 for updating (downgrading) the reservation.

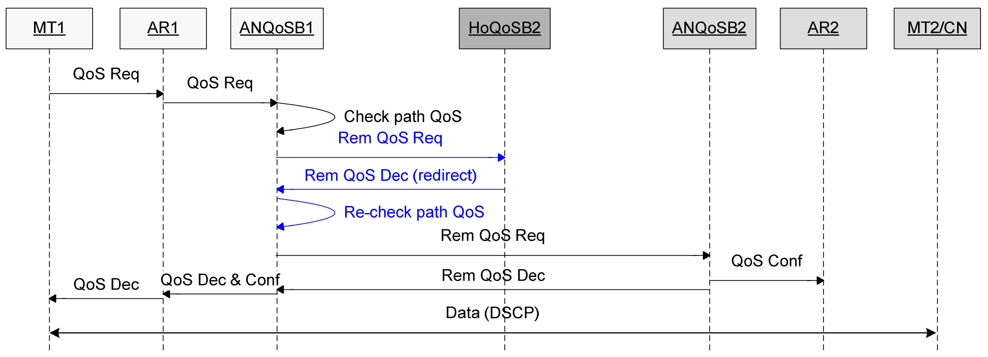


**Figure 6.2: Legacy, QoS-unaware session initiation — MT scenario**

The MT scenario also provides support for applications without an out-of-band signaling protocol. This type of application may be made QoS-aware by adding code that invokes the resource reservation and DSCP marking procedures. However, since synchronization of reservations cannot be performed based on application signaling, it must be performed using QoS Broker to QoS Broker communication. Figure 6.2 illustrates the initiation of such an application, with the callee roaming. As the caller requests resources for the entire path, ANQoSB1 contacts the "home" QoS broker of MT2 (HoQoSB2); however, since the terminal is roaming, the request is redirected to the foreign AN QoS Broker (ANQoSB2, which controls the access network to which MT2 is attached). Packets in the reverse direction, from the callee to the caller, are marked by the former with the DSCP value received in packets from the latter.

### 6.1.2 MMSP Scenario

In the MMSP scenario, signaling is performed through an extended proxy server, capable of parsing QoS configurations, mapping them to network resource requirements and issuing QoS requests. The proxy also enforces policies configured by the operator concerning the services allowed by the user contract (reflected by the respective SVUP). A multimedia conference is initiated in this scenario as shown in fig. 6.3. Upon receiving the *INVITE* message, MMSP1 queries the ANQoSB1 (using COPS) on the resources allowed for that user, in face of the user profile and the load at the access network. After ANQoSB1's answer, the proxy performs service authorization and modifies the *INVITE* message according to the answer (filtering the set of the services and QoS configurations). On receiving the *INVITE* message, MT2 matches the QoS configurations to those it supports, and sends a *200 OK* response with the common set. The MMSPs send requests to the AN QoS Brokers and, based on the decisions, perform service authorization and filter the set of QoS configurations. The *ACK* message contains the final configuration. QoS Report messages inform both QoS Brokers of the amount of resources that will actually be used, triggering the configuration of the ARs.

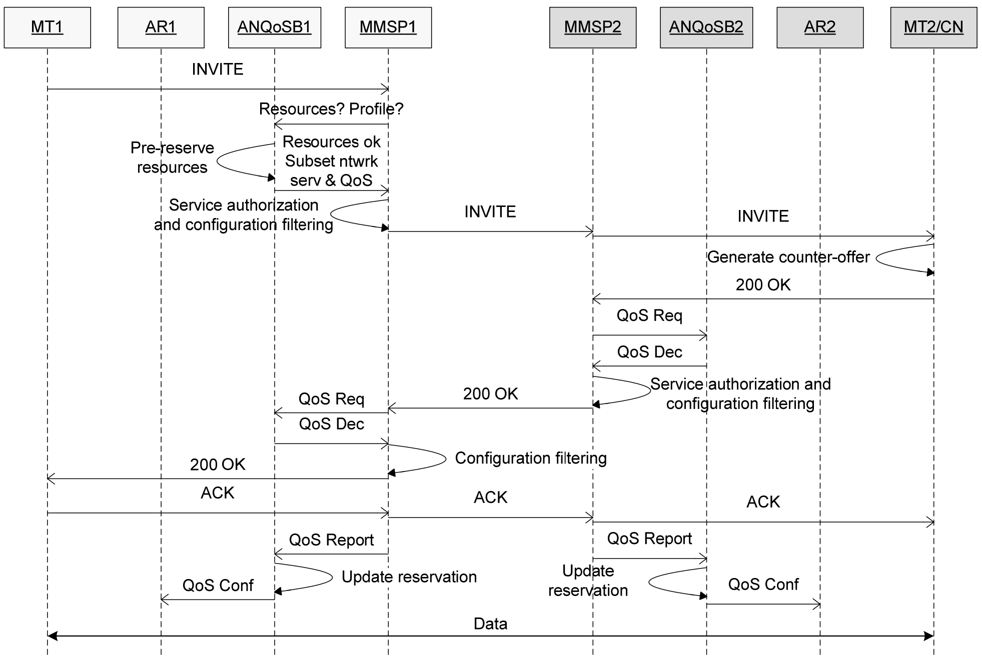


**Figure 6.3: SIP session initiation — MMSP scenario**

## 6.1.3 ARM Scenario

In the ARM scenario, it is the AR that performs application to network level QoS mapping and issues resource reservation requests to the QoS Broker (again, using COPS). Support for multimedia calls is provided a SIP/SDP parser and modification module in the ARM, which works as a simplified and transparent proxy. A multimedia call setup in this scenario is illustrated in fig. 6.4; it is similar to the other above presented scenarios, but in this case it is the ARM that parses the SIP/SDP, translates application-level QoS configurations into network resource requirements, issues the requests to the AN QoS Broker and filters the configurations in the application signaling according to the broker's response. Policy-based service authorization and filtering may still be performed by the MMSP.

Since the AR is always on the data path, legacy, QoS-unaware applications with in-band signaling only are equally supported by the ARM scenario: for example, when the ARM sees a TCP *SYN* packet with destination port 23, it knows a Telnet service is being started; it requests adequate resources (according to the operator's policy) to the QoS Broker, and marks packets belonging to the flow with the appropriate DSCP value. The initiation of a legacy, QoS-unaware application in the ARM scenario is illustrated in fig. 6.5. Notice the similarity to fig. 6.2 — the main difference is that the burden of issuing the resource request to the QoS Broker and marking the packets with the appropriate DSCP (which requires connection tracking capabilities) is now on the ARM; this small but significant difference means that

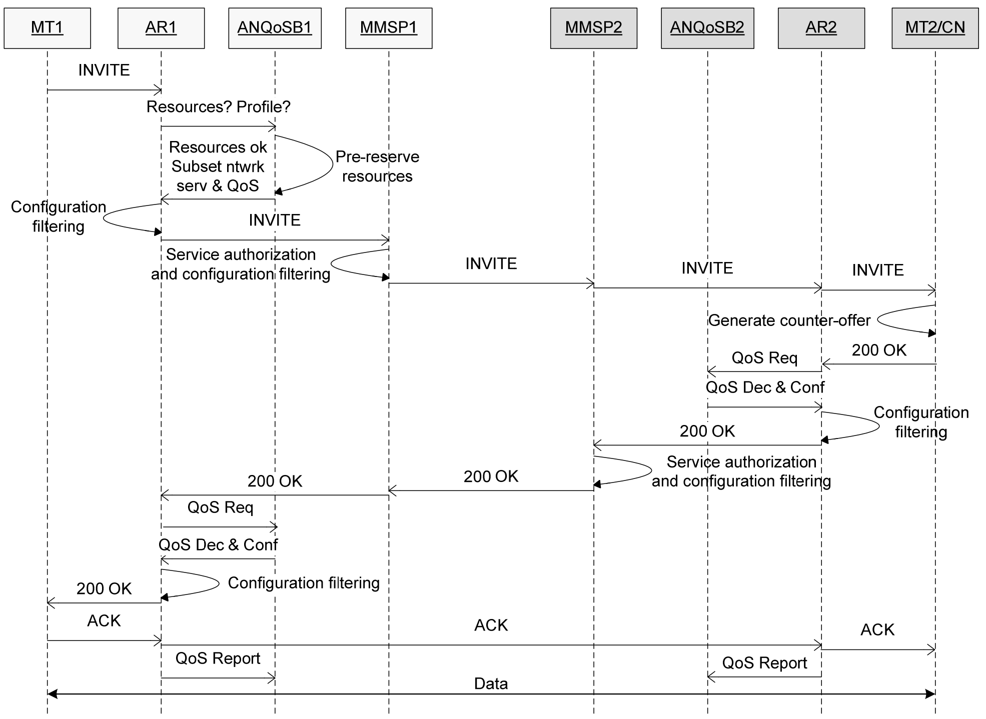


**Figure 6.4: SIP session initiation — ARM scenario**

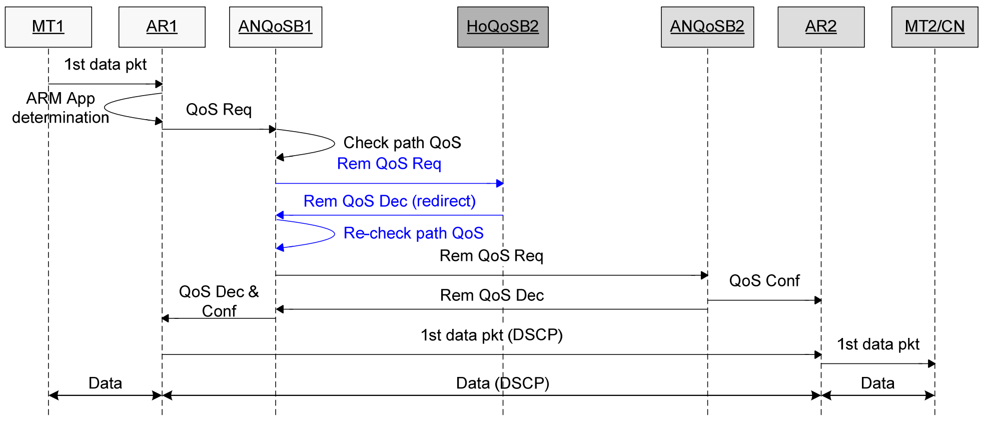


**Figure 6.5: Legacy, QoS-unaware session initiation — ARM scenario**

legacy applications can be used "as is" with the ARM scenario — there is no need to modify or wrap them in code that performs resource reservation.

Contrary to SIP sessions, there is no QoS information in flow signaling; therefore, supporting legacy Applications requires extra intelligence in the ARM. The necessary information to perform these tasks is supplied by the QoS Broker at boot-up of the AR. Information on general QoS profiles for legacy applications comes from the PBNMS, and reflects a mapping of operator business models into network policies.

## 6.1.4 Comparison of the Scenarios

The scenario where network resource reservations are directly requested by the terminal is general enough to support all types of services and applications without requiring special support from the network. Adding a new service is a simple matter of installing the appropriate software on the terminal. On the other hand, the applications must be capable of requesting network resource reservations, preventing the use of *off-the-shelf* applications. The amount of infrastructure required from the operator is minimal, since an operator may provide only a transport service with QoS; operators wishing to upgrade to more advanced services provision may then deploy larger infrastructures. The great flexibility in terms of services provided by this scenario is obtained at the cost of increased terminal complexity, since service intelligence is pushed to the terminals. In terms of privacy, this is a favorable scenario — since the operator is not concerned with the applications, all data may be encrypted, only its destination being known. In handovers, the requirements for context transfer are minimized, as the terminal is the common element in a handover. In business terms, this scenario is especially appropriate if the main service sold by the operators is data transport with QoS.

The scenario where the MMSP controls resource reservations is only appropriate for SIP applications[1]. While many other services, including legacy data transfer services such as FTP, could, in principle, be wrapped in SIP, it is unnatural and cumbersome to do so, adding unnecessary complexity to the terminals and defeating one advantage of this scenario, the simplification of the terminal by performing only application level signaling. For multimedia services, the proxies actively limit the set of QoS configurations inside the application signaling to those allowed by the user profile and for which network resources are available, improving the efficiency of session setup and negotiation. Moreover, with this approach, current and future SIP-based applications may be used with minimal or no need for modifications. The MMSP is a fundamental piece in this scenario, and the network operator itself must deploy it in order to have the complete control over the network and signaling. Very similar, in terms of signaling, to the MMSP scenario is the scenario where application servers perform resource reservations. However, two different providers may be involved in this case: the network operator, providing transport, and the service operator, providing content or a value-added service. A trust (federation) relationship between the two is required, since the application server needs to be able to issue requests to the QoS broker. Service- and content-based charging are easily provided by this model. This approach is efficient for multimedia streaming, file downloading, and all types of applications using central servers, since the servers have good knowledge of the service requirements in terms of network resources. The application providers may deliver content to the user with QoS even if the application itself is QoS-unaware, lowering the requirements on the terminals. Privacy is better handled in this scenario, as the network operator needs not be aware of the retrieved content. Multimedia conference and some other user-to-user services, not necessarily based on SIP, may as well be supported by the application servers.

In the ARM scenario, complexity is pushed to the network edge. Even though scalability in the MMSP may be achieved by load balancing among a number of different servers, a solution where the signaling parsing and reservation initiating entity is as close as possible to the terminals is naturally scalable without special needs for load balancing, since a smaller number of terminals will request its services. Although not as scalable as the MT scenario, it allows the use of simple terminals incapable of performing QoS requests. Regarding signaling complexity, since the AR is naturally in the signaling path, acting as Policy Enforcement Point (PEP) at transport level, less signaling is required in this scenario.

[1] Conceptually, other protocols, such as H.323, could as well be supported, provided that a functional equivalent of the QoS-enabled SIP Proxy (the Gatekeeper in H.323, for example), enhanced with DAIDALOS-specific modules, were deployed.

With the ARM, QoS for legacy applications is easily supported without modifying them or requiring middleware in the terminal, and adding support for another application signaling protocol requires only a software update to the AR (push of a new application translation module). In terms of application support, this model is as flexible as the first: since the ARM is capable of trans-signaling [Gomes04a], minimum support can be provided even for applications with no corresponding ARM module, provided they are QoS-aware. Overall, this is the most flexible scenario, since it provides a choice between dumber terminals using only the limited set of well-known services supported by the ARM and more intelligent and costly terminals that support any application. Both service- and transport-based charging are easily supported. The ARM, however, needs to maintain some state machine consistency with the application signaling, and signaling messages cannot be encrypted, as they may be processed by the ARM. This scenario is preferred when a set of simple well-known services must be universally supported.

## 6.2 Mobility

Mobility plays a central role in 4G networks, and the requirement for seamless handovers is probably the most demanding one in terms of timing. In a heterogeneous network, handovers may be performed between different access technologies; therefore, in the DAIDALOS architecture they are performed at the access-agnostic layer 3. The handover process in this architecture is extended from the fast handover mechanism defined in [RFC4068], associated with the Candidate Access Router Discovery (CARD) protocol [RFC4066], used to propagate information on prospective networks for handover to the MT. The amount of available resources may vary widely in a vertical (inter-technology) handover. In order to fully exploit the resources provided by the different access technologies, the architecture provides support for the coordination between handover and application signaling, allowing for session renegotiation at handover time. This is achieved by including information that session renegotiation is necessary (or advisable) in one of the handover messages, which is used as a trigger for application-level session renegotiation, as will be described below.

Figure 6.6 illustrates a basic user-initiated, intra-domain, inter-AN handover process (procedures specific to inter-technology handovers are discussed later in this section). The terminal begins by sending a *Router Solicitation for Proxy (RtSolPr)* message with an indication of the new network where to perform the handover, selected based on information previously provided by CARD. The old AR (oAR1) sends a handover request message to the old QoS Broker (oANQoSB1), which pushes the NVUP, along with information on the set of active sessions, to the QoS Broker of the prospective network (nANQoSB1). If nANQoSB1 accepts the handover, the new AR is configured, and the decision is communicated first to the oANQoSB1 and then to the terminal by means of the *Proxy Router Advertisement (PrRtAdv)* message. The terminal then sends a *Fast Binding Update (FBU)* message confirming the handover. The *FBU* indicates that the terminal will move and triggers a bicasting process [RFC4068], where each packet sent to MT1 via the old network is duplicated at oAR1 and also sent via the new network. The *Fast Neighbor Advertisement (FNA)* message, sent by the terminal immediately after the attachment to the new AR, tells the new AR1 that the handover is completed. Both QoSBs are informed of the fact, and the bicasting process stops, since the terminal may no longer receive information via the old network; any packet received by oAR1 for MT1 is tunneled to nAR1, which delivers it to the terminal.

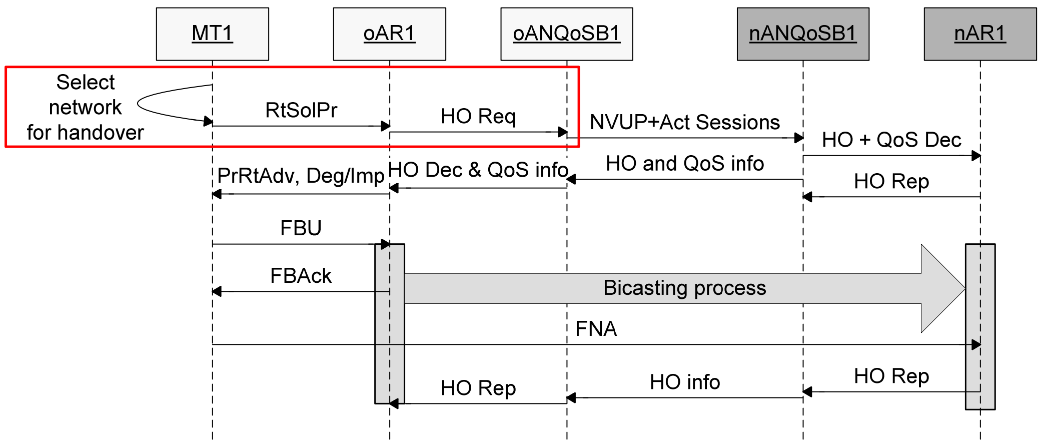


**Figure 6.6: Fast handover process**

Network-initiated handovers are equally possible in this architecture, providing a means for optimizing operator resources. The differences between terminal- and network-initiated handovers are that in the latter the *RtSolPr* and *HO Req* (red box in fig. 6.6) are absent and the *PrRtAdv* message contains an indication that the handover is mandatory.

The most frequent handovers are intra-technology. Usually, no renegotiation is performed in these handovers, but in case of cell congestion some QoS degradation (reduction of reserved resources) may be required; conversely, when leaving the congested cell, QoS may be improved again. With SIP, renegotiation for improvement is initiated by sending a re-*INVITE* together with the *FBU*. The renegotiation process is then performed in parallel with the handover. Since the time to complete the handover is usually much shorter (in the order of 50-100 ms) than the renegotiation process (eventually, up to 1-2 s), the handover completes and the activation of the improved QoS is performed in the new network (otherwise, the activation of changes is delayed until the handover is complete). Renegotiation for degrading QoS is more demanding: if the handover is completed before renegotiation, the new network might be flooded with more traffic than it can handle, but the handover process cannot wait for the renegotiation to complete, since it needs to be fast due to the imminent loss of signal. The solution to this issue is many-sided. A certain amount of capacity can be reserved at the access networks for coping with the extra traffic during a short grace period after a handover, enough for the renegotiation to finish. It is worth noting that differences in the amount of resources available to the user in intra-technology handovers are usually small. Whenever this extra capacity is insufficient, intelligent resource management is used by the QoS broker to temporarily suppress low priority sessions or session components of the user — for example, keep the voice but drop the lower priority video stream — or, if the user has an important profile, to temporarily borrow some capacity from ongoing sessions of lower profile users (when the benefit of keeping the QoS of the high profile user's sessions exceeds the cost of degrading low profile users' sessions).

Although not as frequent, inter-technology handovers are also supported. Indeed, this is one major advantage of 4G networks, as it allows features such as the automatic increase in the quality of a videoconference when arriving at an 802.11 HotSpot, or the dropping of the video component of a multimedia call without dropping the call when leaving the HotSpot

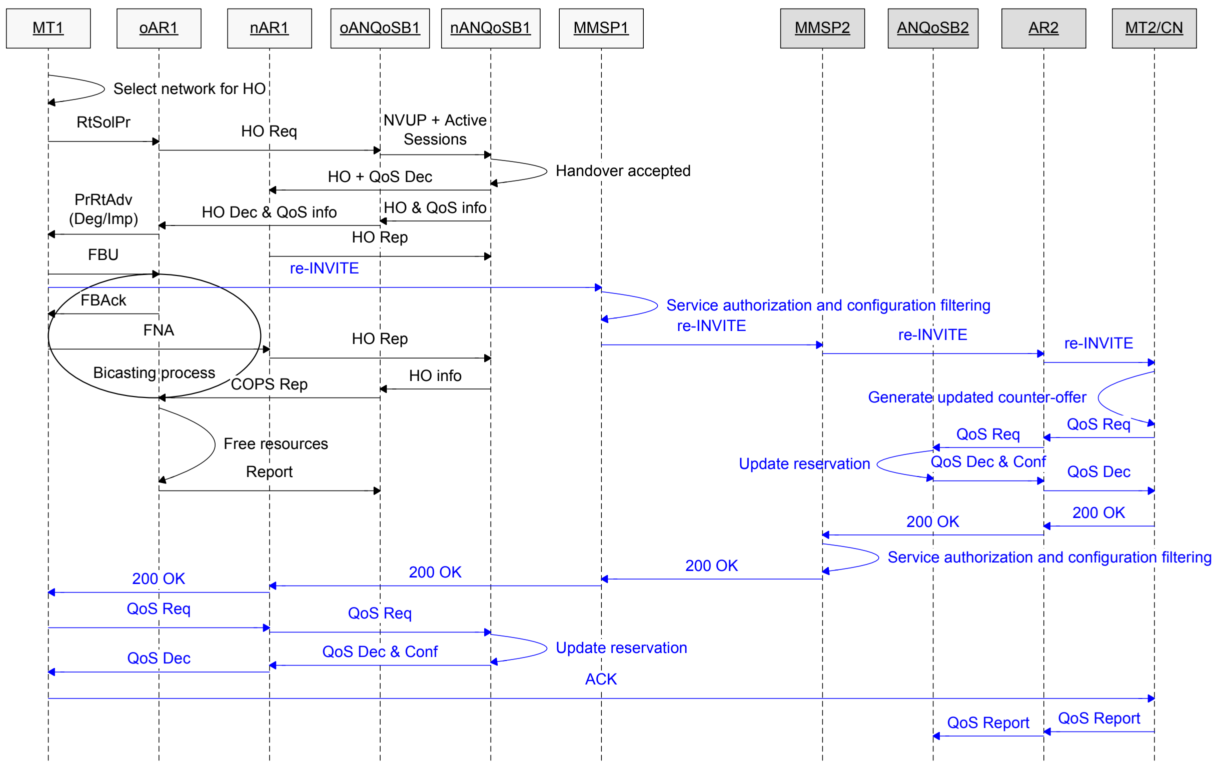


Figure 6.7: Intra-domain, inter-AN handover — MT scenario

and keeping the call via a GPRS connection. In this case, the differences in QoS levels in the different networks are potentially very large. Service improvement poses no problems, and works similarly to the intra-technology case, but for service degrading, large differences in QoS levels prevent the new AN from temporarily supporting the overload. In this case, the solution is to increase the handover time. This approach is feasible in inter-technology handovers since the cell overlapping area is usually much larger, requiring only an adjustment of the signal strength thresholds that trigger the handover in order to give more time for the handovers.

