332:521 – Digital Signal Analytics – Spring 2021

offered remotely in synchronous mode via Zoom – Mondays, 12:00–3:00 PM

Course Description

This course is a basic introduction to digital signal processing. It covers sampling and reconstruction; antialiasing prefilters and anti-image postfilters; discrete-time signals and systems; convolution, block-based and sample-based real-time processing; circular delay-line buffers; digital audio effects; z-transforms; filter realizations; quantization effects; DTFT, DFT, FFT, circular convolution; fast convolution; spectrum estimation, frequency resolution and windowing, periodogram averaging and smoothing; STFT applications, phase vocoder, time and pitch-scale modification; DCT, MDCT and time-domain aliasing cancellation, Princen-Bradley windows, data compression; FIR and IIR filter design methods; normalized lattice filters; introduction to adaptive filters and neural networks; multirate systems, interpolation, decimation, oversampling, polyphase filters, sample-rate conversion, noise-shaping delta-sigma quantizers; sparse modeling, sparse regularization methods, such as lasso/basis-pursuit-denoising.

Topics	Lectures
sampling, reconstruction, prefilters, postfilters	2
LTI systems review, convolution, stability, causality	2
z-transforms, transfer functions, frequency response, transients	2
filter realizations, circular buffers, audio effects	2
DTFT, frequency resolution, windowing, spectrum estimation	2
DFT, IDFT, FFT, circular convolution, fast convolution	2
STFT, phase vocoder, time & pitch scale modification	2
DCT, MDCT, data compression, time-domain aliasing cancellation	2
FIR & IIR filter design methods, normalized lattice filters	3
adaptive filtering and prediction, noise canceling, neural networks	3
multirate systems, interpolation, sample-rate conversion	2
decimation, noise-shaping delta-sigma quantizers	1
sparse modeling, sparse regularization methods	3

Texts

S. J. Orfanidis, *Introduction to Signal Processing*, Prentice Hall, 1996,
available freely from: `https://www.ece.rutgers.edu/~orfanidi/intro2sp/` (2010).

S. J. Orfanidis, *Applied Optimum Signal Processing*, online text, 2018,
available freely from: `https://www.ece.rutgers.edu/~orfanidi/aosp/`.

A. V. Oppenheim and R. W. Schafer, *Discrete-Time Signal Processing*, 3/e, Prentice Hall, 2009.
A low-cost paperback edition is available, ISBN-13: 978-9332535039.

We will be referring to these books as I2SP, AOSP, and O&S, respectively. Additionally, several class notes and papers will be assigned for reading.

Prerequisites

Familiarity with the concepts of LTI systems, convolution, block diagrams, Fourier, Laplace, and z-transforms, stochastic signals, autocorrelation functions, at the level of 16:332:501–Systems Analysis, or an undergraduate linear systems/DSP course. The course emphasizes computational aspects, and familiarity with a high-level programming language, such as MATLAB, is necessary. MATLAB is freely available to Rutgers students. The CVX convex optimization MATLAB package, available from `http://cvxr.com/cvx/`, will also be used.

Course Requirements

The final grade is based on:

1. Final exam
2. Two midterm exams
3. Computer projects

Computer Projects

The computer projects are an essential part of the course and may not be skipped or delayed. Project reports and other related materials, such as audio files, should be uploaded to Sakai Assignments by the due date. Please prepare your reports in PDF format using LaTeX (preferably) or Word, and observe the following guidelines:

a. Include a *discussion section* on the purposes and results of the project.

b. Any numerical and/or theoretical calculations and graphs must be presented in the discussion section.

c. Source code must be attached as an Appendix *at the end* of the report. (Please never attach numerical data listings – unless specifically asked.)

d. Please work alone. Collaboration with other students is not allowed.

