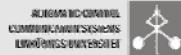


# Welcome to Digital Signal Processing 2011!!



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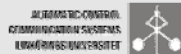


**”Signal Processing is the art of getting what you want from signals”**



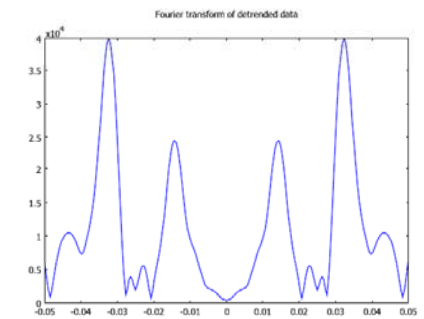
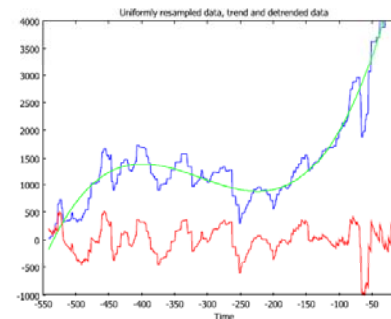
## Outline Lecture 1

1. Introduction and motivation using typical signal processing problems and overview of the chapters in the book
2. Course administration
3. Repetition of frequency domain descriptions
  - a) Fourier series
  - b) Discrete Fourier Transform (DFT)
  - c) Fourier Transform (FT)
  - d) Discrete Time Fourier Transform (DTFT)
4. Repetition of Poisson’s summation formula
5. Repetition of the sampling theorem



## Example 1 – Variation in Species

First thing to do: **Look at the data**



**Given:** Data with the number of species on earth calculated from fossil sedimentations.  
**Problem areas:** Analysis of the frequency components and correlation with ice ages and other climate variations.  
 Preprocessing of data, removing trends, etc.

Brian Hayes. **Life Cycles**. *American Scientist*, 93(4):299-303, July – August, 2005.



## Example 2 – UAV Acceleration Measurements

5

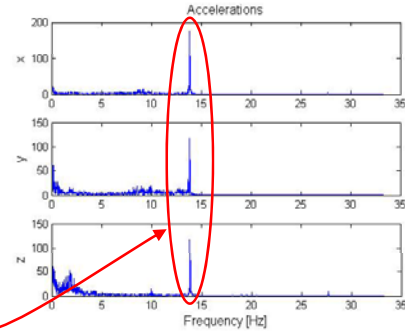
### Sensor signals from an Unmanned Aerial Vehicle (UAV)



Figure courtesy of the Unmanned Aircraft Systems Technologies Lab, Department of Computer and Information Science, Linköping University.

UAV we are working with (CADICS project, [www.cadics.isy.liu.se](http://www.cadics.isy.liu.se) and a new SSF project)

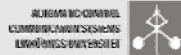
Carries several sensors, for example accelerometers and gyroscopes.



13.85 Hz = 830 rpm (rotor speed)

Frequency domain description of signals (Ch. 2)

Digital Signal Processing, TSRT78  
T. Schön



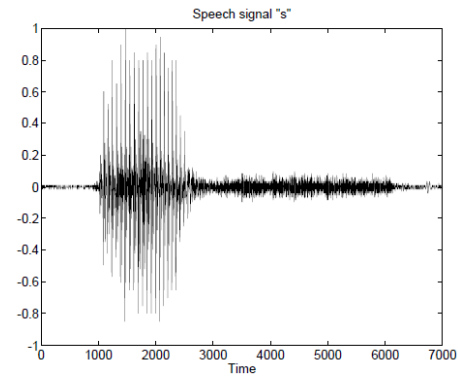
Lecture 1

## Example 3 – Speech

6

**Given:** Sampled speech signal.

**Problem areas:** Analysis, and presentation.  
Transmission and storage.



Digital Signal Processing, TSRT78  
T. Schön

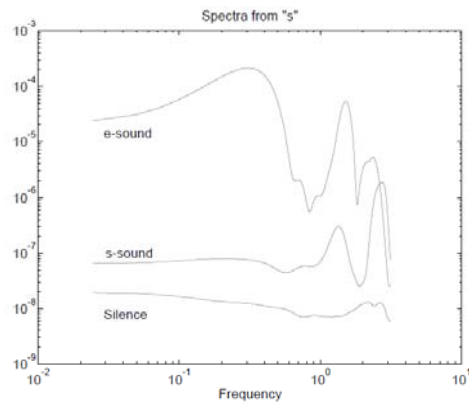


Lecture 1

## Example 3 – Speech (Spectral Analysis)

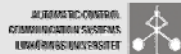
7

Different sounds of speech have different frequency content. Compare the energy spectrum for silence, s-sound and e-sound.