The integration of handover and session renegotiation is achieved by means of the *PrRtAdv* message. When applicable, this message contains indication of the need to perform service degrading or the possibility of service improvement, and is used as a trigger for session renegotiation. Figure 6.7 illustrates a handover process with renegotiation for QoS improvement in the scenario where the MT issues resource requests. The renegotiation process (in blue) proceeds in parallel with the regular fast handover signaling (in black), and the new, improved reservation is only activated in the new AN.

A handover to a different AN implies a change in the core aggregate to the edge router (or to the AN of the correspondent node in intra-domain calls). Therefore, when receiving the message from oANQoSB1 with information on the NVUP and active sessions of the user (figs. 6.6 and 6.7), nANQoSB1 needs to check for available resources in its AN and in the intra- and inter-domain path segments, ensuring that the new path has sufficient resources to accommodate the flows with the required QoS.

Inter-domain handovers are more complex, and usually involve a new complete registration process and the disruption of the active sessions. However, such disruption is

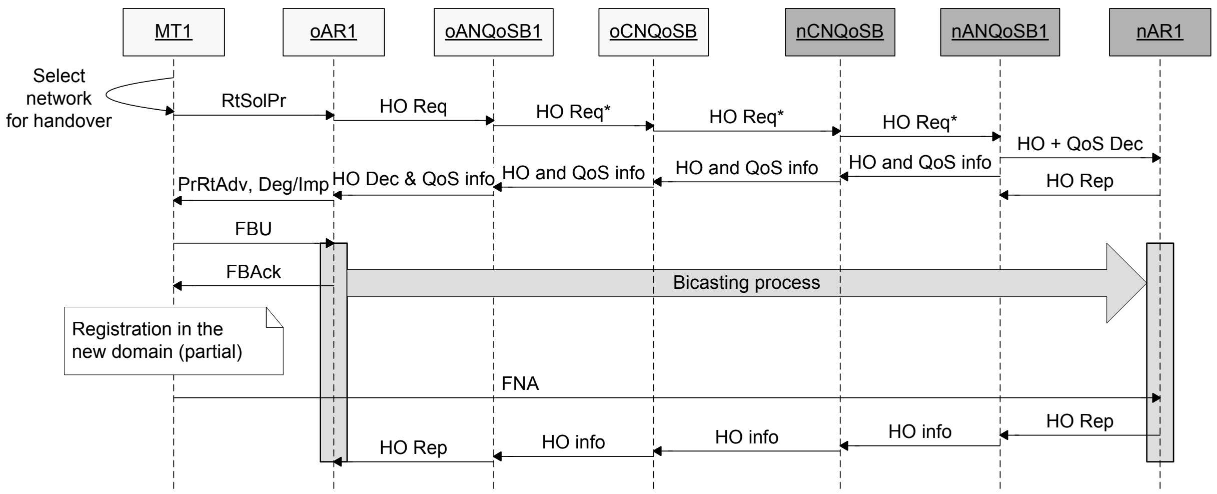


**Figure 6.8: Inter-domain handover between federated domains**

possible to avoid if the old and new domains are federated or have an otherwise appropriate SLA — in this case, the QoS Broker in the old domain adds an assertion with most of the registration information in the *HO Req** message sent to the new domain; due to the inexistence of security associations between AN QoS Brokers in different domains, this message is proxied by the CN QoS brokers. A partial registration, however, is still required to establish a security association between the MT and the A4C in the new domain.

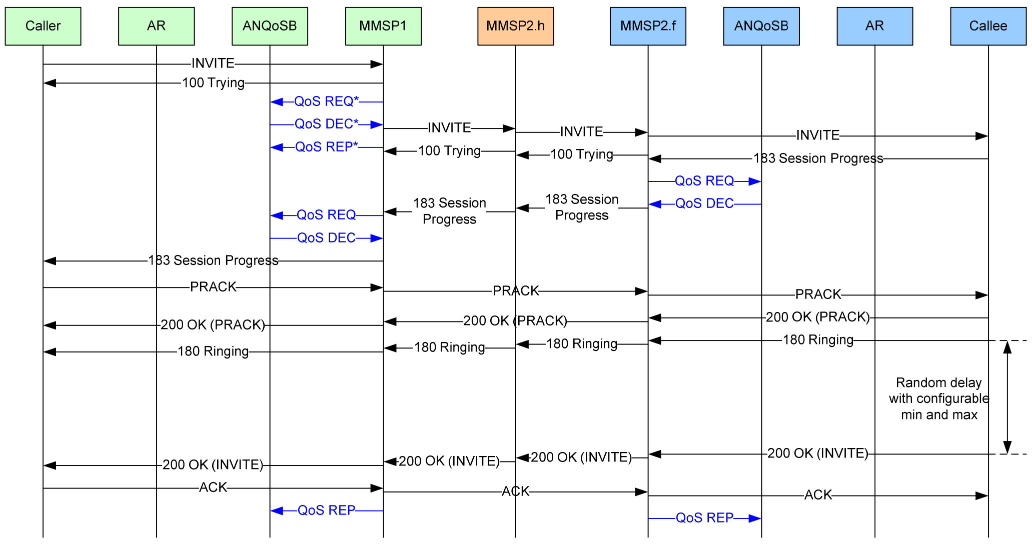
An inter-domain handover between federated domains is illustrated in fig. 6.8. As may be seen in the figure, the process is mostly similar to an intra-domain handover, differing by the (partial) registration in the new domain and by the use of CN QoS Brokers as proxies of the AN QoS Brokers. Another important aspect in which inter-domain handovers differ from their intra-domain counterparts is that they can only be user-initiated; this stems from the need for authorization from the user to perform the handover, as usually services are more expensive (and sometimes have lower quality) in a foreign domain.

## 6.3 Simulation Results

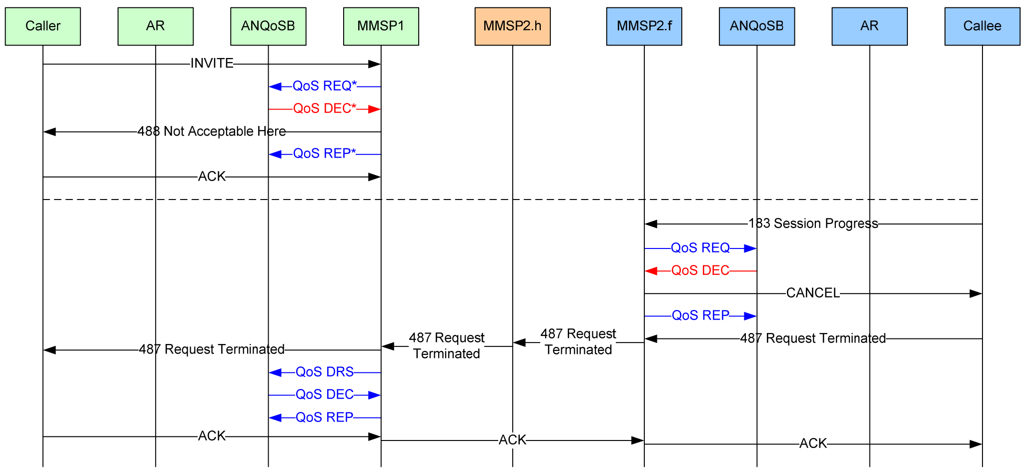
The efficiency of signaling for multimedia call initiation in the QoS signaling scenarios described in section 6.1 was evaluated using the ns-2 simulator [NS2]. We performed several experiments to evaluate the delay in establishing a session, as well as the response to congestion situations (establishment of a massive number of calls) in each of the signaling scenarios. The simulations comprise all possible combinations of (1) caller terminal at the home domain or roaming, (2) callee terminal at the home domain or roaming, (3) caller and callee physically attached to the same or different domains and, in the first case, (4) caller and callee physically attached to the same or different ANs, therefore representing all intra- and inter-domain call scenarios.

The standard distribution of ns-2 does not implement the SIP protocol. Although a third party implementation existed [NISTIPTel], it is incomplete and difficult to extend, and supports only stateless entities. Therefore, we have performed a new implementation of SIP, layered, with stateful entities, supporting user agents (UA) and proxies/registrars, and enhanced to support the QoS-aware UAs and MMSP; it also supports reliability of provisional responses (100rel) SIP extension [RFC3262], used in these simulations. Our ns-2 implementation of the SIP protocol (with the DAIDALOS-specific MMSP and UA features stripped off) is publicly available for download from [PriorNS].

Processing delays in the elements are accounted for in the simulation models. Delay values were extrapolated from measurements in other elements performing similar tasks (e.g., MMSP delays were extrapolated from measurements on SIP proxies). Message processing is performed in a FIFO fashion, meaning that processing of each message can only begin after all previous messages have been processed. Each message type takes a fixed amount of time to process, which is different for the different message types. Processing delays for SIP messages were simulated at both the MT (10 ms) and the MMSP (0.8 ms), with an increment for messages with SDP bodies (10 ms in the MT and 0.8 ms in the MMSP); this increment is larger when the entity performs QoS Broker requests (15 ms in the MT and 1ms in the MMSP). At the ARM, processing delay is considered for SIP messages with SDP bodies (0.2 ms), much larger in the ARM scenario (1 ms). AN QoS Broker request processing is also accounted for (1 ms). The remaining processing delays are considered negligible when compared to these, and ignored in the simulations.



a) Call accepted

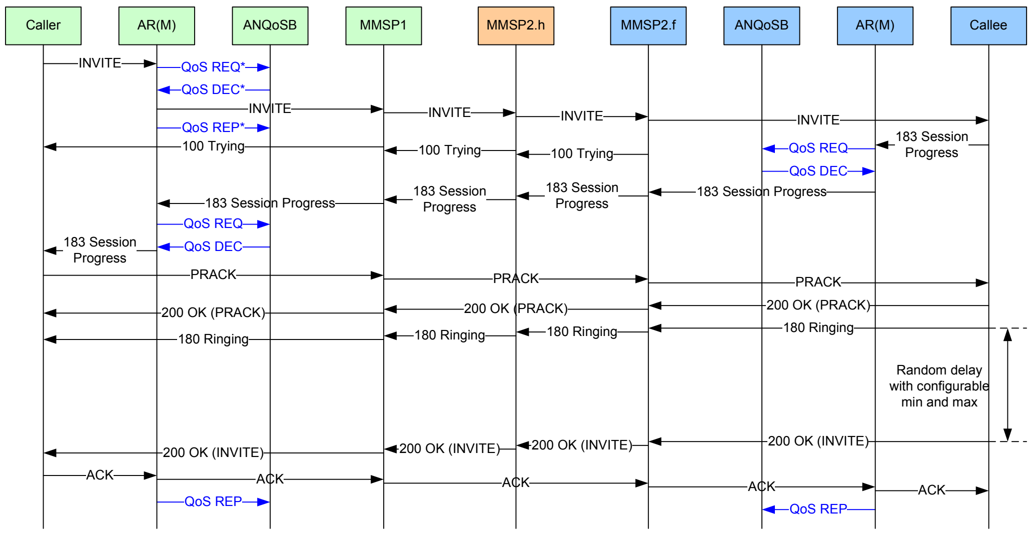


b) Call rejected at the caller side (upper) or the callee side (lower)

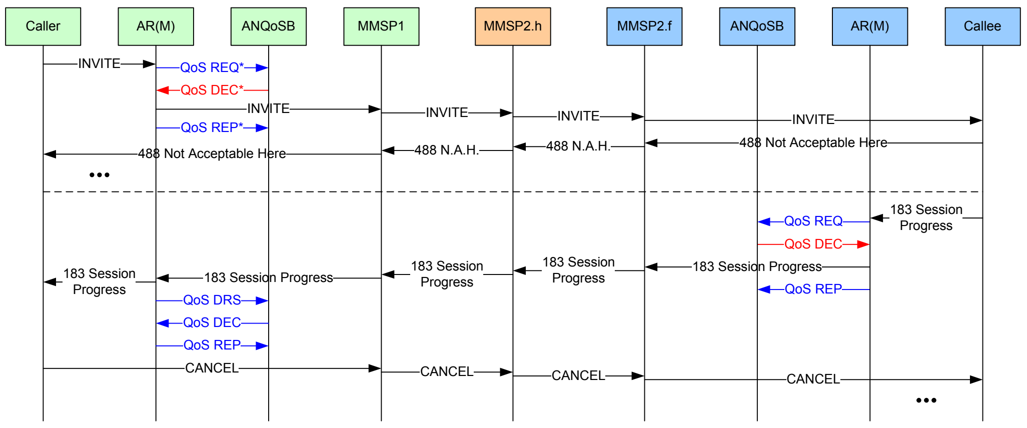
**Figure 6.9: Simulated message sequences — MMSP scenario**

These simulations assume that the terminals are properly registered, meaning that valid NVUP and SVUP are already in place at the AN QoS Broker and the MMSP, respectively, allowing them to act as PDPs for network resources (AN QoS Broker) and multimedia services (MMSP). This assumption allows us to simulate post-paid call initiation scenarios where the accounting/charging messages are not in the critical path of the session setup signaling.

The message sequences are derived from those presented in section 6.1, but use the 100rel extension to avoid ghost rings (calls dropped as soon as the callee picks up the phone due to lack of resources). In the scenario where the MMSP issues the QoS requests, the caller starts by sending an *INVITE* with a configuration offer. The counter-offer is conveyed in a reliable *183 Session Progress* response, and the callee equipment starts ringing on receipt of its confirmation, after which there is a configurable random delay, corresponding to the time



a) Call accepted

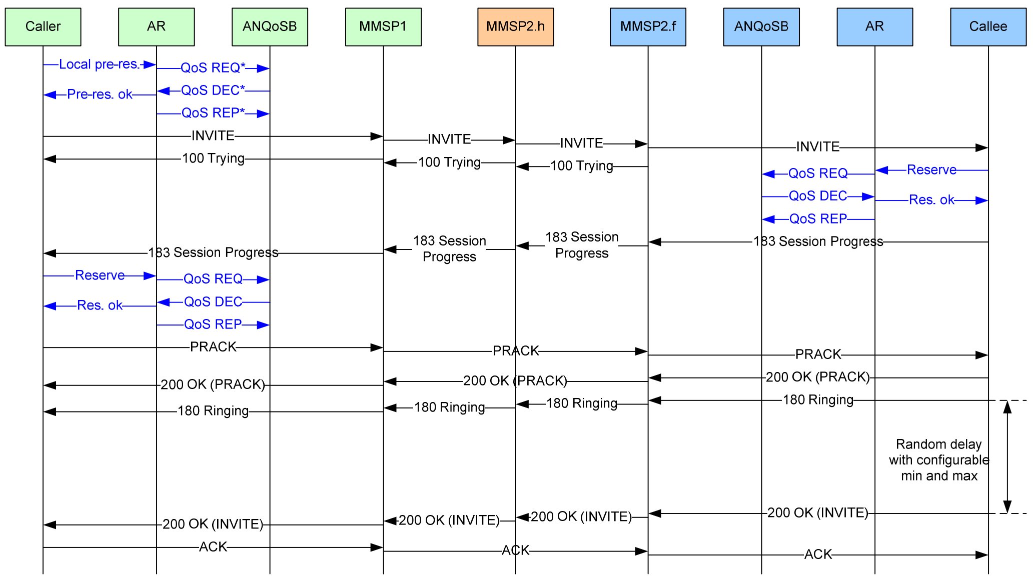


b) Call rejected at the caller side (upper) or the callee side (lower)

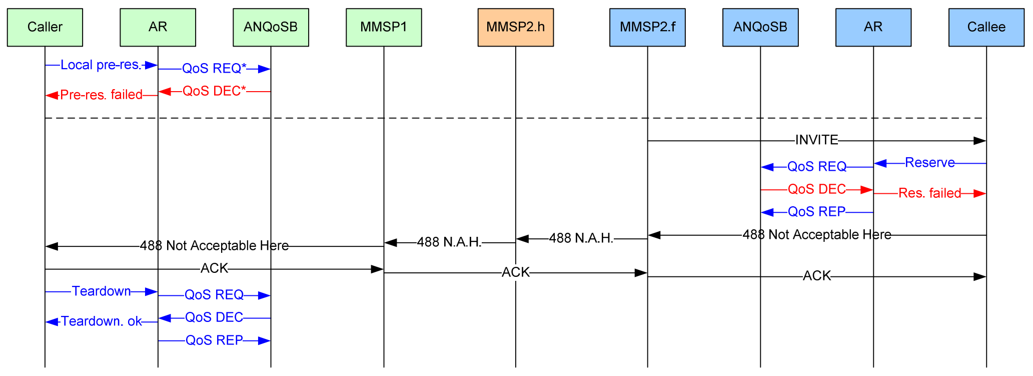
**Figure 6.10: Simulated message sequences — ARM scenario**

it takes for the user to answer the call, before the session is accepted. There are two possible sequences for rejected sessions: if the pre-reservation on the caller side fails, the MMSP of the caller immediately rejects the call with a *488 Not Acceptable Here* response; if it is the QoS request at the callee side to fail, the MMSP of the callee issues a *CANCEL* request to abort the session, resulting in a *487 Request Terminated* response from the callee UA. These sequences are shown in fig. 6.9.

Unlike the MMSP, the ARM is not a full-featured SIP entity. In particular, it does not generate new SIP messages. Therefore, when the ARM is responsible for QoS requests, if the initial request is rejected, the ARM does not generate a *488 Not Acceptable Here* SIP response. Similarly, if a full request is rejected by the AN QoS Broker, it does not generate a SIP *CANCEL* request; instead, it simply modifies the SDP body to indicate that none of the codecs is supported, relying on the SIP UAs on the MTs to react accordingly, aborting the



a) Call accepted



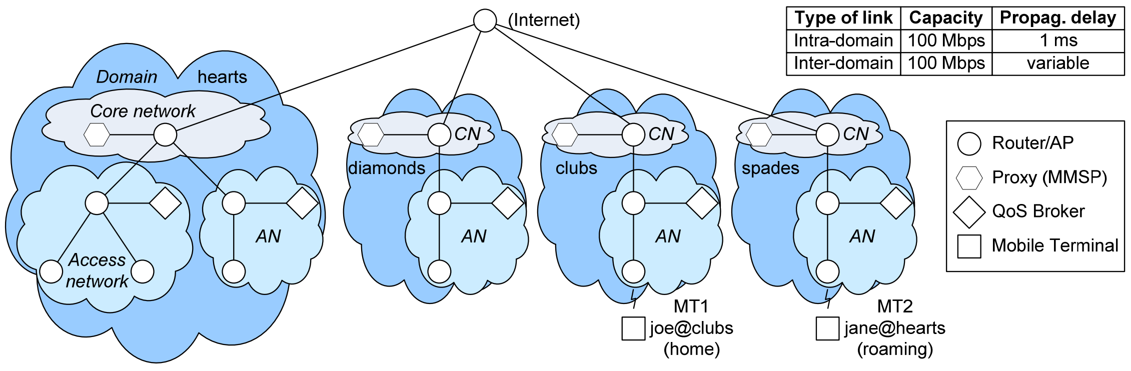
b) Call rejected at the caller side (upper) or the callee side (lower)

**Figure 6.11: Simulated message sequences — MT scenario**

session. Therefore, although the sequence for a successful call is very similar to that of the MMSP scenario, the failure sequences have an additional round-trip time (fig. 6.10).

In the MT scenario (fig. 6.11), if the initial QoS request at the caller side fails, no SIP *INVITE* is ever sent; if the QoS request at the callee side fails, the session is immediately rejected with a *488 Not Acceptable Here* response.

Figure 6.12 shows the topology used in these simulations, containing four domains, the leftmost one containing two ANs, one of which with two ARs. Although very simple, this topology allows us to simulate all possible combinations of roaming and non-roaming terminals: physically attached to the same AR, same AN and different AR, same domain and different ANs, or to different domains.



| Type of link | Capacity | Propag. delay |
|---|---|---|
| Intra-domain | 100 Mbps | 1 ms |
| Inter-domain | 100 Mbps | variable |

**Figure 6.12: Topology used in the QoS signaling simulations**

The implemented AN QoS Broker has topological knowledge of the bandwidth available in each of the interfaces of the ARs it controls. In the current version, however, it considers only access resources; core and inter-domain resource availability is not considered for admission control purposes at this stage (to be considered in the near future).

Some simplifications are assumed in the simulation model, namely the absence of DNS lookups and messages for the translation of home to care-of addresses at the MMSP. The latter is due to the existence of different alternatives to perform the translation, using the A4C or the Home Agent directly. However, it must be highlighted that we can also consider the possibility of integration of the Home Agent and the MMSP which, in fact, do not require any external message exchange to perform the translation, similarly to these simulations.

The efficiency of call setup and teardown signaling is evaluated in the three QoS signaling scenarios for all possible combinations of intra- and inter-domain scenarios. To this end, 32 terminals are uniformly distributed among the different ANs, each terminal having a 50% probability of being at its home domain and 50% of being roaming; random calls are generated between pairs of terminals, with an average duration of 120s and a mean interval between calls generation of 15s, for a simulated time of 24 hours. The roaming scenarios (relative locations of the terminals intervening in a call) are identified by four letters, *abcd*, where *a* indicates if the caller terminal is at its home domain (*a*=*h*) or roaming (*a*=*r*), *b* holds similar information for the callee, *c* indicates if the terminals are connected to the same administrative domain (*c*=*y* or *c*=*n*) and, if it is, *d* indicates if they are connected to the same AN (*y* or *n*). For example, *hryn* means that the caller is at home and the callee is roaming, both are attached to the same domain (that is, the callee is roaming at the caller's home domain), but to different ANs. In the *xrnn* scenarios, the *min* values correspond to the case where the callee is roaming in the caller's home domain; the *max* values correspond to the case of the callee roaming in a different domain from the caller's home.