	Project	Assigned	Due
1.	Sampling, prefilter/postfilter design, quantization & dithering	2/01	2/08
2.	Time-constants, bandwidth, filter-lag, zero-lag filters, noise reduction	2/08	2/15
3.	Dynamic range control, compressors, limiters, expanders, duckers	2/15	2/22
4.	Spectrum estimation, periodogram averaging and smoothing	2/22	3/01
5.	Fast convolution, overlap-add & overlap-save methods	3/01	3/08
6.	STFT, phase vocoder, time-scale and pitch-scale modification	3/08	3/22
7.	Data compression, DCT, MDCT, time-domain aliasing cancellation	3/22	3/29
8.	Filter design methods, normalized lattice, quantization effects	3/29	4/05
9.	Adaptive filtering and prediction, LMS and RLS algorithms	4/05	4/19
10.	Interpolation, decimation, oversampling, delta-sigma ADCs	4/19	4/26
11.	Trend extraction, regularized and sparse methods	4/26	5/10
12.	Sparse modeling, lasso/basis-pursuit-denoising	4/26	5/10

Academic Integrity

By taking this course *you agree* to work alone on your exams and projects, and *you accept and adhere* to the Rutgers academic integrity policy described in:

`http://academicintegrity.rutgers.edu/academic-integrity-at-rutgers/`

Please read also the IEEE code of ethics that should guide your professional life:

`https://www.ieee.org/about/corporate/governance/p7-8.html`

Instructor

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Office hours by email.

