Q2S

Centre for Quantifiable Quality of Service in Communication Systems

Annual Report 2005

Norwegian Centre of Excellence

NTNU
Innovation and Creativity

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The third year of Q2S

The research effort at the Centre is motivated by society's growing dependency on a multitude of networked based services. Despite intensive attention by the world research community for a number of years, there are still many unsolved issues; proposed solutions do not scale or meet reasonable Quality of Service requirements, or it is not known how much quality they will offer. All this makes our research highly relevant.

It is important for Q2S to engage in international activities to address these problems, and a contribution here is the cooperation with EuroNGI, NoE and COST actions. EuroNGI has been given a prolongation until 31 May 2008 (renamed EuroFGI). Joint research activities, workshops and seminars, and PhD courses are important factors that will promote research.

Research activity at the Centre is increasing as many of our PhD candidates become more established. The Centre wants to stimulate research cooperation with other internationally well-known research groups, and in 2005 three of our PhD candidates have had research stays abroad. They will bring back new knowledge, experience, and personal contacts.

Up to now, ERCIM has been our single most important network for recruiting post-docs. The Centre is very pleased to acknowledge the important contribution made by the engaged postdocs several coming from abroad. They are a source of new knowledge and ideas. They have contributed to publications, and some ideas are now considered for patenting. We do not pursue patenting as such, but are very proud when it is possible.

Several of our previous postdocs have also applied for teaching positions at departments at NTNU. Recently Dr. Yuming Jiang was appointed full professor at the Dept. of Telematics.

The laboratories are constantly being updated. The Centre has invested here together with the departments and UNINETT. Some of the equipment is developed and tailored as research goes ahead.

The professors have teaching duties at the departments, and most of the PhD candidates have engagements with a department as teaching assistants. Many scientific problems at the Centre have also been offered this year as projects to fifth year master's students. These student projects are integrated into our ongoing research and give the students challenging and relevant problems.

This coming year the Centre will be evaluated and another five year period may be granted. Even though the Centre has had a "slow start" due to some initial recruitment problems, the targeted research projects have progressed steadily and have, questing the basic goals. This coming year we will reconsider the research projects, but with the same general research goals in mind.

A busy researcher at Q2S: Anne Nevin.

About the centre

The Centre deals with Quality of Service (QoS) issues in heterogeneous, multilayered networks where packet-switching technology is employed. By services, is meant traditional teleservices along with multimedia, messaging, web and information services, as well as location and content aware services. The Centre works within the following areas: multimedia signal processing, dependability, traffic, and security as applied to multiparty communication.

The Centre works with Quality of Service (QoS) in tomorrow's digital packet switched communication systems.

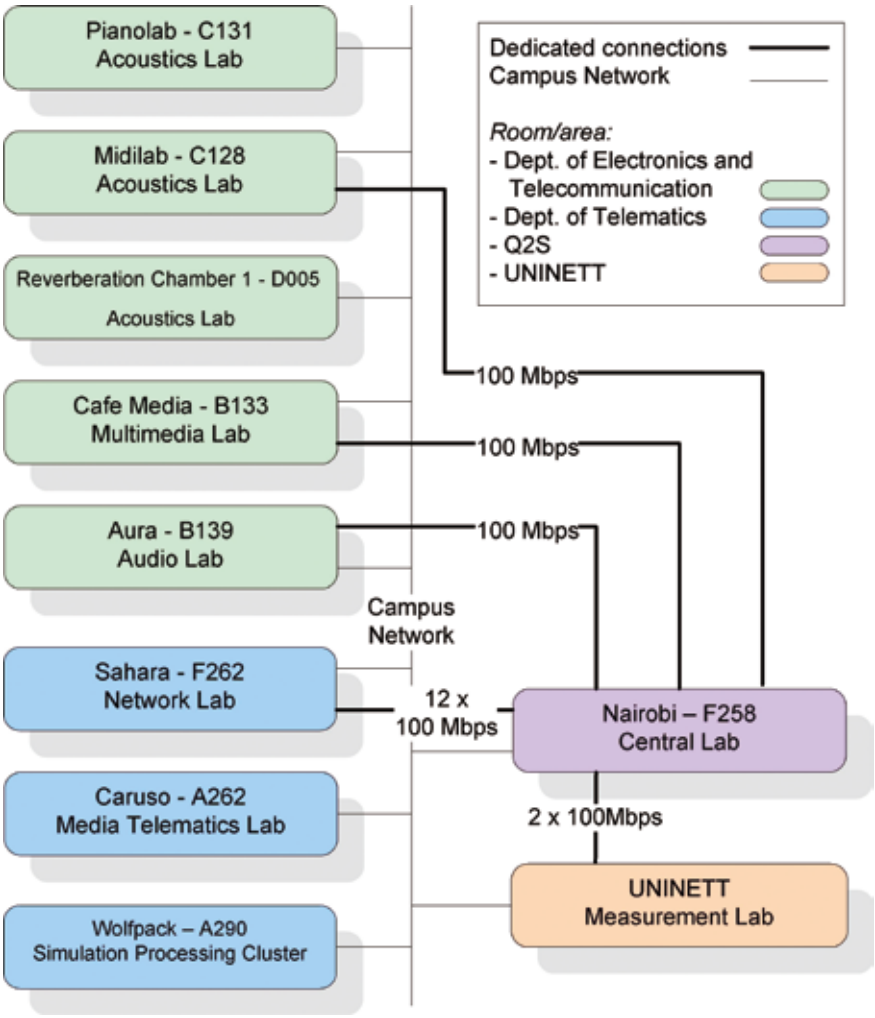
Society at large depends increasingly on digital packet switched networks, the Internet and mobile networks being the most dominant ones. Today, QoS in a broad sense is presently not acceptable in these networks but at best variable.

QoS is a totality and relates to user expectation and satisfaction. In order to assess QoS it is necessary to study the underlying architectures and solutions, make models and case studies, and perform measurements. For this, Q2S applies laboratories located at NTNU's Dept. of Electronics and Telecommunications, Dept. of Telematics, and UNINETT. Facilities available include video and audio capturing and presentation laboratory environments, laboratories for experimental networks, processing clusters for modeling and simulations, and advanced measurement equipment in both experimental and live networks.

In 2003, Q2S engaged in the work of creating an auralization laboratory. The work completed in 2005, resulting in "Aura", a modern and well-designed laboratory with excellent acoustical qualities. Aura is equipped with instrumentation for high quality reproduction of audio material over up to 16 loudspeakers. There are also possibilities for multichannel recordings, and there is audio processing software for advanced audio communication and real-time virtual acoustics, as well as high quality video projection equipment. Two-way experiments are also possible, such as playing music together over networks.

In November 2005 Q2S made the first version of LDAS, a Low Delay Audio Streamer, public. LDAS is a research tool for distributed multimedia interaction and perceived quality of service, and is primarily aimed at demanding applications where very low latency transmission of high quality multichannel audio is needed. LDAS is developed from scratch in order to have a solution that is fully open and enables control over all aspects and parameters of the transmission process.

Q2S has developed a streaming media testbed, which is a fully controlled environment for doing performance measurements of streaming media applications. The testbed includes streaming servers and clients, an IP network simulator, an IP network emulator, and a high-performance network interface monitoring card for capturing and replaying of packet streams. This infrastructure has been used to perform an error robustness evaluation of high-definition H.264/AVC video broadcasting over RTP/IP.



Labs available for Q2S researchers: Six multimedia labs, three network labs and a simulation lab.

UNINETT has deployed 15 measurement systems at customer premises during 2005 in a project called Measurement Beacon (Norw.: Målepåle). The idea is to allow customers to assess the quality of the network and their own connection and equipment, by deploying a multitude of state-of-the-art measurement tools. The Measurement Beacon will allow us to take advantage of the measurement tools developed in Lobster and Geant2 as well as serve as a lab for further research at the Centre.

The Centre is headed by a director who reports to the Board. The director maintains close contact and meets frequently with the five other professors, the UNINETT chief executive director and his director of experimental networks. Each postdoc and PhD candidate reports to one of the professors. Thus everyday research work is centered around each professor. The research is structured in projects where cooperation and coordination are required. One of the postdocs is responsible for laboratory coordination.

FACTS

The Q2S Centre

- A Norwegian Centre of Excellence (CoE) initiated by The Research Council of Norway
- Started 1 January, 2003
- Financed for 5 years by
  - The Research Council of Norway
  - NTNU
  - UNINETT
- Supported by Telenor
- Presently a scientific staff of 36 and 2 administrative/technical personnel.

Research Areas

- Audio over IP* Networks
- Multimedia over IP Networks
- Intradomain Networks
- Interdomain and Overlay Networks
- Trustworthy Multiparty Interactions in Dynamic Networking Environments

The Centre has now 13 internally defined projects in these areas

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* IP stands for Internet Protocol

International Cooperation

- Member of EuroNGI, a Network of Excellence under European Commission's 6th Framework Programme.
- Participates in COST actions COST 276 and COST 290.
- Participates through UNINETT in the EU projects Geant 2, SCAMPI and LOBSTER.
- Contributes through Telenor to the EU project STOLAS.

Personnel at Q2S

Members of the Board

MEMBERS

Nils Holme, Chairman of the Board
Eivind Hiis Hauge, Rector NTNU
Jostein Grepstad, Pro-dean IME, NTNU
Rolf Bjørn Haugen, Chief of Research, TELENOR ASA
Petter Kongshaug, Chief Executive Officer, UNINETT AS

Centre Director

Emstad, Peder J.
Professor at Dept. of Telematics

Administrative/Technical staff

Skotvoll, Anniken

Almåsbakk, Hans

UNINETT personnel

Kongshaug, Petter
Chief Executive Officer

Kvittem, Olav
Director of Experimental Networks

Hellan, Jon Kåre
Scientist

Øslebø, Arne
Scientist

NTNU Professors

Perkis, Andrew
Professor at Dept. of Electronics and Telecommunications

Helvik, Bjarne
Professor at Dept. of Telematics

Svensson, Peter
Professor at Dept. of Electronics and Telecommunications

Knapskog, Svein J.
Professor at Dept. of Telematics

Kure, Øivind
Professor at Dept. of Telematics

Postdocs

Bopardikar, Ajit S.
(01.08.03-11.02.05)

Brekne, Tønnes
(01.06.03-31.12.05)

Fidler, Markus
(01.01.05-31.12.05)

Gligoroski, Danilo
(15.06.05-)

Hoerdt, Mickaël
(03.10.05-)

Hyytiä, Esa
(01.08.05-)

Jetlund, Ola
(01.07.05-)

Jiang, Yuming
(05.01.04-30.11.05)

Kim, Anna Na
(01.10.05-)

Lundqvist, Henrik
(01.12.05-)

Marchand, Corine
(21.02.05-30.04.05)

