ENCLOSURE A:

In-Depth Description of the "IKT-2010" Project Extension Proposal

Bandwidth-Efficient and Adaptive Transmission Schemes for Wireless Multimedia Communications, Part II*

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1 Introduction and motivation

The IKT-2010 project BEATS (Bandwidth-Efficient and Adaptive Transmission Schemes for Wireless Multimedia Communications) was launched on November 1, 2000, having been granted 3.6 MNOK from NFR. As witnessed by the project's WWW page (`http://www.tele.ntnu.no/projects/beats`), the project is proceeding according to plan and has already produced scientific results which have been published internationally.

However, despite the excellent reviews received by the original BEATS proposal from the international expert evaluators, the sum ultimately granted from NFR represented less than 40 % of what the project participants originally applied for, thus necessating a severe reduction of the intended project scope. The number of dr. ing. students was reduced from 3 to 1, our full-time senior researcher Kjell J. Hole could be funded for only three years instead of four as originally intended, and the means for scientific travel, administration, equipment etc. were also reduced.

The present application represents an updated followup proposal, where the BEATS project scope is brought back to its original intentions. However, as will be seen there are also some (relatively minor) changes in the focus of the project. This is natural, since research goals should not be entirely constant over time, but be adjusted according to the results and insights already achieved, to the trends seen in the international research community, and to upcoming fields of interest for the researchers involved.

1.1 Motivation

An acceptable Quality of Service (QoS) for future multimedia services (e.g. high audio and video quality, high reliability, strict real-time constraints) can only be achieved by realizing much *higher information rates* than those available in today's wireless systems. At the same time, *bandwidth* is becoming an ever-scarcer resource as the number of systems, users, and services increase. If real-time, high-quality multimedia services are to be supported in future wireless systems, there is thus a need for novel transmission schemes–i.e. compression, modulation, error control, and access techniques. The schemes must provide bandwidth-efficient, robust communication with low delay, supporting multiple users on broadband wireless channels. Since *time-varying*[1] *channel conditions*–and thus *time-varying channel capacity*–is an important feature of wireless and mobile communication systems, future such systems should exhibit a high degree of *adaptivity* on many levels in order to reach these goals [1, 2]. Examples of such adaptivity are: information rate adaption, power control, code adaption, bandwidth adaption, antenna adaption, protocol adaption.

*Any further questions regarding this proposal may be e-mailed to *G. E. Øien* at *oien@tele.ntnu.no*.

[1] And, to a certain extent, unpredictable.

2 Overall goals and project description

This proposal addresses some critical components in future wireless broadband communication systems, with particular emphasis on important signal processing aspects. The proposal is an extension of the ongoing NFR project BEATS. We focus on the development of *bandwidth-efficient and adaptive transmission schemes for broadband wireless multimedia communications*.

2.1 Scope

The main research ideas covered by the present proposal for project extension are:

- **Design of joint source-channel coding schemes** for optimal efficiency and robustness, particularly in uplink transmission of source-specific, high-capacity parts of multimedia data streams, e.g. video.
- **Design of transcoding schemes in heterogenous systems**, for efficient digitization and coding of jointly source-channel coded signals, e.g. after decoding at the basestation, before further transmission takes place in the system.
- **Design and study of optimized Orthogonal Frequency Division Multiplexing (OFDM) schemes**. We view OFDM as the most suitable modulation/access technology for the currently most promising schemes for bandwidth-efficient digital communications on broadband radio channels (see the original BEATS application for further details).
- **Estimation and analysis of wireless communication channels**, in order to establish realistic benchmarks for practical system performance.

Research on the above topics will complement and enhance the ongoing BEATS research on *adaptive channel coding and modulation* in broadband systems and on *signal processing for smart antennas*.[2]

The figure in Enclosure B depicts how these critical components fit into a proposed architecture for a future wireless multimedia communications terminal (transmitter side[3]). The terminal is highly reconfigurable and may accept multiple analog (e.g. video, images, audio) as well as digital (e.g. text, computer graphics, data files) inputs simultaneously. Working within the general structure of this figure, we aim at giving important contributions to wireless transmission technology supporting a wide range of delay-sensitive, bandwidth-intensive multimedia services.[4]

As seen from the figure, in multimedia communications it is important to distinguish between two fundamentally different classes of source signals:

1. Signals from discrete sources (text, programs, etc.)
2. Signals from analog sources (speech, audio, video)

We advocate that the discrepancies between these two classes of signals call for different communication philosophies. Discrete source signals require transparent transmission, which is sought efficiently handled by the ATCM branches in the figure in Enclosure B. Analog sources, on the other hand, can never be exactly rendered through a transmission system, because it will either suffer from noise when transmitted as an analog entity, or will be contaminated by quantization noise and possibly also channel errors when transmitted digitally. Fortunately, human receivers tolerate noisy conditions for analog signals, partly because the human perception is not perfect, but also because in many situations it is more important to get the message through than to bring a perfect replica of the given signal. For this purpose, joint source-channel coding is an efficient tool.

Although very focused from a technology point of view, our ideas are generic in the sense that they might be exploited both in fixed wireless access and satellite communications, as well as in fourth generation (4G) mobile cellular systems. Also, short-distance wireless communication at much larger rates than those supported by the current Bluetooth standard may be enabled.

A truly optimal system design must start at the physical layer. However, insight into user and market needs, overall system specifications, and network protocols and architectures is of course also important when devising transmission schemes for future communications systems. Some keywords are: foreseen services, demands on QoS, available frequency bands, fundamental technological limitations, standardization issues, and system costs. We will evaluate our ideas also with such perspectives in mind.

[2] Note that the activity on **digital baseband predistortion of nonlinear RF amplifiers** described in the original BEATS application is *not* covered by this extension.

[3] The receiver side will, of course, be the inverse of this structure.

[4] The components covered by our research are depicted as boxes drawn using thick (bold) lines in the figure.

2.2 Personnel

The collaborating BEATS researchers are:

- Professor/Senior Research Scientist, dr. scient. Kjell J. Hole, Coding Theory Group, University of Bergen (UiB)/Signal Processing Group, Norwegian University of Science and Technology (NTNU)
- Professor, dr. ing. Nils Holte, Signal Processing Group, NTNU
- Professor, dr. ing. Tor A. Ramstad, Signal Processing Group, NTNU
- Professor, dr. ing. Geir E. Øien, Signal Processing Group, NTNU (project manager)

Hole and Øien, together with dr. ing. student Bengt Holter, make up the current BEATS research group, and their activities within adaptive coding and modulation and signal processing for smart antennas will continue as already scheduled in the BEATS research plan. Holte and Ramstad completed the team of senior researchers behind the original BEATS application, but their proposed activities ultimately had to be postponed due to the budget cut experienced. Much of the present extension therefore covers research activities to be led by Holte and Ramstad.

Professor Holte has worked extensively on various aspects of OFDM for broadband channels both in wired and wireless systems. Professor Ramstad has contributed significantly to joint source-channel coding research as well as doing fundamental research on signal processing algorithms, digital filter banks, and compression of multimedia information. Filter banks are key components not only in OFDM, but also in realization of joint source-channel coding.

For an overview of Holte and Ramstad's previous publication activities within fields relevant for this proposal, we refer to Enclosure D, which contains a complete list of their relevant publications sorted by topic (complete up to May 2001).

2.3 Organization, dissemination, costs, and industrial support

The project is administrated by The Department of Telecommunications at NTNU. The research covered by the present extension proposal research will be done in cooperation with *four new PhD students*, as well as in interaction with several external collaborators and four dr.ing. students already working at NTNU.[5] For one of the proposed dr. ing. positions, we already have a qualified and interested candidate.

Note that, due to the fact that NTNU's siv.ing. education now lasts for 5 years compared to 4 $\frac{1}{2}$ as in previous years, there will be no siv.ing. candidates graduated in December 2001—the graduation is postponed until May/June 2002. Therefore we do not view it as very realistic to obtain qualified applicants for all four dr.ing. positions from January 1, 2002. In the application form/budget we have therefore suggested that two candidates may be able to start from January 1, whereas two can be in place from July 1, 2002. However, should we be so lucky as to obtain qualified applicants for all four positions who are able to start from January 1, we will apply to NFR for a rescheduling of our funding to take this into account.