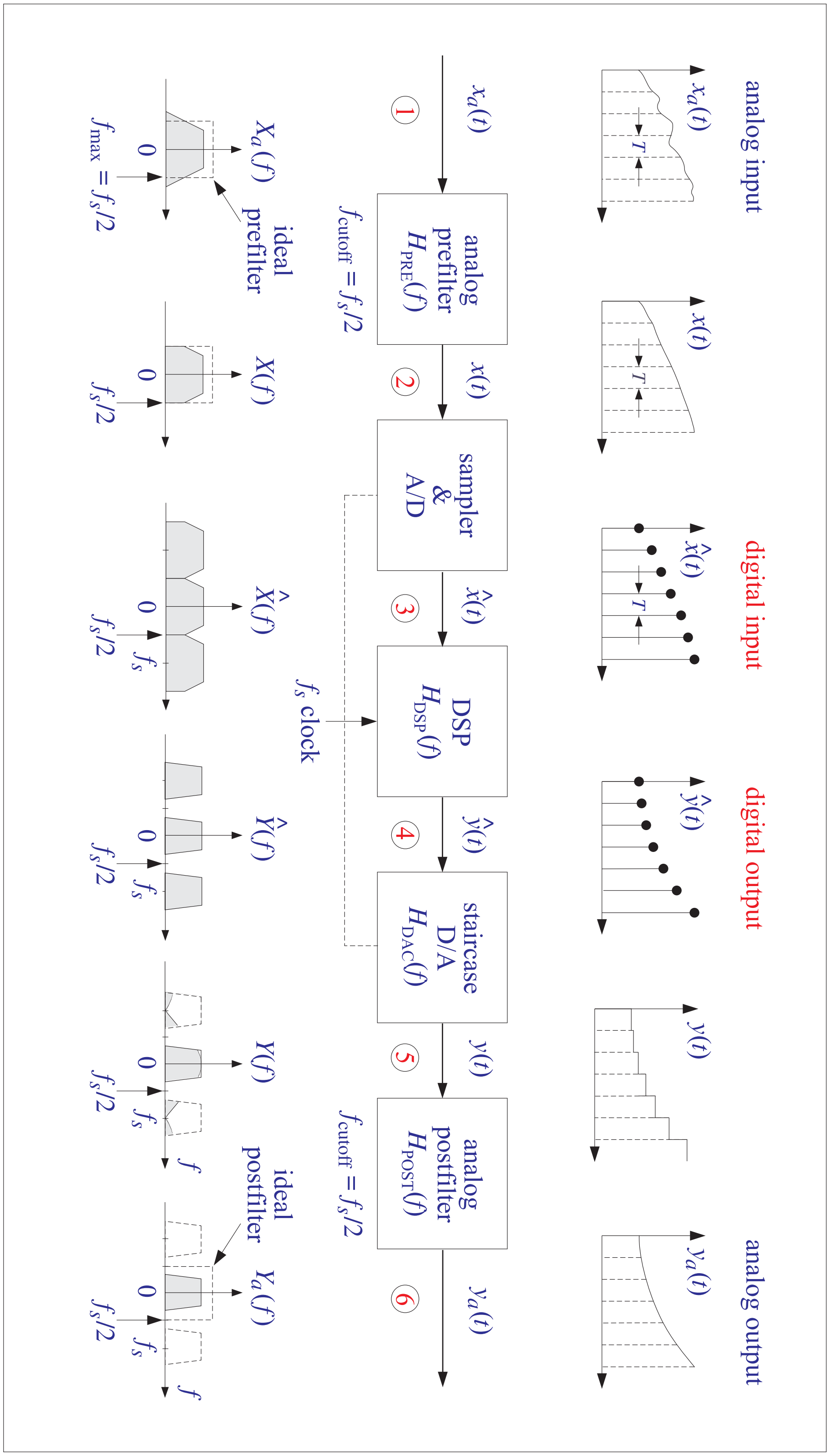


Fig.1 – Conventional DSP system.

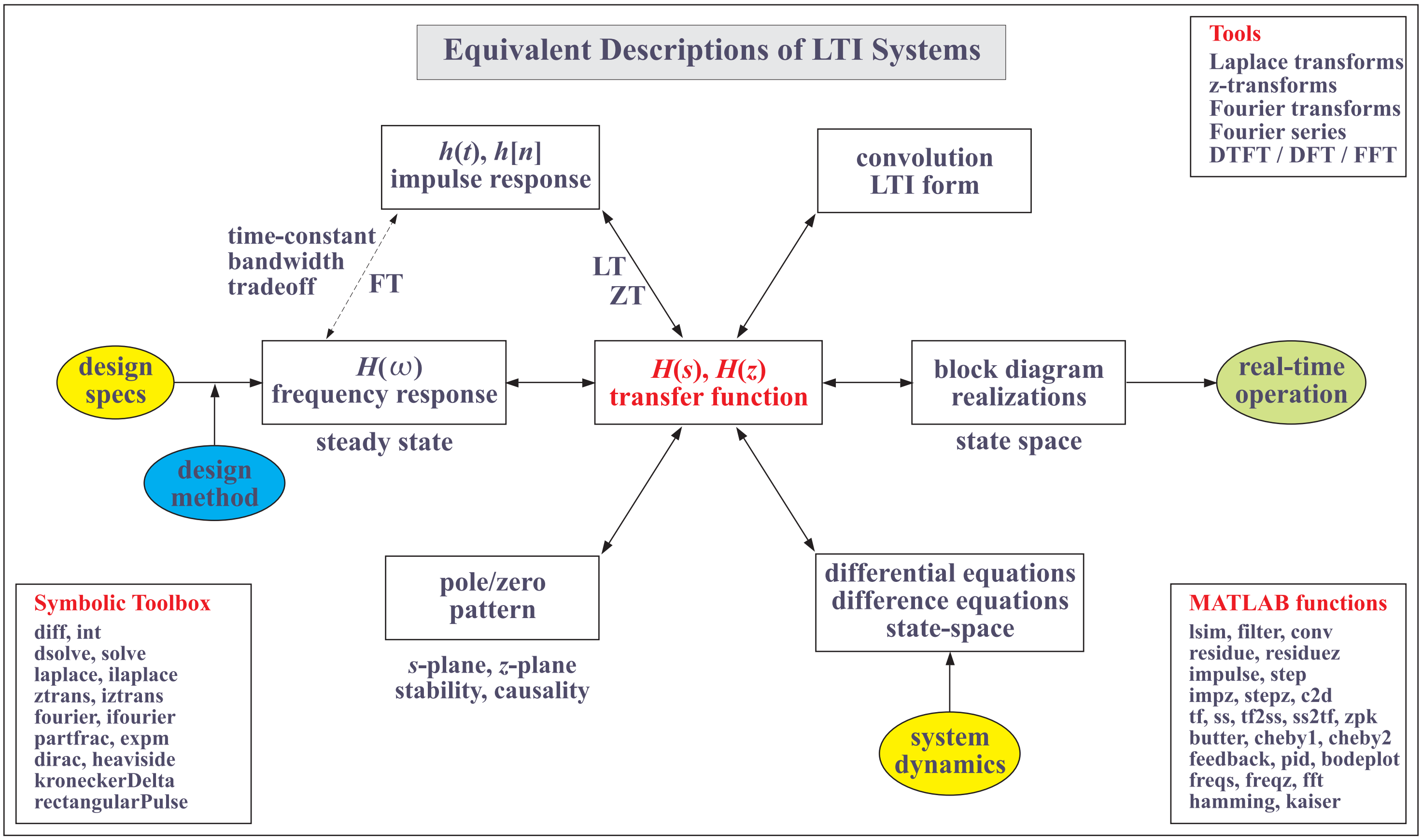


Fig.2 – Equivalent descriptions of continuous-time and discrete-time LTI systems.