The delay results for the successful setup of sessions in light signaling load conditions, simulated by using long calls and non-cumulative processing delays (notice that signaling is performed only at call setup and teardown), are shown in fig. 6.13. These delays are those sensed by the caller, that is, from the instant the caller begins the signaling to the instant it receives the *200 OK* from the callee and responds with the *ACK* (the call answering delay is, obviously, subtracted from this value). Although there are slight differences between the different signaling scenarios, they are of very little significance when compared to those imposed by the roaming scenarios: the dominant factor is the delay inflicted at each inter-domain link, of 30 ms (roughly 6000 km in optical fiber, ignoring router processing) in these simulations. Notice that although the delay at the radio links may be potentially large, it is common to all scenarios, thus not a discrimination factor. The most favorable scenarios are those where both terminals are physically at the home domain of the callee (*xhyx*), where no inter-domain links are traversed. When the callee is roaming, the initial *INVITE*, as well as all its responses, go through its home MMSP even if it is attached to the same domain as the

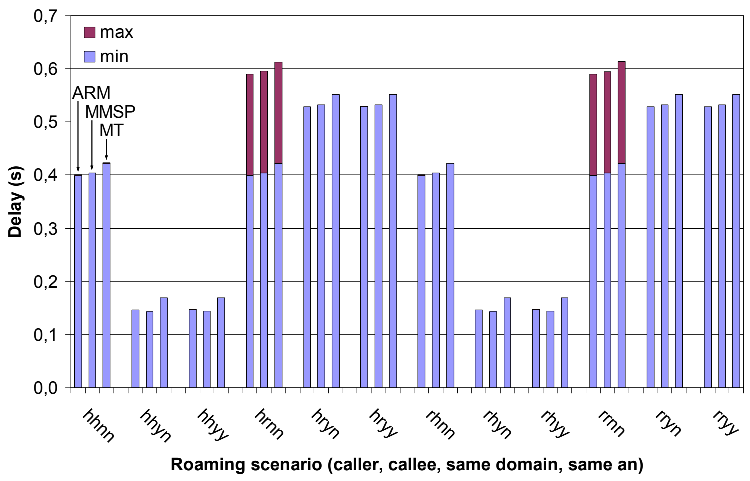


**Figure 6.13: Successful session setup delay**

caller, imposing a much larger setup delay; notice, however, that this does not apply to any further SIP requests, such as *PRACK*, *ACK*, *BYE* and possible *re-INVITE*s for session renegotiation. The case of the *re-INVITE* is particularly important, since we want to minimize session disruption at handover time when the lack of resources at the new network imposes renegotiation to reduce the amount of necessary resources.

The worst scenarios are those where the callee is roaming on a different domain than the caller, and the caller is not at the home domain of the callee (*xrnn max*). In this case two inter-domain paths are crossed by the *INVITE* and its responses; one inter-domain path is crossed by the other SIP messages — if the caller is at the home domain of the callee (*xrnn min*) the *INVITE* crosses only one inter-domain path.

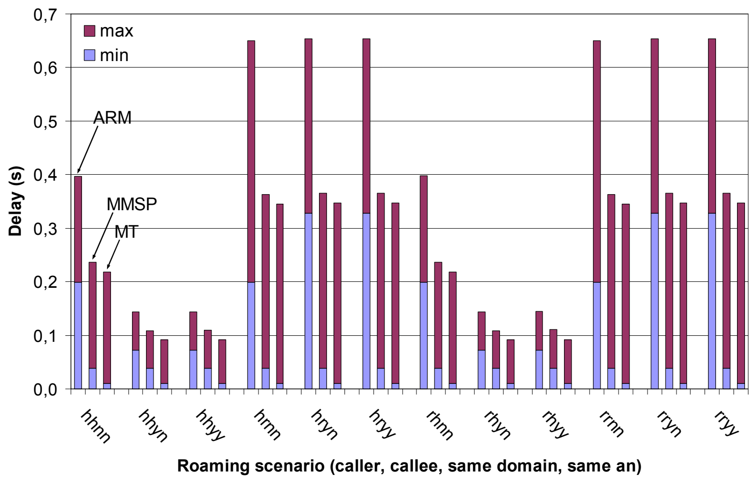


**Figure 6.14: Rejected call setup delay**

Figure 6.14 shows the delay of rejected calls in light signaling load conditions (due to lack of resources at either the caller or the callee network) from the instant the caller begins the signaling to the instant it receives the rejection final response. It is worth noting that, while the availability of resources is related to the number of established calls, the signaling load is related to the number of calls being initiated or terminated. The *min* values shown in the figure correspond to calls rejected at the caller side, and the *max* values correspond to those rejected at the callee side. In this case, the setup delays of the signaling scenarios are inverted, the ARM being much worse than the other two in most roaming scenarios. This stems from the fact that since the ARM is not a fully featured SIP entity, it cannot generate new requests (*CANCEL*) or responses (e.g., *488 Not Acceptable Here*). Rejected calls are, however, a very small minority of the overall attempted calls, meaning that this factor has little relevance in the choice of a signaling scenario.

Figure 6.15 compares the call termination delay. With respect to the roaming scenarios, since the *BYE* is always sent directly between the MMSPs of the domains where the terminals are attached, only their relative physical location (same domain or not) matters, and not the fact that either of the terminals is roaming or not. Regarding the signaling scenarios, the difference is very small.

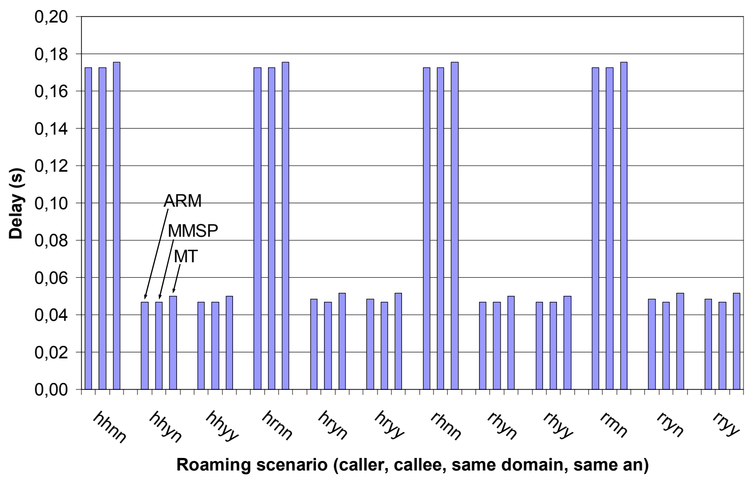


**Figure 6.15: Call termination delay**

From the above presented results, particularly those for successful session setup (the most relevant ones), we conclude that the efficiency, in terms of delay, of the session setup procedure under light load is not a decisive factor in the choice of one of the different signaling scenarios for SIP-based calls. In fact, except for the rejected sessions that take noticeably longer in the ARM scenario but are not much relevant since they will be infrequent, the three scenarios exhibit very similar signaling delays.

In all the scenarios, QoS requests are triggered by responses containing an SDP counter-offer as the message body. Such responses must be sent reliably, that is, they are either provisional responses requiring confirmation by means of a *PRACK* request, as in these simulations, or *200 OK* final responses confirmed by an ACK. In either case, if no confirmation is received within a time interval (defaulting to 500 ms for the first time), they will be retransmitted (this situation has occurred in other simulations). Therefore, care must be taken to avoid these retransmissions, since they are relatively large messages with a counter-offer in the body. If the summed delays from the transmission of the response to the reception of the confirmation, including all processing delays, cannot be consistently kept below 500 ms, this timer should be increased in real scenarios.

In a second experiment we evaluate the distribution of the setup delay of calls in the worst-case *rrnn* roaming scenario, under a medium/high offered signaling load of 70 new calls per second, with an exponential distribution of the time interval between generated calls. We used only 2 ANs in different domains, but a very large number of terminals (3000, 1500 in each AN/domain), in order to support the very large number of simultaneous calls. All the terminals are roaming and belong to the unused domains. The calls are initiated between a terminal attached to the first domain and another one attached to the second domain. Admission control at the AN QoS Brokers was set to always accept the requests.

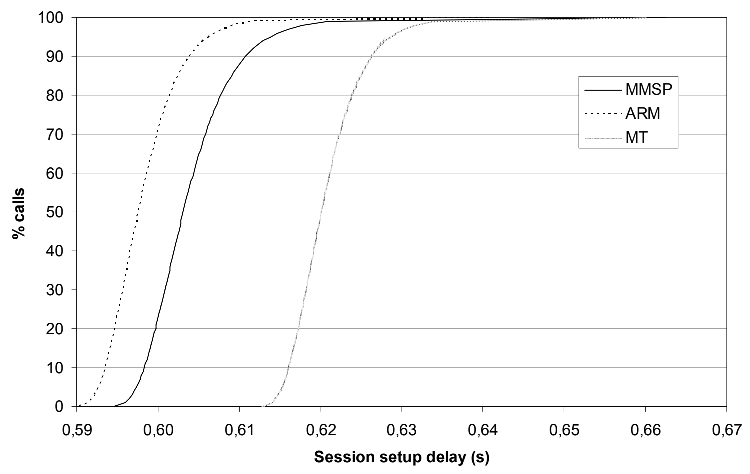


**Figure 6.16: Call setup delay CDF (*rrnn*, 70 calls/s)**

Figure 6.16 shows the results of this experiment by means of the Cumulative Distribution Functions of the setup delay in each scenario (for example, 20% of the calls are established in less than 0.6 s in the MMSP scenario). As can be seen, the setup delay does not vary significantly, even for the few percent calls where its value is larger. The largest measured setup delay exceeds the shortest one by 11% in the MMSP scenario, by 9% in the ARM scenario and by 8% in the MT scenario; for the 99% percentile, the values are 4%, 4% and 3%, respectively. These are average results of 5 simulation runs of 3600 useful seconds

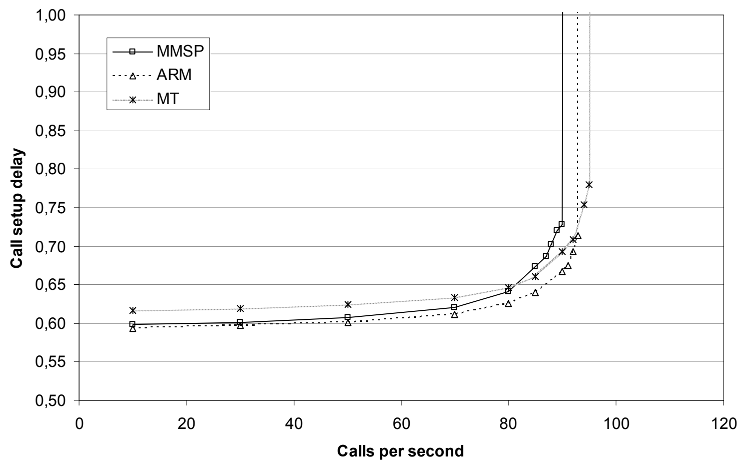


**Figure 6.17: Percentile 99 call setup delay with variable signaling load**

(corresponding to ca. 250000 calls each).

In another experiment we evaluated the limits of the system in the different signaling scenarios by increasing the offered load, in similar conditions to the previous one. The 99th percentile of successful call setup delays for the worst-case *rrnn* roaming scenario is plotted against the average number of generated calls in fig. 6.17 (the reason for using the 99th percentile instead of the average will be explained later on). These results are also the average of 5 simulation runs. As can be seen, the setup delay, approximately constant up to a certain load, grows explosively after that value. This fact is explained by the transaction-stateful character of the MMSPs: at a given point, processing delays accumulate up to a sufficient value for the SIP retransmission timers to expire. Since the retransmitted messages also take time to process, a snowball effect occurs, and delays become so large that calls start failing due to the timeout of the *INVITE* transaction (not to AN QoS Broker rejection by lack of resources). Measures should, therefore, be taken to avoid reaching this unstable state: load balancing between MMSP boxes should be carefully dimensioned for worst-case expected load; additionally, a policy should be implemented in the MMSPs, such that new requests would be ignored (or summarily rejected) as soon as the processing load would exceed a given threshold. It is worth noting that although the MMSP scenario is the first one to reach its limits, as expected since the MMSP is doing more work and is the bottleneck, the values of the other scenarios are very close.

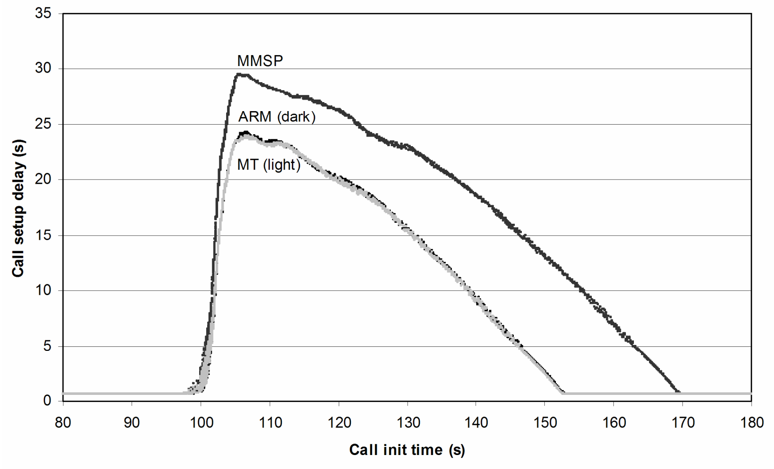


**Figure 6.18: System response under a peak of new calls**

In another experiment we evaluated the system response to a sudden peak of calls. We used the same *rrnn* roaming scenario of the previous experiments, but initiated calls at deterministic generation rates: first, a rate of 10 new calls per second for 100 seconds; then, a peak rate of 200 new calls per second for 5 seconds; lastly, we restored the initial rate of 10 new calls per second. The peak of calls causes processing congestion in the MMSP, triggering the aforementioned snowball effect. However, since the call generation rate after the peak is quite low, the system is able to come back to a stable operation state. In order to evaluate how long it takes for the system to recover, we plotted the call setup delay against the session initiation instant of the calls in the three signaling scenarios. The results are shown in fig. 6.18. As may be seen, the peak causes very large, unacceptable delays in all three scenarios; however, these delays reach higher values and take longer to recover in the MMSP scenario than in the other two. It is worth noting that the results for the ARM and MT scenarios are almost overlapping, since they have the same amount of processing in the bottleneck component — the MMSP.

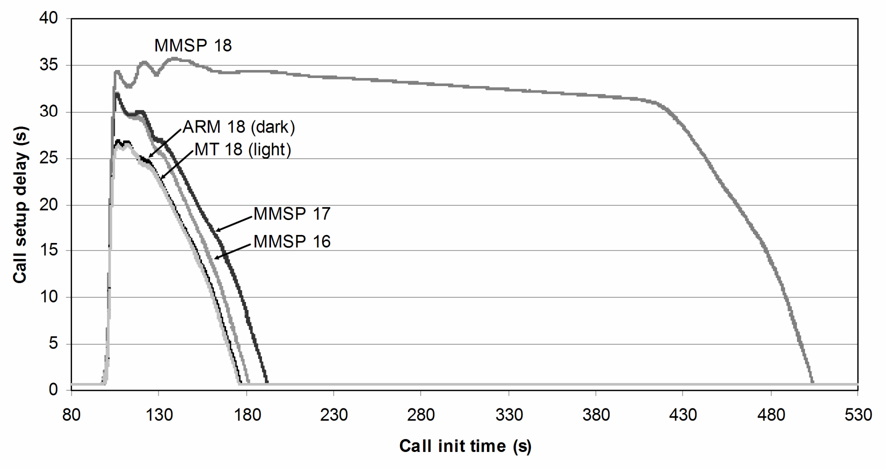


**Figure 6.19: Peak of new calls — varying steady-state load**

As the frequency of new calls in the steady state (before and after the peak) increases, the time it takes for the system to recover after the peak of calls increases. Eventually, the time to recover from the peak rises dramatically; this effect is illustrated in fig. 6.19, where three curves are shown for the MMSP scenario, corresponding to steady-state call initiation rates of 16, 17 and 18 calls per second. In the last case, we may see that there is a slow decay segment in the setup delay curve until a certain point is reached, where the curve begins to decay with a similar pattern to the other cases. For the sake of comparison, curves for 18 new calls per second are also shown for the ARM and MT scenarios, from where we may observe that the critical steady-state load for these scenarios has not yet been reached.

In fig. 6.20 we plot the recovery time from the peak of calls against the steady-state call generation rate for the three signaling scenarios. It may be seen that the recovery time is approximately the same in the ARM and MT, and higher in the MMSP scenario. Additionally, in the last one there is an explosive growth from 17 to 18 new calls per second in steady-state; this growth is more gradual in the other two scenarios, suggesting that the MMSP scenario is somewhat less stable regarding overloads than the others.

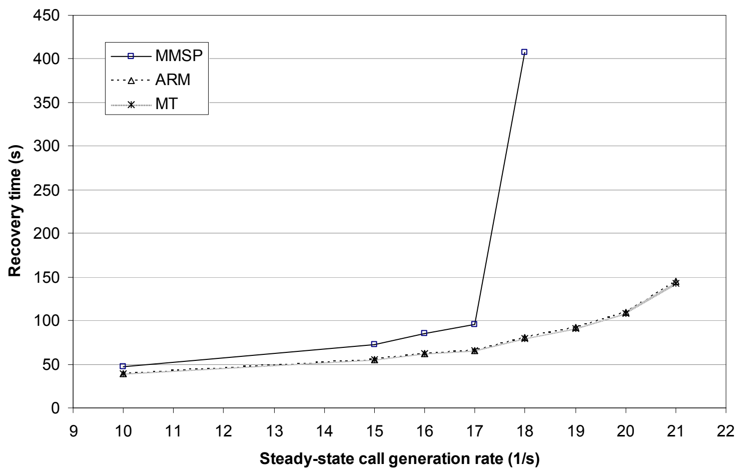


**Figure 6.20: Recovery time from the peak of calls (200/s for 5s)**

In face of the results described in the previous paragraphs, the reason for using the 99th percentile, instead of the average, in fig. 6.17 should become clear. With a sufficiently high number of calls per second, the system cannot recover from the snowball effect, and further calls are rejected. The use of the 99th percentile allows us to capture the effects just before calls start being rejected due to timeouts. With average values, this effect would have been masked out by the large number of calls previously established with very low setup delay. The average of successful and failed calls, however, exhibits a similar (and even more dramatic) effect than the 99th percentile of accepted calls.

The behavior of the system with respect to call setup delay during and after a peak of very high load suggests that the steady-state offered signaling load must be way below what can be handled, on average, by the MMSP, in order to have enough processing slack to absorb the snowball effect caused by the SIP retransmissions. Once again, it is worth mentioning the interest of implementing a policy of summary rejection of calls at the MMSP when the signaling load exceeds a given threshold: in this case, the system would recover much faster from peaks, allowing for a higher average steady-state load, which translates in less cost in hardware for similar load resilience characteristics.

## 6.4 Conclusions

In this chapter we presented and analyzed three different scenarios for interaction between the QoS Brokers and the other QoS-related entities in the DAIDALOS architecture — centered on the terminal (PDA, intelligent cellular phone), on service proxies (e.g. SIP proxies or application servers), or in advanced mechanisms at the access routers — and the corresponding strategies for the interaction between application- and network-level QoS signaling. The MT-centered scenario is appropriate for operators mainly selling data transport with QoS services, the MMSP-centered scenario for operators providing multimedia services and content, and the ARM-centered scenario for operators providing a set of universally available network services. We also described how session renegotiation is coordinated with handover signaling in order to fully explore the resources available in the different network access technologies.

Through a number of simulation experiments, we compared the behavior of the different scenarios. The results indicated that under normal operating conditions the efficiency of the different scenarios is comparable and, therefore, not a relevant factor in the choice of one over the others. However, under heavy load conditions the MMSP scenario exhibits overload problems before the other ones. This was an expected result since more functions are performed by that element, which was the bottleneck component; in the other scenarios, part of these functions is offloaded to the MT or the ARM. Careful dimensioning of the components, especially the MMSP, and the provision of generously dimensioned load balancing is necessary, as the service does not degrade gracefully in overload conditions. Moreover, the fact that excessive signaling load problems are greatly exacerbated by the snowball effect of SIP retransmissions means that a policy of summary rejection of new calls when the processing load reaches a certain threshold at the proxy should be implemented; such policy would prevent the snowball effect and, therefore, improve the resilience to signaling overload without the need for costly additional hardware.

# CHAPTER 7

# MOBILITY OPTIMIZATION

Terminal mobility may be handled at different layers. However, although SIP (Session Initiation Protocol) [RFC3261] may be used for mobility management [Wedlund99, Schulzrinne00], this function is better handled at layer 3 by Mobile IPv6 (MIPv6) [RFC3775] even when SIP is used for session control, for several reasons: (1) applications need not worry about mid-session mobility unless serious changes in available resources force a session renegotiation, e.g., to a lower bitrate codec; (2) layer 3 mobility support must be in place to support non-SIP sessions (HTTP, FTP, etc.), and uniform mobility management is desirable for robustness and flexibility; and (3) seamless mobility may be achieved through the use of MIPv6 extensions like Fast Handovers [RFC4068]. However, due to the duplication of mobility management functions, the joint use of SIP and MIPv6 leads to some inefficiency issues in pre-session mobility, as each protocol is unaware of the other one's mobility management capabilities. These issues are even worse when end-to-end resource reservation must be performed in order to provide appropriate QoS to the session, requiring knowledge of the points of attachment of both terminals; however, end-to-end resource reservation is necessary to provide communication services with the same level of quality users have come to expect from the telephone network. This inefficiency may lead to a significant delay in session setup, especially in the presence of packet loss (not uncommon in wireless links) and of large RTTs (Round-Trip Times). In this chapter we propose a scheme for the minimization of these delays based on simple procedures and cross-layer interactions that make SIP aware of the terminal's physical point of attachment, that is, the Care-of Address (CoA). The gains of the proposal are demonstrated both by a delay analysis and by simulation results. This work has been published in [Prior07e].

The chapter is organized as follows. Sections 7.1 and 7.1.1 contain an analysis of the problem and the proposal of the solution, respectively. Section 7.3 describes the SIP registration procedures. Section 7.4 discusses the relation between the proposed optimizations and the dormancy/paging support for energy saving. An analytical comparison of the standard and optimized procedures is presented in section 7.5, and section 0 discusses simulation results of both. The main conclusions of the work described in this chapter are presented in section 7.7.

## 7.1 Inefficiency of SIP with MIPv6

In order to establish a reservation for a flow ensuring that enough resources are available along the end-to-end path, admission control needs to take into account the available resources in the complete path, including the access, core and inter-domain path segments. To this end, each mobile terminal must be aware of its correspondent's physical location which, in IP terms, corresponds to its CoA. SIP's unawareness of pre-session MIPv6 mobility, as will be seen in the next section, is one of the sources of inefficiency in session initiation signaling. Mid-session mobility, on the other hand, is handled by MIPv6 with a Fast Handover (FHO) extension, as described in chapters 5 and 6, and does not require intervention of SIP unless a large difference in the available resources in the old and new access requires a session renegotiation. It is worth noting that the use of FHO allows for seamless handovers if mid-session mobility is performed at layer 3, which is not possible with SIP mobility — while seamless SIP mobility may be achieved with multihoming, this approach would require two network interfaces, adding to the cost and energy consumption (therefore, to a lower battery life) of the mobile terminals.

In this section we analyze the inefficiencies of the joint use of SIP and MIPv6, particularly in an environment where end-to-end resource reservations must be performed. The message sequence for initiating a call between two roaming terminals is illustrated in fig. 7.1. *100 Trying* SIP responses and *PRACK* requests and responses have been omitted in the figure, since they are not in the critical path of signaling.