Another potential model for improving the recruiting to dr.ing. studies which is currently being pursued at NTNU is the socalled "doctoral school." This implies that students are allowed to start their doctoral education after 4 years of siv.ing. studies (being paid 50 % of ordinary doctoral student wages during the first two years, followed by 100 % pay for two years). They will then finish their siv.ing. thesis after 6 years instead of 5, while working part-time towards a doctoral degree (taking doctoral courses, doing initial research work, and having teaching duties) for the last two years. If this model is developed further at our faculty this may also be one way of recruiting students to BEATS, in which case our budgeting will be changed (not as far as overall amounts are concerned, but as far as the cost scheduling is concerned).

We plan to assign research duties for this project extension (in addition to the duties described in the original BEATS application) as follows:

- Dr. ing. student 1: **Joint source-channel coding** (Supervisor: Prof. Tor A. Ramstad).
  - Candidate: None particular at the moment.
- Dr. ing. student 2: **Transcoding of signals in heterogeneous systems** (Supervisor: Prof. Tor A. Ramstad).
  - Candidate: None particular at the moment.
- Dr. ing. student 3: **Optimization of OFDM schemes** (Supervisor: Professor Nils Holte).

[5] These four students are: Bengt Holter (financed by BEATS), Till Halbach (financed by NTNU; thesis on robust image communications), Anna Kim (financed by NTNU; thesis on joint source-channel coding), and Ola Jetlund (Telenor-financed project; thesis on channel coding for wireless applications).

- Candidate: None particular at the moment.

- Dr. ing. student 4: **Estimation and analysis of wireless channels** (Supervisors: Professor Geir E. Øien and Research Scientist, dr.ing. Lars M. Lundheim (SINTEF Telecom and Informatics)).
  - Candidate: Duc Van Duong (see attached CV for qualifications).

- Nils Holte: **Optimization of OFDM schemes.**

- Tor A. Ramstad: **Joint source-channel coding, transcoding, OFDM, and filter banks.**

- Geir E.Øien: **Capacity analysis, wireless channel estimation, information theory**

- Kjell J. Hole: **Adaptive wireless OFDM systems**

Taking the wireless terminal architecture shown in Enclosure B as our starting point, we aim to integrate all researchers' activities through project seminars and lectures, so that all participants are able to see how their own research has a place in the overall system concept and how it fits in with other research performed in the project. Thus all participants will become knowledgeable about all the components studied in the project, as well as the motivation for studying them. This will also enable cross-fertilization of research ideas between the different sub-topics, and, ultimately, publications where ideas from the different sub-topics are united.

We also expect to supervise about 15 final year siv.ing. students each year within the scope of the project.

The research results will continue to be published in acknowledged international journals and in high-quality (refereed) conference proceedings (internationally and nationally). Furthermore we plan to extend our international activity by actively participating in Working Group 1 "Radio System Aspects" within the newly initiated COST Action 273 *Towards Mobile Broadband Multimedia Networks* (see http://www.lx.it.pt/cost273). *Telenor FoU* are active in COST 273, and we therefore also foresee interesting possibilities for NTNU-Telenor cooperation within this framework.

As has been the case since January 2001, the BEATS homepage at http://www.tele.ntnu.no/projects/beats will continue to present and document all our activities. We refer to point 13 "Project Publication Plan" in the standard NFR application form as well as to Sections 3–6 for further details on our plans for publication.

Our project proposal has been prepared after discussions of future technological needs with relevant industry, and a Letter of Support from *Nera Research* is attached to this proposal (see enclosure F). The total cost of the current project extension is approximately NOK 9.150.000,- over 4 $\frac{1}{2}$ years, of which we apply for NOK 7.500.000,- from The Norwegian Research Council (NFR). A detailed cost plan is submitted in Enclosure C.

3 Joint source-channel coding

Joint source-channel coding[6] is a promising strategy for bandwidth-efficient, reliable wireless communications. Joint source-channel coding may typically be envisaged used e.g. on the uplink in mobile systems. By jointly exploiting the properties of both information source and communication channel, more robust and bandwidth-efficient transmission schemes may be devised, compared to "separable" coding and modulation schemes. I.e., such schemes are specifically optimized with respect to a certain source (signal) model as well as a certain channel model. The challenge lies in *design and optimization* of practical joint source-channel coder structures coming close to fulfilling the theoretical potential.

3.1 Background and state-of-the-art

Traditional communication systems are based on *Shannon's separation theorem* [3] which states that overall optimality can be obtained by cascading an optimal source coder which removes all redundancy and adds imperceivable or acceptable signal noise, and an optimal channel coder which adds redundant bits for explicit error protection. However, optimizing both coders generally require infinite complexity and infinite delay. Thus, a more interesting problem is to find optimal systems within *complexity and delay constraints*.