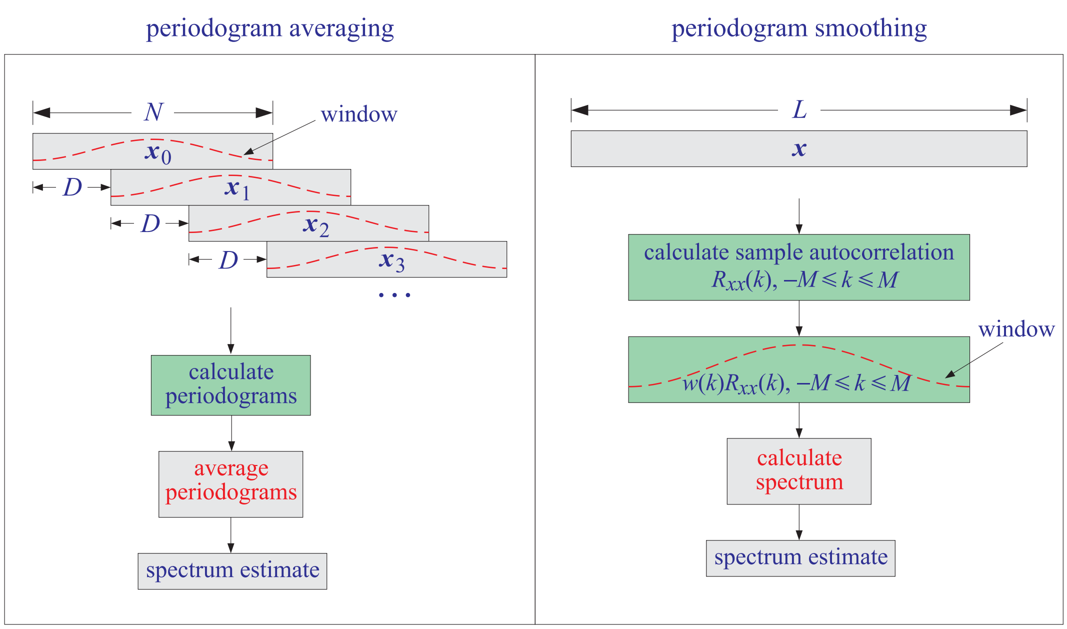


Fig.3 – Spectrum estimation by periodogram averaging and smoothing.

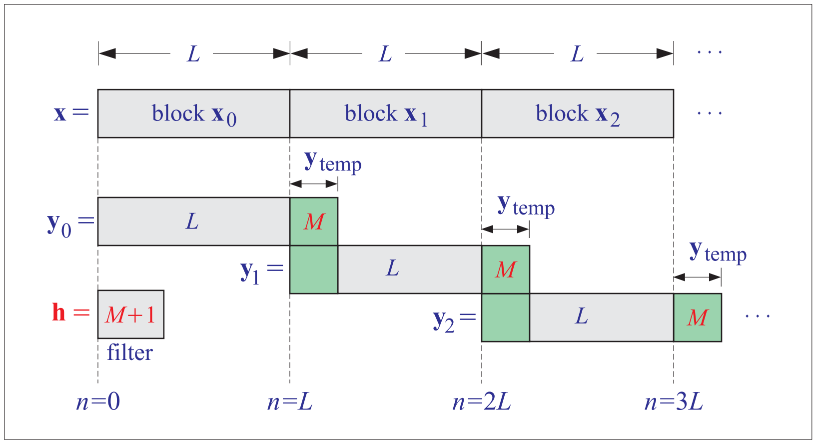


Fig.4 – Overlap-add block convolution.