Sæbø, Asbjørn

Wittner, Otto

Visiting personnel

Associate Professor
Janczewski, Lech
(08.08.05-31.10.05)

Associate Professor
Siew, Chee Kheong David
(20.10.05-01.11.05)

Doctoral candidates

Austreim, Erling

Gan, Qitao

Haslum, Kjetil

Hellerud, Erik

Hillestad, Odd Inge

Johansen, Stian

Libæk, Bjørnar

Lin, Yuan
(11.08.05-)

Moe, Marie E. Gaup

Moseng, Tor Kjetil

Mykkeltveit, Anders

Nevin, Anne

Paquereau, Laurent
(15.08.05-)

Sallhammar, Karin

Solvang, Audun

Undheim, Astrid

Voldhaug, Jan Erik

Zhang, Jijun
(01.05.03-31.10.05)

Årnes, André

Research Assistants

Research Assistant
Johansen, Thom
(02.12.05-24.02.06)

Research Assistant
Michaelsen, Stian
(27.06.05-15.08.05,
12.12.05-23.12.05)

Research Assistant
Solsvik, Fredrik
(27.06.05-15.08.05)

Research projects

Multichannel audio

Audio transmission over IP networks opens the way for advanced multichannel audio formats. Such formats can be used for high quality sound reproduction and virtual acoustics beyond today's stereo and surround sound formats, e.g., for video-conferencing with significantly improved sound quality. For multichannel formats, the coding is a central issue due to the growth in bandwidth for an increase in number of channels. This project studies multichannel audio formats for advanced applications that are suitable for communications over IP Networks. Application-dependent requirements, channel correlation, coding aspects, packet loss handling, psychoacoustic phenomena related to spatial hearing, electro- and room acoustic aspects of end terminals are studied in order to optimize format requirements.

Multi-channel audio in the Aura lab: Peter Svensson.

A focus is placed on the Ambisonics format which has a layered structure. Furthermore, the multidimensional nature of the perception of realistic sound fields requires new quality metrics. Simulations and real perception experiments will be used in this project.

Results this year:
In the on-going exploration of the multichannel format Ambisonics for the representation of sound fields in rooms, the possibilities to exploit spatial masking for the compression of Ambisonics signals was studied [37]. The technique is based on the identification of dominating sources in a complex sound field. Such an identification, which can be done by various techniques, is central to the effective representation of highly complex sound fields.

Relations to external projects and activities:
Collaboration with Bård Støfringsdal, PhD candidate at the Dept. of Electronics and Telecommunications, NTNU, in the project "Sound reproduction in virtual acoustics"

Project coordinator:
Professor Peter Svensson
Postdocs: Asbjørn Sæbø
PhD candidate: Audun Solvang

Audio in advanced video conferencing

Video conferencing can be used for demanding applications such as playing music together over networks, and other distributed multimedia applications. This application is very demanding with respect to end-to-end delay and uncompressed audio and video transmission has been used in demonstrations in order to minimize delays. The purpose of this project is to explore possibilities between this raw format with its low latency and high bandwidth requirement, and standard video conferencing formats with higher latencies but efficient compression. Central questions will be the dependence on the application of the involved compromises, which includes packet loss handling, end-to-end delay, and audio quality. Experimental software for audio streaming, as well as video streaming, will be developed. Real perception experiments will be used in this project.

Results this year:
Audio streaming software running under Linux was developed and made available as open-source software [T17]. Robust transmission with end-to-end delays below 10 ms is possible when the network

delays are minimal and no coding delay is introduced. The perception of various error concealment techniques for packet loss was studied in an initial round of listening tests. A technique was developed for the perceptual optimization of unequal error protection of audio over packet networks, by letting the sender simulate the receiver's error concealment [T7].

Project coordinator:
Professor Peter Svensson
Postdocs: Asbjørn Sæbø
PhD candidates: Erik Hellerud, Jan Erik Voldhaug

Scalable and robust coding schemes for IP networks

Packet delay and loss call for adaptive and error-robust coding schemes for multimedia traffic. Many proposed scalable coding formats require complex bit stream manipulation at intermediate locations in the network. Such requirements are unrealistic in present network nodes, so new solutions are needed. Emerging network handling techniques such as prioritization, intelligent packet dropping and congestion control feedback can be integrated into coding schemes in order to change this situation. We will focus our work on scalable and robust source coding techniques that exhibit graceful degradation under error-prone conditions. In general these coding schemes belong to the adaptive source coding domain which includes layered coding, scalable coding and multiple description (MD). These will be investigated in combination with packet prioritization, rate adaptivity and differentiated packet dropping policies. The relations and dependencies between adaptive encoding/rate control in the application and congestion control will be studied and exploited in a joint source/channel coding setup. We focus on techniques that incur a low end-to-end delay, in order to facilitate interactivity and real-time communication services.

Results this year:
The main results are from studying the interactions between source coding and network traffic handling, especially considering congestion control. A novel method for integrating congestion control and source/channel coding has been developed, in which incentives in terms of quality are provided in order to promote proper rate control for unresponsive sources. Results are being published in 2006. In considering robustness we have developed a method for concealment of packet loss related distortions in video based on structural alignment [12]. The method shows promising results and is far superior on concealing edge distortions. In considering the coding methods work has been concentrated on developing Unequal Error Protection (UEP) schemes [28]. We show how to assign unequal error protection when transmitting embedded data over multiple channels. The proposed extensions to algorithms in the literature perform remarkably well, and it is shown that the performance is consistent for different channel models. The advantage of using UEP is also seen to increase with the number of channels. When a high number of packets are used within each channel, the computationally more efficient EEP scheme may be attractive, as the performance in this case is quite close to that of UEP. The presented results encourage the use of UEP in emerging priority-enabled multichannel packet networks such as DiffServ.

Relations to external projects and activities:
Cooperation with Professor Tor Ramstad. Coordinated with Research

Council project: 140445/13 "Universal Multimedia Access from Wired and Wireless Systems"

Project coordinator:
Professor Andrew Perkis
Professors: Andrew Perkis, Peder J. Emstad, Øivind Kure
Postdocs: Venkatesh Babu Radhakrishnan and Anna Kim
PhD candidates: Stian Johansen, Odd Inge Hillestad, Yuan Lin, Anne Nevin, Bjørnar Libæk

Embedded quality assessment metrics for video presentations

In order to be able to monitor and assess the quality of multimedia over IP we will develop several metrics to measure video quality. In particular no-reference metrics that estimate video quality in the absence of a reference video stream can prove to be very useful in monitoring the quality of streamed video. We need to embed these metrics in the video player in order to improve the perceived quality of video presentations in a Universal Multimedia Access (UMA) enabled system. This leads to our view of QoS driven video players (viewers). At system level we propose to embed these metrics in the delivered bit stream over an IP network so as to use it to enhance the decoded sequence by optimizing de-blocking filters and error concealment for error resilient systems. Our investigations of the embedded metrics will form the basis for setting the requirements of novel multimedia viewers.

Results this year:
We have made some modifications and improvements to our no-reference (NR) metric [33] to use Neural networks for the quality assessments. By having a strong set of NR metrics we have concentrated on finalizing the

High media quality required in networked ensemble playing: (from left) Hans Almåsbakk and Asbjørn Sæbø.

Q2S laboratory testbed for measuring performance evaluations of multi-media services over IP networks [22]. Our initial performance measure has been to evaluate error robustness by modifying coder behavior with regards to parameters setting for the encoding process [T8].

The work within UMA enabled systems has been directed at developing both server and client solutions for adaptive delivery and reception of QoS optimized media resources as well as considering transcoding algorithms and system implementation for transcoding. [32], [46], [15]

Relations to external projects and activities:
Coordinated with Research Council project 140445/13 "Universal Multimedia Access from Wired and Wireless Systems" and COST 276 "Information and Knowledge Management for Integrated Media Communications Systems"

Project coordinator:
Professor Andrew Perkis
Postdocs: Ola Jetlund, Ajit S. Bopardikar, Venkatesh Babu Radhakrishnan
PhD candidates: Odd Inge Hillestad, Jijun Zhang

Traffic measurements

Traffic measurements are made to assess the delay and loss through a packet network, and to find the load and throughput of network elements. This is done by having several measurement points and observing and possibly time stamping packets as they pass by. The time stamping is done by using GPS synchronized clocks providing accuracy better than 100 ns. Comparing measurements from different points in the network enables the time delay and possible loss of packets to be found.
These measurements are passive measurements and are done at high

speed without any disturbance to the traffic. Another type of measurements is active or intrusive measurements. Then probe packets are injected into the network. These packets will inevitably load the network and disturb the traffic. The measurements form the basis for characterizing the traffic offered to the network and for making models. Traffic from video and audio is of particular interest. The metrics used here are different from those used to characterize the perceived QoS of audio and video.

Results this year:
Measurements on a laboratory router have been performed with the objective of making a parameterized model of the router. The idea then is to use this model to assess the router behavior by external measurements. Preliminary results from this work have been presented in [T18].

Measurements and studies of a single video stream through a router competing with real IP-traffic and synthetic traffic have been done [T16].

Efforts have been made to install a high capacity traffic generator to stream recorded traffic to stress a router or a link.

Relations to external projects and activities:
The SCAMPI and LOBSTER projects at UNINETT, EuroNGI, WP.JRA.5.1

Project coordinator:
Professor Peder J. Emstad
Professor: Peder J. Emstad, Øivind Kure
Scientists: Olav Kvittem, Arne Øslebø
PhD candidate: Astrid Undheim

Measurements on optical Gigabit links: Olav Kvittem.

Wireless networks

Achieving QoS is a hard problem in wireless environments. Our focus has been on various aspects of ad hoc networking where terminals must act as routers in addition to source and destinations. New terminals may enter and leave the network at will. They communicate with other terminals within hearing distance and with others by relaying messages through neighbors and so forth. They can connect to the fixed network if they are within range of a base station. Ad hoc networks should offer the same set of services as the ordinary Internet.

In addition to varying loads in a wireless network, the properties of radio channel itself will also vary over time, since the capacity is a function of environment and interference from neighbors. The nodes are likely to move, and the topology, capacity and load will therefore change over time.

Furthermore, as many terminals have access to the media, privacy is of great concern. QoS in wireless networks therefore represents a major challenge.

Results this year:
The work in wireless networks continued with the close collaboration with students and professors at UNIK, Kjeller. The ongoing investigation into enhancing UMTS multicast was put on hold due to Mariann Hauge's maternity leave.

The second line of investigation is directed towards secure routing collaboration between different organizations sharing the same ad hoc network. The focus has been on performance of the architecture [43], [44]. Ad hoc networks require a high average number of neighbors to ensure connectivity. Up to a certain limit, it is beneficial for different organizations to collaborate instead of operating separate networks, provided it can be done in a secure manner. Part of this investigation included work on evaluating the security of the routing protocol itself [42], [41].
A separate collaboration was initiated into QoS aspect of the routing protocols themselves. Ad hoc networks are characterized by high packet loss, often due to broken links, unavailable routes, or packet collisions. The two former ones can be improved by extending the functionality of the routing protocols. We investigated the changes required by using multiple interfaces [30], and the potential benefit of using the link life time as part of routing decision [29].

