Aalto University

School of Electrical Engineering
Department of Signal Processing and Acoustics

S-88.4212 Signal Processing in Telecommunications II

Fall 2013

Lecture 1: Introduction

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Contents of Lecture 1

I Course arrangements
II Review of digital communication channels and systems

Course Arrangements

Course in the Curriculum

- Part of major ‘Signal Processing in Telecommunications’
- Preceding course S-88.2311 *Signal Processing in Telecommunications* I introduced DSP methods (pulse shaping, equalization etc.) for both wireline and wireless transmission
- This course continues with *synchronization, channel estimation* and related topics

Prerequisites

In order to be able to follow and pass the course, you should have passed the following course:

- **S-88.2311 Signal Processing for Telecommunications I**

and its prerequisite

- S-72.1140 Transmission Methods in Telecommunication Systems

or have *equal knowledge* of communications systems and signal processing methods

Goals of the course

Understanding the principles of important DSP functions in receivers and their implementation:

- Maximum Likelihood (ML) estimation and Cramer-Rao bounds
- Carrier frequency estimation
- Carrier phase estimation
- Symbol timing estimation
- Channel estimation principles
- Solutions in practical systems

Related courses

Suggested supporting course:

- S-88.4200 *Statistical Signal Processing* (Spring term)
- Gives solid theoretical background also to topics of our course

Teaching

- *Lectures:* Tuesdays (F201) & Thursdays (I346) 12-14
  - Send e-mail if questions or problems: stefan.werner@aalto.fi
  - Reception Mondays 10-11 or by e-mail appointment (G410)
- *Exercise sessions:*
  - 6 sessions (*some* Tuesdays & Thursdays 14-16)
  - Assistant: Pramod Jacob Mathecken, pramod.mathecken@aalto.fi
  - Reception by e-mail appointment
- *Homework* (voluntary)
- *MATLAB project* (compulsory!)

Time flow

- Lectures available in the web before lecture (hopefully!)
- Problem sheet published before exercise session:
- 1. Simple pre-assignments (no points!)
  - You can solve before exercise session
- 2. Pencil-and-paper exercises
  - solve in the guidance of the course assistant
- 3. Homework to be returned in one week after the session
  - more interesting and slightly more laborious
- MATLAB exercise
  - introduction in a special session, with guide document
  - report deadline next January (details later)

Exercises and Homework

- The homeworks will be graded by the assistant and they contribute positively to the final course grade
- Late homeworks will not be accepted (if you have a good reason, negotiate *beforehand*)
- Double effect of exercises:
  - earn extra points
  - do better in the exam

Final grade

- Exam structure: 5 problems/essays/sets of small questions of 6p each
- Exam (30p + extra 3p):
- Exercise attendance (30p):
- Homework (30p)
- MATLAB project, no bonus (grading pass/fail)
- Grading scale: 14=1, 18=2, 22=3, 26=4, 30=5

- Final grading rule (weighted average):

  Final points: $Pf = Pe + 2/3 \times 4 \times Phw/30 + 1/3 \times 4 \times Pa/30$

Passing as postgraduate course

- The course can be passed as a postgraduate course as well

What You Should Do

- Register using WebOodi
  - enables announcements via e-mail
- Follow web pages for information (notice board updated less frequently)
- Register for the exam (every time!)
- Be active:
  - Questions and comments in the class (if you don't ask the lecturer, the lecturer will ask you!)
  - Attend exercises, do homework, MATLAB project
  - Make questions, comments, feedback also by e-mail

Literature

- Lectures (= slide shows)
- Exercises, with solutions
- Other recommended reading (available in faculty library):
  - U. Mengali and A. N. D'Andrea, *Synchronization Techniques for Digital Receivers*, Plenum Press 1997
  - H. Meyr *et al*, *Digital Communication Receivers*, Wiley 1998
  - S. Kay: *Fundamentals of Statistical Signal Processing: Estimation Theory*, Prentice-Hall 1993
  - J. R. Barry, E. A. Lee and D. G. Messerschmitt, *Digital Communication*. 3rd edition, Kluwer 204
  - J. G. Proakis, *Digital Communication Systems*, 3rd ed., McGraw-Hill, 1995.

