Chapter-1 : Introduction

• Aims/Scope
  Why study DSP ?
  DSP in applications : GSM example

• Overview
  Filter design & implementation
  Filter banks and subband systems
  Optimal and adaptive filters

• Activities
  Lectures : Course notes/literature
  Exercise sessions : Acoustic modem project
  Exam
Why study DSP?

- Analog Systems vs. Digital Systems
  - Can translate (any) analog (e.g. filter) design into digital
  - Going `digital’ allows to expand functionality/flexibility/…
    (e.g. how would you do analog speech recognition? analog audio compression? …?)

Why study DSP?

- Start with one `DSP in applications’ example:
  - DSP in mobile communications (GSM)

- Main message:
  Consumer electronics products (and many other systems)
  have become (embedded) ‘supercomputers’ (Mops…Gops/sec),
  packed with mathematics & DSP functionalities…
DSP in applications : GSM

Cellular Mobile Telephony (e.g. GSM)

• Basic network architecture:
  - Country covered by a grid of cells
  - Each cell has a base station
  - Base station connected to land telephone network and communicates with mobiles via a radio interface
  - Digital communication format

DSP in applications : GSM

• DSP for Digital Communications (‘physical layer’):
  – A common misunderstanding is that digital communications is ‘simple’ …
  – While in practice…

Transmitter 1,0,1,1,0,…

Channel +

Receiver decision

 decision 1,0,1,1,0,…

PS: This is a discrete-time system representation, see Chapter-2 for review on signals&systems
DSP in applications: GSM

- DSP for Digital Communications ('physical layer'):
  - While in practice...
    - This calls for channel model + compensation (equalization)

Transmitter: 1,0,1,1,0, 1,0,1,1,0, ...

Multipath Channel

Received signal

Receiver: noise: 0.59, 0.41, 0.76, 0.05, 0.37, ...

DSP in applications: GSM

- GSM Channel Estimation/Compensation:
  - Multi-path channel is modeled with short (3...5 taps) FIR filter

$$H(z) = a + b z^{-1} + c z^{-2} + d z^{-3} + e z^{-4}$$

Multipath Channel

$\approx$

PS: $z^{-1}$ or $\Delta$ represents a sampling period delay, see Chapter-2 for review on $z$-transforms.
DSP in applications: GSM

- GSM Channel Estimation/Compensation (continued)
  - Multi-path channel is modeled with short (3…5 taps) FIR filter

\[ H(z) = a + b.z^{-1} + c.z^{-2} + d.z^{-3} + e.z^{-4} \]

\[
\begin{bmatrix}
\text{OUT[1]} \\
\text{OUT[2]} \\
\text{OUT[3]} \\
\text{OUT[4]} \\
\text{OUT[5]} \\
\vdots \\
\text{OUT[N]}
\end{bmatrix} =
\begin{bmatrix}
\text{IN[1]} & 0 & 0 & 0 & 0 \\
\text{IN[2]} & \text{IN[1]} & 0 & 0 & 0 \\
\text{IN[3]} & \text{IN[2]} & \text{IN[1]} & 0 & 0 \\
\text{IN[4]} & \text{IN[3]} & \text{IN[2]} & \text{IN[1]} & 0 \\
\text{IN[5]} & \text{IN[4]} & \text{IN[3]} & \text{IN[2]} & \text{IN[1]} \\
\vdots & \vdots & \vdots & \vdots & \vdots \\
\text{IN[N-1]} & 0 & 0 & 0 & 0
\end{bmatrix}
\begin{bmatrix}
a \\
b \\
c \\
d \\
e
\end{bmatrix}
\]

This leads to a least-squares parameter estimation

\[
\begin{bmatrix}
\text{OUT[1]} \\
\text{OUT[2]} \\
\text{OUT[3]} \\
\text{OUT[4]} \\
\text{OUT[5]} \\
\text{OUT[N]}
\end{bmatrix} = \begin{bmatrix}
\text{IN[1]} & 0 & 0 & 0 & 0 \\
\text{IN[2]} & \text{IN[1]} & 0 & 0 & 0 \\
\text{IN[3]} & \text{IN[2]} & \text{IN[1]} & 0 & 0 \\
\text{IN[4]} & \text{IN[3]} & \text{IN[2]} & \text{IN[1]} & 0 \\
\text{IN[5]} & \text{IN[4]} & \text{IN[3]} & \text{IN[2]} & \text{IN[1]} \\
\vdots & \vdots & \vdots & \vdots & \vdots \\
\text{IN[N-1]} & 0 & 0 & 0 & 0
\end{bmatrix}
\begin{bmatrix}
a \\
b \\
c \\
d \\
e
\end{bmatrix}
\]

See PART-IV on ‘Optimal Filtering’

Carl Friedrich Gauss (1777 – 1855)
GSM Channel Estimation/Compensation (continued)

- Channel coefficients (cfr. a,b,c,d,e) are identified in receiver based on transmission of pre-defined training sequences (TS), in between data bits.