Another problem with Shannon's theorem, even if the optimal solution is found, is that it does not guarantee the performance when the channel signal-to-noise ratio (CSNR) decreases slightly below the level for which the system was designed. In wireless communications, the CSNR vary with time, making this a practical weakness. There is a strong need for *error resilience*, i.e. graceful performance degradation for deteriorating channels.

To illustrate the shortcomings of traditional systems, let us consider pulse code modulation (PCM), where the different bits have different significance. An error in the most significant bit may in some cases lead to disastrous errors, while an error in the least significant bit is hardly noticeable in comparison. To cope with this problem many schemes have been suggested. The most obvious solution is unequal error protection (UEP) for

[6] Or *joint source coding-modulation.*

the different source bits to balance the significance of each channel bit. Further improvements can be obtained by redesigning the quantizers and the bit allocation, and in the case of vector quantization, a proper index assignment may improve the performance, especially in combination with multilevel modulation signal sets. A good overview with new contributions can be found in [4]. Important references are [5, 6, 7, 8, 9, 10, 12, 11, 13].

The research team led by Prof. Ramstad at NTNU has throughout the last decade performed some pioneering work which is very innovative, with results closer to theoretical bounds than other known works in the literature. This pertains to performance under complexity and delay constraints. The obtained robustness towards deteriorating channel conditions is quite remarkable and superior to traditional digital systems for image and video communications (See e. g. [14, 15, 16, 17, 18, 19], and Enclosures D, E for further references). The developed methods are based on time-discrete but amplitude-continuous signal parameters and channel symbols. The methods can be viewed as "continuous vector quantization," i.e., approximation of source vectors by means of points in continuous subsets of some multidimensional space, followed by an "index assignment," which in this case means mapping the VQ "index" into the space of modulation symbols. These two stages can be viewed as a single step using dimension-changing nonlinear mappings which combine N source parameters into K channel parameters. When assuming ideal sampling and an ideal Nyquist channel, compression can be defined as $\eta = K/N$, i.e. the bandwidth reduction factor. In practice several mappings are used, in which case $\eta = \mathrm{E}\{K/N\}$, where $\mathrm{E}\{\ \}$ is the expectation operator.

A dimension-reducing mapping, which performs bandwidth compression, will always introduce approximation noise. A dimension-*expanding* mapping is, on the other hand, noiseless. It is introduced to better cope with channel noise, i.e for a purpose similar to traditional channel coding.

3.2 Planned research

A complete source-channel coding system will typically consist of a unit for signal decorrelation, such as a filter bank or a predictor, the mapping devices–usually several of them to adapt to the importance of the different source parameters, and the modulation unit, which e.g. maps the channel parameters to the I- and Q-components in an OFDM system. We refer to Enclosure B for an illustration of how such a system may be fit into the structure of a general multimedia terminal.

The research will be directed towards image and video communications over band- and power-limited channels, in particular those exhibiting non-flat fading.[7] Both signal decomposition methods and development of sets of optimal maps will be considered for different applications and signal statistics.

In the terminal structure in Enclosure B, the signal decomposition and the OFDM synthesis can be implemented using filter banks or linear transforms. The decomposition is done in a bandsplitting analysis filter bank, while the OFDM block performs component merging through a synthesis filter bank. The operations inbetween perform bandwidth change through the nonlinear mappings, and further signal-to-channel adaption by assigning appropriate channel bands to the different source parameters, depending on the importance of the source parameters and on the channel conditions. User-required quality in terms of bandwidth and signal quality can be guaranteed by such a resource allocation mechanism.

A final sub-goal is to integrate data and signal communications through reconfigurable channel constellations. In the figure in Enclosure B such reconfiguration is the task of both the "Adaptive OFDM channel assigner" and the ATCM encoders.

Overall, the expected outcome of the joint source-channel coding research is

1. Algorithms for image and video compression which lend themselves to joint source-channel coding.
2. Nonlinear mappings for dimension reduction and expansion, designed for various source statistics, QoS demands, and channel conditions.
3. Complete simulation programs integrated with the OFDM system.
4. Publication of at least 3 international journal papers, and 6 papers in refereed conference proceedings.