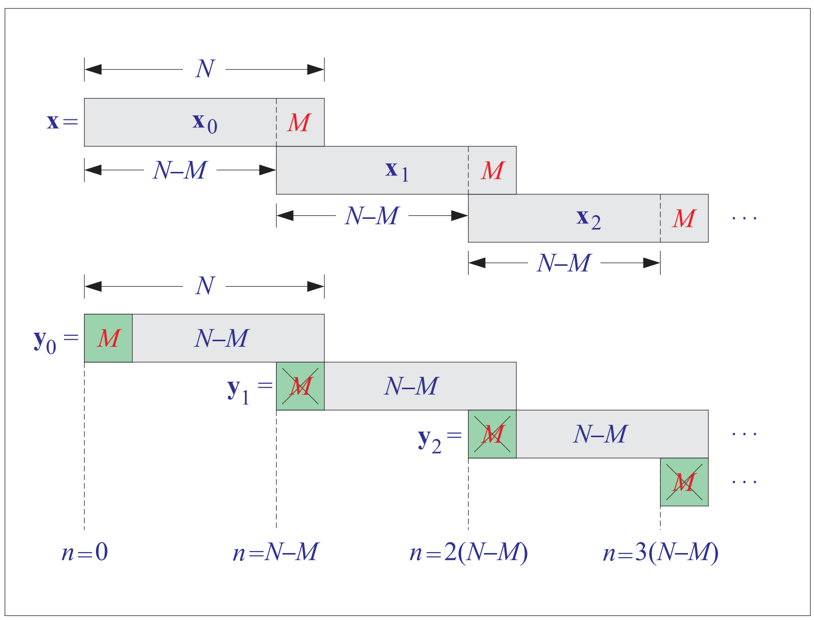


Fig.5 – Overlap-save block convolution.

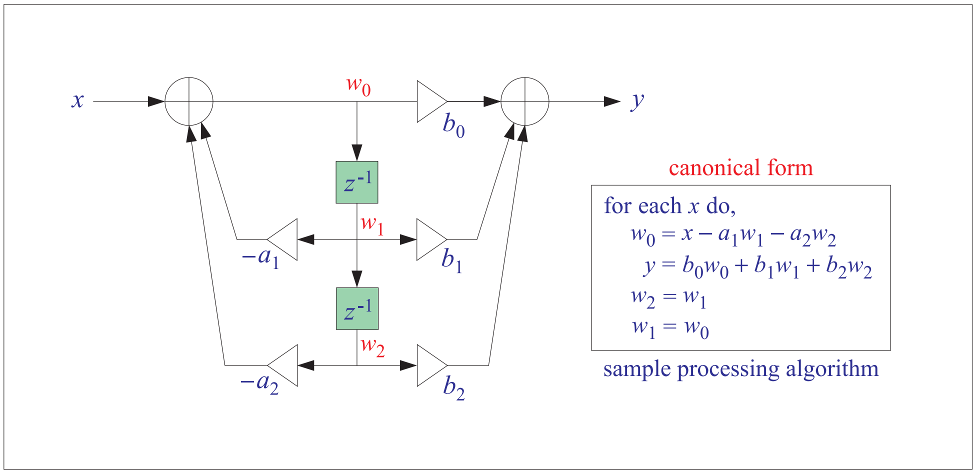


Fig.6 – Canonical realization.

