Digital Signal Processing

DSP

Chapter-1 : Introduction

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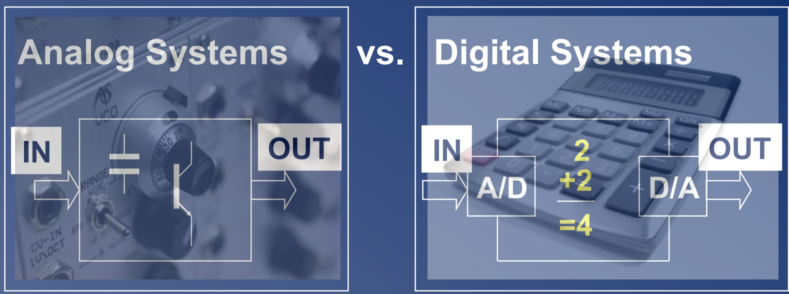
Chapter-1 : Introduction

- **Aims/Scope**
  - Why study DSP ?
  - DSP in applications : Mobile communications example
  - DSP in applications : Hearing aids example
- **Overview**
  - Filter design & implementation
  - Optimal and adaptive filters
  - Filter banks and subband systems
- **Lectures/course material/literature**
- **Exercise sessions**
- **Exam**

Why study DSP ?



- Can translate (any) analog (e.g. filter) design into digital
- Going `digital' allows to expand functionality/flexibility/…
  (e.g. speech recognition, audio compression…)

Why study DSP ?

- **Start with two `DSP in applications' examples:**
  DSP in mobile communications
  DSP in hearing aids

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

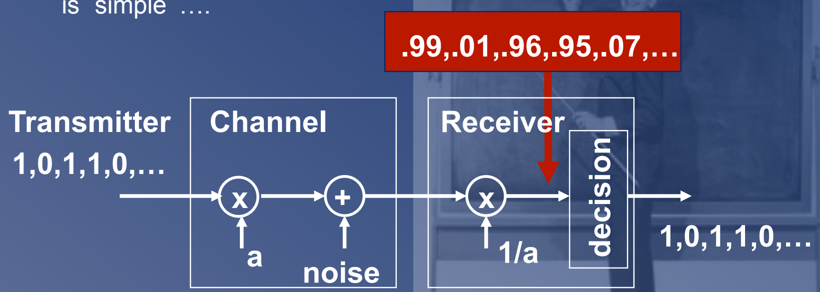
DSP in applications: Mobile Communications 1/10

Cellular Mobile Communications (e.g. GSM/UMTS/4G/...)

- **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: Mobile Communications 2/10

- **DSP for Digital Communications (`physical layer’) :**
  - A common misunderstanding is that digital communications is `simple’ ….

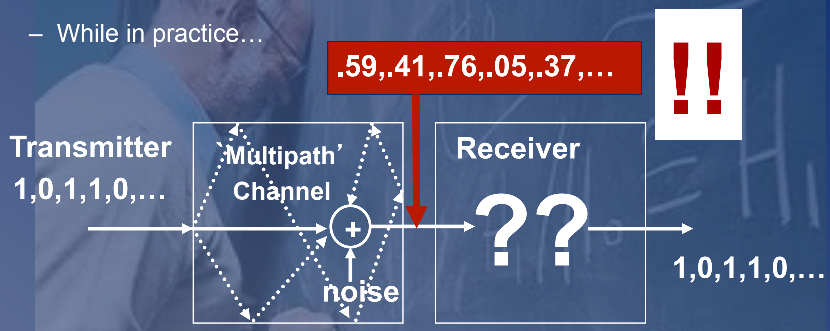


  - While in practice…

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

DSP in applications: Mobile Communications 3/10

- **DSP for Digital Communications (`physical layer’) :**
  - While in practice…



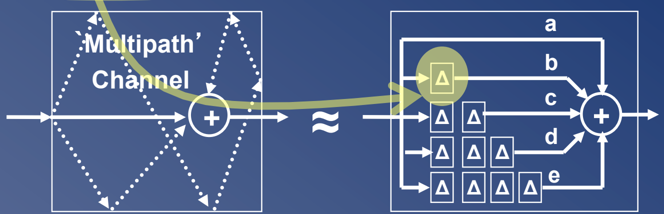
  - This calls for channel model + compensation (equalization)

DSP in applications: Mobile Communications 4/10

- **DSP Challenges: Channel Estimation/Compensation**
  - Multi-path channel is modeled with short (3…5 taps) FIR filter

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

(interpretation?)