- Channel model is then used to design suitable equalizer (‘channel inversion’), or (better) to reconstruct transmitted data bits based on maximum-likelihood sequence estimation (e.g. ‘Viterbi decoding’).

- Channel is highly time-varying (e.g. terminal speed 120 km/hr !) => All this is done at `burst-rate` (+- 100 times per sec).

= SPECTACULAR !!

GSM Speech Coding

- Original ‘PCM-signal has 64kbits/sec = 8 ksamples/sec*8bits/sample
- Aim is to reduce this to <11kbits/sec, while preserving quality!
- Coding based on speech generation model (vocal tract…), where model coefficient are identified for each new speech segment (e.g. 20 msec).
- This leads to a least-squares parameter estimation (again), executed +- 50 times per second. Fast algorithm is used, e.g. ‘Levinson-Durbin’ algorithm.
- Then transmit model coefficients instead of signal samples.
- Synthesize speech segment at receiver (should ‘sounds like’ original speech segment).

= SPECTACULAR !!
**DSP in applications : GSM**

- GSM Channel Estimation/Compensation
- GSM Speech Coding
- GSM Multiple Access Schemes
  - Accommodate multiple users by time & frequency ‘multiplexing’
  - FDMA (freq.division multiple access): 125 frequency channels for GSM/900MHz
  - TDMA (time division multiple access): 8 time slots(=users) per channel, ‘burst mode’ communication
    (PS: in practice, capacity per cell << 8*125 ! )
  See PART-III on ‘Filter Banks & … : Transmultiplexers’

- Etc..

= BOX FULL OF DSP/MATHEMATICS !!
(For only €25)

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**DSP in applications : Other…**

- Digital Communications
  - Wireline (xDSL, Powerline), Wireless (GSM, 3G, Wi-Fi, WiMax, CDMA, MIMO-transmission,..)
- Speech
  - Speech coding (GSM, DECT, ..), Speech synthesis (text-to-speech), Speech recognition
- Audio Signal Processing
  - Audio Coding (MP3, AAC, ..), Audio synthesis
  - Editing, Automatic transcription, Dolby/Surround, 3D-audio,..
- Image/Video
- ...

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Enabling Technology is:

- **Signal Processing**
  - 1G-SP: analog filters
  - 2G-SP: digital filters, FFT’s, etc.
  - 3G-SP: full of mathematics, linear algebra, statistics, etc...
- **Micro-/Nano-electronics**
- ...

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**DSP in applications**

**Signals & Systems Course**

**DSP-I**

**DSP-CIS**

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**DSP-II Aims/Scope**

- **Basic signal processing theory/principles**
  - filter design, filter banks, optimal filters & adaptive filters
  - as well as...
- **Recent/advanced topics**
  - robust filter realization, perfect reconstruction filter banks, fast adaptive algorithms, ...
- **Often `bird’ s-eye view’**
  - skip many mathematical details (if possible... 😊)
  - selection of topics (non-exhaustive)
Overview

• **Part I** : Introduction
  Chapter-1: Introduction
  Chapter-2: Signals and Systems Review
  Chapter-3: Acoustic Modem Project

• **Part II** : Filter Design & Implementation
  Chapter-4: IIR & FIR Filter Design
  Chapter-5: Filter Realization
  Chapter-6: Filter Implementation

Overview

• **Part III** : Filter Banks & Subband Systems
  Chapter-7: Filter Banks Intro/Applications (audio coding/CDMA/...)
  Chapter-8: Filter Banks Theory
  Chapter-9: DFT-modulated Filter Banks
  Chapter-10: Special Topics (frequency-domain processing, wavelets...)

[Diagram of subband processing]
Overview

- **Part IV**: Optimal & Adaptive Filtering
  - Chapter-11: Optimal/Wiener Filters
  - Chapter-12: Adaptive Filters/Recursive Least Squares
  - Chapter-13: Adaptive Filters/LMS
  - Chapter-14: ‘Fast’ Adaptive Filters
  - Chapter-15: Kalman Filters

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Lectures

**Lectures**: 15 * 2 hrs

**Course Material**:

- **Part I-IV**: *Slides* (use version 2012-2013 !!)
  - ...download from DSP-CIS webpage
- **Part IV**: ‘Introduction to Adaptive Signal Processing’
  - (Marc Moonen & Ian.K. Proudler)
  - = support material, not mandatory !
  - ...(if needed) download from DSP-CIS webpage

**PS**: Time budget = (15*2hrs)*4 = 120 hrs
Literature / Campus Library Arenberg

- A. Oppenheim & R. Schafer  
  ‘Digital Signal Processing’ (Prentice Hall 1977)  
- L. Jackson  
  ‘Digital Filters and Signal Processing’ (Kluwer 1986)  
- P.P. Vaidyanathan  
  ‘Multirate Systems and Filter Banks’ (Prentice Hall 1993)  
- Simon Haykin  
  ‘Adaptive Filter Theory’ (Prentice Hall 1996)  
- M. Bellanger  
  ‘Digital Processing of Signals’ (Kluwer 1986)  
- etc...