Spectral analysis and spectral estimation (Ch. 3)

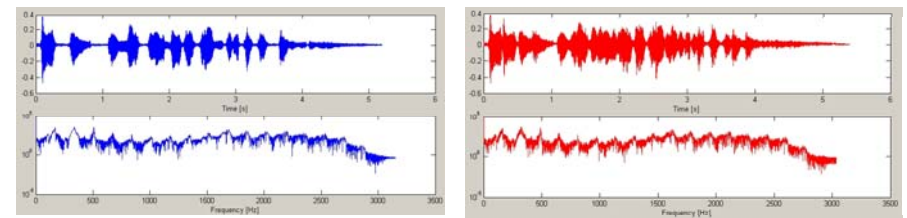
Digital Signal Processing, TSRT78  
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Lecture 1

## Example 3 – Speech (Filtering)

8



"Farbror barbro" original signal

"Farbror barbro" filtered signal (echo with a delay of 0.2 s)

(run demo)

Filter theory (Ch. 4)

Digital Signal Processing, TSRT78  
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Lecture 1

### Example 3 – Speech (Coding in GSM)

9

In the GSM-standard the signal is modelled using the following model

$$y(t) = -a_1y(t-1) - a_2y(t-2) - \dots - a_8y(t-8) + e(t)$$

where  $e(t)$  is a driving "noise". The next signal value can then be predicted using the following equation

$$\hat{y}(t) = -a_1y(t-1) - a_2y(t-2) - \dots - a_8y(t-8)$$

The coefficients are estimated every 20 ms. The eight coefficients and the prediction errors  $y(t) - \hat{y}(t)$  are then transmitted (these are smaller than the signal values itself, hence requiring fewer bits)

(run demo)

Signal models (Ch. 5) and estimating signal models (Ch. 6)

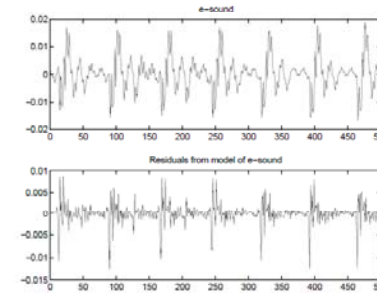
### Example 3 – Speech (Coding in GSM)

10

Estimated signal model for the e-sound:

$$\hat{a} = (-0.0284, 0.5070, -0.6288, -0.0174, \\ -0.5457, 0.6029, 0.1769, 0.3965)$$

gives the residuals:



(run demo)

Signal models (Ch. 5) and estimating signal models (Ch. 6)

### Example 4 – Active Safety Systems in Cars

11

YouTube is a beautiful source for inspiration!

[www.youtube.com/watch?v=bbvUrdKh2gA](http://www.youtube.com/watch?v=bbvUrdKh2gA)

[www.youtube.com/watch?v=LVmi9yXvK4o](http://www.youtube.com/watch?v=LVmi9yXvK4o)

More information about the system can be found here:

<http://www.dn.se/motor/autobromsen-kan-radda-65-liv-per-ar-1.1194456> (DN, Saturday, Oct. 23 2010)

<http://www.control.isy.liu.se/include/images/AndreasDN20110521.pdf> (DN, Saturday, May 21, 2011)