4 Transcoding for heterogenous communication systems

4.1 Background and state-of-the-art

The exploding use of telecommucations is mainly due to the availability of affordable services for large groups of people. For cable transmission the advent of optical fibres and the available technology for exploiting them in communications have provided almost unlimited bandwidth. Possibilities for broadband mobile services on a large scale are, on the other hand, quite limited. This is due to the limited electromagnetic spectrum. To expoit the spectrum fully, several measures should be taken. Antenna adaptivity, optimal and adaptive modulation and coding methods are examples of some such measures. The more general conclusion is that we

[7] I.e., the channels must in this respect be regarded as broadband, even though they are bandlimited.

face two very different scenarios when considering fixed connections and radio-based mobile connections. For point-to-point communication where one or both communicators are mobile, the bottleneck is undoubtably the mobile connection(s). Special measures must therefore be taken to obtain optimal solutions for the mobile radio connection. For communicating *analog* source signals, lossy compression and joint source-channel coding are necessary ingredients which promise to make video as well as voice services available on small mobile devices. The UMTS generation may have a limited availability in this respect, but to really offer interesting quality, further steps must be taken. This will necessitate the use of joint source-channel coding in its more radical form (see previous section).

However, in the high-capacity fixed part of the network the traditional "separation principle" philosophy will prevail, in the sense that a digitized information stream will be transmitted. This stream should be fully error protected, thus making the network transparent. A natural solution is thus to make the very best effort for the mobile channel—implying joint source-channel coding and thus a time-discrete but amplitude-continuous format—and then do re-coding (*transcoding*) of the received analog information in the base station, to produce an efficient digital representation for further transmission.

4.2 Planned research

It is important to study transcoding in the proposed project particularly because it may lead to new optimality constraints on the joint source-channel coding algorithms. In other words, optimality for the complete system may call for joint design (co-design) of the source-channel coding method and the transcoder. Furthermore, complexity and delay issues may also become important.

In the proposed joint source-channel coding systems, nonlinear mappings replace the quantizers used in standard digital compression systems, and provide the baseband channel representation. The transcoding will involve digitization of these channel representations. This process is not straightforward to perform in an optimal way, as the nonlinear mappings already minimize the total noise including mapping approximations and channel noise. The digital bitstream should provide low extra quantization error with a minimum total number of bits. If optimality for the digital bitstream were not so important, then one could simplify the problem by performing a "one-way" transcoder optimization, where the joint source-channel coder representation is given, and a best possible bitstream is constructed from this. However, it is a much more challenging task to produce first an optimal joint source-channel representation, and subsequently an optimal digital representation, such that the end result is as good as what would be possible if the original signal were coded "from scratch". Alternative solutions will be studied and simulated.

Overall, the expected outcome of the transcoding research is

1. Detailed proposals for transcoding algorithms which are suitable for combining with our joint source-channel coding methods.

2. Analysis and simulation of a complete system including joint source-channel coding and transcoding, including analysis and attempted minization of the quality loss induced by the transcoding.

3. Publication of at least 2 international journal papers, and 4 papers in refereed conference proceedings.

5 Optimization of OFDM systems

5.1 Background and state-of-the-art

Today, all mobile and low cost wireless communication systems exploit the frequency spectrum in a relatively inefficient way. Existing single carrier systems use constant envelope modulation limited to a bandwidth efficiency of 2 bit/s/Hz. CDMA systems use BPSK modulation, which means a maximum bandwidth efficiency of approximately 1 bit/s/Hz. Efficient utilisation of the bandwidth can be obtained by using OFDM. OFDM will obtain bandwidth efficiencies close to the theoretical limits by matching different types higher order modulation to the conditions in each frequency band. A tutorial introduction to OFDM is given for instance in [21].

The majority of OFDM systems use square pulseshapes, and channel dispersion is compensated by using a cyclic prefix as described in [22]. This solution has two major drawbacks. Primarily, square pulses give large sidelobes in the frequency domain, which causes un necessary out of band radiation for single user systems. Large sidelobes in the frequency domain prevents an efficient system for the uplink direction of a multiuser system. Secondly, a cyclic prefix means adding redundant information and hence a reduction of the effective bitrate. These drawbacks can be avoided by using pulseshaping in OFDM [23, 25]. Both filter design [25] and efficient implementation of systems with pulseshaping [23] has been solved for distortionless channels. For channels with dispersion, a cyclic prefix can not be used, and other alternative methods must be found for compensation dispersive channels. One open problem is to design optimum pulseshapes which reduce the sensitivity with respect to the dispersion of the channel.

