Helsinki University of Technology
Signal Processing Laboratory

S-38.411 Signal Processing for Telecommunications I

Spring 2000

Lecture 1: Introduction

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Contents of Lecture 1

I Course arrangements
II Models for digital communication channels and systems

Course in the Curriculum

- Part of 'Line Transmission and Signal Processing' path
  - S-72.244 *Modulation and Coding Methods* introduces basic modulation methods and coding
  - S-72.227 *Digital Communication Systems* concentrates on wireline transmission systems
  - S-72.232 *Radio Communication Systems* focuses on wireless systems
- Our course S-38.411 *Signal Processing for Telecommunications* I introduces digital signal processing (DSP) methods for both wireline and wireless transmission

Course in the Curriculum...

- Our course replaces S-38.211 (was in Finnish)
- The course is also part of the new Master's program
- New major 'Signal Processing for Telecommunications' starting Fall 2000

Goals of the course

Understanding the role, the limits and the basic techniques of DSP in practical telecommunication systems

- Models for *communications systems* and *physical channels*
- *Channel capacity* and the limits for DSP
- *Tx&Rx filtering:* matched filter, Nyquist criterion
- *Adaptive filters & equalizers:* principles, implementation
- *Echo cancellation*
- *Optimal nonlinear receivers* (Viterbi algorithm)
- *Practical DSP systems*: 3 guest lectures from industry

Prerequisites

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

- **S-72.244 *Modulation and coding methods***
- S-72.116 *Signals and Systems*
  (or Tik-61.140 *Signal Processing Systems*)
- Tik-61.246 *Digital Signal Processing and Filtering*

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

Teaching

- *Lectures:* Tuesdays 12-14 in S5 (prof. Timo Laakso)
- *Exercise sessions:* (MSc Stefan Werner, stefan.werner@hut.fi)
  - ca. 5 sessions (time & place to be notified later)
  - additional homework problems (improve the grade)
- *MATLAB Project:*
  - implementation of a simple adaptive equalizer for a communication link
  - detailed written instructions will be provided and most of the MATLAB code
  - live demonstration will be arranged
  - affects the final grade

How to Pass the Course

- Registrate with www-Topi
  - enables possible announcements via e-mail
- Grading: Exam + project (+ voluntary homework)
- Course material included in the exam
  - lectures (also the guest lectures!) + exercises
  - lecture notes (by T. Laakso, in preparation)
  - all available through ‘Opetusmonisteet’
  - most of the material also in the course web pages
- Course material assistant: (raise your hand!)

Final grade

- Passed exam *and* project required for passing the course
- Final grade formula:

$$A_{tot} = 0.8 \times A_{exam} + 0.2 \times A_{project}$$

- Exam structure: 5 problems/essays/sets of small questions of 6p each
- Grading: 14=1, 18=2, 22=3, 26=4, 30=5
- Solved homework can gain *extra* ca. 5p for the exam

Literature

- Lectures (= slide shows)
- Exercises, with solutions
- Additional lecture notes (by T. Laakso, in preparation)
- Available in *Opetusmonisteet* (all) and www pages (most)
- Other recommended reading:
  - E. A. Lee and D. G. Messerschmitt, *Digital Communication.* 2nd edition, Kluwer 1994
  - S. Haykin: *Adaptive Filter Theory*, 3rd ed., Prentice-Hall 1996
  - J. G. Proakis, *Digital Communication Systems*, 3rd ed., McGraw-Hill, 1995
  - H. Meyr *et al*, *Digital Communication Receivers*, Wiley 1998.

Lecture Plan and Timetable

Changes possible!

L1	1.2.	Introduction; models for channels and communication systems
L2	8.2.	Channel capacity
L3	15.2.	Transmit and receive filters for bandlimited AWGN channels
L4	22.2.	Optimal linear equalizers for linear channels 1
L5	29.2.	Optimal linear equalizers for linear channels 2
L6	7.3.	Adaptive equalizers 1
L7	14.3.	Adaptive equalizers 2

Lecture Plan and Timetable...

L8	21.3.	Nonlinear receivers 1: DFE equalizers
L9	28.3.	Nonlinear receivers 2: Viterbi algorithm
L10	4.4.	**Guest lecture 1**: DSP for Fixed Networks
		Lic. Tech. Matti Lehtimäki, Nokia Networks
L11	11.4.	**GL2**: DSP for Digital Subscriber Lines
		Dr. Janne Väänänen, Tellabs
L12	18.4.	**GL3**: DSP for CDMA Mobile Systems
		Dr. Kari Kalliojärvi, Nokia Research Center
L13	9.5.	Course review, questions, feedback
E	24.5.	(Wed) 9-12 S4 **Exam**

Exercise Timetable

- To be notified later by Course Assistant Stefan Werner (follow web pages!)
- Homework problems to be solved and returned in 2 weeks after each exercise session
- MATLAB Project info also later (deadline for returning project report ca. 15 June)

End of Course Information

- Questions?

Digital vs. Analog Communications

- *Digital communications* is replacing earlier ‘analog’ techniques in all fields of applications
- *Digital signal processing (DSP)* enables use of powerful algorithms that utilize the transmission capacity of a physical channel efficiently
- *Analog design* is based on use of simple components
  $\rightarrow$ performance often far from optimal
- *Digital design* can be based on a mathematical formulation of the transmission problem and solution with efficient adaptive algorithms
  $\rightarrow$ performance can be close to optimal

Examples of Communication Systems

- PCM (Pulse Code Modulation) Technology in Telephone Networks since 1960’s
- ISDN (Integrated Services Digital Network): first digital subscriber connection
- Digital Subscriber Lines (DSL’s)
  – HDSL, ADSL, VDSL, etc. = xDSL
  – utilize existing telephone lines (copper wiring)
  – transmission capacity 1 … 50 Mbit/s

Examples of Communication Systems...