Lecture Plan and Timetable

Changes possible!

L1	**29.10.**	Introduction; review of communication systems
L2	**31.10.**	ML estimation principles
L3	**05.11.**	Synchronization: Overview
L4	**07.11**	Carrier frequency estimation I
L5	**12.11.**	Carrier frequency estimation II
L6	**14.11.**	Carrier phase estimation I

Lecture Plan and Timetable...

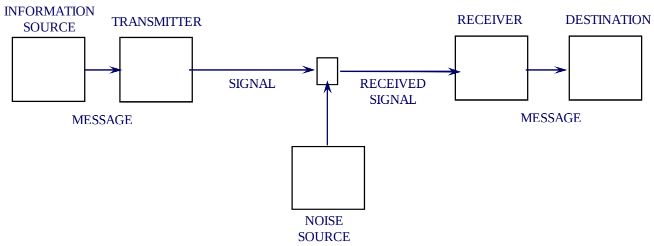
L7	**19.11.**	Carrier phase estimation II
L8	**21.11.**	Symbol timing estimation I
L9	**26.12.**	Symbol timing estimation II
L10	**28.12.**	Channel estimation I
L11	**03.12.**	Channel estimation II, course review
E	**12.12**.	**Thursday** 12-15 in I346

Exercise Plan

- **E1** **05.11** Maximum likelihood estimation
- **E2** **07.11** Synchronization
- **E3** **14.11** Carrier frequency estimation
- **E4** **21.11** Carrier phase estimation
- **E5** **28.12** Symbol timing estimation
- **E6** **03.12** Channel estimation

Models for digital communication channels and systems

Classical communication system model



- Claude Shannon 1948: founding of information theory
- Basic framework for digital transmission of information

Classical communication system...

- Assume digital signal (= sequence of numbers - *sampling* and *quantization* often needed!)

Shannon's basic ideas:

- *Source coding theorem*: Any digital source of certain bit rate can be compressed down to a minimum bit rate = *source entropy* (no loss of information!)
- *Channel coding theorem*: Error-free transmission is possible at a rate of lower than or equal to the *channel capacity*

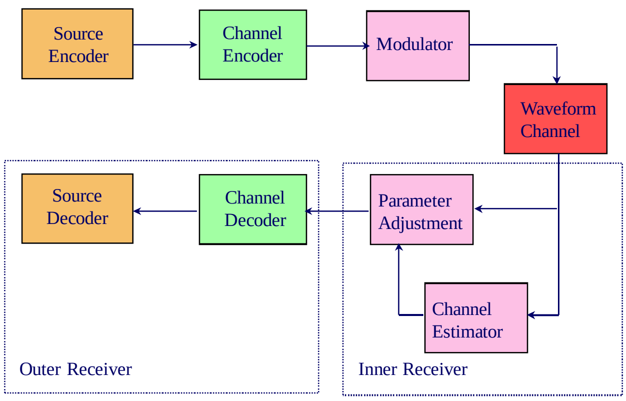
Classical communication system...

- Channel coding theorem implies that, with complex enough processing and long enough delay, the probability of errors can be made as small as desired
- Shannon did not tell *how* to do it (techniques still in development!)

Problem with the classical model:

- The model does not show important functions that are needed in practical systems interfacing analog sources, channels and sinks

Physical communication system model



- Meyr at al.: *Digital Communication Receivers* (1998)

Physical communication system...

Outer receiver:

- Implements the ‘classical part’ of the receiver: channel and source decoding

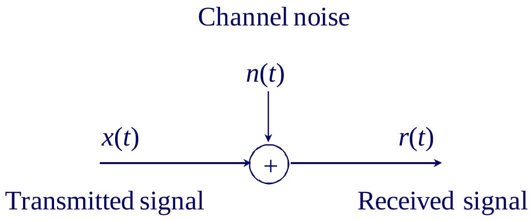
Inner receiver:

- Provides (raw) digital estimates by processing the analog waveform received from the channel
- Necessary preprocessing for the outer receiver (**synchronization, channel estimation,** equalization, removal of noise and interference)
- DSP techniques needed!