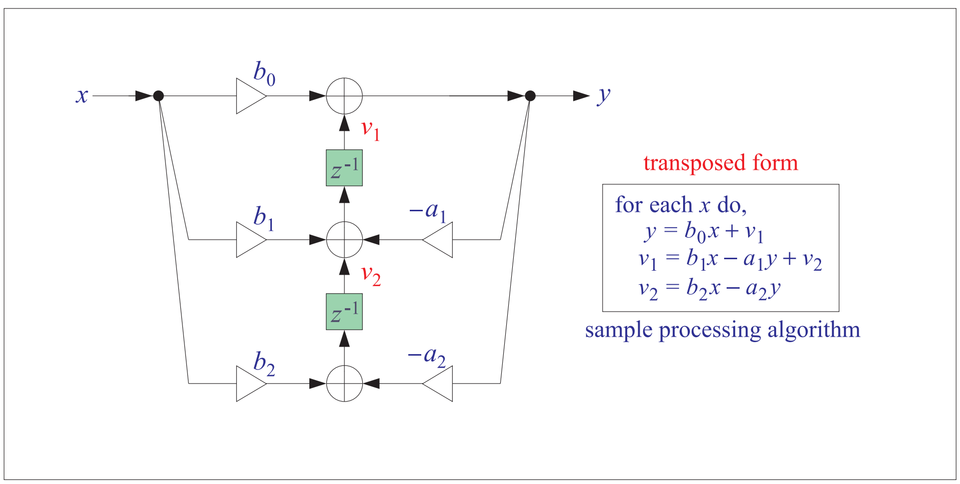


Fig.7 – Transposed realization.

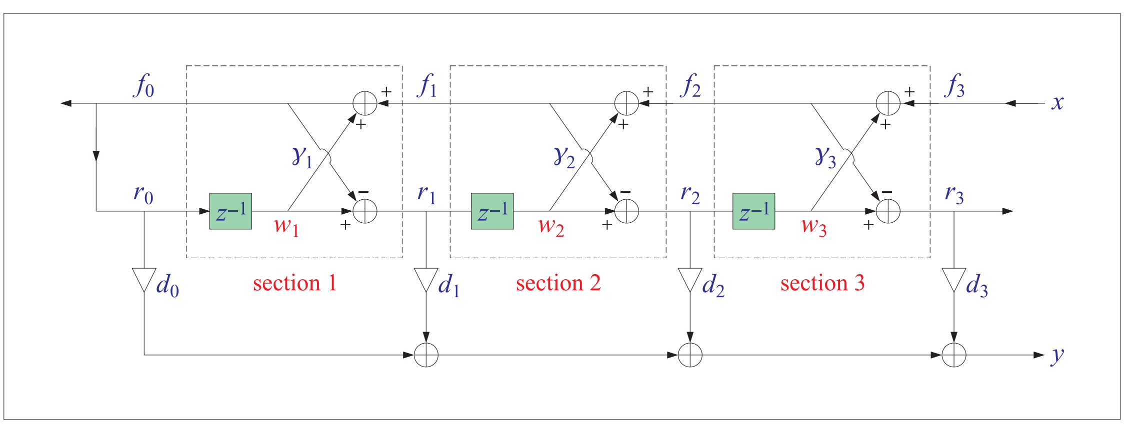


Fig.8 – Lattice realization.

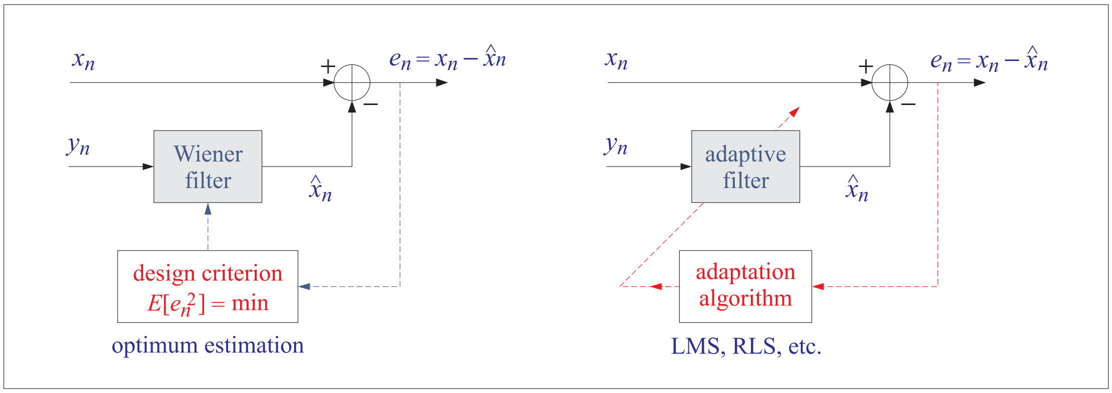


Fig.9 – Optimum Wiener filter and its adaptive implementation.