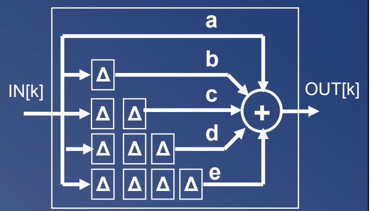
PS: z^{-1} or Δ represents a sampling period delay, see Chapter-2 for review on z-transforms

DSP in applications: Mobile Communications 5/10

- **DSP Challenges: Channel Estimation/Compensation**
  - Multi-path channel is modeled with short (3…5 taps) FIR filter

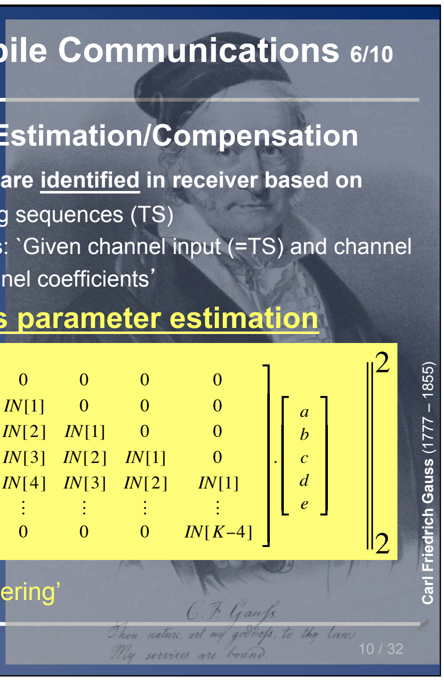
$H(z)= a+b.z^{-1}+c.z^{-2}+d.z^{-3}+e.z^{-4}$

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



=convolution

DSP in applications: Mobile Communications 6/10



- **DSP Challenges: Channel Estimation/Compensation**

Channel coefficients (a,b,c,d,e) are identified in receiver based on transmission of pre-defined training sequences (TS)
Problem to be solved at receiver is: `Given channel input (=TS) and channel output (=observed), compute channel coefficients'

This leads to a least-squares parameter estimation

$$\min_{a,b,c,d,e} \left\| \begin{bmatrix} OUT[1] \\ OUT[2] \\ OUT[3] \\ OUT[4] \\ OUT[5] \\ \vdots \\ OUT[K] \end{bmatrix} - \begin{bmatrix} IN[1] & 0 & 0 & 0 & 0 \\ IN[2] & IN[1] & 0 & 0 & 0 \\ IN[3] & IN[2] & IN[1] & 0 & 0 \\ IN[4] & IN[3] & IN[2] & IN[1] & 0 \\ IN[5] & IN[4] & IN[3] & IN[2] & IN[1] \\ \vdots & \vdots & \vdots & \vdots & \vdots \\ 0 & 0 & 0 & 0 & IN[K-4] \end{bmatrix} \cdot \begin{bmatrix} a \\ b \\ c \\ d \\ e \end{bmatrix} \right\|_2^2$$

See Chapter-6 on 'Optimal Filtering'

DSP in applications: Mobile Communications 7/10

- **DSP Challenges: Channel Estimation/Compensation**
  - Channel coefficients (cfr. a,b,c,d,e) are identified in receiver based on transmission of pre-defined training sequences (TS)
  - 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' (e.g. 100's times per sec)

= SPECTACULAR !!

