Project proposals 2014

As noted, most of the proposed projects are possible to expand into M.Sc. thesis projects. Please let us know if you might be interested in this option as well. You can learn more about each of the projects by asking the project supervisors.

1. Spectral analysis of non-linear signals

In a recent master thesis project at the Department of Mathematical Statistics, a novel technique for non-linear non-destructive testing of materials has been developed (Dahlen, 2013). The proposed method can be used in the field of material characterization and structural health monitoring since it has the potential to sense the very beginning of micro damage, which is not visible to the eye. The proposed method is based on the free reverberation of a sample after a single mechanical impact excitation using a sum of time-varying amplitude polynomial phase signals (parametric model). So far the method has been tested on relatively small concrete samples with low damping properties. In larger samples or other materials with more material damping the decay of the samples resonant frequency is much quicker making the extraction of a non-linear signature more difficult. In this project, alternative spectral analysis techniques are investigated for the extraction of non-linear effects in non-destructive testing of materials.

Supervisors: Andreas Jakobsson and Nils Rydén

2. Improved spectral analysis of Lamb waves

Lamb waves are guided elastic waves propagating in a free infinite plate with finite thickness. At each frequency several modes (defined by phase velocity, wavelength, or wave number along the surface) can propagate simultaneously. The velocities of Lamb waves are dispersive which means that the velocity changes with frequency. Lamb wave dispersion curves can be calculated from the elastic constants of the plate using the Lamb wave equation. In non-destructive testing of plate like structures with unknown thickness or elastic constants, peaks in the 2D spectrum are compared with corresponding theoretical dispersion curves to estimate the unknown plate properties. The normal response of the plate from an impulse source is measured using a linear array of receivers and stored as a matrix with about 40 samples in space and 1000 samples in time. The relatively small number of samples in space results in poor resolution of the spatial periodicity limiting the overall resolution of the measurements. It should be beneficial if multiple modes could be resolved at each temporal frequency resulting in several measured dispersion curves, which can be matched with theoretical modes.

Supervisors: Andreas Jakobsson and Nils Rydén

3. Finding linear chirps

A chirp is a signal where the frequency content changes continuously over time. Chirps are very common in applications such as radar, speech, and bird song. In this project you will be implementing and testing a new approach to estimating linear chirps. In the first part of the project you will get to know constrained optimization and get familiar with the signals. In the second part, you will be given tools to solve the optimization problem and estimate the chirps in the data. To succeed you will need to try different parameter settings in the optimization and evaluate how robust the method is. You will be provided with real bird song data to test your method on.

Supervisors: Johan Swärd and Johan Brynolfsson
4. Birdsong: Syllable comparison in ambiguity domain
Standard methods for analysis of bird songs are based on the intuitive sonogram or the time-frequency spectrum. However, for song analysis based in particular on syllable detection and comparison, the ambiguity domain turns out to be more suitable. A species with an exceptional song capacity is the Great Reed Warbler. Studies of its song are however impaired by the lack of methods, which would automatically, and more objectively analyze the song structure. The method in this project is based on measuring distance between two syllables in the ambiguity domain. The distance between two syllables is calculated from singular vectors of the estimated spectrum and designed to capture dissimilarity in both time- and frequency content. The measure should be evaluated (and optimized) on a larger data set and for different time-frequency algorithms. A possible extension of this project into a Master thesis could be a syllable-based modeling (such as adaptive HMM-type models) and analysis of the warbler’s song.

Supervisor: Mareile Große Ruse

5. Cross-term relocation of dolphin echo-location signals
Recent bio-sonar studies have shown that the measured echolocation signals from bottlenose dolphins (Tursiops truncatus) and beluga whales (Delphinapterus leucat) may contain more than one component. The echolocation signal is time- and frequency-varying and usually heavily disturbed by noise. When estimating the time-frequency distribution of a process containing multiple components one will get the unfortunate effect of cross-terms. A common way to deal with these is to apply some kernel-function, which will remove cross-terms, but the kernel will also smear the auto-terms. Another idea is to simply move the cross terms to some other location in the spectrum where they don’t disturb the interpretation of the true components, meaning that the auto-terms will retain optimal concentration. In this project you will implement code for an imaginary-valued phase-kernel that relocates cross-terms and evaluate and compare the performance on both simulated signals as well as on dolphin echo-location signals. The project is made in collaboration with Josefin Starkhammar, division of Electrical Measurements.