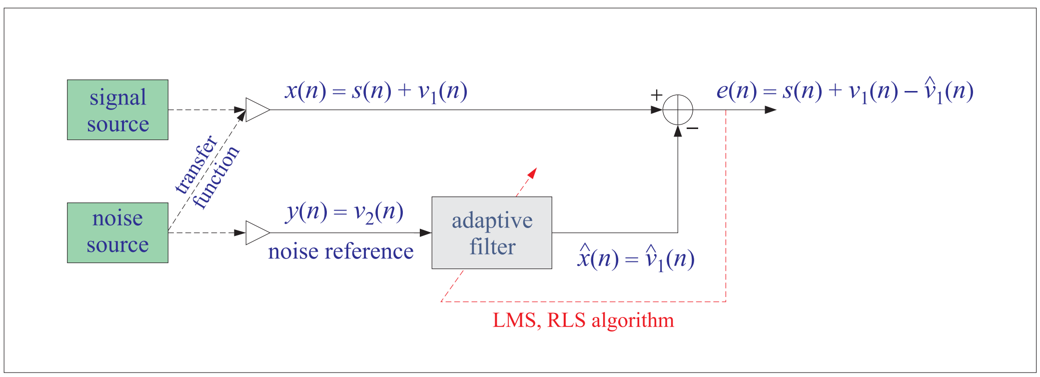


Fig.10 –Adaptive noise canceling system.

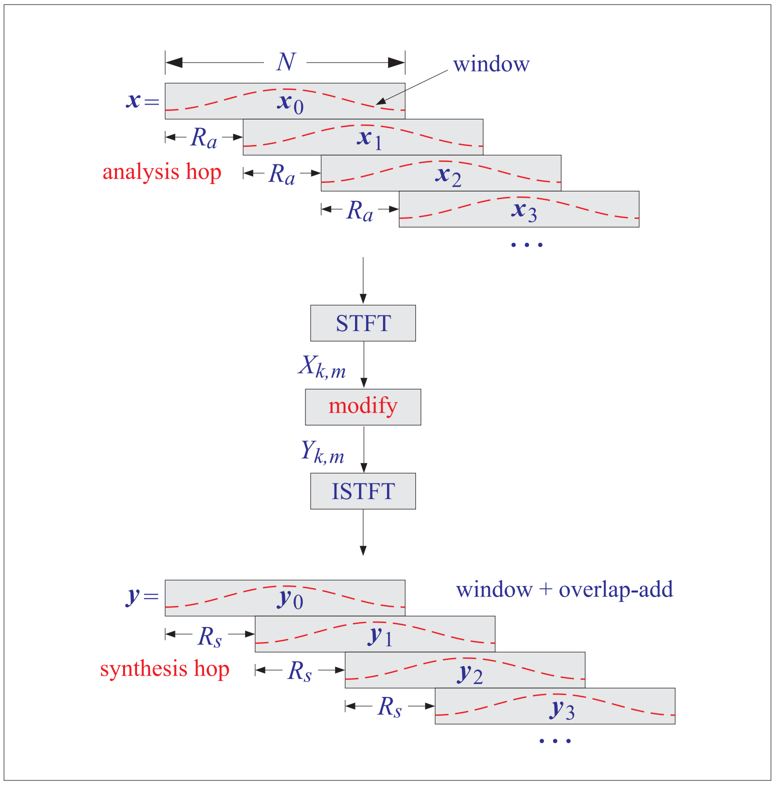


Fig.11 – STFT signal processing system.