Spatial stochastic models for wireless networks have also been studied. This includes modeling the user mobility and characterization of the traffic in a wireless multihop network[T9].

Finally, the investigation into QoS architecture for ad hoc networks was initiated [T12]. The focus will be on the suitability of different proposed architectures, and how they can be incorporated into the QoS architecture in fixed networks.

Relations to external projects and activities:
EuroNGI, WP.JRA. 2.4, COST 290
Collaboration with Professor Pål Spilling, Researcher Frank Li, and PhD candidates Eli Winjum, Mariann Hauge, Department of Informatics, UiO and UNIK; PhD candidate Lars Landmark NTNU/UNIK; Senior scientists Knut Øvsthus FFI/UIO, Reinert Korsnes FFI.

Project coordinator:
Professor Øivind Kure
Professors: Øivind Kure,
Peder J. Emstad
Postdoc: Esa Hyytiä
PhD candidate: Tor Kjetil Moseng

Congestion control and service differentiation

Audio, video and multimedia services impose strict (real-time) requirements on the loss, jitter and delay through the network. Some lesser requirements are set for services such as web browsing, transactions and e-commerce. To meet these requirements, the routers use different traffic handling for the packets. For IntServ this is done on a per flow basis while for DiffServ it is done on a class basis. The latter therefore scales better and it is the preferred architecture in the Internet. With Diffserv, the packet handling is divided into two sets of tools, the scheduling of packets, and the discarding of packets when buffers reach certain watermarks. In order for Diffserv to be used to offer end-to-end QoS, the traffic load must be limited. This can either be done through adaptable applications or admission control. Congestion control signals, either as dropped packets or by the marking of packets, can be used to signal congestion to the sources and thereby let them adapt to the situation. This requires cooperation with the encoding at the source. Audio and video sources are studied for this purpose. Appropriate models of the sources have to be made, policies worked out for service differentiation in a router, and measurements taken.

Results this year:
Analytical studies have been made of rate control policies for streaming traffic. The model is based on a fluid-flow approximation of the traffic, and assumes that the streaming sources can adapt to some explicit notification in the case of congestion. The loss probability, the amount of signaling required, and how these control policies affect the quality at the receiving end have been studied in [10].

At the boundary between administrative domains, there is a need for service level agreement and specification regulating and describing the expected services. A service level specification describes the maximum properties of the accepted traffic and how the service should be monitored. Last year's focus was an investigation into these relationships for highly dynamic networks. The setting was tactical networks, which have similar properties as the dynamic rescue and emergency service networks [2].

Relations to external projects and activities: EuroNGI, WP.JRA. 2.1 and WP.JRA. 2.2,
Collaboration with Ingvild Sorteberg

Project coordinator:
Professor Øivind Kure
Professors: Øivind Kure,
Peder J. Emstad
Postdoc: Henrik Lundqvist
PhD candidates: Bjørnar Libæk, Anne Nevin, Erling Austreim

Security challenges in a mobile and wireless network: Audun Solvang.

Engineering for real-time services in NGI

Next Generation Internet (NGI) will offer a multitude of services, some of which have real-time requirements such as audio, video and multimedia, and should do this with an acceptable QoS. New knowledge is needed to do the required design, planning, dimensioning, and management. Service Level Agreements (SLAs) will be the contracts between network operators/service providers/users and should describe all relevant aspects of QoS that can be expected.

Several architectures like IntServ, DiffServ and MPLS are proposed to achieve these goals, but the proposals need further studies. Service differentiation is a key issue, but how to give end-to-end guarantees with respect to loss and delay needs much further research. In addition to service differentiation, admission control and possibly resource reservation are measures to be taken. The admission control schemes must be studied together with measures taken for congestion control. Laboratory experiments and model-based studies are needed to evaluate the various approaches.

Results this year:
Most of the work is based on Stochastic Network Calculus, which is a min-plus system theory for performance evaluation of queuing systems.

The paper [34] extends the analogy to classical system theory and shows that the Legendre transform plays a similar role in min-plus algebra as the Fourier transform in classical algebra. Owing to this analogy a dual network calculus is developed where min-plus convolution and min-plus deconvolution become simple addition respectively subtraction.

Emerging probabilistic network calculus equivalents aim at utilizing the independence of traffic flows to exploit statistical multiplexing. An additional, relevant gain can be made accessible in case of independent cross-traffic at multiclass schedulers. Respective scheduler models, for example for priority and first-in first-out scheduling, are derived in [51].

The paper [16] investigates the impacts of network topology and routing on the assumption of independence using dependency graphs and establishes a routing algorithm extension which preserves independence.

Based on dynamic priority scheduling (DPS), a measurement-based admission control algorithm has been proposed to support per-flow service guarantees in a flow-aware network [26]. Its analysis is based on the new queuing theory Stochastic Network Calculus. Various results contributing to the development of the stochastic

network calculus have been derived, which include a traffic model [24] and a server model [25]. The calculus has also been applied to service conformance study [3] and long-range dependent traffic analysis [6].

Relations to external projects and activities:
EuroNGI, WP.JRA.2.1, WP.JRA.2.2

Project coordinator:
Professor Peder J. Emstad
Professor: Peder J. Emstad
Postdocs: Markus Fidler, Yuming Jiang
PhD candidates: Anne Nevin, Astrid Undheim

Probabilistic assessment of security services in dynamic environments with non-trusted parties

The security attributes (confidentiality, integrity and availability) of services have traditionally only been expressed as qualitative issues. Some research on applying the dependability paradigm to security has been done during the past decade. In the dependability community, quantitative methods for assessing the attributes of dependability, i.e. reliability, availability and safety, are well-known and considered reasonably effective. A successful application of methods from the dependability area to assess the security attributes of systems will potentially lead to significant progress in the work of adding security to the Quality of Service architecture. This research work will therefore concentrate on how to model and analyze security by means of dependability methodology. New paradigms of ubiquitous integrated computing and communication services have great potential for improving the QoS of existing and new network-based user services, but it also opens new threat scenarios. Adequate network QoS, including security, will be decisive with respect to the establishment and maintenance of trust in fair and equal conditions for partners in a competitive economic system, to protect intellectual property or to contribute to the overall acceptance of the ubiquitous use of ICT among the general public. The study and development of principles and methods for establishing models for handling trust relationships in complex scenarios involving advanced network services in dynamic networks with mobile users are included in the project

Results this year:
Advanced research into the use of quantitative modelling techniques has been successfully conducted. The results [49], [35], [36] are published at several international conferences.

Project coordinator:
Professor Svein J. Knapskog
Professors: Bjarne E. Helvik, Svein J. Knapskog
Postdoc: Tønnes Brekne
PhD candidates: Karin Sallhammar, Kjetil Haslum

Relations to external projects and activities: EuroNGI, WP.JRA.6.3 Security.

Security services in dynamic network environments for mobile users

New paradigms of ubiquitous integrated computing and communication services have great potential for improving the QoS of existing and new network-based user services, but it also opens new threat scenarios. Larger portions of processed data will be considered sensitive, the number of security relevant incidents is steadily on the increase, and the number of possible perpetrators inevitably increases as well. Adequate network QoS, including security, will be decisive with respect to the establishment and maintenance of trust in fair and equal conditions for partners in a competitive economic system, to protect intellectual property or to contribute to the overall acceptance of the ubiquitous use of ICT among the general public. This research project will focus on principles and methods for establishing appropriate security services in complex scenarios involving advanced dynamic networks with mobile users.

Results this year:
Advanced research into the use of cryptographic services in networks has been successfully conducted. The results [5], [4], [52], [48], [17], [18] are published in international journals and at international conferences.

Project coordinator:
Professor Svein J. Knapskog
Professor: Svein J. Knapskog
Postdoc: Danilo Gligoroski
PhD candidates: Marie Elisabeth Gaup Moe, André Årnes

Relations to external projects and activities:
EuroNGI, WP.JRA.6.3 Security.

Other relevant activity: [55], [61], [53]

Patent filed: Gligoroski, Markovski, Knapskog: United Kingdom Patent Application No. 0522038.9 Enhancing Security of Hash Functions

QoS management by emergent behavior

The project explores the principles for, the feasibility and the quantitative properties of applying emergent behavior to provide QoS. As a basis for this, efficient resource utilization in

distributed communication systems is required. (Emergent behavior, in this context, is global behavior seen from the individual service users and the network as a whole, emerging from a (large) number of simple, loosely coordinated actions/behavior by agents/network elements/etc.) This may be regarded as a distributed stochastic optimization problem. From the rare event theory based on the cross-entropy optimization principle, simple agent behavior is derived, which finds good solutions to NP-hard problems, and with added heuristics, this deals well with QoS management issues.

Preparing for simulations: Henrik Lundqvist.

Results this year:
The previous work on the issue is made available in a reference work on dependable computing systems [54]. The applicability of the principles is further substantiated through simulation experiments with dynamic networks [19], the ability to provide network protection by p-cycles [45] and ongoing experiments with monitoring. The ns2 simulator, which is the vehicle for these experiments are thoroughly, revised (rewritten) during 2005. We have succeeded in dealing with the exponential growth of the number of states and complexity in the analytical/ numerical analysis of such dynamic networks [T6]. The findings are used for dependability modeling of networks with emergent and traditional routing schemes.Work has been started on dependability differentiation, [50], and emergent QoS handling in mesh access networks.

Relations to external projects and activities:
EuroNGI, WP.JRA. 2.3 and 3.3; Informal cooperation with Future & Emerging Technologies project (IST-2001-38923) BISON (Biology-Inspired techniques for Self-Organization in dynamic Networks)
Project coordinator: Bjarne E. Helvik
Professors: Bjarne E. Helvik, Victor F. Nicola
Postdoc: Otto Wittner
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QoS in burst and packet-switched optical networks

DWDM optical networks offer a huge increase in capacity and combined with optical switching, very efficient transport services may be provided. Research on burst and packet-switched optical (OBS and OPS) networks is initiated to overcome the lack of flexibility in OCS (optical circuit switched) networks. In these networks the design constraints from the optical technology are different from those traditionally met in packet-switched networks. For instance, buffering of data in the optical domain is at best inflexible and cumbersome and the switching costs become relatively much larger than transmission costs compared to "electronic" networks. This raises new challenges in comprehending as well as in the design and dimensioning of these networks to achieve a required QoS cost efficiently. The project addresses these challenges.