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Literature / DSP-CIS Library

- **Collection of books is available to support course material**
- **List/reservation via DSP-CIS webpage**
- **Contact:** beier.li@esat
Exercise Sessions: Acoustic Modem Project

- Digital communication over an acoustic channel (from loudspeaker to microphone)
- FFT/IFFT-based modulation format: OFDM (as in ADSL/VDSL, WiFi, DAB, DVB, ...)
- Channel estimation, equalization, etc...

Digital Picture (IN)

Tx

D-to-A
+filtering
+amplif.

A-to-D
+filtering
+

Rx

Digital Picture (OUT)

Exercise Sessions: Acoustic Modem Project

- Runs over 8 weeks
- Each week
  - 1 PC/Matlab session (supervised, 2.5hrs)
  - 2 ‘Homework’ sessions (unsupervised, 2*2.5hrs)

PS: Time budget = 8*(2.5hrs+5hrs) = 60 hrs

- ‘Deliverables’ after week 2, 4, 6, 8
- Grading: based on deliverables, evaluated during sessions

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  pepe.gilcacho@esat (English+Spanish)

PS: groups of 2
Will consider digital communications over acoustic channel:

- Discrete-time transmit signal (sampling rate $F_s$, e.g., 10kHz)
- Discrete-time receiver signal (sampling rate $F_s$, e.g., 10kHz)

This will be the easy part...
Will consider digital communications over acoustic channel:

Discrete-time transmit signal (sampling rate $F_s$, e.g., 10kHz)

...straightforwardly realized (in Matlab/Simulink with 'Real-Time Workshop', see below)

Means we do not have to deal with hardware issues, components, etc.

Acoustic Modem Project – Preview 3/8

...and will be modeled by a linear discrete-time transfer function

Acoustic Modem Project – Preview 4/8
• Will consider digital communications over an acoustic channel:

Acoustic Modem Project – Preview

- Will use OFDM as a modulation format

Orthogonal frequency-division multiplexing
From Wikipedia, the free encyclopedia

Orthogonal frequency-division multiplexing (OFDM), essentially identical to discrete multi-tone modulation (DMT), is a frequency-division multiplexing (FDM) scheme used as a digital multi-carrier modulation method. A large number of closely-spaced orthogonal sub-carriers are used to carry data. The data is divided into several parallel data streams or channels, one for each sub-carrier. Each sub-carrier is modulated with a conventional modulation scheme (such as quadrature amplitude modulation or phase-shift keying) at a low symbol rate, maintaining total data rates similar to conventional single-carrier modulation schemes in the same bandwidth. OFDM has developed into a popular scheme for wideband digital communication, whether wireless or over copper wires, used in applications such as digital television and audio broadcasting, wireless networking and broadband internet access.

- OFDM/DMT is used in ADSL/VDSL, WiFi, DAB, DVB …
- OFDM heavily relies on DSP functionalities (FFT/IFFT, …)
**Target:**
Design efficient OFDM based modem (Tx/Rx) for transmission over acoustic channel

**Specifications:**
- Data rate (e.g. 1kbits/sec), bit error rate (e.g. 0.5%),
- channel tracking speed, synchronisation, ...

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**Work Plan**
- Week 1: Introduction Matlab/Simulink
- Week 2: Acoustic channel measurement & modeling
  *deliverable*
- Week 3-4: OFDM transmitter/receiver design
  *deliverable*
- Week 5-6: OFDM over acoustic channel
  *deliverable*
- Week 7-8: OFDM with adaptive equalization
  *deliverable*
Exam

• Oral exam, with preparation time
• Open book
• Grading:
  5 pts for question-1
  5 pts for question-2
  5 pts for question-3
  +5 pts for Acoustic Modem Project evaluation (p.24)

= 20 pts

September Retake Exam

• Oral exam, with preparation time
• Open book
• Grading:
  5 pts for question-1
  5 pts for question-2
  5 pts for question-3
  +5 pts for question-4 on Acoustic Modem Project

= 20 pts
Website

1) TOLEDO
2) homes.esat.kuleuven.be/~pepe/dsp-cis/2012-2013
   • Contact: pepe.gilcacho@esat
   • Slides
   • Project info/schedule
   • Exams
   • DSP-CIS Library
   • FAQs (send questions to pepe.gilcacho@esat or marc.moonen@esat)

Questions?

1) Ask Teaching Assistants (during exercises sessions)
2) Send questions to pepe.gilcacho@esat or marc.moonen@esat
3) Make appointment marc.moonen@esat
   ESAT Room 01.69