The sequence is initiated by the caller sending an *INVITE* with a message body containing an offer with a list of codecs supported by itself, along with the corresponding ports (at the caller end only); this message is sent via the outbound proxy, MMSP1.f. If the binding cache of MMSP1.f is not up to date with the caller's current CoA, this *INVITE* is

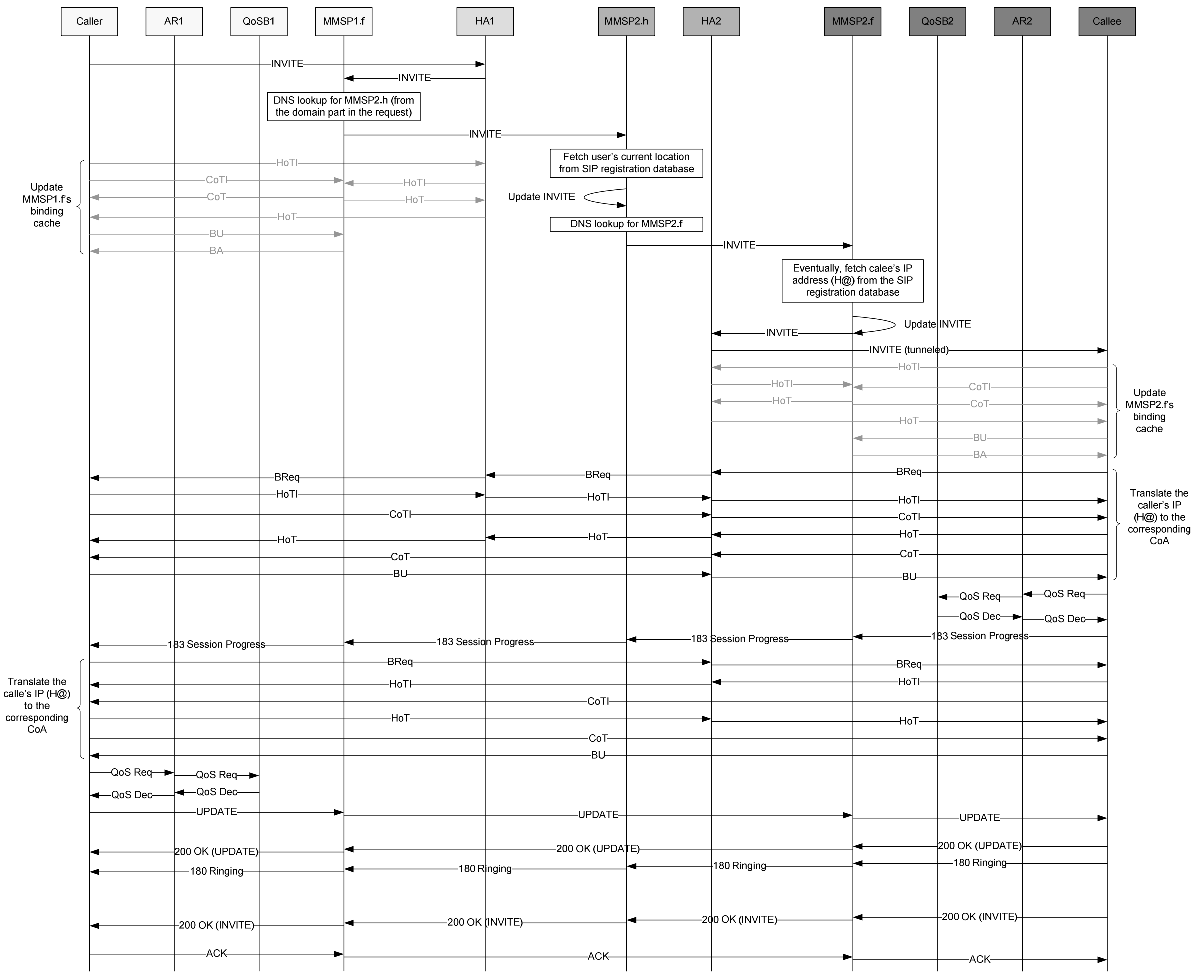

Caller
AR1
QoSB1
MMSP1.f
HA1
MMSP2.h
HA2
MMSP2.f
QoSB2
AR2
Callee
INVITE
DNS lookup for MMSP2.h (from the domain part in the request)
Fetch user's current location from SIP registration database
Update INVITE
DNS lookup for MMSP2.f
Update MMSP1.f's binding cache
HoTI
CoTI
CoT
HoT
BU
BA
Eventually, fetch calee's IP address (H@) from the SIP registration database
INVITE (tunneled)
Update MMSP2.f's binding cache
BReq
Translate the caller's IP (H@) to the corresponding CoA
QoS Req
QoS Dec
183 Session Progress
Translate the calle's IP (H@) to the corresponding CoA
UPDATE
200 OK (UPDATE)
180 Ringing
200 OK (INVITE)
ACK

**Figure 7.1: Inter-domain call without optimization (both terminals roaming)**

tunneled to its Home Agent (HA1), from where it is sent to MMSP1.f, introducing an additional delay corresponding to one RTT between the caller's home and foreign domains. The caller may then initiate a Return Routability Procedure (RRP — grayed out since it is not in the critical path of signaling) to MMSP1.f so that further messages between them are optimally routed. If the *Home Keygen Token (HKT)* has not expired since registration, only the *Care-of Test Init/Care-of Test (CoTI/CoT)* exchange is necessary, but otherwise a full RRP must be performed. Notice that mobility-unaware applications use Home Addresses (HoAs) as endpoints in order for layer-3 mobility to be transparent.

When the *INVITE* request arrives at MMSP1.f, it must find out the proxy responsible for the callee to send the *INVITE*. To this end, a Domain Name Service (DNS) lookup is performed, involving a round-trip to a root DNS server, another one to a top level DNS server and one or two[1] to the home domain of the callee, unless the entries are already cached. On receiving the *INVITE*, MMSP2.h looks up the registration database and finds out that the user

[1] At least an SRV lookup; however it is usually preceded by a NAPTR lookup.

(callee) is roaming; DNS lookups are performed to find out the proxy for the foreign (visited) domain. Notice that service authorization is mandatory, therefore MMSP2.h cannot send the *INVITE* directly to the callee, as packet filtering mechanisms would drop it.

MMSP2.f receives the *INVITE* and fetches the callee's IP address from its registration database (for the sake of simplicity, we assume that the callee has registered itself with the IP address rather than a hostname). Since regular SIP is not layer-3-mobility-aware, this IP address is a HoA; therefore, the message must go to the callee's home agent (HA2), from where it is tunneled to the callee.

When the callee receives the *INVITE*, it builds a list of the common codecs. In possession of the IPs and ports at both ends (the caller and itself), the callee may request resources to/from the caller (*QoS Req*). However, if resources are reserved for more than the wireless link, as in our case, the reservation must be made according to the physical points of attachment of the terminals, that is, their CoAs. The callee knows its own CoA, but not the caller's, therefore a *Binding Request (BReq)*[2] must be issued to the caller, which will trigger an RRP and a *Binding Update (BU)* from the caller to the callee, adding two RTTs between them. Since all of these messages must go through the HA of the callee (the caller has no binding for the callee yet) and some of them (*BReq*, *HoTI* and *HoT*) through the HA of the caller, this translates in 11 inter-domain traversals (considering that *HoTI/HoT*, and not *CoTI/CoT*, are in the critical path of signaling, as is most common).

The callee also initiates a return routability procedure and binding update to MMSP2.f, so that future messages need not be tunneled; however, the *183 Session Progress* response must still go through the HA (otherwise MMSP2.f would drop it, since it has no binding for the callee).

When the caller receives the *183 Session Progress*, it knows its own CoA, but not the callee's; therefore, it must send a *BReq* to the callee (symmetrical of the previously mentioned procedure). Two additional RTTs are, therefore, added (corresponding to 7 inter-domain traversals, not 11 as the previous one, since the callee already has a binding for the caller). Only now the HoA and CoA of the callee are known at the caller side, therefore only now a fully-formed QoS request may be performed at this side. As the amount of available resources may be less than what was reserved at the callee side, an indication of the final codec configuration (counter-answer) must be sent in an *UPDATE* request in order to synchronize the reservations. Hopefully, by this time all the binding caches are updated, meaning that the

[2] Notice that the above mentioned Binding Requests are not entirely compliant with the Binding Refresh Requests defined in [RFC3775]. Please refer to the note on Binding Requests in section 7.1.1 for details.

*UPDATE* (as well as all further signaling) travels through optimal paths. The media packets will also use the optimized path, since each terminal has a binding for the media address of the other one (which is usually the same as the signaling address, except in some multi-homed terminals).

It is worth noting that many of the inefficiencies (namely, RTTs between foreign and home domains of the caller, of the callee, and between the foreign domains of both) are due to the SIP protocol's unawareness of layer-3 mobility, and to the need to perform end-to-end resource reservations combined with this unawareness. As we will see in the next section, this scenario can be significantly improved by means of very simple procedures and cross-layer interactions.

### 7.1.1 Note on the Use of Binding Requests

Binding Requests (BReqs) in figure 7.1 behave somewhat differently from the Binding Refresh Requests (BRRs) defined in [RFC3775], which states that a mobile node should not respond to BRRs for addresses not in the Binding Update List (BUL). Although it is possible that the CN will respond with a BU if a packet or sequence of packets of any type (e.g., dummy packets) is sent to its HoA when it is roaming, we cannot rely on such solution because:

- There is no guarantee that it will do so.
- If the CN is at home, no BU would ever be received.

Therefore, with the behavior recommended by [RFC3775], it is not possible for a terminal to know for sure the physical location of the CN in order to perform an end-to-end reservation.

The reason for rejecting BRRs from nodes not in the BUL is to avoid being subject to a denial of service attack, since state maintenance is required for the RRP at the MT side. Unfortunately, it is not possible to make the procedure completely stateless: while the home and care-of init cookies could be implemented in such a way that the MT would not need to keep them, the first received keygen token (home or care-of) must be stored until the other one arrives. It is possible, though, to implement binding requests as used in this work by having up to a limited number of low priority entries in the BUL used for replying to binding requests from nodes not in the BUL. These low priority entries are promoted to regular entries only when the conditions that would normally trigger a BU are met. Due to their limited number, these low priority entries do not affect the normal operation of the BUL even under a

DoS attack (only a service which is not anyway provided by [RFC3775] may be denied); under normal conditions, this additional and potentially useful service is provided.

## 7.2 Optimizing the Use of SIP with MIPv6

The call initiation scenario illustrated in the previous section can be much improved by means of very simple procedures and cross-layer interactions. In this section we propose a series of optimizations that allow for a significant reduction in the session setup time, as will be shown in sections 7.5 and 0.

The first optimization consists on eliminating the need for the *INVITE* message between the caller and MMSP1.f to go through the HA. While this could be easily accomplished by having the terminal keep the MMSP's cache updated all the time, such approach would lead to a lot of unnecessary signaling, since most of the time it is not actually communicating, and would limit its ability to conserve energy using the paging features of the system. Therefore, we propose a different approach: using the CoA as source IP address of the packet containing the *INVITE* message. Notice that the *INVITE* message itself still uses the HoA. Responses to the *INVITE* will be delivered to the CoA since the proxy adds a *received* parameter with the source IP address of the packet to the *Via* header of a received request, whenever the *sent by* parameter in the *Via* header does not match that source IP address (in our case it contains the HoA). The terminal may then perform the RRP, which is not on the critical path of signaling, and then maintain MMSP1.f's binding cache updated for the whole duration of the call, so that future requests (*PRACK*, *UPDATE*, *ACK*, *re-INVITE*s, etc.) and their respective responses will always use the optimized path.

The goal of the second optimization is to eliminate the need for the *INVITE* message between MMSP2.f and the callee to go through the callee's HA. Contrary to the previous case, the message is not generated at the mobile terminal. In order to use the callee's CoA as the destination address, MMSP2.f must have knowledge of the mapping between the callee's HoA and CoA, as the HoA is the one used by the application layer. In order to provide this information, we introduce a cross-layer interaction at MMSP2.h: after retrieving the IP address (HoA) of the callee from the registration database, MMSP2.h queries the HA to find out the callee's current CoA. The Uniform Resource Identifier (URI) in the request line is then changed to the IP address (HoA), as usual, but with tag containing the current CoA (e.g., "coa=FF1E:03AF::1") appended. Using the CoA from the tag in the request line as the destination IP address of the packet, MMSP2.f may send the INVITE directly to the callee. Notice that the use of the CoA tag by MMSP2.f for direct forwarding does not add any security issue to standard SIP, since the same would occur if MMSP2.h had placed the CoA directly in the request line of the forwarded *INVITE*.

The third optimization concerns the elimination of the DNS lookup at MMSP2.h when forwarding the *INVITE* request: if the registration for redirection includes the IP address of the inbound proxy where to forward an incoming *INVITE* (MMSP.f, in this case), no DNS lookup to find this proxy is necessary. The use of the *Path* header field described in [RFC3327] is recommended for conveying this information, while also providing a simple means of enforcing the traversal of an MMSP at the foreign domain, necessary to perform service authorization and filtering.

The fourth optimization is related to the need to perform network resource reservations concerning more than the wireless link. Since the requests are performed for a path-optimized flow, they must be performed between the physical locations (that is, the CoA) of both terminals. Once again, we rely on the transport of CoA information in the application signaling. However, since there is no guarantee that the media will use the same IP addresses as SIP signaling (particularly with multi-homed terminals), the CoA information used to this end is conveyed not in SIP, but in the protocol used for session negotiation. Inclusion of CoA information in Session Description Protocol (SDP) [RFC2327] and SDP new generation (SDPng) [Kutscher05] is discussed in section 7.2.1.

One might argue that the inclusion of layer 3 mobility information in an application protocol such as SIP should not be done because it breaks the layering principle; it is worth noting, however, that not only does the standard SIP already include layer-3 information (IP addresses) in its headers, but also that cross-layer information would be required by any protocol with similar characteristics to SIP, namely regarding independence between the signaling and media interfaces.

### 7.2.1 Inclusion of CoA Information in SDP(ng)

Media negotiation and configuration for the sessions is performed using either the SDP or its “new generation” successor, SDPng. The inclusion of CoA information requires simple extensions to these protocols.

In SDP, the IPv6 address of the media endpoint is conveyed in the *c*= field [RFC2327, RFC3266]. Since it is not possible to change the definition of this field without breaking backward compatibility, it contains the HoA only. For conveying CoA information we resort to a newly defined attribute, the standard way of extending SDP. This attribute, named *coa*, has a similar definition to that of the *c*= field:

```
a=coa: <network type> <addr type> <connection addr>
```

The use of the *coa* attribute is illustrated in the following example:

```
c=IN IP6 FF1E:03AD::7F2E:172A:1E24
a=coa:IN IP6 FF1E:03AF::1
```

In SDPng, the HoA is conveyed by an *rtp:ip-addr* element. The CoA is conveyed by a newly defined element, *rtp:ip-coa*, as illustrated in the following example:

```
<rtp:udp name="rtp-cfg1" ref="rtp:rtpudpip6">
   <rtp:ip-addr>FF1E:03AD::7F2E:172A:1E24</rtp:ip-addr>
   <rtp:ip-coa>FF1E:03AF::1</rtp:ip-coa>
   <rtp:rtp-port>9456</rtp:rtp-port>
   <rtp:pt>1</rtp:pt>
</rtp:udp>
```

### 7.2.2 Optimized Initiation Sequence

Figure 7.2 shows the message sequence for an optimized multimedia call with both terminals roaming (messages not in the critical path of signaling are omitted). Since the *INVITE* is sent with the CoA as source IP address, it goes directly to MMSP1.f. A DNS lookup is performed (there is no way to avoid it) and the message is forwarded to the callee's home proxy. MMSP2.h changes the request line from the URI to the HoA of the callee, adding a *coa* tag with the callee's current CoA (retrieved from HA2) to the request line of the *INVITE*. Although in the optimal case MMSP2.h and HA2 would be integrated, with the respective location databases merged, even if they are not, communication between them is fast and efficient, since they belong to the same domain and are located close to one another. This communication, however, requires new messages, *Binding Query (BQ)* and *Binding Response (BR)*, since the standard BRR and BU messages are exchanged with the MT, not the HA.

Since MMSP2.h has the IP address of the callee's outbound/inbound proxy, MMSP2.f, there is no need for a DNS lookup. Using the information from the *coa* tag on the request line, MMSP2.f is able to send the INVITE directly to the callee without the need to go through its HA. When this message arrives, the callee retrieves the caller's HoA and CoA from the SDP, and uses this information to request network resources.

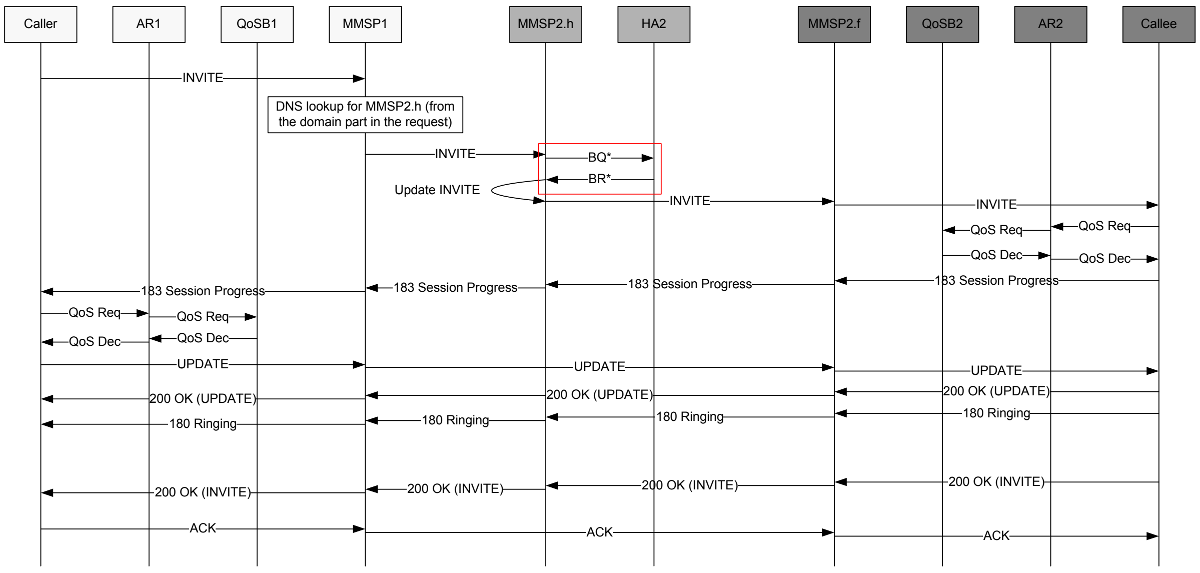


**Figure 7.2: Optimized inter-domain call (both terminals roaming)**

After receiving the reservation response, the callee sends a 183 Session Progress response, containing an answer with the set of common codecs and their respective ports at both ends, to the caller. Information on the callee's CoA is included in the SDP; this information is used by the caller to perform the resource reservation on its side.

Usually, by the time the *UPDATE* is sent, both terminals have already established bindings with their respective proxies. However, the caller may include a *coa* tag with the CoA of the callee to the request line, lest MMSP2.f not have yet a binding for MT2: if this is the case, MMSP2.f uses the tag to send the request directly to the CoA, as it has previously done with the *INVITE*; otherwise, the tag is ignored. Notice that the *UPDATE* (and all further requests) does not traverse the home proxy of the callee, since only the local (foreign) proxies, which have responsibilities in service control, have added themselves to the *Record-Route* header of the initial *INVITE*.

On receipt of the *UPDATE* with the final configuration, the callee knows that the reservations have been successfully performed and that network resources are available, therefore it may start ringing.

It is worth noting that bindings must still be established between the terminals for the media sessions, meaning that the overhead of both solutions will be comparable (except for a few encapsulated packets in the standard signaling); however, these message exchanges are moved out of the critical path of signaling in the optimized case.

Mid-session mobility is handled exclusively at the layer 3 by MIPv6 (with Fast Handover extensions, in our case). SIP sessions are handled similarly to non-SIP ones based on UDP or TCP, and no re-INVITE message is sent unless a session renegotiation (e.g., for changing the codec or bit rate) is necessary; this way, it is possible to seamless support both mobile multimedia and non-multimedia applications in the same architecture.

## 7.3 SIP Registration

A user at home registers normally with the local (home) MMSP using its Address of Record[3] in the *To* header and the IP address (HoA) in the *Contact* header. A roaming user must register itself with the foreign MMSP, since it will be performing service control, but also with the MMSP of its home domain for location purposes; therefore, MMSP2.f forwards the *REGISTER* request to MMSP2.h. In the standard case, the user registers itself with MMSP2.f as user@home.com, using the IP address as *Contact*; MMSP2.f changes the *Contact* to user%40home.com@foreign.com and forwards the registration request to MMSP1.h. In the optimized case, the user registers itself with MMSP2.f as user@home.com using the IP address as *Contact*, similarly to the standard case (fig. 7.3). However, MMSP2.f does not change the *Contact*: instead, it adds a *Path* header [RFC3327] with its own IP address, forcing incoming requests from MMSP1.h to traverse it. Though the *Path* header is an extension to the basic SIP protocol, it is a standard one.

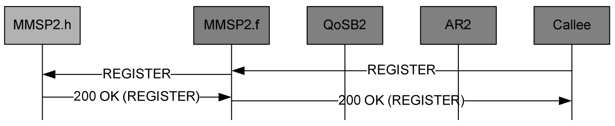


**Figure 7.3: Registration procedure**

It is worth noting that, contrary to the standard registration approach where any proxy of the foreign domain may be traversed by an incoming request, the optimized approach associates the terminal with a given proxy. However, being tied to a particular proxy does not degrade fault tolerance when compared to having pool of *N* proxies: if the probability of failure of each proxy is $p_f$, then the probability of failure of the one chosen by DNS lookup among a pool of *N* is the sum of the probability of choosing each of the proxies times the failure probability of that particular proxy, that is, $\sum_{i=1}^{N}(\frac{1}{N}p_f) = p_f$, the same as that of any individual proxy. Moreover, if the MT periodically contacts the proxy in order to check that it is "alive," failure will be detected, and the MT may re-register with a different one; in this case, resilience is actually increased by sticking to a particular inbound proxy. On the other hand, proxy pooling could still be achieved by providing MMSP2.h an anycast address

[3] Sometimes erroneously called Network Access Identifier (NAI)

instead of the regular unicast address of MMSP2.f at registration time, using a method like the one proposed in [Engel98].

## 7.4 Issues with Dormancy/Paging

Energy is a scarce resource in mobile terminals, particularly in the smaller and lighter-weighted ones. Therefore, any architecture where small and low-power devices are foreseeable must provide some mechanism for dormancy/power saving. In our architecture, support for dormancy is provided by the Paging Controller (PC). This entity provides an alternate CoA to the MT. When packets arrive, the PC buffers them and informs the terminal that it must wake up; when the MT wakes up, the buffered packets are delivered and the MT starts using its new, real CoA (more details in [Banchs05]).