Supervisor: Johan Brynolfsson

6. Using spatial information to localize microphones, speakers, and/or to “hear” the shape of a room
Lots on current research focuses on localizing either the sound sources or the microphones of a general microphone array. Using only a couple of microphones placed in an arbitrary geometrical pattern and a known sound, is it then possible to determine the 3D geometry of a convex polyhedral room? Alternatively, in a room of a known shape, can one localize the position of a single microphone using the room reverberation of a single speaker? These questions are related to some current research projects in the field. To our help, we have different techniques fetched from linear algebra, geometry and statistical signal processing, as well as measurement microphones, speakers, a sound card, and supplementary acoustic hardware to perform real-life measurements in controlled environments.

Supervisors: Ted Kronvall and Simon Burgess
7. **Modeling inharmonicities in voiced speech**

It is well known that several forms of string-based instruments, such as a piano or a guitar, exhibit inharmonicities in the harmonic line spectral structure. This means that the overtones appear slightly higher than a multiple of the fundamental frequency, a fact that needs to be considered when constructing pitch estimation algorithms. In this project, we will examine similar frequency shifting in voiced speech, and try to determine how this could be modelled as a function of frequency.

**Supervisor:** Andreas Jakobsson

8. **Making reassignment adjustable**

The idea of the reassigned spectrogram is to move the value of the spectrogram at a certain time-frequency location to another time-frequency location, which is more representative as the localization of the signal energy. For a sinusoid or a chirp-signal, this relocation procedure results in a sharp ridge. In the basic algorithm there is no possibility to adjust these relocations, as they are controlled by the derivative of the phase of the short-time Fourier transform. In this project the Levenberg-Marquardt algorithm is used to allow for adjustment in the relocation, allowing a strong or weak reassignment. A Gaussian window will be used in the short-time Fourier transform of the spectrogram which allow for easy implementation of the different derivatives. The aim is to implement the algorithm and to investigate advantages and disadvantages compared to the basic reassignment procedure.

**Supervisor:** Maria Sandsten

9. **Modeling neuronal cells changes**

In a biological experiment neuronal cells are placed on a plate. After about three weeks these cells have connected into a network, which is oscillating synchronously. These oscillations are manifested as variation of Ca^{2+}-ions in the cells. Different drugs with the potential of becoming pharmaceutical treatments of diseases in the brain can be given to these cells. The oscillations are thereby altered in different ways with a structure that is often obviously non-linear and non-stationary. With genetic modification, genes that are associated with schizophrenia can be altered in some cells. Differences in drug response between "normal" cells and genetically modified cells have the potential of pointing towards new treatments of schizophrenia. The challenge is to describe the changes in the time series that are associated with differences in drug response. Empirical Mode Decomposition (EMD) is a data-driven time-frequency method with the aim to find time-varying basis functions that are not necessarily harmonically related. In this project, some EMD-algorithms should be implemented and investigated for possible classification. The project is made in collaboration with Lars Arvastson, Lundbeck A/S, Denmark.

**Supervisors:** Maria Sandsten and Andreas Jakobsson

10. **Cramér-Rao lower bound for inharmonic data**

Speech and music signals are often modeled frame-by-frame as a sum of harmonically related sinusoids. However, short-term Fourier transforms of such signals often reveal that the peaks of the harmonics are not at exact integer multiples of the fundamental frequency, a phenomenon called inharmonicity. An important problem in audio and array signal processing is to find accurate estimates of the signal parameters despite such modeling mismatches. In this project, we will derive and analyze the Cramér-Rao Lower Bound (CRLB) for the estimation of parameters in inharmonic data. The analysis will help answer fundamental questions about dealing with inharmonicity. Knowledge gained through this project would be useful for further work in audio signal processing and multi-sensor data processing.

**Supervisor:** Naveed Butt
11. Robust spatial estimation for near-field signals.
Many of the spatial methods in “Spectral Analysis of Signals” by Stoica and Moses assume far-field sources, i.e., that the incoming sound waves are planar, and also, that the signal transmitted contains only a single sinusoid, having the carrier frequency. In this project, we want to loosen these assumptions and yet use the same techniques, although allowing for robustness in the estimations. Also, we want to allow the signals to be broadband, i.e., to have a larger frequency contents than assumed. This year we have proposed a joint estimation of pitch (such as a musical sound note) and direction-of-arrival using a sparse estimation technique. It is however non-robust and assumes far-field sources, but a possible project could be to test it on real data measured near-field in an anechoic chamber, to see how well it performs. Another project application would be to perform measurements in the same anechoic chamber and perform analysis with robust beamforming techniques, starting out with the ones in the course literature, after which steps towards implementations with beamformers for broadband signals can be taken.

Supervisor: Ted Kronvall

The projects are meant to be performed in groups of two, but can be done individually if desired. By Friday, January 31st, please let us know your group’s project preferences, ranking 1st, 2nd, and 3rd choice to allow us to distribute the projects fairly. If you have your own project, please let us know as well, giving a brief description of the project. If you have any questions, please just get in touch!

Andreas & Maria