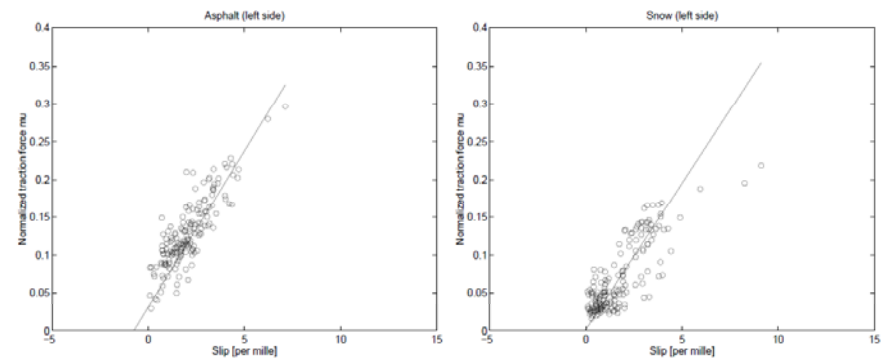
State-space models (Ch. 5) and Kalman filtering (Ch. 8)

### Example 5 – Estimating Tire – Road Friction

12

**Given:** Measurements of engine torque and wheel slip and a linear model.

**Problem areas:** Estimate the friction between tire and road surface as the slope of the curve.



Adaptive filters (Ch. 9)

## Example 5 – Estimating Tire – Road Friction (model)

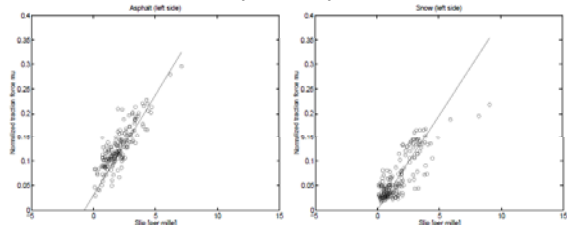
13

Measurements of the engine torque  $M(t)$  and the wheel slip  $s(t)$  are given (wheel slip =  $\omega r/v - 1$ ). Signal model:

$$s(t) = a + bM(t) + e(t)$$

where  $e(t)$  is stochastic noise, representing model errors and errors in the measurements.

The slope of the straight line that approximates the measurements is proportional to the friction. Compare asphalt and snow below.



Signal Models (Ch. 5) and estimating signal model (Ch. 6)

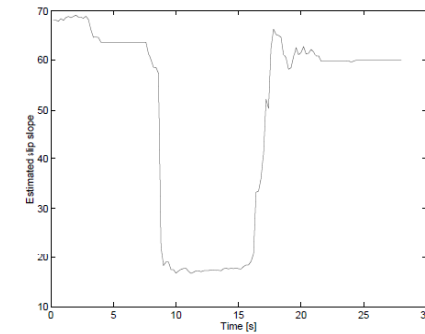
Lecture 1

## Example 5 – Estimating Tire – Road Friction (adaptation)

14

The parameters are time varying. Let  $\theta(t) = (a(t), b(t))^T$   
Use model adaptation (Kalman filter, RLS)

$$\hat{\theta}(t) = \hat{\theta}(t-1) + K(t)(s(t) - \hat{s}(t))$$



Kalman filters (Ch. 8) and adaptive filters (Ch. 9)

Lecture 1

## Signal Processing in a bigger context

15

Signal processing plays an important role in itself (as we have seen already),

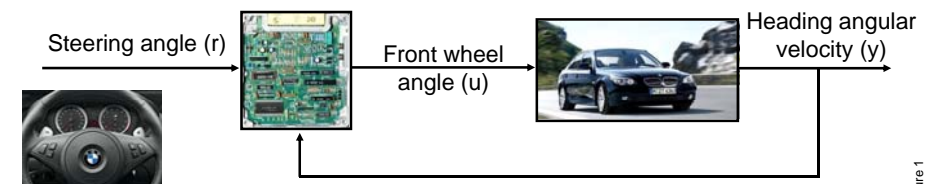
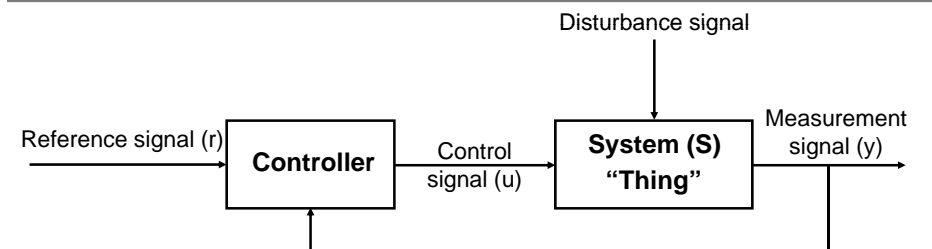
**but** it also plays a very important role as a part of a bigger system, we will provide two examples of this,

- Control systems
- Autonomous systems

Lecture 1

## Signal Processing within a Control Setting (I/II)

16



Lecture 1

## Control Goal: Stabilize the heading angle

Standard manoeuvre, double lane change at 100 km/h. The controller affects the angle of the front wheel (active steering).