Results this year:
Most of the activities have been carried out within the STOLAS project, which concluded in May with a discussion on the viability of the optical label switch concept [47], [7]. The architecture with multihoming of the electronic in- and egress unit for tolerance of node and link failures is further studied [8] and work is done on fault tolerance in optical burst switching nodes [9], [11].

Relations to external projects and activities:
Essential technologies and infrastructure project (IST-2000-28557), STOLAS (Switching Technologies for Optically Labeled Signals), WP.1.2 in co-operation with Telenor R&D and Dept. of Telematics, NTNU.

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Dependability assurance in distributed systems

Distributed computing has the potential of providing highly dependable services by having replicas of the server processes on several nodes in the network. This technology is relevant for network internal services (e.g. network management) as well as external services (e.g. e-commerce). Keeping consistency between replicas is demanding and introduces performance penalties. Fault transparency, one of the virtues of distributed computing, also requires management functionality, which is critical for the QoS, both during fault handling and normal operation. The project studies the means to provide trustworthy services by this approach and the quantitative evaluation of the QoS obtained. The work has so far been focused on QoS related management in the Jgroup/ARM platform and experimental evaluation.

Results this year:
The experimental validation of the Jgroup/ARM system has continued with failure injection experiments. Multiple failures are randomly introduced in a in a cluster of service providing nodes. A method, based on post-stratification of the observations, is developed to relate the results, i.e. the service availability, to cases with ordinary scaled failure intensities [21].

Relations to external projects and activities:
The Jgroup/ARM Project, in cooperation with University of Bologna and University of Stavanger.

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Postdoc: Corine Marchand

Multimedia over IP Networks

Network media handling

Abstract

Multimedia communication over IP networks can either be one-way from a sender to one or more receivers (e.g. streaming) or two-way or interactive multi-way between two or more communicating parties (e.g. video or audio conferencing). Both these scenarios rely on audiovisual coding systems and network protocols such as TCP, UDP and RTP. The coding systems introduce coding distortions while the networks may introduce loss or delay of the information resulting in distortions in the decoded signal. This distortion needs to be quantified in order to be able to measure the perceived quality of the media presentation. Once these measurements are available they can be used in a feedback loop through the network to adapt the coding systems in order to enhance the user's perceived quality. This paper gives an introduction to the field, identifying the major challenges ahead and some of the solutions chosen to solve them. The case studies investigated are streaming media and audiovisual conferencing.

Introduction

The simplicity and flexibility of packet switched communication using the IP protocol has played an important role in its emergence as the method of choice for multimedia delivery. However, multimedia over IP and wireless networks face many challenges due to network variability and lack of service guarantees with respect to available bandwidth and delay and jitter. These result in packet-loss or, for streaming services, in packets being delivered after they are required. Clearly, the effect of such losses on video depends on how the video stream has been coded and how it has been mapped into flows and packetized.

In the current best-effort Internet service model, no service guarantees with respect to packet loss, delay jitter and available bandwidth can be made. Packet loss most often occurs due to congestion in network nodes; more and more packets are dropped by routers in IP networks when congestion increases. While packet loss is one of the things that makes the TCP protocol efficient and fair for non-real time applications communicating over IP networks, the effect of packet loss is a major issue for real-time applications such as streaming of audiovisual media using the RTP protocol over UDP/IP. Even delay jitter manifests itself as packet loss, since packets received after the intended playout/presentation times are not useful.

Currently, providers and creators of multimedia presentations have to create multiple formats of content and deploy them in a multitude of networks in order to meet the increasing demand from consumers for high quality interactive multimedia. Also, no satisfactory automated configurable way of delivering and consuming content exists that scales automatically to different terminal and network characteristics, device profiles or QoS. The quest to represent, deliver and present such interactive multimedia with the ultimate experience in multimedia entertainment and conferencing in such a multimedia framework means that the boundaries between the delivery of audio (music and speech), accompanying artwork (graphics), text (lyrics), video (visual) and synthetic spaces will become increasingly blurred.

In Section 2 we illustrate how some different scenarios for multimedia communication are handled in an IP

network. Section 3 gives an overview of video coding, focusing on new video coding techniques including H.264/AVC (H.264 named by ITU viz. MPEG-4 Part 10 Advanced Video Coding named by ISO/MPEG) and scalable coding. Audio coding in the form of AAC coding is described briefly, and audio conferencing is discussed. Section 4 deals with perception, specifically quality as perceived by a receiver/ listener or by the communicating parties, of the resulting sound fields and of the audio based communication process.

Network architecture for audiovisual communications

Many important challenges arise when considering audiovisual communication over packet-switched networks such as multiservice IP networks. Available resources in these networks are shared among competing flows with highly varying characteristics, and even if the network supports some QoS scheme (such as service differentiation) individual media flows will typically experience varying delay jitter and occasional packet loss during periods of network congestion.

Audiovisual communication is in the form of a few common services, as listed in Table 1. Downloading of multimedia content is basically no different from other file downloading, and is handled by the TCP protocol since no real delay requirements exist. One-way streaming of speech, audio or audio+video can be a close-to-real-time service if speech, audio and/or video is encoded and consumed in real time. The two-way conferencing service includes IP telephony. Audiovisual communication may also employ virtual audio and/or video in network games, where participants move around and interact in virtual environments that are rendered at the end terminals. Short command-like messages are sent in real-time, and for some games delay requirements might be very strict.

In order to prevent packet loss from corrupting an entire file or session, application protocols like FTP or HTTP employ TCP, a reliable byte-oriented transport protocol which uses acknowledgements to explicitly state which byte segments that have been properly received. Lost segments are retransmitted automatically. While this mechanism obviously does not work well for the transport of media with real-time constraints, the unreliable real-time alternative, RTP over UDP, might employ a similar approach on the application level for retransmitting lost packets. Typically, such an automatic repeat request (ARQ) mechanism would only be used for retransmitting packets belonging to frames that could still arrive in time for decoding and presentation. Recently the SIP protocol has emerged for configuration of the service prior to set up. It is widely used for Voice over IP and also considered for peer-to-peer systems.

For an audiovisual streaming system, the codecs being used and their configuration are of fundamental importance. They set the upper limit for the reconstructed quality and level of distortion introduced. Depending on the application scenario (e.g. live broadcast, two-way conversational or on-demand streaming), additional encoder decisions with regard to rate control and coding mode, error control and error resilience tools (e.g. Forward Error Correction, FEC, data portioning), rate shaping and packetization all have an important effect on the audiovisual quality and application performance during a video streaming session. [Wu01].

Second, the perceived quality heavily depends on the ability of the system to adapt to changing network conditions and show graceful behavior in case of lost or delayed packets. For instance, packets that are delayed beyond the time they should be available for decoding at the receiving end, are of no use and inflict the same damage as loss of packets in congested router queues. This situation can be alleviated

Table 1 Some audiovisual communication scenarios.

SERVICE	DELAY REQUIREMENTS	PROTOCOLS	EXAMPLES	BANDWIDTH REQUIREMENTS
Downloading	Practically none	TCP - no difference from general file downloading	Purchase of recorded music, films, TV shows	
Streaming (one-way)	Relaxed	TCP or UDP+ARQ	Internet radio - live or archived Video on demand	Radio: 32-128 kbps TV: 2 -10 Mbps HD >10 Mbps
Conferencing (two- or multiway) audio/ video	Voice over IP: 150 ms Music playing: 20 - 50 ms	UDP	IP telephony Video conferencing Distributed music playing	Voice over IP: 16 kbps Video conf: 144 kbps - 2 Mbps
Multi-party with synthetic signals	Strict-to-medium	UDP	Multiplayer network games	

by increasing the size of a playout buffer (also known as jitter buffer) in the streaming media client, but this increases the end-to-end delay and memory requirements. Another recent proposal to prevent buffer underflow is adaptive media playout, in which the playout rate of decoded frames is altered in accordance with the current buffer fullness [Kalman04].

Intelligent retransmission schemes can also be used to recover lost packets and help prevent loss of media data. However in situations of heavy network congestion, packet loss is destined to occur. If packet loss occurs, the aspect of graceful behavior is reflected in the decoder's error resilient behavior; the use of proper error detection and error concealment techniques make sure that the visual appearance of the media resource shows a gracefully degrading visual reconstruction. [Wang98]

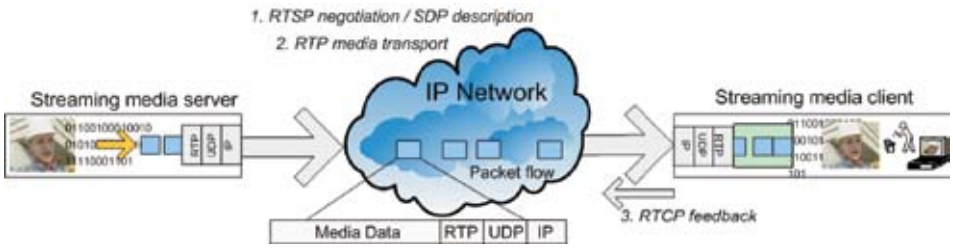

Figure 1 Streaming media architecture.

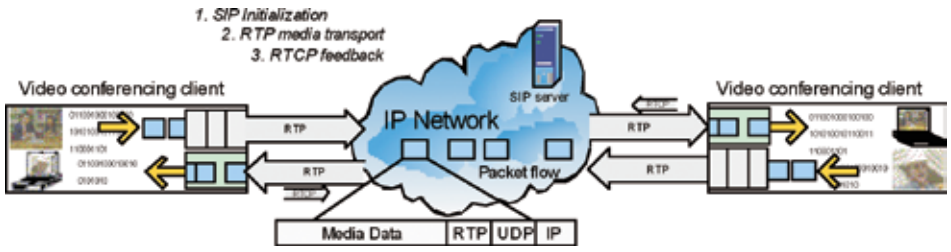

Figure 2 Conferencing architecture.

Figure 1 depicts a typical streaming session, in which a piece of media content is delivered from a server to a client on-demand. The client requests the media using RTSP, and receives a description of how to access and decode the media flows for this session. The media is transported using RTP/UDP, and RTCP may be used to feed reception statistics back to the server. [RTP96] At the receiving end, the streaming media client retrieves media packets from the playout buffer, detects packet loss, decodes available media data and tries to conceal missing parts of the media representation. To measure media quality on the receiver side, and help make right decisions in adapting the delivery, objective quality metrics could be applied to the reconstructed media in order to estimate end-user perceived quality, as described in Section 4. Furthermore, the sending end might employ unequal error protection (UEP) which can be optimized against packet loss.
The two- or multi-way service of IP telephony or video-conferencing is illustrated in Figure 2. A typical session here involves the request by one of the parties to set up a session. This might be done using the SIP protocol, and an SIP server sets up the two-way streaming which typically uses RTP or proprietary protocols on top of the UDP layer. The handling of packet loss and varying network delays might be similar to one-way streaming sessions. Video conferencing is commonly used for meetings where speech conversation is complemented by video. More advanced uses include playing music together, as has been demonstrated across the American continent [Woszczyk05]. Very high quality video and audio transmission has been used in such music-playing examples.