Support for dormancy/paging disallows keeping fresh the binding cache of correspondent entities, namely MMSPs: the terminal only acquires a real CoA when it wakes up; therefore, only after waking up it can update the correspondents' binding caches. If the terminals are idle for long periods of time, as usually happens with mobile phones, the probability that the newly acquired CoA differs from the one the terminal had before going asleep is pretty high, even with slow mobility patterns. Dormancy/paging also introduces an additional delay in any message received, corresponding to the time it takes for the terminal to be located and waken up. Other than these, dormancy/paging has no issues: the alternate CoA provided by the PC may be used for location purposes as does the real CoA. Notice that this dormancy/paging affects only the reception of the initial *INVITE*.

## 7.5 Delay Analysis

In this section we perform a comparative analysis of the dial-to-ringtone delay with standard and optimized signaling for a call between two roaming terminals (processing delays at the nodes are not accounted for). In order to simplify our analysis, the following assumptions have been made:

- Inter-domain delays are symmetrical, i.e., it takes about the same time to go from A to B as from B to A.
- Compared to the delay a message suffers in the wireless link or in inter-domain trips, the delay in intra-domain wired links is minimal and may, therefore, be neglected.
- In the RRP, the *HoTI/HoT* exchange takes longer than the *CoTI/CoT*.
- An RRP from a terminal to a local MMSP takes less time than the same procedure to a remote terminal, provided the HA is the same in both cases.

- The *BU* from the Caller arrives at MMSP1.f before the 183 Session Progress in the sequence of fig. 7.1, meaning that the 183 S.P. will not go through the HA even in the standard case. This is almost always true, failing only if the inter-domain delay between the home and foreign domains of the caller is disproportionately large compared to the other delays.

In this analysis we will use the following notation: $T_{W1}$ and $T_{W2}$ are the delay at the wireless links of the caller and the callee, respectively; $T_{F1F2}$, $T_{F1H1}$, $T_{F1H2}$, $T_{F2H1}$ and $T_{F2H2}$ are the inter-domain one way trip delays (between combinations of the Foreign and Home domains of the caller — 1 — and the callee — 2); $T_{DNS1}$ and $T_{DNS2}$ are the delays for DNS lookups of the home and the foreign domains of the callee, respectively. Notice that if the entries are not cached, the DNS lookups imply at least one RTT to the DNS registrar, to find out the DNS server of the domain to be resolved, and another one or two to that domain, to find out the address of a SIP proxy (*SRV* record and, eventually, *NAPTR* record).

## 7.5.1 Standard Case

With standard, non-optimized signaling (refer to fig. 7.1), the *INVITE* takes $T_{Inv}$ (eq. 7.1) to go from the caller to the callee. The QoS request can only be initiated after the caller's CoA has been found, which takes $T_{CoA1}$ (eq. 7.2). The QoS request/response at the callee side takes $T_{QoS1}$ (eq. 7.3). Then, the 183 Session Progress takes $T_{SP}$ (eq. 7.4) to go from the callee to the caller. Finding out the callee's CoA takes $T_{CoA2}$ (eq. 7.5). The QoS request/response at the caller side takes $T_{QoS2}$ (eq. 7.6). Finally, the *PRACK* is sent to the callee with the SDP counter-answer, after which it may start ringing. Until the *180 Ringing* arrives at the caller, there is an additional $T_{Pra}$ (eq. 7.7). Adding all these delays, we obtain a total dial-to-ringtone delay of $T_{Std}$ (eq. 7.8).

$$T_{Inv} = T_{W1} + 2T_{F1H1} + T_{DNS1} + T_{F1H2} + T_{DNS2} + 3T_{F2H2} + T_{W2} \tag{7.1}$$

$$T_{CoA1} = 4T_{W1} + 4T_{W2} + 4T_{F2H2} + 3T_{H1H2} + 3T_{F1H1} + T_{F1H2} \tag{7.2}$$

$$T_{QoS1} = 2T_{W2} \tag{7.3}$$

$$T_{SP} = T_{W1} + T_{W2} + T_{F2H2} + T_{F1H2} \tag{7.4}$$

$$T_{CoA2} = T_{W1} + 4T_{W2} + 3T_{F1H2} + 3T_{F2H2} + T_{F1F2} \tag{7.5}$$

$$T_{QoS2} = 2T_{W1} \tag{7.6}$$

$$T_{Pra} = 2T_{W1} + 2T_{W2} + T_{F1F2} + T_{F2H2} + T_{F1H2} \tag{7.7}$$

$$T_{Std} = 14T_{W1} + 14T_{W2} + 2T_{F1F2} + 5T_{F1H1} + 7T_{F1H2} + 12T_{F2H2} + T_{DNS1} + T_{DNS2} \tag{7.8}$$

### 7.5.2 Optimized Case

With our proposed optimizations (refer to fig. 7.2), the *INVITE* takes $T_{Inv}$ (eq. 7.9) to go from the caller to the callee. Then, the QoS request takes $T_{QoS1}$ (eq. 7.10). Then, the 183 Session Progress takes $T_{SP}$ (eq. 7.11) to go from the callee to the caller. The QoS request/response at the caller side takes $T_{QoS2}$ (eq. 7.12). Finally, the *PRACK* is sent to the callee, after which it may start ringing. Until the 180 Ringing arrives at the caller, there is an additional $T_{Pra}$ (eq. 7.13). Adding these delays, we get a total dial-to-ringtone delay of $T_{Opt}$ (eq. 7.14) for the optimized signaling case.

$$T_{Inv} = T_{W1} + T_{DNS1} + T_{F1H2} + T_{F2H2} + T_{W2} \tag{7.9}$$

$$T_{QoS1} = 2T_{W2} \tag{7.10}$$

$$T_{SP} = T_{W1} + T_{W2} + T_{F2H2} + T_{F1H2} \tag{7.11}$$

$$T_{QoS2} = 2T_{W1} \tag{7.12}$$

$$T_{Pra} = 2T_{W1} + 2T_{W2} + T_{F1F2} + T_{F2H2} + T_{F1H2} \tag{7.13}$$

$$T_{Opt} = 6T_{W1} + 6T_{W2} + T_{F1F2} + 3T_{F1H2} + 3T_{F2H2} + T_{DNS1} \tag{7.14}$$

### 7.5.3 Comparison

If we consider $T_{W1} = T_{W2} = T_W$, $T_{DNS1} = T_{DNS2} = T_{DNS}$ and all inter-domain traversal delays equal to $T_{ID}$, the dial-to-ringtone delays in the standard and optimized cases become those of equations 7.15 and 7.16, respectively. As can be seen, there is always a more than twofold improvement.

$$T_{Std} = 28T_W + 26T_{ID} + 2T_{DNS} \tag{7.15}$$

$$T_{Opt} = 12T_W + 7T_{ID} + T_{DNS} \tag{7.16}$$

Figure 7.4 illustrates the variation of the dial-to-ringtone delay with standard and optimized signaling when all inter-domain delays for one-way trip are equal and assume values from 2 ms to 64 ms. The DNS lookups take twice that value plus 5 ms (RTT to the DNS registrar). Delay on both wireless links is 10 ms. The delay with optimized signaling is reduced to about one third of that obtained with standard, non-mobility-aware session signaling, a significant improvement. For example, with an inter-domain delay of 64 ms, the dial-to-ringtone delay is 2.2 s in the standard case and only 0.7 s with our proposed optimizations.

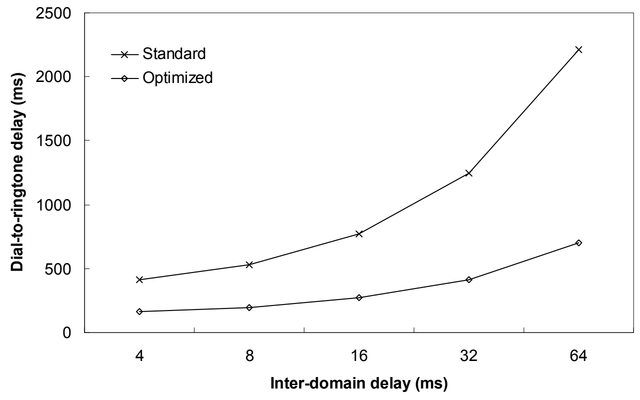


**Figure 7.4: Dial-to-ringtone delay with varying inter-domain delay**

## 7.6 Simulation Results

The efficiency of the standard and optimized signaling scenarios for the initiation of a mobile multimedia call was evaluated using the *ns-2* simulator [NS2] under Linux. The simulations comprise all possible combinations of: (1) caller terminal at the home domain or roaming; (2) callee terminal at the home domain or roaming; (3) caller and callee physically attached to the same or different domains and; (4) in the first case of (3), caller and callee physically attached to the same or different ANs, therefore representing all possible intra- and inter-domain call scenarios.

The standard *ns-2* simulator supports neither MIPv6 nor SIP. MIPv6 support was provided by the *MobiWan* extension [MobiWan226], which we further improved by adding several features (reverse encapsulation, RRP, etc.) it did not support. We have also modified our implementation of SIP [PriorNS], introduced in chapter 6, in order to integrate it with MIPv6, supporting the two above mentioned scenarios (standard and optimized).

Some processing delays are accounted for in the simulation model. Message processing is performed in a FIFO fashion, meaning that processing of each message can only begin after all previous ones have been processed. Processing delays for SIP messages were simulated at both the terminals (15 ms) and the MMSP (0.8 ms), with an increment for messages with SDP bodies (10 ms in the terminals and 0.8 ms in the MMSP). QoS request processing at the QoS brokers is also accounted for (1 ms). The remaining processing delays are considered negligible when compared to these, and thus ignored in the simulations. DNS lookups were not simulated for lack of a realistic model for DNS caching. Moreover, since our purpose is the evaluation of signaling, no actual session data was simulated.

The topology used in the simulations is the same as the one used in the simulations of chapter 6, illustrated in fig. 6.12. It contains four domains, the leftmost one containing two

ANs, one of which with two ARs. Notice that the total inter-domain delay is twice the inter-domain link delay. Though very simple, this topology allows us to simulate all possible combinations of roaming and non-roaming terminals: physically attached to the same AR, same AN and different AR, same domain and different ANs, or to different domains. 128 terminals were uniformly spread among the access networks, each terminal having a 50% probability of being at its home domain and 50% of being roaming. Random calls were generated between pairs of terminals, with an average duration of 120s and a mean interval between generated calls of 15 s, for a simulated time of 24 hours (86400 s). Several runs of each simulation were performed with different pseudo-random number generator (PRNG) seeds; different streams of the standard ns-2.27 PRNG were used for generating independent events.

In a first experiment we evaluated the call setup delay with different values of propagation delay for the inter-domain links. The setup delay is evaluated at the caller side, that is, from the moment the *INVITE* is sent to the moment the *200 OK* for the *INVITE* is received and the *ACK* transmitted, subtracting the time it takes for the callee to answer the call (delay from sending the *183 Session Progress* to sending the *200 OK*). The results from this experiment are shown in fig. 7.5 for both the standard and optimized sequences, in three different roaming scenarios (relative locations of the terminals intervening in a call). The roaming scenarios are identified by four letters, *abcd*, where *a* indicates if the caller terminal is at its home domain (*a=h*) or roaming (*a=r*), *b* holds similar information for the callee, *c* indicates if the terminals are connected to the same administrative domain (*c=y* or *c=n*), and *d* indicates if they are connected to the same AN (*y* or *n*). For example, *hhnn* means that both the caller and the calle are at their home domains, which are different (they are connected to different domains).

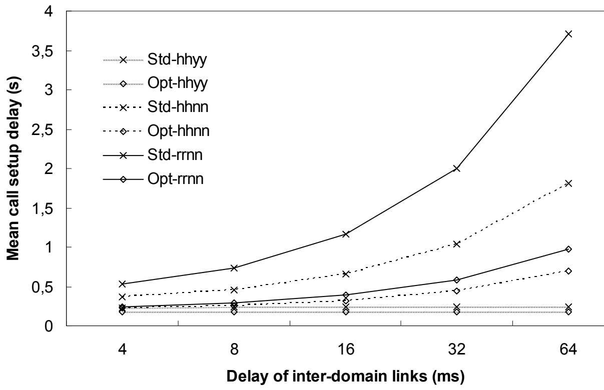


**Figure 7.5: Call setup delay with varying inter-domain link propagation delay**

As expected, the setup delay does not vary with the propagation delay of inter-domain links in the *hhyy* scenario, since all signaling is performed intra-domain in this case. The worst scenario in terms of call setup delay is the *rrnn*, where both terminals are roaming and physically attached to different domains (as in figures 7.1 and 7.2). In this scenario, the difference in call setup delay between standard and optimized signaling is large, and increases with the propagation delay of inter-domain links: with 64 ms of propagation delay at the inter-domain links, the call setup delay with standard signaling is about 4 times larger than with optimized signaling. The 95% confidence intervals for the mean (5 runs), omitted in the figure for clarity, were less than ±3% of the mean in all cases.

In a second experiment we fixed the inter-domain propagation delay at 16 ms and introduced a varying loss probability at the wireless links; 802.11 MAC layer retransmissions were disabled so that losses were not compensated for. The results of this experiment are shown in fig. 7.6 for the different roaming scenarios, including 95% confidence intervals (10 runs).

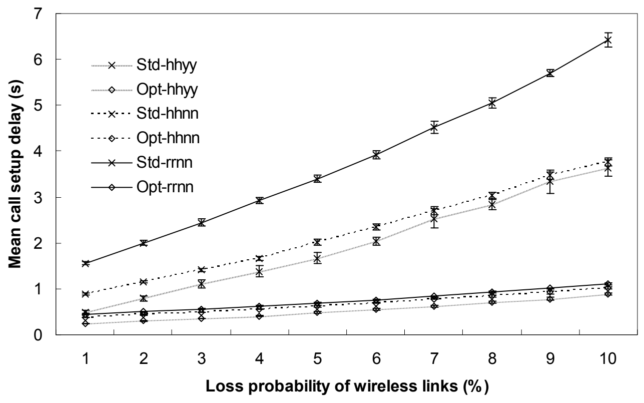


**Figure 7.6: Call setup delay with varying loss probability in the wireless links**

The figure clearly shows that the non-optimized scenario is much more severely affected by packet losses than the optimized one; this behavior stems from the much larger number of exchanged messages. It is worth noting that, even with a packet loss ration of 1%, the mean setup delay of the most favorable roaming scenario (*hhyy*) with standard signaling was larger than that of the least favorable one (*rrnn*) with optimized signaling, a gap that is largely widened as the loss probability increases.

The above presented results show the clear advantage, in terms of call setup delay, of the optimized signaling method over the standard one. The improvement is even more dramatic for long distance calls (larger inter-domain propagation delays) and/or in the presence of packet loss in the wireless links, even though small.

## 7.7 Conclusions

In this chapter we identified the sources of inefficiency with the joint use of SIP and Mobile IPv6 (the probable protocols for session initiation and mobility support, respectively, in the next generation telecommunication systems) for the initiation of mobile multimedia applications, particularly when end-to-end resource reservations must be performed for the media. This inefficiency generally stems from SIP/SDP's unawareness of layer 3 mobility, and from the need to perform resource reservations accounting for the physical points of attachment of the terminals combined with that unawareness. A solution for these inefficiencies was proposed, based on the direct use of the Care-of Addresses in some messages (namely in the short-lived message transactions in call initiation) and on cross-layer interactions (use of layer 3 location information in session setup signaling).

The advantages of the proposed optimizations in session establishment were analyzed, and simulation results have demonstrated that the session initiation sequence is much faster with the optimizations than in the standard case, particularly in the presence of larger inter-domain link propagation delays (long distance calls) or packet loss in the wireless links.

# CHAPTER 8

# INTER-DOMAIN QOS

The provision of multimedia services with real-time requirements in the Internet across domain boundaries is conditioned by the ability to ensure that certain Quality of Service requirements are met. The introduction of QoS routing mechanisms able to select paths with the required characteristics is of major importance towards this goal. Though much attention has been paid to QoS in IP networks, most of the effort has been centered on intra-domain; much less has been done in the scope of inter-domain, a much more complex problem, for a number of reasons. The Internet is a complex entity, comprised of a large number of Autonomous Systems (ASs) managed by very diverse operators. If it is to be widely deployed, an inter-domain QoS routing mechanism must be capable of handling the heterogeneity of the Internet and impose minimum requirements on intra-domain routing, in order to be appealing to the different operators. The introduction of QoS metrics should not disrupt currently existing inter-domain routing: the QoS and non-QoS versions should interoperate, allowing for incremental deployment among the different networks, and the stability of the routes should not be overly affected by the QoS mechanisms. A final requirement is scalability: a solution that does not scale to the dimension of the Internet cannot be deployed widely enough to be useful.

In this chapter, we address the problem of inter-domain QoS routing, part of the overall solution to the problem of end-to-end QoS, introduced in chapter 5. Our proposal is based on Service-Level Agreements (SLAs) for data transport between peering domains, using virtual-trunk type aggregates. The problem is formally stated and formulated in Integer

Linear Programming (ILP), and proof is given that routes obtained through the optimization process are cycle-free. We propose a practical solution for inter-domain QoS routing based on both static and coarse-grained dynamic metrics: it uses the light load delay and assigned bandwidth (both static) in order to improve the packet QoS and make better use of network resources, and a coarse-grained dynamic metric for path congestion, intended to avoid overloaded paths. We define the QoS_INFO extension to the Border Gateway Protocol (BGP) [RFC4271] to transport these QoS metrics and the algorithm to use them for path selection. Using the ns-2 simulator [NS2], we compare the proposed protocol with standard BGP and with BGP with the QoS_NLRI extension [Cristallo04] conveying static one-way delay information (expected route delay in light load conditions). Optimal solutions for the same topology and traffic matrix, obtained using the ILP formulation in a MIP (Mixed Integer Programming) code, are also used as baselines for comparison. Results show that the QoS parameters of the route set obtained with QoS_INFO are the closest to those of the optimal route set. Specifically, we show that congestion and packet losses are much lower with QoS_INFO than with standard BGP or with QoS_NLRI. Parts of this work have been published in [Prior06a], [Prior07b], [Prior07c] and [Prior07d].

The rest of the chapter is organized as follows. Next section contains the formal description of the problem and its formulation in ILP. Section 8.2 presents the proposed protocol and the associated path selection algorithm. In section 8.3 we compare the optimal results with simulation results from standard BGP, QoS_NLRI and QoS_INFO. Finally, section 8.4 contains a summary of the conclusions of this chapter.

## 8.1 Inter-Domain QoS Routing with Virtual Trunks

In this section we formally describe the problem of inter-domain routing with virtual trunks and formulate it as an ILP problem. In section 8.3, this formulation will be used in a MIP solver to obtain optimal route sets, against which we compare the results of our proposal.

### 8.1.1 Virtual Trunk Model of the Autonomous Systems

Though the use of some inner information of the ASs is important for inter-domain QoS routing, information on the exact topology and configuration of the ASs should not be used for inter-domain routing for two reasons: (1) the level of detail would be excessive, complicating the route computation task and, most important, (2) network operators usually want to disclose the minimum possible amount of internal information about their networks.

In the work presented in this chapter, we use a “black box” model where only externally observable AS information is disclosed. The intra-domain connections between edge routers are replaced by virtual trunks with specific characteristics interconnecting the peering ASs. Each virtual trunk corresponds to a particular (*ingress link, egress link*) pair, and has a specific amount of reserved bandwidth and an expected delay. These values depend on the internal topology of the AS, on the intra-domain routing and on resource management performed by the operators, and usually reflect SLAs established between the operator of the AS they traverse and the operators of the peering[1] ASs.

The virtual trunk model is an edge-to-edge transport service that can be implemented resorting to DiffServ, the most widely used QoS framework for IP networks. In traffic engineered Multiprotocol Label Switching (MPLS) networks [RFC2702, RFC3031], it is implemented by assigning the packets that belong to a given virtual trunk to a Label-Switched Path (LSP) and reserving the corresponding amount of bandwidth for that LSP. This feature is supported by major network equipment manufacturers, and is frequently used to implement "virtual leased line" or similar services. The virtual trunk model, however, is especially appropriate for Dense Wavelength Division Multiplexing (DWDM) transit networks, where conversion between the electrical and optical domains happens only at the edges, and lightpaths provide edge-to-edge transport pipes with given capacities. In such networks, the virtual trunk model is not only easily implemented, but also a natural management model.

The virtual trunk model of ASs is illustrated in fig. 8.1. A Service-Level Specification (SLS) between domain S1 and domain T1 specifies that an amount X of traffic may flow between S1 and domain T3; an SLS between domain T1 and domain T3 specifies that Y volume of traffic may flow between T1 and domain D1. Aggregates are managed internally within each (transit) domain, ensuring that enough resources are assigned, and no imposition is made regarding mechanisms used to this end.

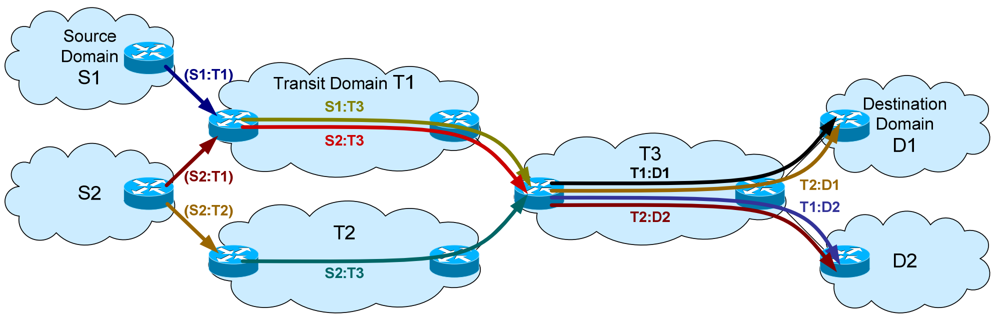


**Figure 8.1: Virtual trunk type SLSs**

[1] The word peering is used here in a loose sense, and includes customer/provider relationships in addition to strict peering interconnections.

The configuration of the virtual trunks must be consistent with the inter-domain links. In particular, the summed bandwidth of all virtual trunks traversing AS *j* and going to AS *k* must be less than the bandwidth of the inter-domain link connecting ASs *j* and *k*; similarly, the summed bandwidth of all virtual trunks coming from AS *i* and traversing AS *j* must be less than the bandwidth of the inter-domain link connecting ASs *i* and *j*.

## 8.1.2 Problem Statement

Let $G = (V, E)$ be an undirected graph with edge capacities $c_{i,j}$ and edge delays $w_{i,j}$. Each node represents an AS, and the edges correspond to the inter-domain links. Additionally, let us define a set $F$ of aggregate flows between pairs of nodes and a corresponding matrix of traffic demands $a_{s,d}$ for all $(s,d) \in V^2$, where $s$ and $d$ denote the source and destination nodes, respectively.