Without controller

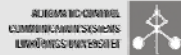


With controller

The movies are used with permission from ZF Lenksysteme



Lecture 1



## Sensors

- Joint displacements
- Force
- Current
- 3D Gyroscopes
- 3D Accelerometers
- Temperature (engine, oil)
- Oil flow
- Oil pressure
- Engine RPM
- Stereo vision
- LIDAR (laser)

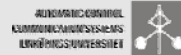
Big dog



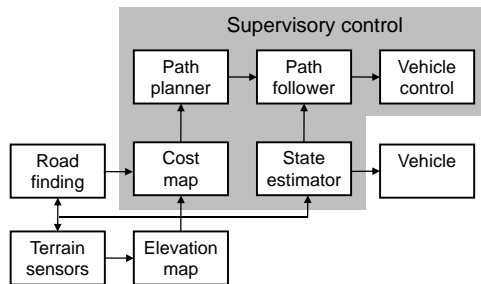
Boston Dynamics

All the measurements from these sensors need signal processing in one form or another!

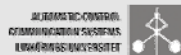
www.bostondynamics.com



Lecture 1



www.darpa.mil/grandchallenge



Lecture 1



Basic idea is to make use of inertial sensors and signal processing (Kalman filter) to estimate the position and orientation (pose)

## Inertial Measurement Unit (IMU)

- 3 accelerometers (specific force),
- 3 gyroscopes (angular velocity),
- 1 magnetometer.

State-space models (Ch. 5) and Kalman filtering (Ch. 8)

(Illustrating the close connection to the course in Sensor Fusion (TSRT14))



Lecture 1

The **aim** of this course is to show the most important **methods and algorithms** for signal processing

and to show how these can be **applied on signals** of various kinds.



Lecture 1

- You: ~65 students
- Us:
  - Lecturer and examiner: Thomas Schön
  - Teaching assistant: Fredrik Lindsten
  - Teaching assistant: Saikat Saha

Lecture 1

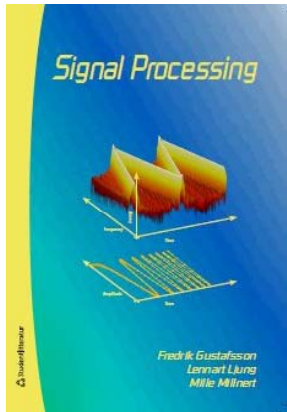
- Examiner and lecturer: Thomas Schön
- 14 lectures (theory, examples and guests from industry)
- 12 exercise sessions (7 in front of a computer) (solve problems and **discuss, ask questions**)
- 2 labs (lab course 1hp)
  1. Fundamental signal processing. Give you practical experience with the theory and algorithms that are discussed in the course book, **free work, examined by a report and a review report (deadline exercise session 9 and 11, respectively).**
  2. Active noise control (in our lab, Laboteket), suppressing disturbing sound in real time. Standard 4h lab with **preparations.**
- Register for the labs via the course web-site (soon available).

Lecture 1

The course is rather **different** to most other courses, and more **similar** to the situation at companies, when it comes to the use of computers:

- The use of computers is highly integrated in the course and more than half of the exercise sessions takes place in computer labs.
- The computer and MATLAB are "allowed aids" during the exam. This also explains why the exam takes place in ISY's computer labs.
- MATLAB can be downloaded from the student portal or bought at Bokakademien in Kårallen at a very low cost (basically the cost of the disc).

Lecture 1



Introduction (Ch. 1)

**Non-parametric signal processing**

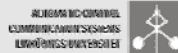
Transforms (some repetition FT, DTFT, alias), DFT (Ch. 2)  
 Stochastic signals and spectral estimation (Ch. 3)  
 Filtering (some repetition and practical aspects) (Ch. 4)

**Parametric signal processing**

Parametric signal models (Ch. 5)  
 Estimating signal models (Ch. 6)  
 Linear estimation (Wiener and Kalman filtering) (Ch. 7-8)  
 Adaptive signal processing (Ch. 9)

If you understand Swedish you can use the older Swedish version, but the English version is recommended since it is updated.