A service which has seen a significant growth is network games. Some types

might involve hundreds of participants and the idea of playing computer games together over local networks, and later the Internet, has been around since the mid-90s. Depending on the type of games, delay requirements might be strict or relaxed. Often, a central server collects and distributes commands to and from participants, but commands might be very compact. Environments and sounds are rendered at the end terminals. Peer-to-peer network games also exist.

At Q2S, a dedicated multimedia test bed and IP networks are used to measure the performance and characteristics of streaming media applications such as the ones described above. Besides a streaming server and a streaming media client, it consists of an IP network emulator that enables us to subject our application to varying delay and packet loss in real-time. In addition, the test bed is equipped with high-performance packet monitoring and capture devices, and a packet flow regenerator that enables us to replay media streams and recreate specific scenarios and network conditions. [Hillestad05].

Audiovisual coding for IP networks

The possibilities to compress speech, audio, images and video have been explored ever since digital techniques were introduced. There has been a strong emphasis to minimize the bit-rates of such media content in order to use limited bandwidth and/or storage space resources efficiently for rapidly increasing communication demands. For some applications, such as two-way communication, the effort to achieve efficient compression algorithms has been balanced by the requirement to keep processing delays low. The delay requirements are especially strict for real-time audio communication. Another central factor is the robustness to errors. For speech, a number of codecs (coder/decoder) have been developed and used in, e.g., mobile telephone systems, ranging from simple logarithmic quantization to advanced speech modeling codecs in use today that are based on predictive coding. In audio, the compact disc was the breakthrough for digital audio but notably it uses uncompressed audio for maximum quality but also with a high robustness to drop-out errors. In the 1990s, compression of audio started to be used in applications like digital radio broadcasting and digital audio distribution formats such as Sony's Minidisc. Compressed audio got a real breakthrough in the form of music file transfer on the Internet. The well-known MP3 format which is one example of ISO/MPEG coding [MPEG-1 1991, MPEG-2 1994] has lead to a wide acceptance of such compression. A rapid development has been possible because of the ease to introduce new formats using software plug-ins, such as Real Audio and Windows Media Player. Consequently, the bit-rate needed for what is judged to be adequate quality has been decreasing steadily.

Compression of audio and video signals

At present, most of the successful techniques for video coding are in some way a part of either the MPEG standards or the Society of Motion Picture and Television Engineers (SMPTE) standards. SMPTE is currently in the process of standardizing a low rate video codec referred to as VC1, which again is based on the Windows Media Video (WMV) codec developed by Microsoft. Vendor technologies are also included in the MPEG standards. The most current of these is the MPEG-4 standard. There also exists several open source video codecs based upon MPEG-4, e.g. OpenDivX and XviD. In classic literature, coding or more specific video compression is separated into four categories [Ebrahimi 1998]; waveform, object based, model based, and fractal coding. The first is also commonly referred to as transform coding. With the exception of fractal coding all four categories are included in the MPEG standards. All in all the MPEG standards comprise a large knowledge base of video coding techniques and the ongoing efforts by the Moving Pictures Experts Group provide state-of-the art video coding techniques.

MPEG-1 covers video compression with target bit rates up to 1.5 Mb/s and consists of techniques for synchronization and multiplexing of audio and video, compression of non-interlaced video, and a compression codec for audio designed for perceptual coding. The MPEG-2 standard is typically used for encoding audio and video broadcast signals for target bit rates between 1.5-35 Mb/s. Unlike MPEG-1 this standard also includes interlaced video. Enhanced versions of MPEG-2 are used as the video codec in DVD movies and in most HDTV transmission systems. The MPEG-4 includes many of the features in MPEG-1 and MPEG-2 [MPEG-4 1997]. However, this standard also includes support for digital rights management, object-based coding, interactivity, 3D-rendering, and target bit rates as low as 8 Kb/s and up to 35 Mb/s. MPEG-4 consists of numerous parts dealing with system description for synchronization and multiplexing, but also with state-of-the-art coding techniques for audio and video such as; Advanced Audio Coding (AAC), Advanced Video Coding (AVC) and carriage on IP networks.

The newest video coding standard is a joint effort between ISO and

ITU resulting in many names. The standard is commonly referred to as either MPEG-4 Part 10 Advanced Video Coding or H.264. For simplicity we will refer to the standard as H.264/AVC. H.264/AVC has achieved a significant improvement in compression performance, error resilience and a "network- friendly" video representation.

For audio coding within the MPEG standards, the first generation included MPEG 1 Layer 3, or MP3, which uses a sub-band technique as well as a finer-resolution transform and bit-rates of 32 - 320 kbps. The next generations, MPEG 2 and MPEG 4, introduced the AAC technique. This employs a transform technique but no sub-bands, and is not backwards compatible with MP3. Evaluation of the perceived quality shows that AAC offers somewhat better sound quality than MP3 does at the same bit-rates. Alternatively, a lower bit-rate with AAC can offer for the same perceived quality as somewhat higher bit-rate with MP3. For most listeners, a bit-rate of 128 kbps with AAC leads to transparency, that is, no difference can be heard from a CD original [Herre02]. It should be noted, however, that this is for typical music material, and for typical listeners. Notably, the AAC coding technique with a bit-rate of 128 kbps has been selected by the iTunes Music Store for online sales of music [Apple05].

As indicated in Table 1, conferencing applications might require low end-to-end delays, and improved delay performance can be observed in the development of the AAC audio codec may also serve as an example of improvement due to latency requirements. The original AAC had an algorithmic delay of several hundred milliseconds. The later AAC-LD (Low Delay AAC) has an algorithmic delay of 20 ms, and a reported end-to-end implementation latency of about 45 ms [Hilpert00]. The recent Ultra Low Delay codec from Fraunhofer supports an algorithmic delay of 6 ms [Hirsch04].

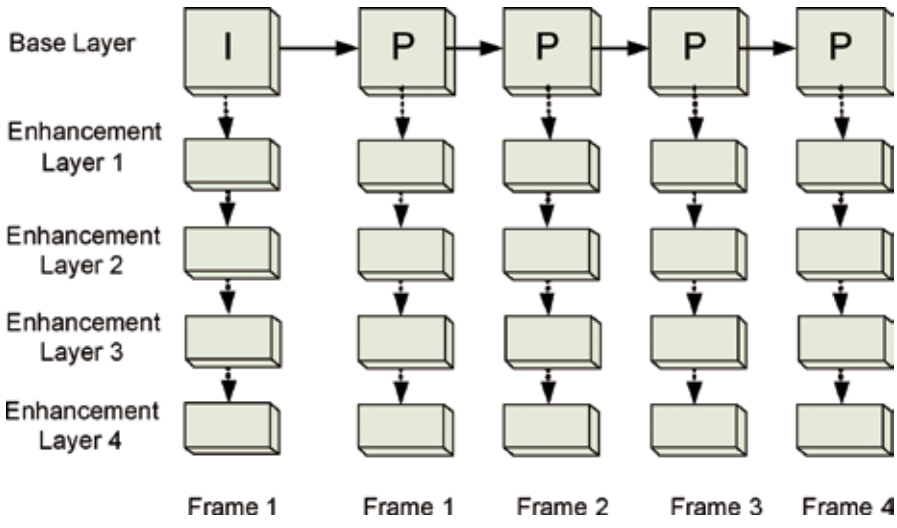

Figure 3 Scalability by base layer and enhancement layer.

Scalable coding and compression for lossy transmission

The development of efficient scalable coding schemes is motivated mainly by the possibility of adapting the encoded data to match channel/network conditions, terminal capabilities or business models. The concept of scalability is shown in Figure 3, where the encoded bit stream for each frame is built up by a base layer and several enhancement layers. Such an approach has been adopted by MPEG-2 and also in the upcoming scalable version of AVC/H.264 called SVC (Scalable Video Coding). The scalability can either be by spatial or temporal resolution or by quality (SNR).

Furthermore, since the generated bit stream is embedded, effective unequal error protection (UEP) schemes can easily be applied. These schemes utilize the fact that the importance of each bit is (conceptually) monotonically decreasing as one move from the start to the end of the bit stream. Different parts of the bit stream are assigned different strength error correction codes, dependent upon the relative importance of the bits and the channel/network conditions. Novel results include [Albanese1996] [Mohr1999] [Puri1999], and fast algorithms for assigning UEP have been proposed [Stankovic2002]. Recently, this concept has been extended to transmitting over multiple parallel channels with possibly different characteristics [Johansen2005]. For audio signals, UEP strategies have been suggested that protect transient parts of the audio signal since error concealment is less successful for transients [Korhonen 05]. More generally, automatic algorithms for estimating the perceived quality of error concealment, at the receiver in case of packet loss, can be employed by the sender for choosing when to apply UEP [Hellerud2005].

Impairment	Rating
Imperceptible	5
Perceptible, but not annoying	4
Slightly annoying	3
Annoying	2
Very annoying	1

Figure 4 Illustration of a subjective test and a five-point rating scale that might be used to judge the degradation of an audio signal compared to a reference signal.

Quality metrics and perception

QoS may be defined as user satisfaction with the service, that is, the perceived quality by the end user for a specific service. This is a far-reaching concept since it involves direct effects such as perceived quality of the sound and video when multimedia content is streamed. Furthermore, indirect effects that affect the perceived quality could include price and expectation of the service, as well as the preconditioning of the customer. It is virtually impossible to derive a model that would predict a single-number perceived quality taking all these factors into account, in particular since many of the factors have strong individual weights. What is possible, however, is to quantify quality within well-defined subsets of this complex and with many of the factors controlled. The use of quality metrics can be very different in a laboratory setting, with the aim of developing a coding scheme, compared to a live traffic setting, with the aim to adapt codec settings to current traffic properties. A distinct difference exists between so-called Full-reference (FR) metrics, and No-reference (NR) metrics. FR metrics can be used in cases where both an original signal and a degraded signal are available whereas NR metrics must be used in cases where the original is not available. Both types of metrics might be used during real-time streaming. The sender might employ FR metrics for predicting the perceived quality at the receiver in case of packet loss, for example [Hellerud 2005], and use this information for optimizing the encoding. At the receiver end, NR metrics might be applied for the decoding process.

Perceived quality will depend on the type of service. For audio signals, there are two distinctly different situations: listening-only (streaming) and conversational (conference) situations. Even if the real application is a conversational situation, such as in telephony, the listening-only situation is often used in tests since such a test situation is easier to control. The result from a listening-only test does not translate perfectly to a conversational situation but a strong correlation exists.