Given any triplet $(i, j, k)$ of nodes such that $i$ is directly connected to $j$ and $j$ is directly connected to $k$ (that is, $\{\{i, j\}, \{j, k\}\} \subset E$), there may be a traffic contract (SLS) stating that $j$ provides a virtual trunk between $i$ and $k$ with reserved capacity $r_{i,j,k}$. The volume of data transported from $i$ to $k$ via $j$ per time unit is, therefore, bounded by $r_{i,j,k}$. If no such contract exists, we say that $r_{i,j,k} = 0$. Since each virtual trunk is mapped to an actual path inside the AS, it has an associated delay $y_{i,j,k}$, corresponding to the delay of that path. Call $L$ the set of all virtual trunks $(i, j, k)$.

The virtual trunks must satisfy the conditions $o_{j,k} + \sum_i r_{i,j,k} \le c_{j,k}$, where $o_{j,k}$ is the capacity reserved for traffic originated at node $j$ and destined to or traversing node $k$, and $t_{i,j} + \sum_k r_{i,j,k} \le c_{i,j}$ where $t_{i,j}$ is the capacity reserved for traffic destined to node $j$ and originated at or traversing node $i$.

The expected total delay suffered by packets of a given flow is the sum of the $w_{i,j}$ and $y_{i,j,k}$ parameters along the path followed by the flow. Our goal is to find the set of hop-by-hop routes that minimizes the delay while guaranteeing that inter-domain link and virtual trunk capacities are not exceeded.

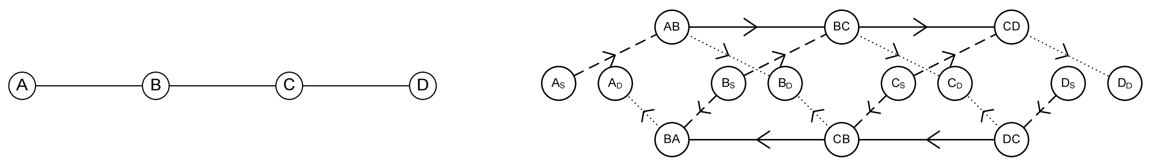
## 8.1.3 Problem Statement Transform

In order to formulate the stated problem as an ILP problem, we first transform the original graph into a transformed graph where the virtual trunks are explicitly accounted for.

#### 8.1.3.1 Transform Graph

While it is possible to transform the graph into a directed multigraph where each edge corresponds to a virtual trunk, by doing so it would be difficult to account for the delays of all links in the original graph (inter-domain links) without counting some of them twice. For this reason, and since graphs are easier to deal with than multigraphs, we add virtual nodes to the directed multigraph in order to obtain a resulting directed graph.

Virtual trunks are established between an entry link and an exit link. Therefore, we add two virtual vertices per link of the original graph, one for each direction, and virtual trunks are represented by edges connecting these virtual nodes. Moreover, in order to forbid a node of the original graph from being traversed directly (instead of via a virtual trunk), we split each original node into two: one source virtual node, with outgoing edges only, and one destination virtual node, with incoming edges only. Flows on the transform graph exist between source virtual nodes and destination virtual nodes.



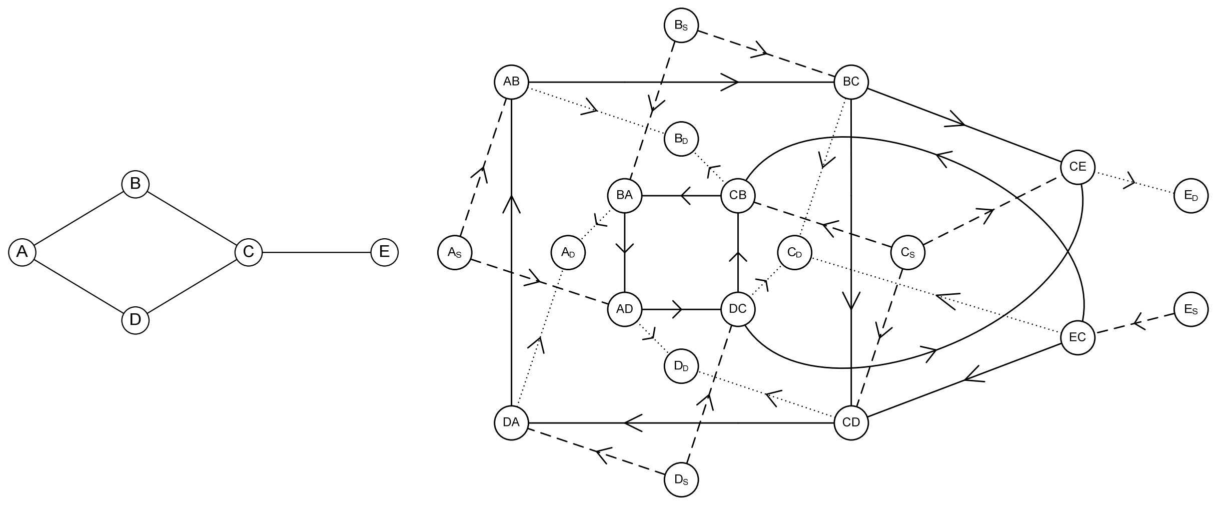
a) Original graph b) Transform graph

**Figure 8.2: Simple network with 4 nodes**

A very simple example of an original graph and its transform with all possible virtual trunks is shown in fig. 8.2. Link (A,B) on the original graph is represented by virtual nodes AB and BA; virtual trunk (A,B,C) is represented by an edge connecting AB to BC; node A is represented by the virtual source and destination nodes $A_S$ and $A_D$; and flow (A,D) is represented by flow ($A_S$,$D_D$), for example.

The solid edge connecting the virtual nodes $ij$ and $jk$ correspond to the virtual trunk for sending traffic from node $i$ to node $k$ via node $j$, and has capacity $r_{i,j,k}$ (the capacity of the virtual trunk), and delay $y_{i,j,k} + w_{j,k}$, where $y_{i,j,k}$ is the internal delay of the virtual trunk and $w_{j,k}$ the delay of the inter-domain exit link. Each dashed edge $(j_S, jk)$ corresponds to the inter-domain exit link from node $j$ to node $k$, and has delay $w_{j,k}$ and capacity $o_{j,k}$. Each dotted edge $(ij, j_D)$ corresponds to the inter-domain entry link in node $j$ from node $i$, and has zero delay and capacity $t_{i,j}$. Notice that even if there is no explicit $o_{j,k}$ and $t_{i,j}$ values, they may be computed using $o_{j,k} = c_{j,k} - \sum_i r_{i,j,k}$ and $t_{i,j} = c_{i,j} - \sum_k r_{i,j,k}$ .

In the example of fig. 8.2 there is only one possible path from A to C, corresponding to the virtual trunk through node B. Traffic sent from A to C is subject to a delay equal to the sum of $w_{a,b}$ (from the dashed edge $A_S \rightarrow AB$) and $y_{a,b,c} + w_{b,c}$ (from the solid edge $AB \rightarrow BC$); the dotted edge $BC \rightarrow C_D$ has zero delay. Regarding bandwidth, it is constrained by $o_{a,b}$, $r_{a,b,c}$, and $t_{b,c}$, all of them shared with other traffic.



a) Original graph b) Transform graph

**Figure 8.3: Cyclic network with 5 nodes**

Figure 8.3 provides a slightly more complex example — a cyclic graph and the respective transform containing all possible virtual trunks. Though the transform graph looks overly complex when compared to the original one, the number of variables and constraints in the ILP formulation is not increased, since a formulation based on the original graph would require variable unfolding in order to be linear. Also keep in mind that an undirected graph has half the number of edges of the equivalent directed graph.

### 8.1.3.2 Generation of the Transform Graph

In this section we present an algorithm for the generation of the transform graph $G' = (V', E')$ from the original graph and the set of virtual trunks, informally described above. The algorithm is as follows:

```
1. For each node $i \in V$
    1.1. Add node $i_S$ to the set $S$ of sources and to the set $V'$ of nodes
    1.2. Add node $i_D$ to the set $D$ of destinations and to $V'$
2. For each (undirected) edge $\{i, j\} \in E$
    2.1. Add node $ij$ to $V'$
    2.2. Add node $ji$ to $V'$
    2.3. Add edge $(ij, j_\mathrm{D})$ to the set $E'$ of edges
        2.3.1. Set capacity $c'_{ij,j_D} = t_{i,j}$
        2.3.2. Set delay $w'_{ij,j_\mathrm{D}} = 0$
    2.4. Add edge $(ji, i_\mathrm{D})$ to $E'$
        2.4.1. Set capacity $c'_{ji,i_\mathrm{D}} = t_{j,i}$
        2.4.2. Set delay $w'_{ji,i_\mathrm{D}} = 0$
    2.5. Add edge $(i_\mathrm{S}, ij)$ to $E'$
        2.5.1. Set capacity $c'_{i_\mathrm{S},ij} = o_{i,j}$
        2.5.2. Set delay $w'_{i_\mathrm{S},ij} = w_{i,j}$
    2.6. Add edge $(j_\mathrm{S}, ji)$ to $E'$
        2.6.1. Set capacity $c'_{j_\mathrm{S},ji} = o_{j,i}$
        2.6.2. Set delay $w'_{j_\mathrm{S},ji} = w_{j,i}$
3. For each (directed) virtual trunk $(i, j, k) \in L$
    3.1. Add edge $(ij, jk)$ to $E'$ and to the set $L'$ of virtual trunk edges
        3.1.1. Set capacity $c'_{ij,jk} = r_{i,j,k}$
        3.1.2. Set delay $w'_{ij,jk} = y_{i,j,k} + w_{j,k}$
4. For each flow $(i, j) \in F$
    4.1. Add flow $(i_\mathrm{S}, j_\mathrm{D})$ to the set $F'$ of flows
    4.2. Set traffic demand $a'_{i_\mathrm{S},j_\mathrm{D}} = a_{i,j}$
```

When the algorithm finishes, we have the transform graph $G' = (V', E')$, the associated edge capacity and edge delay matrices $C'$ and $W'$, a set $L' \subset E'$ of virtual trunk edges, a set $S \subset V'$ of source nodes and a set $D \subset V'$ of destination nodes, a set $F'$ of flows, and the respective traffic demand matrix $A'$.

#### *8.1.3.3 Complexity of the Transform Graph*

The number of nodes and edges of the transform graph $G'$ is related to the original (undirected) graph $G$ and the set of virtual trunks in the following way. The number of nodes is two per node of the original graph (one source and one destination, e.g., $\mathrm{A_S}$ and $\mathrm{A_D}$) plus two per edge of the original graph (one for each direction, e.g., AB and BA). The number of edges is four per edge of the original graph (combinations of source/destination and transmission/reception, e.g., ($A_S$,AB), (AB, $B_D$), ($B_S$,BA) and (BA,$A_D$)) plus one per virtual trunk (e.g., (AB,BC)).

In the example of fig. 8.3, the original graph has 5 nodes, 5 edges and 12 possible virtual trunks. The transform, therefore, has 20 nodes (2*5+2*5) and 32 edges (4*5+12).

### *8.1.3.4 Conversion of Routes from the Transform to the Original Graph*

A route $p'$ on the transform graph may be converted back to a route $p$ on the original graph by analyzing the traversed edges. Each traversed edge on the transform graph corresponds to a traversed node on the original graph, according to the rules of table 8.1. For example, the route ($B_S$,BC,CE,$E_D$) on the transform graph of fig. 8.3.b) corresponds to the route (B,C,E) on the original graph.

| Edge on the transform graph | Node on the original graph |
|---|---|
| $(i_S, ij)$ | $i$ |
| $(ij, jk)$ | $j$ |
| $(jk, k_D)$ | $k$ |

**Table 8.1: Route conversion from the transform to the original graph**

## 8.1.4 Problem Formulation in ILP

We now formulate our bandwidth-constrained global route and delay optimization hop-by-hop routing problem as an ILP problem with boolean variables using the transform graph. Formulation in the transform graph is somewhat simpler, as some constraints are already enforced by the topology: since there are no incoming edges in source nodes, it is not necessary to add a constraint disallowing incoming traffic for flows originated at those nodes (similarly for destination nodes).

Our objective is to minimize the global delay while respecting the bandwidth limits, assuming that the network has enough capacity to satisfy all demands. In addition to the transform data obtained by the above described algorithm, let us define a set of positive flow weights $b_{s,d}$ for all $(s,d) \in F'$. Two different kinds of optimization may be obtained by using different weight values. The first alternative uses $b_{s,d} = 1, \forall (s,d) \in F'$, stating that all flows have equal importance; in this case, optimization is performed on a per-route basis. The second alternative uses $b_{s,d} \propto a'_{s,d}, \forall (s,d) \in F'$, stating that a flow's importance is proportional to its traffic demand; in this case, optimization is performed on a traffic volume basis.

Let us define the boolean decision variables $x_{ij}^{sd}$ which take the value 1 if the flow $(s,d) \in F'$ is routed through the edge $(i,j) \in E'$ and the value 0 otherwise.

The problem can, thus, be formulated as follows:

$$\textit{Minimize} \sum_{(i,j)\in E'} \sum_{(s,d)\in F'} b_{s,d} w'_{i,j}\, x_{i,j}^{s,d} \text{ subject to}$$

$$x_{i,j}^{s,d} \in \{0,1\}, \forall (s,d) \in F', (i,j) \in E' \tag{8.1}$$

$$\sum_{(s,d)\in F'} a'_{s,d}\, x_{i,j}^{s,d} \le c'_{i,j}, \forall (i,j) \in E' \tag{8.2}$$

$$\sum_{(j,k)\in E'} x_{j,k}^{s,d} - \sum_{(i,j)\in E'} x_{i,j}^{s,d} = 0, \forall (s,d) \in F', j \in \left(V' - \{s,d\}\right) \tag{8.3}$$

$$\sum_{(s,j)\in E'} x_{s,j}^{s,d} = 1, \forall (s,d) \in F' \tag{8.4}$$

$$\sum_{(i,d)\in E'} x_{i,d}^{s,d} = 1, \forall (s,d) \in F' \tag{8.5}$$

$$\sum_{j\in V':(i,j)\in E'} \sum_{\substack{t\in S\\ t\neq d}} x_{i,j}^{t,d} \le |S| \cdot x_{s,i}^{s,d}, \forall (s,d) \in F', i \in S : (s,i) \in E' \tag{8.6}$$

$$\sum_{(i,j)\in E':(j,d)\in E'} \sum_{\substack{s\in S\\ s\neq d}} x_{i,j}^{s,d} = \sum_{j\in V':(j,d)\in E'} \sum_{\substack{s\in S\\ s\neq d}} x_{j,d}^{s,d}, \forall d \in D \tag{8.7}$$

Constraint set (8.1) imposes boolean decision variables, meaning that flows cannot be split over multiple paths.

Constraint set (8.2) states that the sum of all flows traversing an edge will not exceed its capacity.

Constraint sets (8.3), (8.4) and (8.5) are the "mass balance" equations: (8.3) means that each flow entering a node that is neither source nor destination for that flow must leave it and vice-versa; (8.4) means that each flow leaves the source node once and, similarly, (8.5) means that each flow enters the destination node once.

Constraint set (8.6) means that if a flow from a source to a destination traverses a given virtual node directly connected to that source, no other flows to the same destination may traverse a different virtual node connected to the same source. On the original graph it means that if the flow from a given node to a certain destination leaves that node by a given link, no flow to the same destination traversing that node may leave it by a different link — in other words, it imposes hop-by-hop routing.

Finally, constraint set (8.7) prevents routing loops at the destination nodes of flows in the original graph by forcing flows arriving at a node directly connected to their destination virtual node to use that direct path. Failing this, a flow would be counted twice (or more) on the left hand side and only once on the right hand side, invalidating the equality.

*Theorem 1:* Paths obtained through this optimization procedure are guaranteed to be cycle-free on the transform graph.

*Proof:* Satisfaction of constraints (8.4) and (8.5) implies that each flow leaves the source virtual node exactly once (8.4) and arrives at the destination virtual node exactly once (8.5), therefore the source and destination virtual nodes belong to the path.

There are no incident edges to source virtual nodes, therefore these nodes cannot be in a cycle. Conversely, there are no incident edges from destination virtual nodes, therefore these nodes cannot be part of a cycle either.

Now let $p$ be a path from a given source $s \in S$ to a given destination $d \in D$ containing a cycle and $p^*$ the same path with the cycle removed. The cycle may only include intermediate nodes, since source and destination nodes cannot be part of cycles. If the above constraints (notably the capacity constraints) are satisfied with path $p$ from $s$ to $d$, then they are also satisfied with path $p^*$ from $s$ to $d$. Since $b_{s,d} > 0, \forall (s,d) \in F'$ and $w'_{i,j} > 0, \forall (i,j) \in E' : i \notin S \wedge j \notin D$, the cost of using $p^*$ would be lower than the cost of using $p$, therefore $p$ could not be in the optimal route set, as a route set including it would not minimize the cost function. ■

*Theorem 2:* Paths obtained through this optimization procedure are also guaranteed to be cycle-free on the original graph.

*Proof:* The proof is based on three lemmas, regarding cycles without the destination node, cycles with the destination node and the preceding edge, and cycles with the destination node without the preceding edge.

*Lemma 1:* A path satisfying the above conditions cannot contain cycles that do not include the destination node on the original graph.

*Proof:* Let $p_1$ be a path on the original graph from source $s$ to destination $d$ containing a cycle that does not include node $d$, and $p'_1$ the equivalent path in the transform graph. Since the cycle on $p_1$ does not contain the destination $d$, it must contain a node $i$ left by the flow twice by different edges, $(i, j)$ towards the cycle and $(i,k)$ towards the destination node. Therefore, in the transform graph, $p'_1$ must contain virtual nodes $ij$ and $ik$. Constraint (8.6) implies that $x_{i,j}^{i,d} = 1$ and $x_{i,k}^{i,d} = 1$. However, this violates constraint (8.4); therefore $p_1$ can only contain cycles that include the destination node $d$. ■

*Lemma 2:* A path satisfying the above conditions cannot contain cycles that include the destination node and the preceding edge on the original graph.

*Proof:* Let $p_2$ be a path on the original graph from source $s$ to destination $d$ containing a cycle that contains node $d$ and the edge incident to $d$, $(i,d)$; let $p'_2$ be the equivalent path in the transform graph. In this case, the flow enters $d$ twice by the edge $(i,d)$, from the source and from the cycle. Therefore, in the transform graph, $p'_2$ encloses a cycle containing the virtual node $id$. This contradicts theorem 1, which states that $p'_2$ is guaranteed to be cycle-free on the transform graph, meaning that a cycle containing node $d$ and the edge incident to $d$ cannot exist in the returned route set. ■

*Lemma 3:* A path satisfying the above conditions cannot contain cycles that include the destination node but not the preceding edge on the original graph.

*Proof:* Let $p_3$ be a path on the original graph from source $s$ to destination $d$ containing a cycle that includes node $d$ but not the edge incident to $d$; let $p'_3$ be the equivalent path in the transform graph. In this case, the flow enters node $d$ twice by different edges, $(i,d)$ and $(j,d)$, from the source and from the cycle. Therefore, in the transform graph, $p'_3$ contains virtual nodes $id$ and $jd$, contributing two units to the left hand side of constraint (8.7). According to constraint (8.5), the destination virtual node can only be entered once, therefore this flow's contribution to the right hand side of constraint (8.7) can only be one unit. Since the same applies to all flows, no flow on the original graph can have a cycle containing the destination node $d$ but not the edge incident to $d$. ■

The preceding lemmas cover all possible cycles on the original graph, therefore all paths satisfying the above conditions are guaranteed to be cycle-free on the original graph. ■

#### 8.1.4.1 Reducing the Number of Variables

The transform graphs have particular characteristics that allow us to significantly reduce the number of decision variables. Notice that the source virtual nodes have only out-edges (dashed edges in fig. 8.3.b) and the destination virtual nodes have only in-edges (dotted edges in fig. 8.3.b). As such, the following conditions hold:

$$x_{i,j}^{s,d} = 0, \forall (s,d) \in F', (i,j) \in E' : i \in (S - \{s\}) \tag{8.8}$$

$$x_{i,j}^{s,d} = 0, \forall (s,d) \in F', (i,j) \in E' : j \in (D - \{d\}) \tag{8.9}$$

Therefore, we may restrict the domain of $(i,j)$ in variables $x_{i,j}^{s,d}$ to $L' \cup \{(i,j) \in E' : i = s \vee j = d\}$.

### 8.1.5 Variant Formulation

The problem formulation above assumed that a certain amount of capacity is reserved at each inter-domain link for traffic generated at the transmitting AS and destined to or traversing the receiving AS (next hop), therefore not corresponding to any virtual trunk; it also assumed that a given amount of capacity is reserved for traffic destined to the AS itself at the ingress policing. In terms of constraints, they are represented by the capacities of the edges incident from the virtual source nodes (dashed) and by the capacities of the edges incident to virtual destination nodes (dotted), respectively.

A perhaps more reasonable assumption is that traffic outside the virtual trunks may use all the capacity left unused by traffic inside the virtual trunks (including reserved but unused virtual trunk capacity). Adapting the problem formulation to this new assumption involves the following changes:

1. Restricting the domain of $(i,j)$ in constraint (8.2) to $L'$, the set of virtual trunk edges, instead of $E'$, the set of all edges, therefore removing the fixed capacity constraints outside the virtual trunks.
2. Introducing an additional constraint (8.10) at each virtual node $i$ corresponding to an inter-domain link in the original graph, where $c_i$ is the capacity of the corresponding inter-domain link in the original graph. This constraint set states that the sum of all flows traversing (leaving) the inter-domain link must be less than the capacity of that link. Usually, $c_i = \sum_{i \in V' : (i,j) \in E'} c'_{i,j}$ .

$$\sum_{j \in V':(i,j)\in E'} \sum_{(s,d)\in F'} a'_{s,d}\, x_{i,j}^{s,d} \le c_i, \forall i \in (V'-S-D) \qquad (8.10)$$

## 8.2 Proposed Protocol and Associated Algorithms

While the ILP formulation of the inter-domain QoS routing problem presented in the previous section is useful as a baseline for comparison with real protocols in controlled environments where all the input data is known, it cannot be used in the implementation of a real protocol itself for several reasons: first, the problem of 0-1 integer programming is known to be NP-complete [Karp72]; second, because it requires knowledge of the traffic matrix, which is not easy to obtain in real utilization scenarios; and third, because it requires knowledge of the virtual trunk SLAs, which are usually disclosed only to the involved peers. In this section we propose a practical virtual-trunk-aware inter-domain QoS routing protocol, based on an extension of BGP, for deployment in real internetworks.

### 8.2.1 QoS Routing

As mentioned in section 2.4, inter-domain routing in the Internet is performed using BGP. The most common policy for path selection in BGP is the minimum number of “AS hops” in the *AS_PATH*. Even though the standard BGP does not provide any support for QoS-based routing (the *AS_PATH* length metric bears only a very loose relation to QoS parameters), it can easily be extended to convey virtually any kind of relevant QoS information through the use of optional path attributes. The decision processes may also be changed to use the QoS information (if present) for path selection without breaking backward compatibility. We extended BGP to transport and use three QoS metrics: assigned bandwidth (static), path delay under light load (static) and a dynamic metric for path congestion described below.