Also in this course, we focus on the inner receiver

Channel models

1. Additive Noise Channel



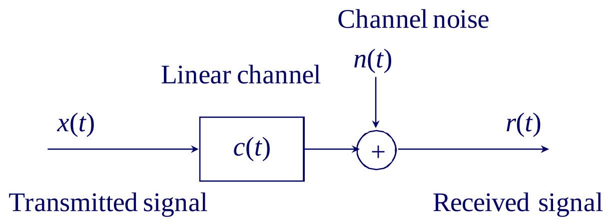
- Simplest channel model: noise is added in the channel
- Often Gaussian (normal) amplitude distribution assumed
- AWGN: additive white Gaussian noise: spectrally white (= consecutive samples completely uncorrelated)

1. Additive Noise Channel...

- Add attenuation factor *a* in the model
- Corresponding equation:

$$r(t) = ax(t) + n(t)$$

2. Linear Filter Channel



- Characterized by linear (continuous-time) filter with impulse response $c(t)$
- Also additive noise included

2. Linear Filter Channel...

- The received signal can be expressed with *convolution*

$$r(t) = \int_{-\infty}^{\infty} c(\tau)x(t-\tau)d\tau + n(t)$$

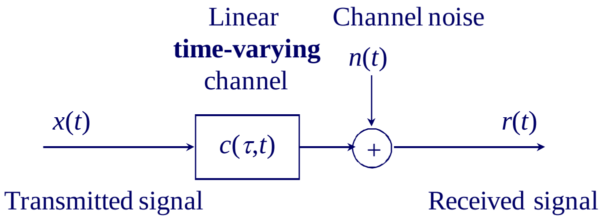
$$\equiv x(t) * c(t) + n(t)$$

- In the frequency domain:

$$R(f) = C(f)X(f) + N(f)$$

$$C(f) = \int_{-\infty}^{\infty} c(t)e^{-j2\pi ft}dt$$

3. Linear Time-Varying Filter Channel



- Characterized by linear filter with *time-varying* impulse response $c(\tau,t)$
- $c(\tau,t)$ is the response of the channel at time t to an impulse applied at time $t-\tau$ (= τ sec *before*)

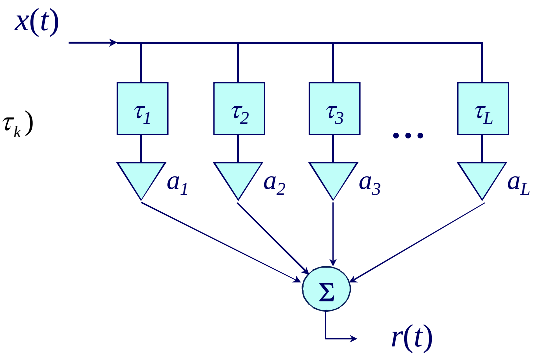
3. Linear Time-Varying Filter Channel...

- If the channel is *slowly varying*, we can define (average) impulse and frequency responses which are valid for a certain period time with certain precision
- Accurate modelling of fast varying channels is complicated
- Common *multipath model*:

$$c(\tau;t) = \sum_{k=1}^{L} a_k(t)\delta(\tau - \tau_k)$$

3. Linear Time-Varying Filter Channel...

$$c(\tau;t)=\sum_{k=1}^{L}a_k(t)\delta(\tau-\tau_k)$$



$a_k(t)$ = time-varying attenuation (fading) factors

τ_k = time delays (possibly time-varying)

Questions and tasks for the course

- Find useful signal and channel models, including relevant sync parameters
- Baseband and passband transmission
- ML Estimation principle for derivation of algorithms
- Theoretical limits (Cramer-Rao bounds etc.)
- Use of training signals and frame structure
- Evaluation of algorithm performance
- Examples of practical system solutions

Summary

Today we discussed

- Course arrangements
- Models for digital communication systems and channels

Next time:

- Maximum likelihood (ML) estimation principles