For the listening-only situation, the most used subjective assessment method is a five-point point impairment scale used in a so-called absolute category rating where a scale of categories is used. For speech tests, this is the technique standardized by ITU [ITU-T P.800] and the quantity that results from such a test is called "mean opinion score", abbreviated MOS. Versions of this might compare a stimulus to a reference for an increased sensitivity to small differences, so-called Degradation Category Rating, as illustrated in Figure 4. For audio codec evaluation, this test is standardized by ITU [ITU-R BS.1116].

These subjective tests have become standard ways to evaluate speech codecs and audio codecs. Using subjects is, however, time-consuming so methods have been developed for objective evaluation of the sound

quality of systems. One approach is to compare a degraded signal with a reference and to evaluate these two using some perceptual model. This so-called Full Reference (FR) approach is used in the PESQ method (Psychoacoustic Evaluation of Speech Quality), [ITU-T P.862] and PEAQ (Perceptual Evaluation of Audio Quality), [ITU-R BS.1387]. In these methods, a perceptual model of hearing is used to determine the difference on hearing-related scales (for frequency and amplitude) between a reference signal and a signal that has been degraded by a system. This is illustrated in Figure 5, and it can be noted that such an evaluation is of the FR type since the degraded audio signal is compared to the reference input signal. A number of parameters have been adjusted to make the quality predictions fit as well as possible to a large data set of perception results from experiments with human subjects. For the PESQ model it has been possible to reach a high correlation between the objective predictions and the rating from subjective evaluations and consequently it can be used for automatic evaluation of speech codecs. However, such an approach will always be less accurate for evaluating cases that are very different from the data set used for adjusting the model.

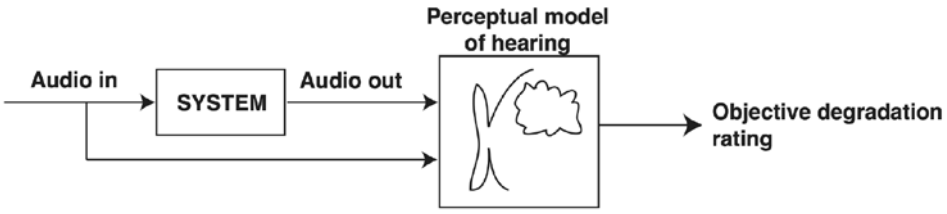

Figure 5 Illustration of an objective quality evaluation of the Full-Reference type.

For communication over packet networks the effects of delays and packet losses are central. The so-called E-model includes the delay in an attempt to model the conversational quality [ITU-T, G.107] whereas the PESQ does not. Studies of conversational quality have indicated quite variable results because the conversational style and the experimental setting will have a lot of influence on the perceived degradation due to delays. The ITU recommendation is that a one-way delay should be below 150 ms for "essentially transparent interactivity" [[ITU-T, G.114]. Still, some applications might be affected by even lower delays. An example of this is distributed music playing, which was referred to above, where it is possible to measure the effect of musicians' timing for delays as low as 20-40 ms.

For IP networks, the deterioration in perceived quality is typically due to packet loss [Feamster 20002, Boyce 1998, Hillestad 2004, Bopardikar 2005]. The other major source of distortion and degradation of perceptual quality in multimedia communication is because of the inevitable coding and compression of media sources. In particular, for block-based video compression schemes such as the ISO/IEC and ITU standards (e.g. MPEG-1/2/4, H-261/3/4) the main forms of distortions include block impairment effects, blurring, ringing and the DCT basis image effect [Wu 1996, Wu 1998].

NR metrics that have been proposed, in general try to quantify the effects of these distortions [Caviedes 20003, Marziliano 2002, Babu 2004] but the emphasis of research on NR metrics has been predominantly on quantifying the effects of block impairment artifacts [Wu 1998, Winkler 2001, Wang 2002, Gao 2002]. This is because block impairment artifacts tend to be perceptually the most significant of all

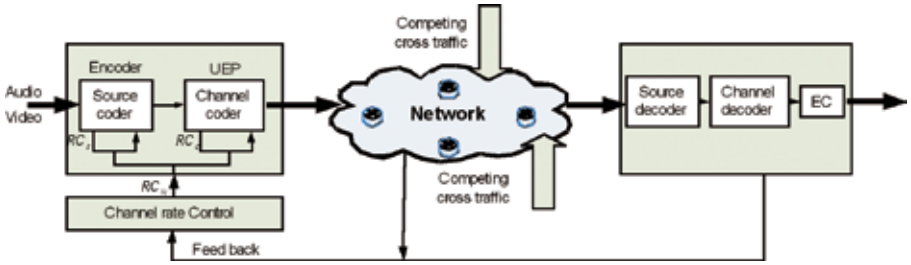

Figure 6 Multimedia communication.

coding artifacts. With the Video Quality Experts Group working towards their standardization [VQEG], NR metrics remain a topic of great research interest.

Challenges

The effects of delay and packet loss in packet-based transmission have introduced the challenge of handling delays and packet losses. Existing standards for speech coding and video conferencing have been extended to include some form of packet loss concealment but more development can be expected in this area. The delay that is introduced by the packet-based transmission is irrelevant for most one-way transmission applications but is crucial for two-way communication. From one end to the other end, the total delay includes the physical transmission delay as well as the buffer delays at both ends. These buffers are necessary because of delay variations, packet loss concealment algorithms, and coding/decoding processing delays. Notably, the average physical transmission delay might be a small part of the total delay.

In current multimedia communications there is already a digital media market place, where we all consume - our mobile phones, Internet-based services or broadcasting. The major challenges arising are in transcoding between formats or even transmoding between media modalities such as speech, audio and video. This requires new ways of content representation, where the methods and functionalities must consider interactivity, and adaptive representations of the media. Figure 6 illustrates a system for multimedia communications where the focus is on the transmitter and receiver sides. The adaptation to the network feedback requires Rate Control (RC) mechanisms at the transmitter. This is indicated by separate methods for rate control for the Source code – RCs – the Channel coder – RCc and the Channel rate – RCn. For quantifying quality we need quantitative measures for perception of digital media and the ability to measure this in IP networks. The measure can be used for Error Concealment (EC) or as the feedback from the receiver to the transmitter to improve the quality in a dynamic way.

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Further reading

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International collaboration

EuroNGI

EuroNGI, Design and Engineering of the Next Generation Internet, is a Network of Excellence under the European Commission's 6th Framework Programme.

The Next Generation Internet (NGI) will offer multiservice/multimedia, mobility, convergence (services and fixed-mobile), Quality-of-Service and variable connectivity as the norm. On the one hand, future high-speed wireline and wireless access technologies provide instant high bandwidth connectivity, which makes it difficult to forecast traffic and thus to apply existing traffic engineering methods. On the other hand, the technology diversity explodes and mastering such a heterogeneous environment becomes essential for the network designer. This requires investigation into new multi-technology architectures. It is understood today that new design, planning, dimensioning and management principles are needed.

In this context, the main topics addressed by the NoE are:

- Mastering the technology diversity (vertical and horizontal integration) for the design of efficient and flexible NGI architectures.
- Providing required innovative traffic engineering architectures adapted to the new requirements and developing the corresponding appropriate quantitative methods.

EuroNGI was evaluated by the European Commission after its first year of operation and received the best possible feedback: "Proceed as is". The network has applied for a prolongation of its period, and is granted funding until 31 May 2008, under the acronym Euro-FGI, Design and Engineering of the Future Generation Internet.

For further information, see the website of the network http://www.eurongi.org/.

Contributions from Q2S:

The Centre has attended organizational meetings, workshops, and made several presentations. There have been made contributions to work packages listed below.

- WP.JRA.1.1 Horizontal and vertical integration of fixed and mobile networks
- WP.JRA.2.1 Mechanisms and protocols for controlled bandwidth sharing
- WP.JRA.2.2 Traffic management in a multi-provider context
- WP.JRA.2.3 Traffic engineering for a cost effective network
- WP.JRA.2.4 QoS in multiservice wireless networks
- WP.JRA.3.3 Study of methods for achieving network resilience/ robustness
- WP.JRA.5.1 IP traffic characterization, measurements and statistical methods
- WP. JRA.6.3 Creation of trust by advanced security concepts

Q2S arranged a two-day working group meeting for WP. JRA.6.3 in Trondheim in March 2005.

There were several contributions from the Centre at the EuroNGI 2005 Conference. PhD candidates Karin Sallhammar and André Årnes contributed with presentations at the "Spring School on Security" in Marseille, France in April 2005.

PhD candidate Anders Mykkeltveit followed the PhD course "Multi-commodity flow networks – modeling and optimization" given by Politecnico di Milano. Most of the course was taught during one focused week in November 2005.

PhD candidates Erling Austreim and Bjørnar Libæk contributed with presentations at the "Workshop on QoS and Traffic Control" in Paris in December 2005.

COST 276 "Information and Knowledge Management for Integrated Media Communication"

The main objective of the Action is to develop advanced multimedia data and knowledge management technologies for personal multimedia communication systems and services, including specific signal processing and implementation techniques for users' personal terminals. In addition, key system aspects will be considered, such as: system integration, personification of services, usage trials and demonstrations of advanced personal services.

For further information see the websites: http://cyberspace.mht.bme.hu/cost/cost276/Home.htm http://www.cordis.lu/cost/src/276_indivpage.htm

Contributions from Q2S:
The 8th and final COST276 workshop [56] was organized by Q2S and held on 27-28 May 2005 in Trondheim. 22 papers were presented in 5 oral sessions and 1 poster session. Dr. Jan Bormans from IMEC gave an invited talk on "Quality of Experience for Personal Multimedia Communication Devices" and Professor Torbjørn Svendsen from NTNU held an invited talk on "Speech technology - why, what and where?"

COST 290 "Traffic and QoS Management in Wireless Multimedia Networks"

The main objective of the Action is to develop and to investigate new approaches, techniques, methods, models, strategies and tools for the analysis, design, control and evaluation of future advanced Multiservice Wireless Networks (MWNs) supporting user mobility, multimedia applications, and internet working. Special attention must be given to QoS and related aspects in both access networks and core networks in the presence of mixed multimedia traffic. New analytical tools, software implementations and prototypes have to be invented to accomplish these tasks.

For further information see the website: http://www.cost290.org/

Contributions from Q2S:
We have participated in the all COST 290 meetings in 2005 presenting ongoing work in the wireless project.

SCAMPI - Scalable monitoring tools for the Internet

SCAMPI is a European project to develop a scaleable monitoring platform for the Internet. It also aims to promote the use of monitoring tools for improving services and technology.

SCAMPI involves ten partners from commercial, research and academic sectors and represents a total investment of EUR 5.5 million; 50% of which will come from the project partners themselves, and 50% from the Information Society Technologies Programme of the European Commission. The project started on 1 April 2002 and ended 30 April 2005.