#### *8.2.1.1 Metrics*

Virtual trunk information is explicitly included using BGP to carry information on the amount of bandwidth contracted between two domains regarding data transport to a third one. The assigned bandwidth, reflecting traffic contracts, is essentially static. It is updated along the path to be the minimum, that is, the bottleneck bandwidth (concave metric). Notice that our model does not require explicit and quantified agreements, only that transport operators assign a certain capacity for data transport between their connected peers; explicit SLAs are just a means to guarantee that reasonable assignments are performed.

Information on the expected delay in light load conditions (a lower bound for the expected packet delay) is also carried. Minimization of this metric by the path selection mechanism allows not only for better packet QoS (smaller delays), but also for a more rational use of the network resources. This is because in high capacity links with significant length, such as those found in today's inter-domain connections, path delay consists mostly on the sum of propagation delays [Papagiannaki03], directly proportional to the traversed span of fiber, as long as there is no congestion. The light load delay metric is static, and is summed along the path (additive metric).

The third QoS metric conveyed by our proposed extension is path congestion. The concept of congestion is deliberately vague and may, therefore, be translated into a coarse objective metric, minimizing the overhead in message exchange and path re-computation typical of dynamic metrics. The congestion alarm is expressed by an integer with three possible values, whose meaning is the following: 0 — not congested; 1 — very lightly congested; 2 — congested. This metric is updated along the path to the maximum value (convex metric). In the most basic version, congestion may be inferred from the utilization of the aggregates; a more advanced version would use additional parameters, such as packet loss, average length of traversed router queues or measured delay, as inputs for computing the alarm level of virtual trunk aggregates. The main requirement for the congestion alarms, the sole dynamic metric in our proposal, is that changes should be infrequent, for scalability and stability reasons; hysteresis and related techniques may be applied in the assignment of alarm levels to this end.

An effective value of the congestion alarm is used for path selection instead of the received value, with the objective of reducing the fluctuations in the usage of the aggregates. This effective value is the same as the received value, unless the received value is 1 and the route is already in use, in which case the effective alarm is 0. In practice, this means that when level 1 (light congestion) is reached, the route should not be used to replace a previously established one, but if it were already in use, no switch to a different route should be performed unless a higher congestion level is reached. This behavior is meant to avoid the synchronized route flapping problem.

#### 8.2.1.2 Path selection algorithm

The three above mentioned QoS metrics are conveyed in the *UPDATE* messages by a newly defined Path Attribute, *QoS_INFO*, which is optional and transitive (meaning that ASs which do not support the extension simply forward the received value), and are updated by the BGP-speaking routers at each transit domain, taking into account the virtual trunks between the domain to which the route is advertised and the "next hop domain" for the route. Notice that these virtual trunks are shared among different source to destination routes: in fig. 8.4, for example, all traffic transported from T1 to D1 via T3 shares the T1:D1 virtual trunk, independently of being originated at S1 or S2.

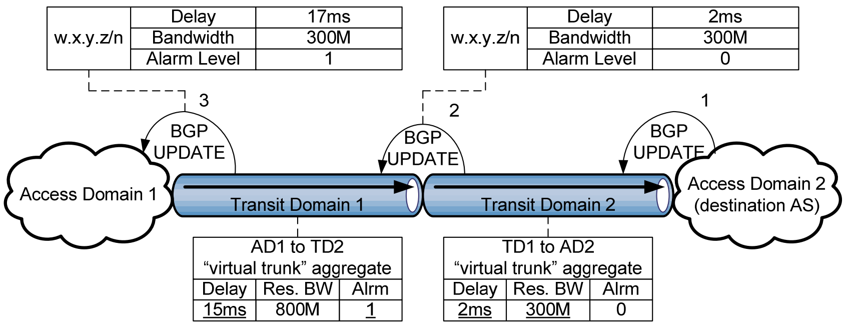


**Figure 8.4: Propagation of metrics in the QoS_INFO attribute**

Figure 8.4 illustrates the propagation of the delay, bandwidth and congestion alarm metrics in the *QoS_INFO* attribute of *UPDATE* messages. When the destination AS (AD2) first announces the route to an internal network, it may omit the *QoS_INFO* attribute if this network is directly connected to the announced *NEXT_HOP*. On receiving the *UPDATE*, the edge router at transit domain TD2 creates (or updates, if already present) the *QoS_INFO* attribute with metrics of the outgoing link, for route selection purposes (this step is omitted in the figure). If the route is selected, it is propagated to all peering domains; the *QoS_INFO* attribute sent to the different upstream domains is different, since the metrics are updated with respect to the virtual trunk aggregates. The same process is repeated at transit domain TD1. Notice that the delay metric in the *UPDATE* sent from TD1 to AD1 (17 ms) is the sum of the delays of the concatenated virtual trunks (2 ms and 15 ms), the reserved bandwidth is the minimum along the path (300 Mbps), and the congestion alarm is the maximum (1). The virtual trunk values that contribute to the final values received by AD1 for this route are underlined in the figure.

Figure 8.5 shows the route comparison algorithm used in the decision processes in pseudo-code. Delay information is used to select the fastest/shortest route. The benefits of doing this, as previously mentioned, are twofold: packets will suffer lower delays and network resource usage will be more rational. The information on the reserved bandwidth is used to eliminate, from the set of possible choices, routes with insufficient bandwidth to support the current outgoing traffic aggregate from the local AS to the destination (measured by monitoring at the edge routers, including flows generated at the local AS and flows

```
set Traffic to dest = Local traffic to dest + Transit traffic to dest
for both routes
   if $Alarm_{rcv}$ = 1 and route in use, set $Alarm_{eff}$ = 0
   else set $Alarm_{eff}$ = $Alarm_{rcv}$
if both routes have Assigned BW < traffic to dest,
   choose the one with larger Assigned BW
else if one route has Assigned BW < traffic to dest,
   choose the other one
else if $Alarm_{eff}$ is different,
   choose the route with lower $Alarm_{eff}$
else if Delay is different,
   choose the route with least Delay
else if Assigned BW is different,
   choose the route with larger Assigned BW
else use normal BGP rules (AS_PATH length, etc.)
```

**Figure 8.5: Route comparison/selection function**

traversing it); it is also used as tie breaker when two routes for the same destination have the same announced delay. The alarm levels are used to eliminate congested routes from the set of possible choices. Elimination of routes with insufficient capacity from the set of possible choices prevents, to a certain degree, congestion of those routes, contributing to lower message and processing overheads and to increased route stability.

#### *8.2.1.3 Route Aggregation*

A very important aspect in inter-domain routing is the possibility of aggregating routes. Without the deployment of route aggregation and Classless Inter-Domain Routing (CIDR) [RFC1519] in the 1990s, routers would not have been able to support the increasing number of advertised routes. Paradoxically, little attention is given to aggregation in inter-domain QoS routing proposals, in general.

The use of a metric as coarse-grained as the congestion alarm in this proposal is aggregation-friendly. While the introduction of new metrics reduces the possibilities of aggregation compared to the standard, non-QoS-aware BGP, congestion alarm values will almost always be either 0 or 1, meaning that much aggregation is still possible. This is particularly true if congestion is introduced in transit domains, since it is common to all routes sharing the congested virtual trunk. The light load expected delay metric may easily be made compatible with aggregation if assumed to be an indicator rather than an exact value — in this case, two routes may be aggregated if the smaller delay is more than a certain fraction (say 75%) of the larger one, announcing the larger value for the aggregated route. The hierarchical structure of the Internet allows for a large degree of aggregation with this approach. The bandwidth metric, however, is more difficult to deal with, even with a hierarchical structure.

Perhaps a solution to be deployed in the Internet at large would require the use of a very coarsely grained bandwidth metric, or even entirely giving up the use of this metric. However, a meaningful assessment of the tradeoff between the bandwidth metric and route aggregability would require a much larger scale evaluation and access to information available only to data transport operators, thus exceeding the scope of this thesis.

## 8.3 Simulation Results

In this section we present simulation results obtained in ns-2 [NS2] of the QoS_INFO proposal for inter-domain QoS routing, implemented as an extension of an existing BGP module [Feng04]. These results concern the performance, in terms of delay, loss probability and inter-domain links congestion, of QoS_INFO when compared to standard BGP, to BGP with the QoS_NLRI extension conveying static one-way delay information (the expected delay of the route in light load conditions), and to optimal solutions obtained using the ILP formulations of section 8.1 (both the original, *opt-r*, and the variant, *opt-nr*) in a MIP code (Xpress-MP from Dash Optimization [DashOpt]). They also concern the number of updates required to provide inter-domain QoS and the stability of the routes. Note that the QoS_NLRI extension can be used to convey QoS parameters other than delay, and that the extension does not specify whether the delay information is static or dynamic. In fact, [Cristallo04] is focused on the BGP extension for the transport of QoS information, not specifying the way that information is to be used by BGP in the path selection process. Therefore, in this comparison we used the scenario therein illustrated.

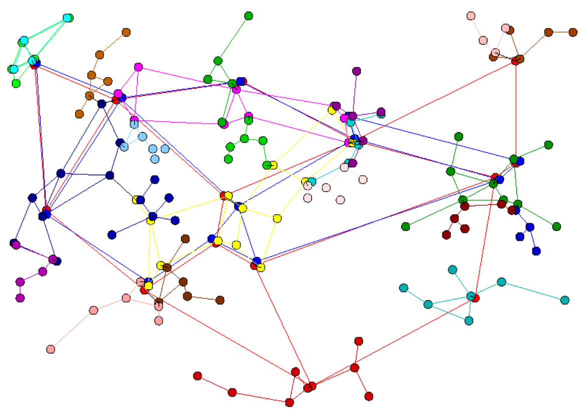
The amount of traffic in inter-domain scenarios is extremely high, making it very difficult to complete simulations with realistic parameters within a reasonable time span. For this reason, in our implementation we have chosen to simulate the signaling protocol normally at the packet level, but not the data traffic, which was mathematically simulated using the well-known M/G/1 queuing model with three different packet sizes: 50% of packets with 40 bytes (representing 4% of the traffic volume), simulating SYN, ACK, FIN and RST TCP segments; 20% of packets with 80 bytes, simulating packetized voice (3% of traffic volume); and 30% of packets with 1500 bytes, simulating full size TCP segments (93% of traffic volume). These packet sizes reflect the bimodality currently observed in internet traffic [Sinha05], complemented with voice packets, whose frequency tends to increase. Queuing delays were obtained using the Pollaczek-Khintchine formula [Bertsekas92], $W_Q = \frac{\lambda E[S^2]}{2(1-\lambda E[S])}$ where $W_Q$ is the queuing delay, $\lambda$ is the traffic arrival rate and $S$ is the service time; the computation of total packet delays was based on the Kleinrock independence approximation [Bertsekas92].

### 8.3.1 Simulated Scenario

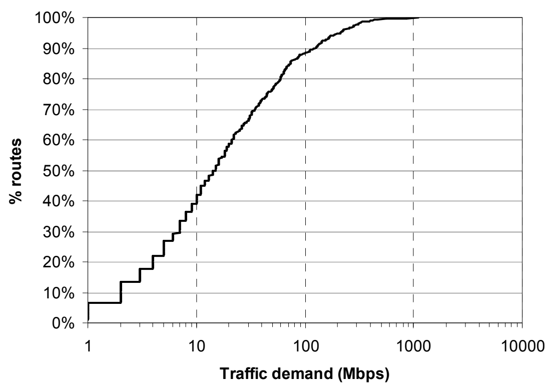
In this subsection we describe the scenario used in the simulations. To have meaningful results, a realistic topology and traffic matrix is required. We have used a hierarchical topology (fig. 8.6.a) containing two large transport providers with broad geographical coverage, four regional providers and 19 local providers. Abstracted at the AS level, the topology has 25 nodes (ASs) and 36 inter-domain links. The traffic demand for each route (source-destination pair) is constant during the simulation. The distribution of traffic demand values for the different routes is summarized in fig. 8.6.b, having a maximum of 1.1 Gbps, an average of 45 Mbps and a standard deviation of 90 Mbps. The link bandwidth was assigned based on expected demands. The configuration of the virtual trunk type SLSs in our proposed model was performed automatically, based on the link bandwidth, the traffic matrix and a set of feasible routes (proportional distribution of link bandwidth). Not all triplets (*a*,*b*,*c*) such that *a* is connected to *b* and *b* to *c* have a corresponding SLS — whenever this is the case, traffic between *a* and *c* should use intermediate nodes other than *b*. Traffic that does not match an established SLS or that exceeds its assigned capacity is discarded at the ingress routers of the ASs.

Thresholds for setting alarm levels on path usage were 35% of the SLS bandwidth for level 1 and 80% for level 2, except where otherwise stated.

We ran simulations for 8200 simulated seconds, discarding data for the first 1000 in order to filter out transient effects. We evaluated link usage, route optimality, route stability, QoS parameters and signaling overhead.



a) Topology



b) Cumulative distribution function of traffic demands

**Figure 8.6: Topology and traffic distribution**

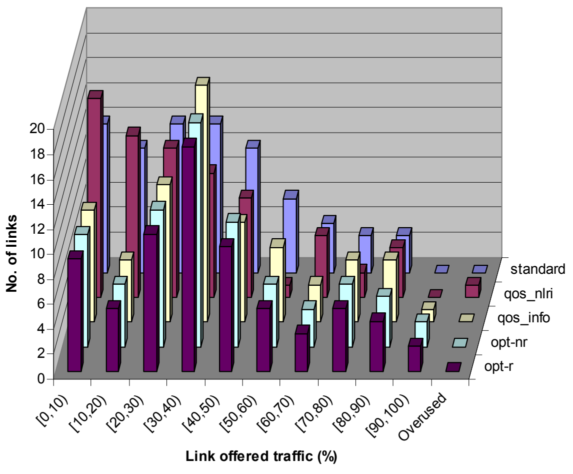
## 8.3.2 Link Usage, Route Optimality and QoS Parameters

In the first experiment we compare the three inter-domain routing mechanisms: standard BGP, BGP with QoS_NLRI and our proposed QoS_INFO with respect to link usage, route optimality and QoS parameters.

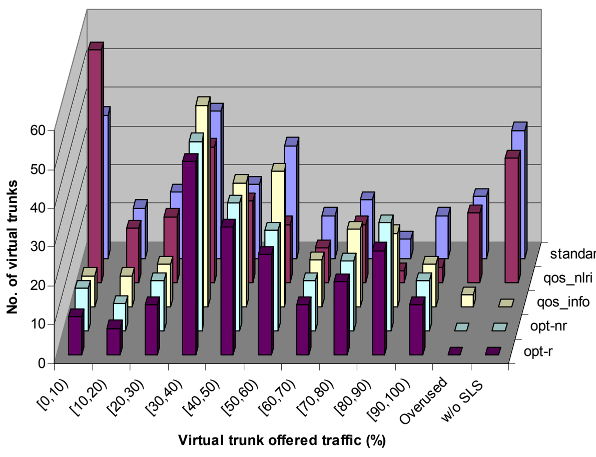
Figure 8.7 shows histograms with the distribution of the offered traffic for the links and the virtual trunks in the three approaches, averaged out of the 7200 useful simulation seconds. The same results are also provided for the optimized route sets. The *overused* class corresponds to links/virtual trunks having an offered load above their capacity, and the *w/o SLS* class in fig. 8.7.b to AS triplets (*a*,*b*,*c*) with traffic but without an established SLS; in both cases, a significant portion of packets is consistently discarded due to link capacity limitation or SLS policing (not only a very small portion due to sporadic queue overload).

With standard BGP, routes are normally chosen based on the lowest number of elements in the *AS_PATH*, not taking into consideration path delay or congestion. As a result, 22% of the routes were sub-optimal in terms of expected light load delay. Regarding utilization, 16 out of the 211 virtual trunks (7.6%) were overused and, even worse, there was traffic on 33 triplets without established SLSs (15.6% compared to the number of SLSs). As a consequence, packet losses were 17.1% of the total traffic demand.

With the QoS_NLRI BGP extension carrying light load path delay information (static), all routes are optimal in terms of expected light load delay. Congestion, however, is even worse than with standard BGP: 18 of the virtual trunks (8.5%) are congested, and there is traffic on 32 triplets without established SLSs (15.2% compared to the number of SLSs). Additionally, 1 inter-domain link was overused (1.4%). Notice that since the sum of the SLSs



a) Per link



b) Per virtual trunk

**Figure 8.7: Offered traffic distribution**

containing a link is always less than the link's capacity, link overuse is caused only by first hop traffic, that is, traffic originated at the AS transmitting through the inter-domain link. As a result of these factors, the overall packet loss figure was 28.2%. The fact that congestion is worse in QoS_NLRI than in standard BGP is probably related to the fact that, by minimizing the number of AS hops, standard BGP tends to exploit the hierarchical character of the network by preferring a more logical path comprising a small number of transport operators with broad geographical coverage to a path consisting on a large number of operators with small coverage[2] that may, nevertheless, have a lower light load delay value.

With our proposed QoS_INFO approach, there was no traffic on AS triplets without a corresponding SLS, and only 3 SLSs were overused (1.4%). The overall packet loss, of only 0.4%, was much lower than in both of the previous cases. The reason for this is that the system reacts to congestion by changing the affected routes. Obviously, both optimized results had no overused virtual trunks or inter-domain links, and neither did they have traffic on AS triplets without corresponding SLSs.

Figure 8.8 shows the packet loss probability Cumulative Distribution Function (CDF) for the routes at the end of the simulation[3] in the different scenarios. Again, our proposed QoS_INFO approach yields better results, with 96.5% of the routes having a negligible packet loss probability, contrasting to only 58.8% in QoS_NLRI and 68.0% in the standard BGP. In both optimized cases, 100% of the routes had no packet losses.

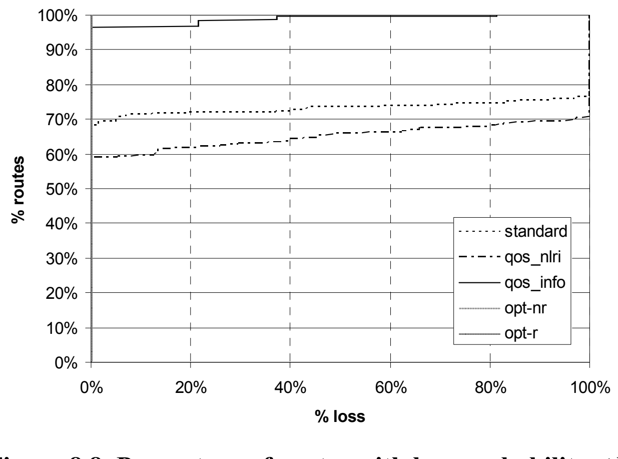


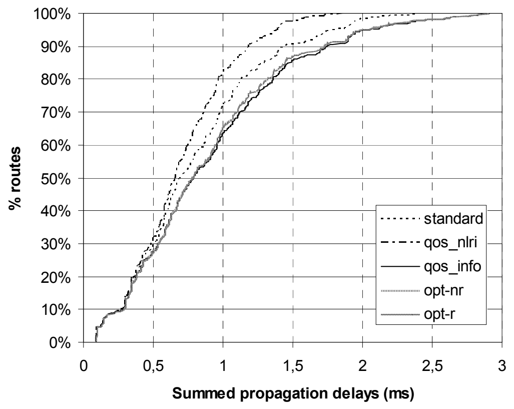
**Figure 8.8: Percentage of routes with loss probability ≤ X**

Figure 8.9.a shows CDFs of the summed propagation delays for the routes in the three scenarios (in the cases of QoS_NLRI and QoS_INFO, they correspond to the announced

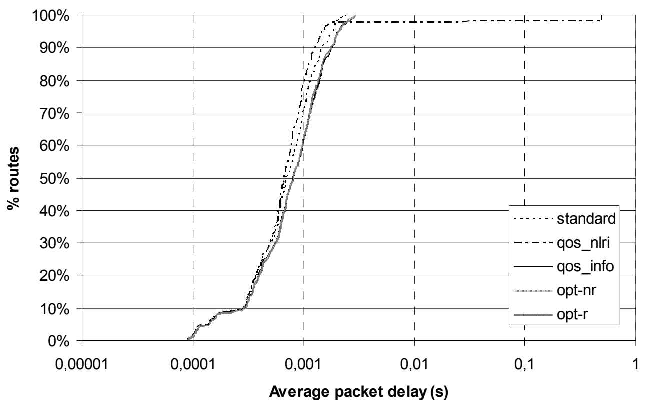
[2] In non-hierarchical topologies standard BGP performed worse than QoS_NLRI with respect to congestion.

[3] Since routing with QoS_INFO is based on dynamic information, routes do change in the course of the simulations; in standard BGP and BGP with QoS_NLRI all routes are stable during the useful simulation period.

delay values). As expected, QoS_NLRI performs better in this respect, even better than the optimizations, since the routes with the lower delay metric are always chosen, ignoring virtual trunk capacity and congestion. Interestingly, standard BGP also does better than the optimal solutions in this respect, since it also ignores capacities and congestion. The QoS_INFO curve follows the optimal curves very closely.



a) Percentage of routes with light load delay ≤ X



b) Percentage of routes with average packet delay ≤ X

**Figure 8.9: Percentage of routes with delay ≤ X**

It is worth noting that the light load expected delay holds little significance if routes are congested (heavily loaded); therefore, a much more meaningful parameter is the expected packet delay for the routes (sum of propagation and transmission delays with the expected queuing delays along the path), plotted in fig. 8.9.b. Since policing is performed on the virtual trunks and their assigned capacity is consistent with the capacity of the inter-domain links they traverse, there was no link congestion in most of the cases, therefore route delays were kept low. Nevertheless, 2.2% of the routes in QoS_NLRI traversed a congested link and suffered large delays. Except for these routes, packet delays are close in all cases, with the QoS_INFO curve practically overlapping those of the optimizations. Table 8.2 shows a summary of the above discussed results.

| | Overloaded | | AS triplets with traffic and no SLS (% of SLS) | Packet losses | Routes with losses | Routes traversing congested | | Routes traversing AS triplets with no corresponding SLS |
|---|---|---|---|---|---|---|---|---|
| | i.d. links | v. trunks | | | | i.d. links | v. trunks | |
| Standard | 0.0% | 7.6% | 15.6% | 17.1% | 32.0% | 0.0% | 14.3% | 24.2% |
| QoS_NLRI | 1.4% | 8.5% | 15.2% | 28.2% | 41.2% | 2.2% | 21.3% | 30.2% |
| QoS_INFO | 0.0% | 1.4% | 0.0% | 0.4% | 3.5% | 0.0% | 3.5% | 0.3%[†] |
| Optimal NR | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% |
| Optimal R | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% | 0.0% |

[†] Corresponding to routes without traffic, therefore not relevant. It would be trivial to modify BGP with QoS_INFO in order not to propagate routes traversing triplets without corresponding SLSs.