One Ph.D. thesis [24, 25, 26, 27, 28] and several diploma theses [29, 30] have been carried out on OFDM systems with pulseshaping under supervision of Professor Nils Holte. The plan is to continue this research with a Ph.D. student, putting particular emphasis on channels with dispersion.

Both the activity on adaptive modulation and the activity on joint source and channel coding within this project are based upon using OFDM. Hence, further work on the basics of OFDM systems will be beneficial for the entire project.

5.2 Planned research

The plan is to find efficient methods to compensate for channel dispersion in OFDM systems with pulseshaping. One of the main strategies will be to find pulseshapes which minimise the sensitivity to channel dispersion and also have desired spectral properties. This is mainly a theoretical optimisation problem, but can also be investigated by simulation. Many aspects of OFDM systems are too complicated to be handled by theoretical analysis, so that efficient simulation tools are important. Our current simulation programs for OFDM are limited to a few subchannels. It is our plan to develop an efficient simulation program based upon MATLAB which is able to handle several hundreds of subchannels. The planned outcome of our research can be summarised as follows:

1. Methods for finding optimum pulseshapes in OFDM for channels with dispersion.
2. Efficient simulation programs for OFDM systems based on MATLAB.
3. Minimum 2 journal papers and 3 papers at international conferences.

6 Estimation and analysis of wireless channels

6.1 Background and own contributions

The success of adaptive transmission is ultimately dependent on good knowledge of the channel dynamics at the transmitter. One important research challenge is therefore to extract the best possible estimate of the overall channel conditions at any given time, given a realistic statistical model of the channel variations in any subchannel. This channel state estimate must then be used to adapt the transmitter in the best possible way (i.e., such that overall information throughput is maximized, subject to the condition that the QoS demands are still met). For this purpose, optimal channel estimation and prediction techniques will be investigated and analyzed. There will always be a nonzero degradation of system performance compared to the ideal case (i.e., that of perfect channel knowledge) when only imperfect channel state information is at hand [31]. We will focus on making this degradation as small and controlled as possible. Analysis of the available channel capacity under various practical system constraints is also a problem of great interest, although also of great complexity. Some preliminary work has already been done by Øien and Hole in these areas [32, 33].[8]

6.2 Planned research

This research will build on existing information and estimation theoretic results for idealized narrowband channel models, but attempt to take more into account channel estimation errors, practical return channel delays, QoS demands on transmission delay, more complex channel models etc. Methods and models for efficient channel prediction and estimation will be investigated. The ultimate aim is the design and analysis of such methods in an OFDM setting where antenna diversity is used. Particular emphasis will be put on providing a rate-adaptive transmitters with the information they need to choose codes and modulation constellations, and to control power assignments, in order to maximize throughput subject to BER and delay constraints.

We expect to publish at least 2 journal papers and 4 papers in refereed international conference proceedings on topics related to channel estimation and analysis.

7 Collaborations and possible future follow-up activities

7.1 Present collaborations

As stated in the enclosed Letter of Support (see Enclosure F), Nera Research possesses a strong interest in our research. Holte, Øien, and Hole have all been acting as consultants for Nera Research. One outcome of the Nera collaboration is a joint research paper by Øien, Hole, and researchers *Bjarne Risløw* and *Pål Orten* at Nera Research (see the BEATS homepage). We will continue the contact with Nera throughout the project.

[8] Interestingly, it also turns out that some of the results from capacity analysis for wireless channels are directly transferrable to pair cable systems limited by crosstalk. Prof. Holte is therefore currently extending our research on wireless channels to include capacity analysis of pair cable systems.

Risløw and Orten are already listed as project collaborators, and are now joined as such by research scientists *Per-Hjalmar Lehne* from Telenor FoU, and *Lars Lundheim* from SINTEF Telecom and Informatics. We foresee interaction with Telenor FoU particularly through COST Action 273, where Lehne serves as vice-chairman, whereas Dr. Lundheim will co-supervise the dr.ing. student working on channel estimation together with Prof. Øien.