DSP in applications: Mobile Communications 8/10

- **DSP Challenges: 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)

DSP in applications: Mobile Communications 9/10

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  - 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

    See Chapter-6 on 'Optimal Filtering'
  - Then transmit model coefficients instead of signal samples (!!!)
  - Synthesize speech segment at receiver (should `sound like' original speech segment)

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DSP in applications: Mobile Communications 10/10

- **DSP Challenges: Multiple Access Schemes**

  Accommodate multiple users by time & frequency `multiplexing'
  - FDMA: frequency division multiple access
  - OFDMA: orthogonal frequency division multiple access
  - TDMA: time division multiple access
  - CDMA: code division multiple access

- **etc..**

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

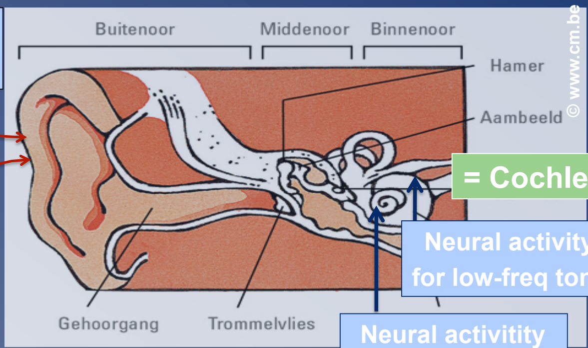
DSP in applications: Hearing Aids 1/10

Hearing

- Outer ear/middle ear/inner ear
- Tonotopy of inner ear: spatial arrangement of where sounds of different frequency are processed

Low-freq tone

High-freq tone



= Cochlea

Neural activity for low-freq tone

Neural activity for high-freq tone

DSP in applications: Hearing Aids 2/10

Hearing loss types:

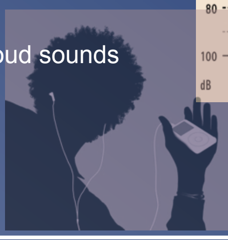
- Conductive
- Sensorineural
- Mixed

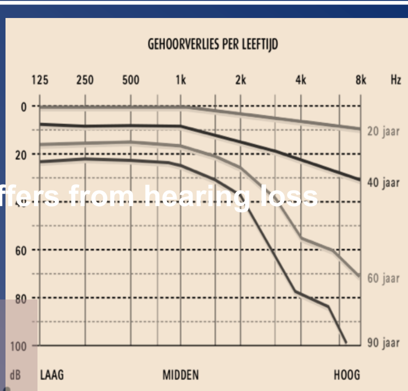
One in six adults (Europe) suffers from hearing loss

…and still increasing

Typical causes:

- Aging
- Exposure to loud sounds
- …





[Source: Lapperre]

DSP in applications: Hearing Aids 3/10

➔Hearing Aids (HAs)

- Audio input/audio output (`microphone-processing-loudspeaker')
- 'Amplifier', but so much more than an amplifier!!
- History:
  - Horns/trumpets/…
  - `Desktop' HAs (1900)
  - Wearable HAs (1930)
  - Digital HAs (1980)
- State-of-the-art:
  - MHz's clock speed
  - Millions of arithmetic operations/sec, …
  - Multiple microphones

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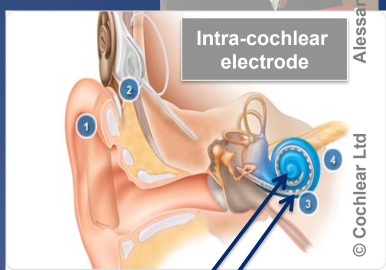
DSP in applications: Hearing Aids 4/10

➔Cochlear Implants (Cis)

- **Audio input/electrode stimulation output**
- Stimulation strategy + preprocessing similar to HAs
- **History:**
  - Volta's experiment…
  - First implants (1960)
  - Commercial CIs (1970-1980)
  - Digital CIs (1980)
- **State-of-the-art:**
  - MHz's clock speed, Mops/sec, …
  - Multiple microphones



➔Other: Bone anchored HAs, middle ear implants, …

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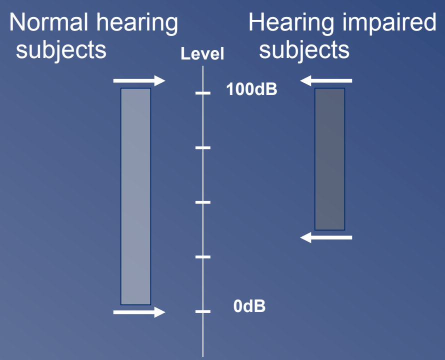
DSP in applications: Hearing Aids 5/10