Objectives
- Development of a high-performance intelligent monitoring adapter, initially at speeds of 10 Gbps.
- Development of an open and extensible architecture for network monitoring.
- Development of monitoring and measurement tools for denial-of-service detection, SLS auditing, quality-of-service, traffic

Advanced media coding: Yuan Lin.

engineering, traffic analysis, billing and accounting
- Investigate strategies and methodologies for monitoring systems operating at 100 Gbps and beyond
- Collaborate with other monitoring activities and standardization bodies.
- Dissemination of results

For further information see the web-site: http://www.ist-scampi.org

The SCAMPI project ended in 2005, and the final evaluation was excellent.

Fruitful discussions: (from left) Arne Øslebø, Olav Kvittem, Jon Kåre Hellan and Petter Kongshaug.

LOBSTER-Large-scale Monitoring of Broadband Internet Infrastructures

Accurate network monitoring systems enables a variety of new applications, including

- early warning for the detection of Internet worms as soon as they start to spread,
- detection of distributed denial-of-service attacks,
- better traffic characterization, and
- "traffic weather" service for GRID-enabled applications.

Although accurate network monitoring is increasingly important for reliable and efficient operation of cyber-infrastructure, current traffic monitoring systems in Europe do not provide the information needed to support the above-mentioned applications. Indeed, current systems focus either on collecting lossy flow-level statistics, or on active measurement of latency, bandwidth, error rate, and similar properties of network links.

LOBSTER proposes the design and deployment of a pilot European Infrastructure for accurate Internet monitoring. Based on passive monitoring, and capitalizing on previous experience, the LOBSTER infrastructure will be unique in Europe and among the only two similar infrastructures that exist in the world today. Thus LOBSTER will make a significant contribution to "IST 2.3.5 Research Networking test-beds", whose objective is "to integrate and validate ... state-of-the-art technology that is essential for preparing the future upgrades in the infrastructure deployed across Europe".

LOBSTER will provide a passive monitoring infrastructure across multiple administrative domains and

to succeed it is important to have a versatile and good anonymization mechanism. To solve this problem LOBSTER will create a configurable anonymization framework that allows administrators to configure what kind of anonymization that is needed on a per user basis.

For further information see the web-site: http://www.ist-lobster.org/

Contributions from Q2S:
UNINETT is a principal contractor and actively involved in the development process.

Postdoc Tønnes Brekne, PhD candidate Andre Årnes, and Arne Øslebø from UNINETT have analyzed how secure the existing anonymization techniques are and have come up with a new more secure method. This work [14] was presented at the Workshop on Privacy Enhancing Technologies 2005.

STOLAS- Switching Technologies for Optically Labelled Signals

STOLAS is a research project within the framework of the European Commission's IST programme. The project period is extended until end of April 2005.
The main objective of the STOLAS project is to improve the throughput of packet-switched networks by novel optical routing techniques based on stacked optical labels, supported by:

- theoretical evaluation of the network possibilities of optical label switching
- high-speed modulation and fast wavelength switching of widely tunable lasers
- development of multichannel 2R regenerators in a hybrid technology
- development of optical-label-controlled cross connect and add/drop nodes
- study of monitoring and control aspects of optical-label-routed networks
- construction of a limited network testbed and validation of the key functionalities in a laboratory testbed and in a field trial
- contributions to standardization processes

For further information see the web-site: http://www.ist-stolas.org/

The project was concluded on 1 May, 2005

Geant2

Geant2 is an EU project to build a fiber/lambda-based research network infrastructure in Europe going from 2005 to 2009. The Performance Measurement and Management activity (JRA1) is developing a multidomain network measurement architecture (perfSonar) in cooperation with Internet2. The goal is to allow advanced users to be able to perform new and study existing measurements to assess the quality of the network service end-to-end in Europe.

For further information see the web-site: http://www.geant2.net/

Contributions from Q2S:
UNINETT is taking part in developing passive measurements and flow analysis as well as the Topology Service and visualization clients. We have supervised master's theses in inter-domain measurement protocols and flow characterization to support this work. UNINETT is also taking part in JRA5 on the development of eduGain, a pan-European authentication infrastructure allowing easy global access by one ID.

Statement of accounts for the year 2005

Statement of accounts for the year 2005

(Amount in NOK 1000)

OPERATING INCOME	Notes	Accounts
Research Council	1	9 400
NTNU		7 237
UNINETT		2 231
Telenor		555
Other	2	115
Amount operating income		**19 538**

OPERATING COSTS		
Wages and social costs	3	12 752
Equipment and investments		1 488
Other operating costs		3 582
Amount operating costs		**18 745**

For allocation	**793**

Incoming balance 2005	17 731
Allocation 2005	793
Outgoing balance 2005	18 524

Note 1 In accordance with approved financing plan.

Note 2 Other are EU projects.

Note 3 The amount includes salary and social costs. Further specification, see below.

Amount teaching assistance by PhD students: NOK 946'

Specification for note 3.

PhD candidates	Persons	Labour year 2005
Sum labour year PhD	19	16,3

Postdoc	Persons	Labour year 2005
Sum labour year postdoc	12	6,6

Professor (contributions)	Persons	Labour year 2005
Sum labour year professors	6	2,2

Visiting professor	Persons	Labour year 2005
Sum labour visiting professor	2	0,3

Adm	Persons	Labour year 2005
Sum labour year adm	3	2,4

Miscellaneous activities

Supervision of master's theses

The supervision is of students at either the Department of Electronics and Telecommunications or at the Department of Telematics.

Alnes, Per Andreas. *Profitability in deploying WiMax Networks.* (Kure, Øivind/Lie, Harald Wium)

Alonso, Alberto Cuevas. *Universal Mobile Telecommunications System (UMTS) Security.* (Knapskog, Svein J/Knapskog, Svein J)

Andreassen, Erling J. *In Situ Measurement of Absorption Coefficients.* (Svensson, Peter/ Morset, Lars Henrik (Morset SD))

Bech, Rudi. *Intrusion Detection.* (Knapskog, Svein J/Knapskog, Svein J)

Berntzen, Jørn Andre Nese. *Internet connectivity and load balancing for a mobile ad hoc network.* (Kure, Øivind/Hafslund, Andreas (UNIK))

Brevik, Bård Kuvås. *Implementation and Test of Error Concealment Algorithms in Compressed Video.* (Perkis, Andrew /Zhang, Jijun)

Brørs, Kristin. *Effects of Combined Audio-visual Feedback on Binaural Reproduction.* (Svensson, Peter)

Carjaval, Borja Sacristan. *Simulation and Study of Virtual Audio Environments.* (Svensson, Peter)

Caro Cantero, Jose Maria. *Using game theory to model and analyze attacks.* (Knapskog, Svein J/Knapskog, Svein J)

Ellefsen, Jarle. *Line Array Loudspeaker System.* (Svensson, Peter)

Embretsen, Johan Herman. *Identification, valuation and threat analysis of information resources in organizations.* (Knapskog, Svein J/Knapskog, Svein J)

Fagerlie, Erling Andreas. *Implementing a load balancing algorithm for ad hoc networks in the J-sim simulator.* (Kure, Øivind/Kure, Øivind)

Faureng, Anne-Margrethe. *Binaural Strategies for Reverb Reduction Methods.* (Svensson, Peter/ Bjerkvik, Aslak)

Fossen, Espen Andre. *Principles of Internet Investigations: Basic Reconnaissance, Geopositioning and Public Information Sources.* (Knapskog, Svein J./Årnes, Andre)

Heimdal, Oddbjørn Gundersen. *Security aspects in open broadband networks (midlertidig).* (Knapskog, Svein J/Tøndel, Inger Anne)

Hodne, Erik. *Use of passive measurements to characterize network performance at the end systems.* (Emstad, Peder J./ Helland, Jon Kåre, Kvittem, Olav)

Karlsen, Johan Rise. *Edge Diffraction Modelling of Loudspeaker Enclosures.* (Svensson, Peter)

Koksrud, Andreas. *Transcoding Algorithms for H.264/AVC.* (Perkis, Andrew /Johansen, Stian)

Kveldstad, Eigil Lien. *Web Services's security.* (Knapskog, Svein J/ Knapskog, Svein J)

Larsen, Christian Nesheim. *Using honeypots to document the threats from the blackhat-community.* (Knapskog, Svein J/Årnes, Andre)

Longva, Peter. *Security aspects of RFID based e-payment.* (Knapskog, Svein J/Brede, Steinar; Jakobsson, Sune)

Michaelsen, Stian. *Measurements on IP-packets from video sources.* (Emstad, Peder J./Øslebø, Arne; Hillestad Odd Inge; Kvittem Olav; Hellan, Jon Kåre; Emstad, Peder J.)

Munkeby, Solveig. *Modeling of Experience.* (Perkis, Andrew /Hillestad, Odd Inge)

Nyre, Åsmund Nergård. *Increasing survivability by dynamic deployment of honeypots.* (Knapskog, Svein J/Jaatun; Martin Gilje)

Ordaz, Oscar Peidro. *Dependability evalutation of an IP backbone network.* (Helvik, Bjarne E./Gan, Qitao)

Presenting new ideas: Jon Kåre Hellan.

Palacios, Sandra Rodiño. *Virtual Environment and Ensemble Performance, a Study of Perception.* (Svensson, Peter)

Pedersen, Trond Iver R. *Optimized Loudspeaker for Room Acoustic Measurements Based on the Sine-swept.* (Svensson, Peter/Morset, Lars Henrik (Morset SD))

Perez, Pablo Abella. *Basis for Modelling Intrusion Detection with coloured Petri Nets.* (Knapskog, Svein J/Knapskog, Svein J)

Peveri, Magnus. *Implementation of the MOLSR protocol in J-SIM.* (Kure, Øivind/Kure, Øivind)

Sandvik, Stine Iren. *Statnett's future network management system: Transferring tasks from CA Unicenter to HP Openview.* (Helvik, Bjarne E./Sharifabadi, Kamran)

Simonsen, Audun. *Evaluation of message transfer on concatenated tactical links.* (Kure, Øivind/Jodalen, Vivianne)

Sjølingstad, Torgeir Berge. *Distrubution of Sound Fields in Rooms at Low Frequencies.* (Svensson, Peter/ Morset, Lars Henrik (Morset SD))

Solsvik, Fredrik Hildershavn. *Measurements on IP-packets from video sources.* (Emstad, Peder J./Øslebø, Arne; Hillestad Odd Inge; Kvittem Olav; Hellan, Jon Kåre; Emstad, Peder J.)