Internationally, we have a long history of contact with some of the leading research institutions within signal processing, digital transmission, source and channel coding, and wireless communications—e.g., Washington State University (Prof. Thomas R. Fischer), University of San Diego (Prof. Paul H. Siegel), University of Santa Barbara (Prof. Sanjit K. Mitra), California Institute of Technology (Prof. P. P. Vaidyanathan), Chalmers Institute of Technology (Prof. Arne Svensson), University of Minnesota (Ass. Prof. Mohamed-Slim Alouini), RWTH Aachen (Prof. Henrich Meyr) and several others. These contacts have resulted in joint publications, guest researcher exchanges and sabbaticals, student exchange at the graduate and undergraduate level, mutual participation in doctoral defense committees, invited lectures, *et cetera*. We will continue to build on the above and other international contacts in the future, in order to keep strong links between our research and the international state-of-the-art.

Specifically, BEATS is currently establishing a close collaboration with the highly regarded AWACCS[9] group at The University of Minnesota, MN, USA (cf. http://www.ece.umn.edu/users/alouini/awaccs/index.htm). BEATS intends to finance a two-month stay as guest researcher at NTNU for the AWACCS group's director, Ass. Prof., PhD *Mohamed-Slim Alouini* (see attached CV), during Fall 2002. The goals of this collaboration are related to joint research and publications within adaptive coded modulation, hierarchical modulation schemes, and channel estimation issues, as well as to co-supervision of a PhD student at AWACCS, Mr. *Pavan K. Vitthaladevuni*. We are currently looking towards ways to fund a 1-year stay at NTNU for Mr. Vitthaladevuni. Prof. Alouini has also offered to give an intensive doctoral course on modern performance analysis of wireless communication systems during his planned stay at NTNU.

7.2 Possible future activities

A natural follow-up to the present basic research project would be a more implementation-oriented project. Such a project should concentrate on practical realization of the ideas that are deemed most promising in our research. The focus of such a follow-up project could be e.g. software radio technology with a particular emphasis on the ATCM/OFDM combination, combined with the use of higher frequency bands (typically 40–60 GHz), reconfigurable networking, adaptive antennas, and digital predistortion. This may lead towards the realization of advanced, adaptive solutions suitable for future low-cost telecommunications products for the mass market. In such a future project, *SINTEF Telecom and Informatics* would be a natural partner since they possess the necessary expertise on microwave technology and software radio implementation. The *Department of Physical Electronics* at NTNU should also be a partner in such a project.

References

[1] H. Meyr, "Algorithm design and system implementation for advanced wireless communications systems", invited paper in Proc. *International Zurich Seminar on Broadband Communications*, February 2000, Zurich, Switzerland.

[2] V. Bose, D. Wetherall, and J. Guttag, "Next century challenges: RadioActive Network", in Proc. *MOBICOM-99*, August 1999.

[3] T. A. Cover and J. A. Thomas, *Elements of Information Theory*, Wiley Series in Telecommunications, 1991.

[4] S. B. Zahir Azami, *Joint Source/Channel Coding. Hierarchical Protection*, Ph. D. thesis, Ecole Nationale Supérieure des Télécommunications (ENST), Paris, 1999.

[5] T. Fine, "Properties of an optimum digital system and applications", *IEEE Trans. Inf. Theory*, pp. 287–296, Vol. IT-10, October 1964.

[6] A. Kurtenbach and P. Wintz, "Quantizing for noisy channels", *IEEE Trans. Commun.*, pp. 291–302, vol. COM-17, April 1969.

[7] H. Kumazawa, M. Kasahara, and T. Namekawa", "A construction of vector quantizers for noisy channels", *Electron. and Eng. in Japan*, pp. 39–47, vol. 67-B, 1984.

[8] N. Farvardin and V. Vaishampayan, "Optimal quantizer design for noisy channels: An approach to combined source-channel coding", *IEEE Trans. Inf. Theory*, pp. 827–838, vol. IT-33, November 1987.

[9] E. Ayanoğlu and R. M. Gray, "The design of joint source and channel trellis waveform coders", *IEEE Trans. Inf. Theory*, pp. 855–865, vol. IT-33, November 1987.

[10] K. Zeger and A. Gersho, "Pseudo-Gray coding", *IEEE Trans. Commun.*, pp. 2147–2158, vol. 38, December 1990.

[11] P. H. Skinnemoen, *Robust Communication with Modulation Organized Vector Quantization*, Ph. D. Thesis, Norwegian Institute of Technology, Dept. of Telecommunications, 1994.