DSP Challenges: Dynamic range compression

Dynamic range & audibility

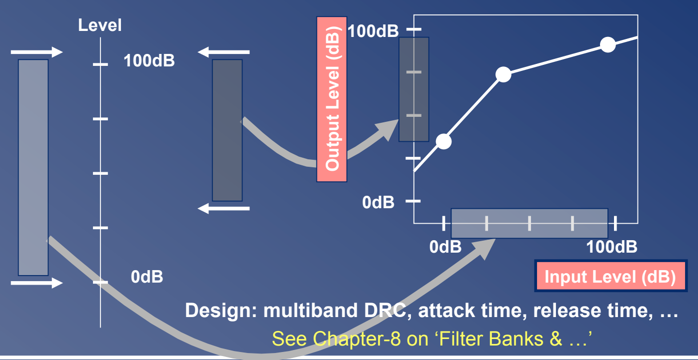


DSP in applications: Hearing Aids 5/10

DSP Challenges: Dynamic range compression

Dynamic range & audibility

need `signal dependent amplification'

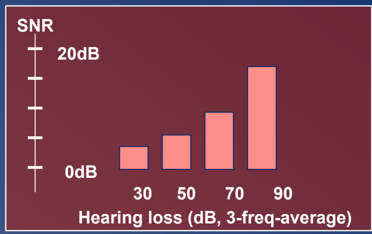


Design: multiband DRC, attack time, release time, …

See Chapter-8 on 'Filter Banks & …'

DSP in applications: Hearing Aids 6/10

- However: Audibility does not imply intelligibility
- Hearing impaired subjects need 5..10dB larger signal-to-noise ratio (SNR) for speech understanding in noisy environments

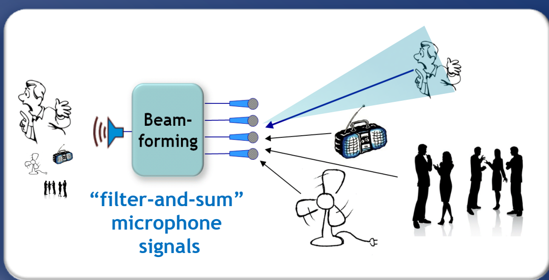


- Need for **noise reduction** (=speech enhancement) algorithms:
  - State-of-the-art: monaural 2-microphone adaptive noise reduction
  - Near future: binaural noise reduction (see below)
  - Not-so-near future: cooperative HAs wih multi-node noise reduction

DSP in applications: Hearing Aids 7/10

DSP Challenges: Noise reduction

Multimicrophone 'beamforming', typically with 2 microphones, e.g. 'directional' front microphone and 'omnidirectional' back microphone

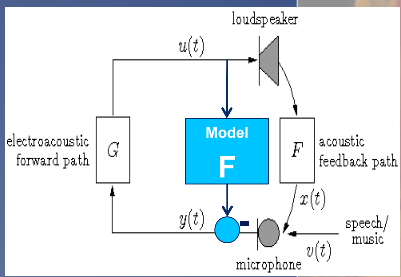


See Chapter-3 on '(Spatial) Filter Design'

DSP in applications: Hearing Aids 8/10

DSP Challenges: Feedback cancellation

- Problem statement: Loudspeaker signal is fed back into microphone, then amplified and played back again
- Closed loop system may become unstable (howling)
- Similar to feedback problem in public address systems (for the musicians amongst you)



Similar to echo cancellation in GSM handsets, Skype,…

but more difficult due to signal correlation

See Chapter-6 on 'Adaptive Filtering'

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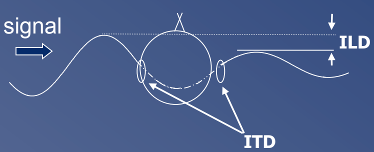
DSP in applications: Hearing Aids 9/10

Binaural hearing: Binaural auditory cues

- **ITD (interaural time difference)**
- **ILD (interaural level difference)**



- Binaural cues (ITD: f < 1500Hz, ILD: f > 2000Hz) used for
  - Sound localization
  - Noise reduction

    =`Binaural unmasking' ('cocktail party' effect)

    0-5dB

DSP in applications: Hearing Aids 10/10

DSP Challenges: Binaural hearing aids

- Two hearing aids (L&R) with wireless link & cooperation
- Opportunities:
  - More signals (e.g. 2*2 microphones)
  - Better sensor spacing (17cm i.o. 1cm)
- Constraints: power/bandwith/delay of wireless link
- Challenges:
  - Improved localization through ‘localization cue’ preservation
  - Improved noise reduction + benefit from ‘binaural unmasking’
  - **Signal selection/filtering**, **audio coding**, **synchronisation**, …

= SPECTACULAR !!