Solvang, Audun. *Removal of Spatial Irrelevancy in 3D Audio Utilizing Ambisonics and the Continuity Illusion.* (Svensson, Peter)

Tronslin, Anne. *Sikkerhetstjenester for egensikring av nett.* (Knapskog, Svein J/McInally, Thomas (Statoil))

Trøite, Torgeir. *Local Quality of Service Reservations.* (Kure, Øivind/Kvittem, Olav)

Tveit, Olav Tarkjell. *Aspects of realization and evaluation of a price model in which quality of service is provided by contracted reservation of capacity.* (Helvik, Bjarne E./Øverby, Harald)

Wikshåland, Sigbjørn. *Use of passive measurements to characterize network performance at the end systems.* (Emstad, Peder J./ Helland, Jon Kåre; Kvittem, Olav)

Refereeing activities

Professor Peder J. Emstad
Technical Program Committee member of:
HET-NETs '05, Ilkley, U.K.
WWIC 2005, Xanthi, Greece.
2nd EuroNGI Conference, Valencia, Spain.

PhD evaluation committee:
Niklas Widell. *Active Resource Management in Middleware and Service-oriented Architectures.* Lund Institute of Technology, Lund University, Sweden.

Professor Bjarne E. Helvik
Member of:
Program committee Globecom 2005, St. Louis, Missouri, USA

Reviewer for:
IEEE/ACM Transactions on Networking. LANMAN 2005, Chania, Crete, Greece. NIK 2005, Bergen Norway. Het-Net05, Ilkley, U.K.
2nd EuroNGI Conference, Valencia, Spain.

PhD evaluation committee:
Mohammad Zib Beiroumi; *High Availabile and Fault Tolerant Mobile Communications Infrastructure.*
Technical University of Denmark. Lyngby, Denmark.

Professor Svein J. Knapskog
Technical Program Committee member of:
SEC2006, Karlstad, Sweden.
CNIS2005, Phoenix, USA.
NORDSEC2005, Tartu, Estonia.
Steering committee NORDSEC.

PhD evaluation committee:
Henric Johnsson; *Towards Adjustable Lightweight Authentication for Network*

Access Control, Blekinge Institute of Technology, Karlskruna, Sweden.

Stefan Lindskog; *Modeling and Tuning Security from a Quality of Service Perspective;* Chalmers University of Technology, Göteborg, Sweden.

Rajan Shankaran; *Security Issues in Mobile IP and Mobile Ad Hoc Networks.* University of Western Sydney, Penrich South, Australia

Reviewer for:
The Cor Bayen Award 2005.
CCN2005, Marina del Rey, CA, USA.
ACISP05, Brisbane, Australia.
AISW-NetSec 2006, Hobart, Tasmania.
IRMA2006, Washington, D.C., USA.
NIK2005, Bergen.

Evaluator for Swedish Professor position:
Sead Muftic, KTH.

Professor Øivind Kure
Member of:
Program Committee WSNET-05, Oslo, Norway.

Reviewer for:
Journal on Computer Communications
IEEE Communications Magazine.

PhD evaluation committee:
Fredrik Davik, *Fairness aspects of buffer-insertion rings in general and resilient packet rings in particular*, University of Oslo, Department of Information

Professor Andrew Perkis
Member of:
Program Committee ICETE 2005, Reading, UK.
Program committee of Eurescom Summit 2005 - Ubiquitous Services and Applications - Exploiting the Potential, 27 – 29 April 2005, Heidelberg, Germany www.eurescom.de/summit2005.
UNINETTS fagråd.

Reviewer for:
IEEE Transactions on Image Processing.
ETRI Technical Journal.
EUSIPCO 2005, Antalya, Turkey.

Professor Peter Svensson
Associate Editor for:
Acta Acustica united with Acustica in the field of Computational Acoustics.

Member of:
The Board of the Acoustical Society of Norway (Chairman).

Reviewer for:
Acta Acustica united with Acustica.
IEEE Transactions on Speech and Audio Processing.
Journal of the Acoustical Soceity of America.

PhD evaluation committee:
Guillaume Potard, 3D-Audio Object Oriented Coding, University of Wollongoong.

Evaluator for Swedish "Docent" position (Associate Professor):
Jens Forssen, Department of Applied Acoustics, Chalmers University of Technology.

Postdoc Ajit S. Bopardikar
Reviewer for:
IEEE Transactions on Image Processing.
Journal of Electronic Imaging.

Markus Fidler
Technical Program Committee member of:
IEEE Globecom 2005, St. Louis, USA.
IEEE/IFIP AGNM 2005, Barcelona, Spain.

Reviewer for:
Elsevier Performance Evaluation.

Postdoc Danilo Gligoroski
Reviewer for:
IEE Proceedings - Circuits, Devices & Systems. IEEE ISCAS-2006, Island of Kos, Greece.

Postdoc Esa Hyytiä
Reviewer for:
Optical Switching and Networking (OSN) Magazine.

Postdoc Yuming Jiang
Technical Program Committee member of:
IEEE Globecom 2005, St. Louis, USA.
IEEE ISWCS 2005, Siena, Italy.
IFIP WOCN 2005, Dubai, UAE.

Vice Chair of:
IEEE Globecom 2005 – General Conference Symposium, St. Louis, USA.

Reviewer for:
IEEE/ACM Transactions on Networking.
IEEE Journal on Selected Areas in Communications.
IEEE Journal of Lightwave Technology.
IEEE Communication Letters.
EURASIP Journal on Wireless Communications and Networking.
International Journal on Wireless & Optical Communications.
IEEE ICC 2005, Seoul, South Korea.
IEEE Globecom 2005, St. Louis, USA.
IEEE WCNC 2005, New Orleans, USA.
IEEE HPSR 2005, Hong Kong.
IEEE ISWCS 2005, Siena, Italy.
IFIP WOCN 2005, Dubai, UAE.

Postdoc Otto Wittner
Reviewer for:
NIK 2005, Bergen.
Globecom 2005, St. Louis, Missouri, USA.

UNINETT researcher Stig Venas
Co-Chair of:
IETF working group on Dynamic Host Configuration (DHC).

Dissemination

Seminars and colloquia at Q2S

Seminar Thursday 9 June, 0830-1600 at NTNU
Internal working seminar consolidating research efforts within the Centre. Ongoing research was presented, and new ideas discussed.

Lunch seminar series
Internal seminar series with informal presentations, talks, discussions, 16 in total in 2005.

Lectures in the "Q2S Colloquium" series 2005
The series has been managed and organized by Otto Wittner.

Fidler, Markus, Q2S, NTNU. *A parameter-based admission control for aggregate scheduling networks*. 24 January 2005.

Solvang, Audun, Q2S. *Removal of spatial irrelevancy in 3D audio utilizing Ambisonic and the continuity illusion.* 11 February 2005.

Øverby, Harald, Dep. of Telematics. *Packet loss rate- and delay jitter differentiation in asynchronous optical packet switched networks.* 1 April 2005.

Meling, Hein, Univ. of Stavanger. *An Approach to Experimentally Obtain Service Dependability Characteristics of the Jgroup/ARM system.* 11 April 2005.

Marchand, Corine, Q2S. *Adaptation of Distributed Algorithms in Highly Dynamic Environments, Experimentation in Wireless Ad-hoc Networks.* 29 April 2005.

Bormans, Jan, IMEC, Belgium. *Quality of Experience for Personal Multimedia Communication Devices.* 26 May 2005.

Jiang, Yuming, Q2S, NTNU. *Stochastic Network Calculus and its Impact.* 27 May 2005.

Sallhammar, Karin, Q2S. *Incorporating Attacker Behavior in Stochastic Models of Security.* 1 June 2005.

Larsen, Christian, Dept. of . *Honeypots*. 20 June 2005.

Hillestad, Odd Inge, Q2S, NTNU. *Performance Evaluation of Multimedia Services over IP networks*. 5 August 2005.

Gligoroski, Danilo, Q2S. *Using quasigroups for PRNGs, Cryptographic Hash Functions, Stream Ciphers, One-way Functions, One-way Permutations, and Random Error Correcting Codes.* 10 August 2005.

Alvestrand, Harald, IETF. *The IETF vision of the Internet of the future.* 19 August 2005.

Austreim, Erling, Q2S. *Rate Control of Traffic Sources Based on Level Crossing in a Congested Buffer.* 24 August 2005.

Aune, Frank, UNINETT. *An introduction to the IPPM framework.* 26 August 2005.

Hyytiä, Esa, Q2S, NTNU. *Random Waypoint Mobility Model and its Applications.* 2 September 2005.

Janczewski, Lech, Q2S.NTNU. *Research Proposals.* 9 September 2005.

Lundqvist, Henrik, ERCIM, Q2S NTNU. *An evolutionary approach to quality of service.* 30 September 2005.

Wittner, Otto, Q2S, NTNU. *Internet Failure Protection using Hamiltonian p-Cycles found by Ant-like Agents.* 14 October 2005.

Siew, Chee-Kheong (David), ICIS Singapore. *Open issues in QoS provisioning.* 26 October 2005.

Hoerdt, Mickaël, Q2S, NTNU. *Source Specific Multicast (SSM): Toward an extensible group communication service.* 4 November 2005.

Fidler, Markus, Q2S, NTNU. *Probabilistic Network Calculus: A System Theory for the Internet.* 16 November 2005.

Paquereau, Laurent, Q2S. *Combining performance monitoring and location data in wireless networks.* 18 November 2005.

Lie, Arne, NTNU Dept. of Telematics. *Trace driven simulation of rate adaptive MPEG-4 video.* 2 December 2005.

Talks at conferences, seminars and meetings

[T1] Aarnes, André. *Anonymization of IP monitoring data.* Spring School in Security, Marseille, 2005.

[T2] Aarnes, André. *Anonymization of passive monitoring data.* UNINETT and Q2S Network Measurement Seminar, 2005.

[T3] Aarnes, André. *Distribuert nettverks-overvåkning.* Seminar, Forskningsmessige utfordringer innen dataetterforskning og elektroniske spor, Nye Kripos, 2005.

[T4] Aarnes, André. *Elektroniske spor og etterforskning på Internett.* Medlemsmøte Dataforeningen Trøndelag "Jakten på elektroniske spor", Faggruppen Informasjonssikkerhet, 2005.

[T5] Aarnes, André. *LOBSTER: Large Scale Monitoring of Broadband Internet Infrastructure.* Sikkerhet og sårbarhet 2005, Dataforeningen Trønderlag.

[T6] Gan, Qitao; Helvik, Bjarne Emil. *Limiting the State Space Explosion as Taking Dynamic Issues into Account in Network Modelling and Analysis.* The second Norwegian Network Research Seminar, Fornebu, Norway, 27-28 October 2005.

[T7] Hellerud, Erik; Voldhaug, Jan Erik; Svensson, Peter. *Perceptually Optimized Error Protection for Audio Streaming over IP Networks.* Norwegian Network Research Seminar 2005, Oslo, Norway, October 2005.

[T8] Hillestad, Odd Inge; Jetlund, Ola; Perkis, Andrew. *Error Robustness Evaluation of High Quality H.264/avc Broadcast Services over RTP/IP using Network Emulation.* Norwegian Network Research Seminar, 27-28 October 2005.

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