Lecture 1



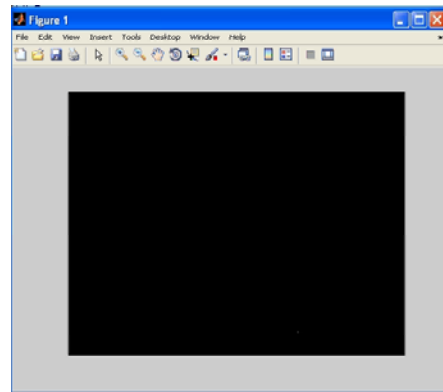
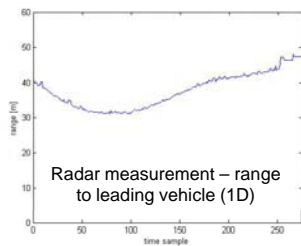
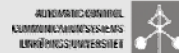
- All course information including lecture material is available from the course home page

[www.control.isy.liu.se/student/tsrt78/](http://www.control.isy.liu.se/student/tsrt78/)

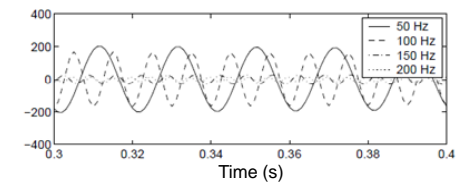
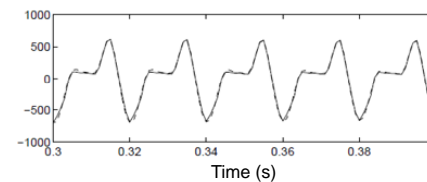
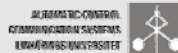
- Exam: The book with normal study notes is an "allowed aid". The exam is in front of a computer, which is also an "allowed aid". Special exam accounts are used during the exam.

- **Ask questions!**

Lecture 1



Lecture 1



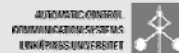
Solid line – measured current in a 400 kV transformer (Söderåsen) when it is switched on.  
 Dashed line – Approximation based on the fundamental frequency and the 3 first overtones (harmonics).

Showing the fundamental frequency (50 Hz) and the three first overtones.

$$i(t) \approx \sum_{n=1}^4 a_n \sin(2\pi 50nt + \phi_n)$$

Hence, a compact and simple representation in the frequency domain.

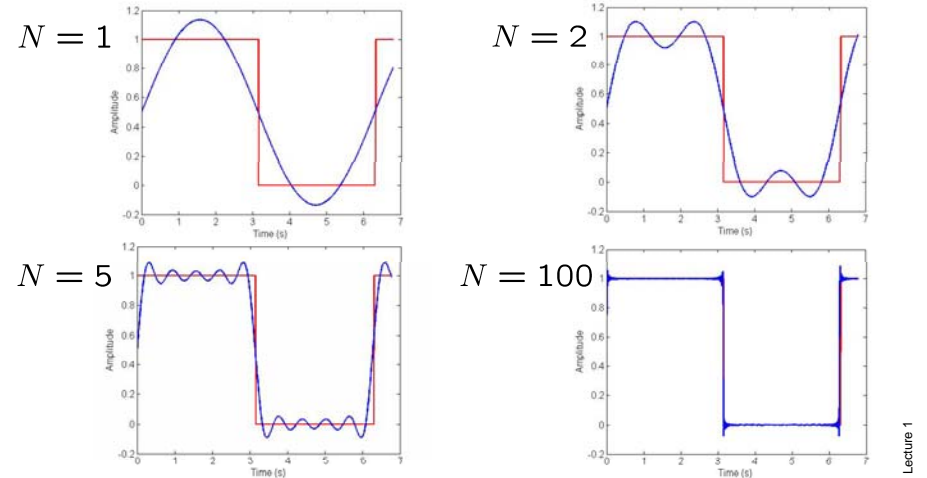
Lecture 1



Signal	Periodic	L1-signal	General
Continuous time	FS $X_{FC}[n]$	FT $X(i\omega)$	Laplace $X(s)$
Discrete time	DFT $X[n]$	DTFT $X_T(e^{i\omega T})$	ZT $X(z)$

Lecture 1

Using more and more sinusoidals gives a better approximation as expected.