DSP in applications : Other…

- **Digital Communications**

  Wireline (xDSL,Powerline), Wireless (GSM, 3G, 4G, 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**
- **…**

Aims/Scope

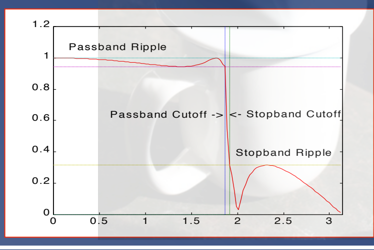
- **<u>Basic</u> signal processing theory/principles**
  Filter design, filter banks, optimal filters & adaptive filters

…as well as…

- **Recent/<u>advanced</u> 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)
- Prerequisites: Signals & Systems (sampling, transforms,..)

Overview

- **<u>Part I</u>** : Introduction
  Chapter-1: Introduction
  Chapter-2: Signals and Systems Review
- **<u>Part II</u>** : Filter Design & Implementation
  Chapter-3: Filter Design
  Chapter-4: Filter Realization
  Chapter-5: Filter Implementation

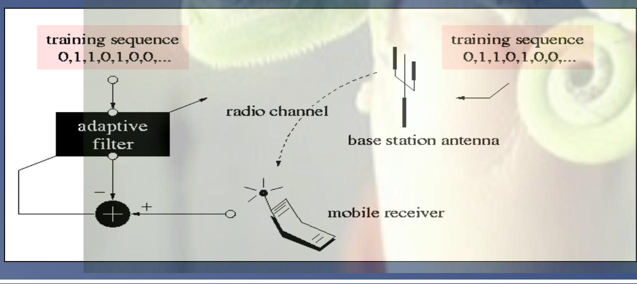


Overview

- **Part III** : Optimal & Adaptive Filtering

  Chapter-6: Wiener Filters & the LMS Algorithm

  Chapter-7: Recursive Least Squares Algorithms

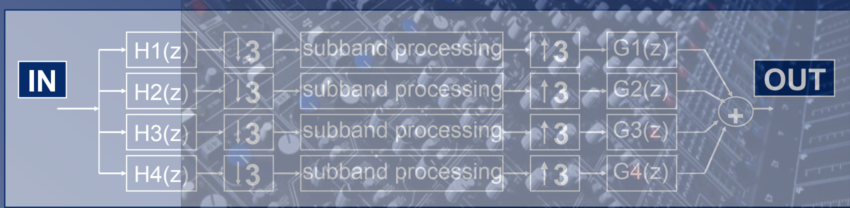


Overview

- **Part IV** : Filter Banks & Subband Systems

  Chapter-8: Filter Bank Preliminaries/Applications

  Chapter-9: Filter Bank Design



Lectures

Lectures: 2 + 8*1.5 hrs = 14 hrs

Course Material:

- Slides

http://homes.esat.kuleuven.be/~dspuser/DSP-CIS/2016-2017:

  – Optional reading: `Introduction to Adaptive Signal Processing' (Marc Moonen & Ian.K. Proudler)
  – Lectures 2 & 4 with audio

Literature

- **A. Oppenheim & R. Schafer (*)**
  `Digital Signal Processing' (Prentice Hall 1977)
- **L. Jackson**
  `Digital Filters and Signal Processing' (Kluwer 1986)
- **Simon Haykin**
  `Adaptive Filter Theory' (Pearson Education 2014)
- **P.P. Vaidyanathan**
  `Multirate Systems and Filter Banks' (Dorling Kindersley 1993)
- **M. Bellanger**
  `Digital Processing of Signals' (Kluwer 1986)
- **etc...**

(*) MOOC www.edx.org/course/discrete-time-signal-processing-mitx-6-341x-1