**Table 8.2: Summary of results**

## 8.3.3 Signaling Overhead and Route Stability

The drawback of the QoS_INFO approach, as usual with dynamic QoS routing approaches, is increased signaling load and decreased route stability. In the performed simulations we measured an average of 6.16 updates per second for the whole topology, or 0.246 per node. These updates, however, do not affect all ASs equally, since some routes are very stable, while others oscillate. The distribution of the frequency of sent and received updates is shown in fig. 8.10.

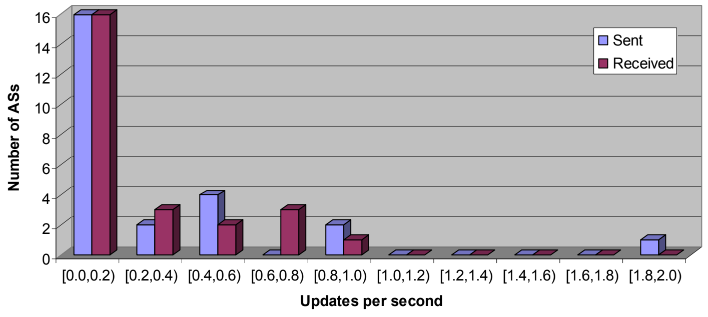


**Figure 8.10 - Distribution of the number of updates per second in ASs**

With the other models all routes are stable as long as there are no topology changes (due, e.g., to link failures). It is worth noting that if the delay information conveyed in the QoS_NLRI extension were dynamic, based on measurements, then route oscillations would also occur in that model; on the other hand, link overloads would be reduced. Regarding route stability, with the QoS_INFO approach, 572 out of a total of 600 routes in the topology (ca. 95%) were stable, meaning that they did not change during the useful simulation period; the other 5% did change, though with varying frequency. For example, 16 ASs sent less than 0.2 updates per second, whereas 2 ASs sent between 0.8 and 1.0 updates per second.

Since the choice of a new route is triggered by changes in the alarm levels, the SLS utilization thresholds used to assign a given alarm level have strong influence in route stability. In order to evaluate this influence, we evaluated route stability in simulations using alarm level 1 utilization threshold values ranging from 20% to 65% of the bandwidth assigned to the SLSs (*x* axis), and alarm level 2 threshold values ranging from 70% to 90% of that bandwidth (different curves). The results of this experiment are shown in fig. 8.11.a. We may see that relatively low values of alarm level 1 threshold (*th1*) tend to improve the route stability, especially for lower values of alarm level 1 threshold (*th2*). As *th1* gets close to *th2*, route stability decreases. Higher values of *th2* also tend to improve route stability: the highest value achieved was 98.8% for *th1*=55% and *th2*=90%. However, even though such values did not lead to increased packet losses (0.3%) or to the use of non-established virtual trunks, and only increased the number of overused virtual trunks to 4 in 211 (1.9%), they would have to be lowered in a practical deployment, since traffic demand is much more variable.

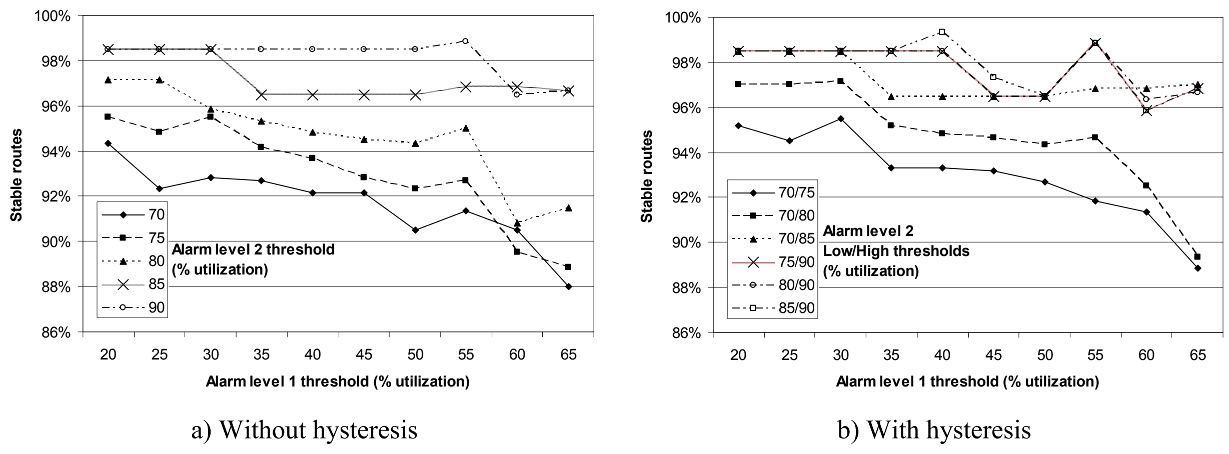


**Figure 8.11 - Route stability vs. alarm level thresholds**

Route stability is related to the frequency of update messages, since they are triggers for the BGP decision processes. Some reduction in the number of updates may be achieved by the introduction of hysteresis in the assignment of congestion alarm levels. We have introduced hysteresis by using two different values for each threshold — a change from an alarm level to a higher one occurs only when the high value is crossed, but in order to return to the lower alarm level, the utilization must drop below the lower level. Figure 8.11.b shows the stability of routes with different lower and higher levels for *th2*[4]. Compared to fig. 8.11.a, the results are generally slightly better, with a higher percentage of stable routes.

Figure 8.12 shows the average number of updates per second per node without and with hysteresis. Similarly to route stability, results of the number of updates with hysteresis are slightly better than without it, with a generally lower number of updates and somewhat

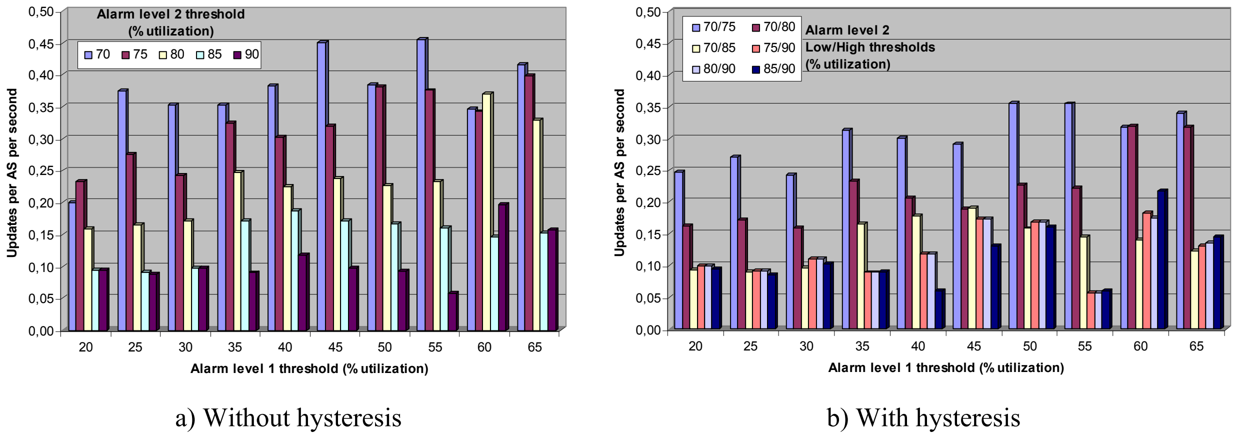


**Figure 8.12 - Frequency of BGP updates**

[4] The introduction of hysteresis in *th1* has much lower relevance, as selected routes with *th1*=1 are treated as if *th1*=0.

less sensitive to the threshold values. Even though a more conclusive comparison would require simulations using many different topologies, it is worth noting that the interest of hysteresis in a practical deployment with dynamic traffic demands is higher than in these simulations with a static traffic demand matrix, as the average thresholds would have to be lower due to constant variations in traffic demand.

## 8.4 Conclusions

This chapter addressed the problem of inter-domain QoS routing, part of the overall proposed solution to the problem of end-to-end QoS. Our approach is based on the use of virtual trunk type aggregates for the indirect transport of traffic between two different administrative domains across a third one. These virtual trunks are usually, though not necessarily, defined by means of Service Level Agreements between the operators of peering domains, and each inter-domain path consists on a concatenation of virtual trunks. Each virtual trunk, defined by a triplet of Autonomous Systems, is shared by all active routes containing that sequence of ASs in the *AS_PATH*. We formally stated the problem of SLA-aware inter-domain QoS routing and formulated it as an Integer Linear Programming optimization problem, providing a proof that routes obtained through the optimization process cannot contain cycles.

As a practical solution, we proposed the QoS_INFO extension to the Border Gateway Protocol, using of a combination of three different metrics (assigned bandwidth, expected light load delay and congestion alarm) in order to simultaneously achieve different and conflicting goals: finding non-congested paths that satisfy the QoS requirements of the data flows, minimizing the network resources used to transport the flows, and minimizing the message exchange, path computation overheads and route instability. Although one of the metrics, the congestion alarm, is dynamic, its coarse granularity and the rules for its use in the path selection algorithm are such that the impact overhead in message exchange and path re-computation is minimized, and route stability preserved to a large degree.

Simulations were performed to evaluate our proposal and compare it to standard, QoS-unaware BGP, to BGP with the QoS_NLRI extension, and to the optimal case. The results show that, while QoS_NLRI represents an improvement over standard BGP in choosing routes with the lowest light load delay, routing using only static QoS parameters is also unable to avoid path congestion, leading to considerable packet losses. With our QoS_INFO proposal, congested paths and their consequences on QoS are avoided. Even though there is a penalty in overhead and route stability in doing this, most of the routes are stable, especially if the thresholds for alarm setting are appropriately selected. The introduction of hysteresis in the alarm level assignment seems to improve stability and overhead, and is expected to have an even higher relevance in a real deployment of the protocol, with dynamic traffic demands.

# CHAPTER 9

# CONCLUSIONS

This thesis dealt with the problem of scalable QoS provisioning in packet switching networks, and involved work on many different subjects (low level architectural issues, algorithms, signaling, etc.). This chapter concludes the dissertation with a summary of the work done and our main results, and with an identification of open problems that could be used as a starting point for further research in these areas.

## 9.1 Thesis Summary

The work presented in the previous chapters had two main parts, corresponding to two different lines of our research. The first line concerned distributed models for providing end-to-end QoS in the Internet, where the routers along the path of the flows autonomously perform both data plane and control plane functions. The second line of research concerned the QoS subsystem of a next-generation IP-based mobile telecommunications network architecture, more specifically the one proposed in the DAIDALOS Integrating Project, in the scope of which the second part of our work was performed.

In the first part, we started by performing an evaluation of the RSVP Reservation Aggregation model for scalable QoS provisioning in high-speed backbone networks. We gave an overview of the model and of our implementation in the ns-2 simulator, which required the definition of a policy for aggregate bandwidth management, not provided by the standard. Simulation results have shown that the RSVP Aggregation model can provide RSVP/IntServ Controlled Load service semantics but, contrary to the latter, in a scalable fashion. Packet classification and scheduling are performed per class, based on the DiffServ framework, and the signaling load is greatly reduced — even with a small test topology, the signaling load was reduced to less than 1/10th that of RSVP/IntServ. The simulation results have also shown the existence of a tradeoff between signaling load reduction and lower network utilization: holding more unused capacity in an aggregate increases the probability that new flows can be accepted into that aggregate without signaling the core nodes, but also increases the probability that bandwidth increases in other aggregates, necessary to accept new flows in those aggregates, are rejected, thus reducing network utilization. The operating point in this tradeoff is defined by setting two tunable parameters of the bandwidth management policy — the *bulk* size and the hysteresis time.

With the goals of providing scalable end-to-end QoS with higher resource utilization than with RSVP Aggregation and an additional service class with stricter guarantees than Controlled Load, we proposed the Scalable Reservation-Based QoS (SRBQ) architecture, a model where scalability is achieved by using exclusively low computational complexity algorithms whose execution time is independent on the number of simultaneous flows. SRBQ provides two service classes: the Guaranteed Service class, with strict QoS guarantees, and a Controlled Load class, providing only soft guarantees but more tolerant to burstiness. It combines the use of DiffServ style per class aggregation on the data plane (packet classification, scheduling, policing and shaping) with a scalable signaling model for per-flow resource reservation. Some techniques were developed for making per-flow resource reservation signaling scale to a very large number of simultaneous flows, namely a label mechanism that provides routers with direct access to resource reservation information of the flows, and an efficient implementation of soft state expiration timers. Though developed for SRBQ, these techniques could be used with other types of signaling.

Using our implementation of SRBQ in ns-2, we analyzed the QoS metrics and per-class network resource utilization in different experiments, using both synthetic and real-world multimedia streams. The results have shown that SRBQ can support both strict and soft QoS guarantees in the GS and CL classes, respectively, and provides adequate isolation of in-profile flows from out-of-profile ones; out-of-profile CL flows are more penalized in terms of loss probability, while out-of-profile GS flows are essentially penalized in terms of delay, due to ingress shaping. Based on average rates, the CL class is much more tolerant of bursts, and is appropriate for flows accepting some packet loss. The GS class supports flows intolerant to packet losses and having well defined traffic envelopes which they do not exceed, but heavily penalizes non-conformant flows. A comparative analysis of SRBQ (CL class) against RSVP

Aggregation has shown that although both models are able to provide adequate QoS in a scalable way, most QoS results are favorable to SRBQ and, more important, with SRBQ they are achieved with a much improved usage of network resources.

In the second part of the thesis, we started with a short introduction to the DAIDALOS project, identifying its major goals, key concepts and development guidelines. We then described the proposed QoS subsystem of the architecture, identifying its main components and explaining their interactions for providing end-to-end QoS support to flows across heterogeneous access technologies. We also gave a high-level view of the layered approach to resource management (performed per-flow in the access and in an aggregate basis in the core and inter-domain connections) and how they work together, providing context for the detailed partial descriptions in the following chapters.

With respect to per-flow resource management in the access, we proposed and analyzed three different scenarios for the interaction between QoS Brokers and the other QoS-related entities — centered on the terminal (PDA, intelligent cellular phone), on service proxies (e.g. SIP proxies or application servers), or in advanced mechanisms at the access routers — and the corresponding strategies for the interaction between application signaling and network-level resource reservation signaling. The MT-centered scenario is appropriate for operators mainly selling data transport with QoS services, the MMSP-centered scenario for operators providing multimedia services and content, and the ARM-centered scenario for operators providing a set of universally available network services. We also described how session renegotiation is coordinated with handover signaling, allowing the applications to adapt to the resource levels available in the different network access technologies. The different signaling strategies were evaluated in a number of simulation experiments. We concluded that the efficiency of the strategies is comparable under normal operating conditions, but the MMSP-centered strategy exhibits overload problems before the other ones due to the concentration of functions that in the other strategies are offloaded to the MT or the ARM. Since service degradation under overload conditions is quite abrupt, careful dimensioning of the components and load balancing mechanisms must be performed based on the worst-case expected load, particularly for the MMSP. We found out that excessive signaling load problems are greatly exacerbated by the snowball effect of SIP retransmissions, and concluded that a policy of summary rejection of new calls when the processing load reaches a certain threshold at the proxy should be implemented — such policy would prevent the snowball effect and thus improve the resilience to signaling overload with a lower investment in hardware.

Still regarding signaling in the access, we identified some sources of inefficiency with the joint use of SIP and Mobile IPv6 in a scenario where end-to-end resource reservation for the media must be performed. We proposed a solution for these inefficiencies based on the direct use of the Care-of Addresses in some messages (namely in the short-lived message transactions in call initiation) and on cross-layer interactions (use of layer 3 location information in session setup signaling). Based on a delay analysis and simulation results we demonstrated that with our proposed optimizations, session initiation is much faster than in the standard case, particularly in the presence of larger inter-domain link propagation delays (long distance calls) or packet loss in the wireless links.

The last problem we addressed in the DAIDALOS architecture was inter-domain QoS routing. We proposed a model where transit Autonomous Systems (ASs) are treated as "black boxes" providing virtual trunks for data transport between pairs of other ASs to which they are connected. Each virtual trunk is defined by a triplet of Autonomous Systems, and is shared by all active routes containing that triplet as a subsequence of the AS path. Virtual trunks may be defined as part of peering Service Level Agreements. We formulated the problem of virtual-trunk based inter-domain QoS routing in Integer Linear Programming (ILP), allowing us to obtain the optimal set of routes for a given network configuration and traffic matrix using a Mixed Integer Programming (MIP) code. This solution is a good benchmark, but cannot be used directly in practice, because it is centralized and requires global knowledge of the topology and traffic matrix, and because 0–1 integer programming is a NP-complete problem. Therefore, we proposed a practical solution based on a QoS extension to the Border Gateway Protocol (BGP). Our solution uses a combination of metrics (assigned bandwidth, expected light load delay and a three-level congestion alarm) for simultaneously achieving different and conflicting goals: finding non-congested paths that satisfy the QoS requirements of the data flows, minimizing the network resources used to transport the flows, and minimizing the message exchange, path computation overheads and route instability. We used ns-2 simulations to validate our proposal and compare it to standard BGP, to BGP with the QoS_NLRI extension, and to the optimal route set provided by the MIP optimizer. The results indicated that, contrary to the alternatives, with our proposal, congested paths and their consequences on QoS are avoided; even though there is a penalty in overhead and route stability in doing this, most of the routes are stable, especially if the thresholds for setting the alarm values are appropriately selected. It is worth noting that even though the proposed inter-domain QoS routing model was developed for use in DAIDALOS, it is not tied to the architecture, and can be used independently.

## 9.2 Topics for Further Research

"*Now this is not the end. It is not even the beginning of the end. But it is, perhaps, the end of the beginning.*" These are words by Winston Churchill on the first major victory during World War II, but they might just as well have been said of a PhD work. This section concludes the dissertation with some directions for the further development of our work.

With respect to the Scalable Reservation Based QoS architecture, the setup of a testbed using a real world implementation would allow for a better evaluation of its merits and eventual shortcomings; in particular, it would allow for a precise measurement of the processing load imposed by the traffic control mechanisms and by signaling processing. The signaling module would be implemented as a user level process, and the traffic control support of the Linux kernel has all the necessary pieces to build the traffic control module; communication between the signaling module and the kernel could be based on the LTCM library [Santos05].

With respect to signaling in the DAIDALOS architecture, one possible topic for further research is the evaluation of the possibility of using the forthcoming 802.21 standard for media independent handovers, and whether it could improve the handovers and their integration with session renegotiation. Another interesting development would be the integration of multicast support, as the interest in multicast applications is re-emerging. The support of broadcast transmission technologies where the return channel is absent or has different properties from the main channel would be another important topic for further work, since the generous capacity and sharing possibilities provided by broadcast channels have innumerous practical applications.

Regarding inter-domain QoS routing, it would be interesting to formulate a theoretical analysis of the convergence conditions for the inter-domain QoS routing algorithm, along with an evaluation of the practicality of imposing those conditions on the Internet. Devising methods for mitigating the effects of residual route instability would also be useful, particularly if the conditions for convergence of all routes are found to be impractical. As we already mentioned in chapter 8, it would be interesting to conciliate reasonably accurate QoS routing with the possibility of performing large scale route aggregation. The congestion alarm metric is aggregation-friendly, as is an approximate (instead of exact) light load delay metric. The inclusion of the bandwidth metric regarding the entire path or a subset of it, or its exclusion would have to be studied. The evaluation of inter-domain QoS routing with aggregation would require large scale simulations. Regarding the more general topic of inter-domain resource management, the development of methods and tools for automatic negotiation and establishment of Service Level Specifications according to the dynamic traffic demands would be a large step in the direction of providing QoS in the Internet.

# APPENDIX A

# ADMISSION CONTROL

In a resource-managed network providing QoS guarantees to flows, a mechanism is required that can estimate the amount of resources each new flow will need, and whether the network has enough available resources to service the flow, that is, an admission control mechanism: if enough resources are available, the new flow is admitted; otherwise, it is rejected.

The simplest form of admission control is the Parameter-Based Admission Control (PBAC). Under this scheme, each new flow tries to reserve specified amounts of a set of network resources, typically bandwidth and buffer space. The mechanism combines the amount of resources requested by the new flow with the amount of resources already granted to other flows, and decides whether the new flow is admitted by comparing this value with the total amount of existing resources. The simplest example of PBAC is the Simple Sum algorithm [Jamin97], using bandwidth only as the reserved resource: a new flow $i$ requesting a rate $r_i$ is admitted iff $\nu + r_i < \mu$, where $\nu$ is the sum of reserved rates and $\mu$ is the link capacity. A slightly more complex algorithm is used by SRBQ, where a maximum profile for the aggregate is characterized by the token bucket $(R_{GS\max}, B_{GS\max})$; a new flow requesting a token bucket profile $(r_i, b_i)$ is accepted iff $R_{sum} + r_i \leq R_{GS\max}$ and $B_{sum} + b_i \leq B_{GS\max}$, where $(R_{sum}, B_{sum})$ is the summed reserved profile of the already admitted flows. Parameter-Based Admission Control, combined with reservations that must be upper bounds on the amount of resources actually used by the flows, is required for providing hard QoS guarantees.

PBAC, however, has a drawback: in presence of bursty flows reserving traffic envelopes that are significantly higher than their average transmission characteristics, hard QoS guarantees come at the price of a low utilization of network resources. Many applications, however, have looser QoS requirements, and network service providing statistical QoS guarantees is sufficient for their needs. For this type of network service, Measurement-Based Admission Control (MBAC) algorithms are more appropriate: since they base admission control decisions on measurements of existing traffic, rather than on worst-case bounds on traffic behavior, MBAC algorithms can achieve higher network utilization figures than their parameter-based counterparts while still providing acceptable QoS. Evidently, traffic measurements are not always good predictors of future behavior, and so the measurement-based approach to admission control can lead to occasional packet losses or delays that exceed desired levels. However, such occasional failures are acceptable given the relaxed nature of the service commitment provided by soft QoS network services.

In addition to the Simple Sum PBAC algorithm, three MBAC algorithms were described and analyzed in [Jamin97]. One of these algorithms, named Measured Sum (MS), is a direct extension of the Simple Sum concept: a new flow $i$ requesting a rate $r_i$ is admitted iff $\hat{v} + r_i < \upsilon\mu$, where $\hat{v}$ is an estimate of the traffic load of existing flows and $\upsilon \leq 1$ is a target utilization factor that prevents the system from having an operating point too close to its full capacity, which would lead to instability and loss of QoS. The traffic load estimator most commonly used with the MS algorithm is Time Window, which computes the average rate during $T$ time intervals of duration $\tau$, and then uses the largest of these values as an estimation of the used bandwidth.

The MS algorithm was used in our ns-2 implementation of RSVPRAgg; the MBAC algorithm used in the CL class of SRBQ is an adaptation of MS for 3 drop probability levels, where the estimated traffic for each level is added to the estimated traffic of the lower drop probability ones.

Several other MBAC algorithms and their relation with different estimators are analyzed in [Breslau00a].

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