- Mobile communications
  - 2nd generation markets growing fast:
  - GSM = Global System for Mobile Telecommunications
  - speech services + little data
  - 3rd generation in development:
  - UMTS = Universal Mobile Telecommunication System
  - WCDMA= Wideband Code Division Multiple Access
  - much improved data services and terminals (?)
  - multimode terminals and networks

Examples of Communication Systems...

- Digital Audio Broadcasting
  - replaces FM transmission early 2000's

- Digital Television and Video
  - digital terrestrial and satellite radio transmission
  - cable TV networks
  - cable modems: 2-10 Mbit/s data transmission: alternative to xDSL techniques

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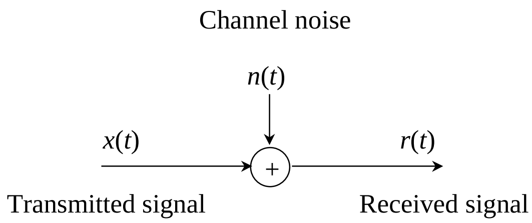
Channel models

1. Additive Noise Channel



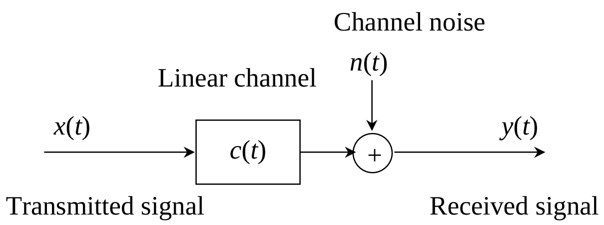
- Simplest channel model: noise is added in the channel
- Often Gaussian (normal) amplitude distribution assumed
- AWGN: additive white Gaussian noise: spectrally white (= consecutive samples completely uncorrelated)

1. Additive Noise Channel...

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

$$r(t) = \alpha x(t) + n(t) \qquad (1.1.)$$

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*

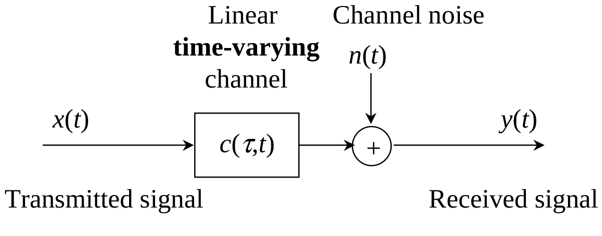
$$
\begin{aligned}
r(t) &= \int_{-\infty}^{\infty} c(\tau)x(t-\tau)\,d\tau + n(t) \\
&\equiv x(t) * c(t) + n(t)
\end{aligned}
\qquad (1.2)
$$

- In the frequency domain:

$$R(f) = C(f)X(f) + N(f) \qquad (1.4)$$

$$C(f) = \int_{-\infty}^{\infty} c(t)e^{-j2\pi ft}\,dt \qquad (1.3)$$

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*)

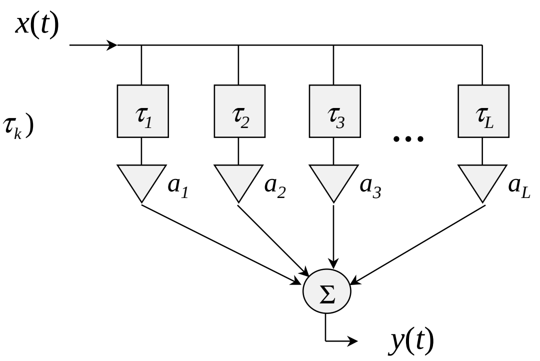
3. Linear Time-Varying Filter Channel...

- 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*:

$$c(\tau;t) = \sum_{k=1}^{L} a_k(t)\delta(\tau - \tau_k) \qquad (1.6)$$

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)

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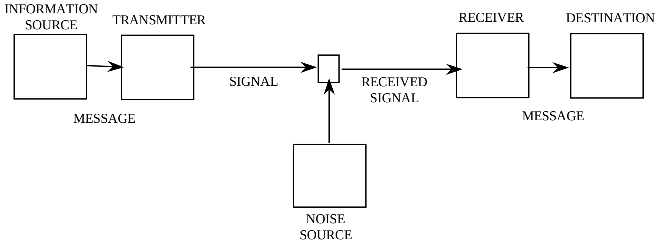
Communication system models

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* 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 the rate of lower than or equal to the *channel capacity*

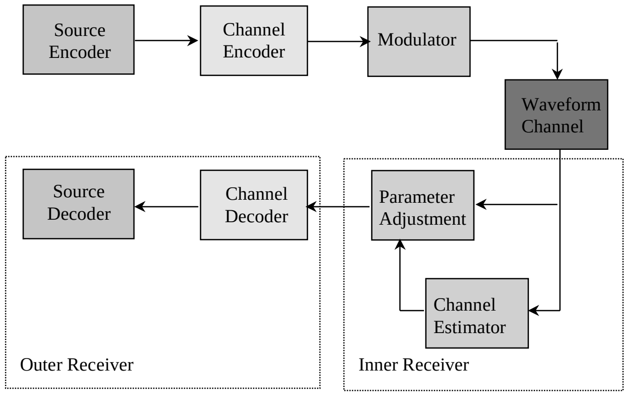
Classical communication system...

- Channel coding theorem is 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 et 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
- This is where (most of the) DSP is!

In this course, we focus on the inner receiver

Summary

Today we discussed

- Course arrangements
- Models for digital communication channels and systems

Next time:

- Channel capacity