[12] P. Knagenhjelm, "A recursive design method for robust vector quantization", pp. 948–954, Proc. *Int. Conf. on Signal Processing Applications and Technology*, 1992, Boston, MA, USA,

[9] Analysis of Wireless and Cellular Communication Systems.

[13] M. Wang and T. R. Fischer, "Trellis-coded quantization designed for noisy channels", *IEEE Trans. Inf. Theory*, pp. 1792–1802, vol. 40, November 1994.

[14] T. A. Ramstad, "Efficient and robust communication based on signal decomposition and approximative multi-dimensional mappings between source and channel spaces", *2nd IEEE Nordic Signal Processing Symposium*, 1996. 34 pages.

[15] J. M. Lervik, A. Fuldseth, and T. A. Ramstad, "Combined image subband coding and multilevel modulation for communication over power- and bandwidth-limited channels", Proc. IEEE VSPC, 1994.

[16] J. M. Lervik, A. Grøvlen, and T. A. Ramstad, "Robust digital signal compression and modulation exploiting the advantages of analog communications", Proc. *GLOBECOM-95*, p.1044–1050, 1995.

[17] J. M. Lervik and T. A. Ramstad, "Robust image communication using subband coding and multilevel modulation", Proc. *SPIE Visual Communications and Image Processing Conference*, pp. 524–535, 1996.

[18] A. Fuldseth and T. A. Ramstad, "Combined video coding and multilevel modulation", Proc. *IEEE International Conference on Image Processing*, pp. 941–944, 1996.

[19] A. Fuldseth and T. A. Ramstad, "Bandwidth compression for continuous amplitude channels based on vector approximation to a continuous subset of the source signal space", Proc. *IEEE International Conference on Avoustics, Speech, and Signal Processing*, pp. 3093–3096, 1997.

[20] T. A. Ramstad, "Combined source coding and modulation for mobile multimedia communication", pp. 415–429 in D. Bull, N. Canagarajah, and A. Nix (ed.): *Insights in Mobile Multimedia Communications* (Academic Press, 1998).

[21] J. A. C. Bingham, "Multicarrier modulation for data transmission: an idea whose time has come," *IEEE Commmunications Magazine*, vol. 28, pp. 5–14, May 1990.

[22] E. A. Lee and D. G. Messerschmitt, "Digital Communication", 2nd ed. Kluwer Academic Publishers, Boston, 1994.

[23] G. Cariolaro and F. C. Vagliani, "An OFDM scheme with half complexity", *IEEE Journ. on Sel. Areas in Comm.*, vol. 13, no. 9, Dec. 1995, pp. 1586-1599.

[24] A. Vahlin, "Optimisation and analysis of OFDM communication systems." Ph.D. thesis, NTH, Trondheim, 1995.

[25] A. Vahlin and N. Holte, "Optimal finite duration pulses for OFDM", *IEEE Trans. Commun.*, vol. 44, pp. 10–14, Jan. 1996.

[26] A. Vahlin and N. Holte, "OFDM for broadcasting in presence of analogue co-channel interference", *IEEE Trans. Broadcast.*, vol. 41, no. 3, pp. 89–93, 1995.

[27] A. Vahlin and N. Holte, "Use of a guard interval in OFDM on multipath channels", *IEEE Electronics Letters*, vol. 30, no. 24, pp. 2015–2016, Jan. 1996.

[28] A. Vahlin and N. Holte, "A new class of OFDM systems", Proc. *1st IEEE Nordic Signal Processing Symposium*, pp. 274–277, 1994.

[29] J. O. Hansen, "Multiband digital transmisjon," Siv.ing. thesis, NTH, 1992.

[30] H. B. Jahren, "Pulsforming for multibånd digital modulasjon," Siv.ing. thesis, NTNU, 1999.

[31] J. Baltersee, G. Fock, and H. Meyr, "An information theoretic foundation of synchronized detection", submitted to *IEEE Trans. Commun.*, April 2000.

[32] G. E. Øien and K. J. Hole, "Maximum average spectral efficiency of slowly varying Rayleigh fading channels with pilot-symbol-assisted channel estimation," in Proc. *EPMCC-2001*, February 2001, Vienna, Austria.

[33] K. J. Hole, H. Holm, and G. E. Øien, "Analysis of adaptive coded modulation with antenna diversity and feedback delay," to be published in Proc. *IST-2001*, September 2001, Barcelona, Spain.