Lecture 1

Some of the most important symmetry (**duality**) relations between the time and the frequency domain for the FT are

$$x(t - \tau) \leftrightarrow e^{-i\omega\tau} X(i\omega)$$

$$e^{i\omega_0 t} x(t) \leftrightarrow X(i\omega - i\omega_0)$$

$$\int_{-\infty}^{\infty} x(t - \tau)y(\tau)d\tau \leftrightarrow X(i\omega)Y(i\omega)$$

$$x(t)y(t) \leftrightarrow \frac{1}{2\pi} \int_{-\infty}^{\infty} X(i\omega - i\phi)Y(i\phi)d\phi$$

Lecture 1

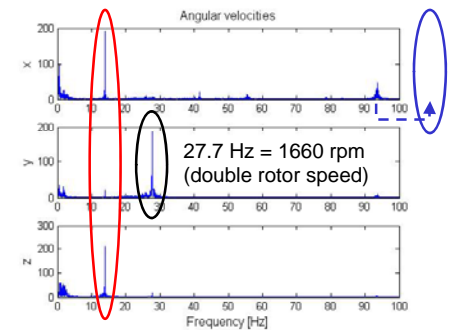
These are not all valid for the DTF. This is a problem, since we always are working with finite data in practice. However, if we understand the problem, we can handle it.



Figure courtesy of the Unmanned Aircraft Systems Technologies Lab, Department of Computer and Information Science, Linköping University.

Investigating the measurements from the gyroscopes (angular velocities)

13.85 Hz = 830 rpm (rotor speed)      Aliasing from 106.4 Hz = 6384 rpm (Engine rpm)

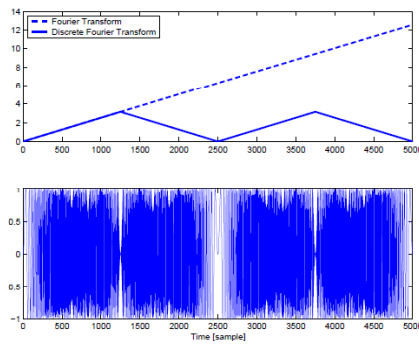


Nyquist frequency (half the sampling frequency): 100 Hz

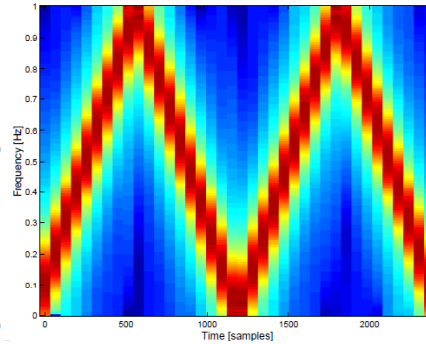
Lecture 1

The chirp signal is defined as  $y[k] = \sin\left(\frac{2\pi k}{N}k\right)$ ,  $k = 0, 1, \dots, N - 1$

```
N = 15000;
y = sin(2*pi*(0:N-1).^2./N);
sound(y);
```



Spectrogram



The instantaneous frequency is  $4\pi k/N$   
We expect a straight line here

Lecture 1

**Signal Processing:** “The art of getting what you want from signals”.

**Signals:** A sequence of numerical values.

**Discrete Fourier Transform (DFT):** Practically speaking this is the most commonly used Fourier transform.

**Poisson’s summation formula:** Explains the relationship between the Fourier transform and the DTFT.

**The sampling theorem:** The DTFT and the FT are identical if the continuous-time signal’s energy content is between  $-\omega_s/2$  and  $\omega_s/2$

**Nyquist frequency (folding frequency):** Half the sampling frequency.

**Aliasing (folding):** If the sampling theorem does not hold, frequencies appear under a fake name (aliasing) since they get folded, as explained by Poisson’s summation formula.

**Anti-alias filter:** Low-pass filter used to remove frequency contents in a signal that otherwise would lead to alias, i.e., get folded and fool the user.

Lecture 1