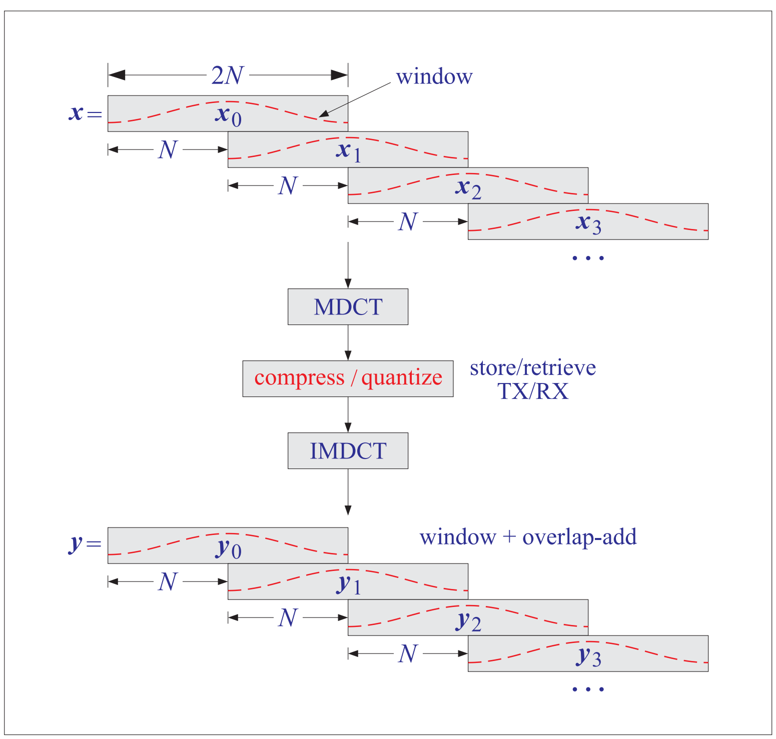


Fig.12 – MDCT/TDAC signal compression system.

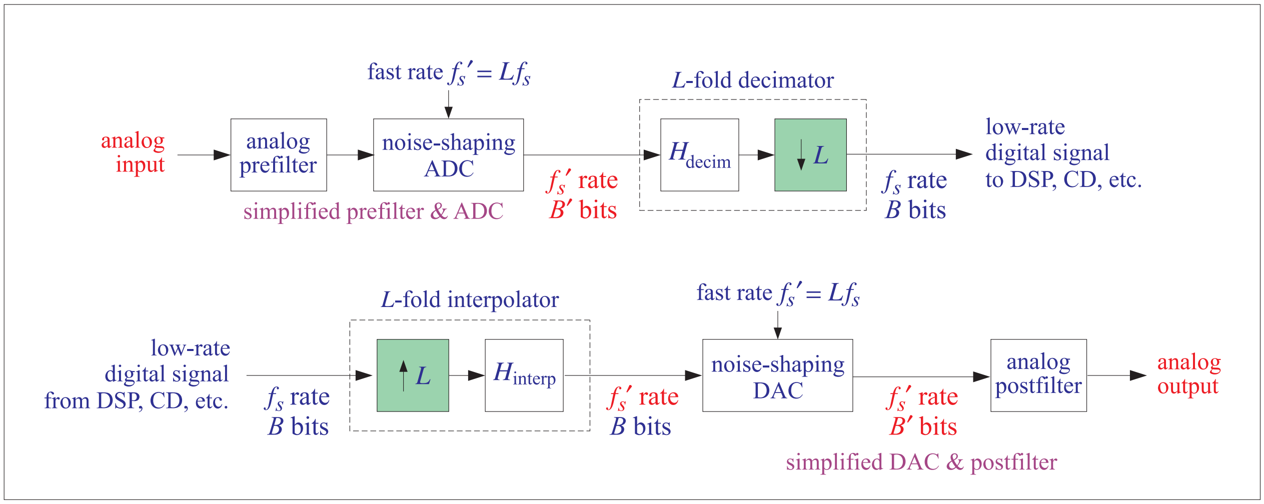


Fig.13 – Oversampled DSP system with noise-shaping quantizers.

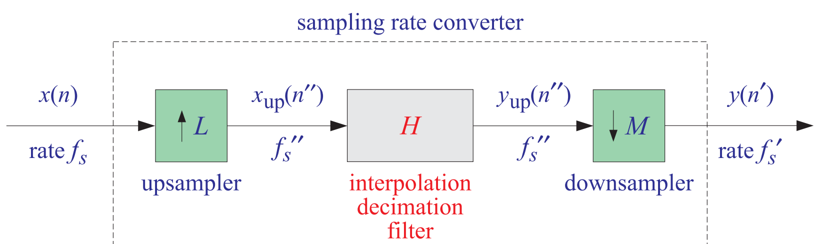


Fig.14 – Sample rate converter, $f_s' = \dfrac{L}{M} f_s$.

Tools

Laplace transforms
z-transforms
Fourier transforms
Fourier series
convolution
filter design
state-space realizations
DTFT, DFT, FFT
STFT, DCT, MDCT

MATLAB functions	**Symbolic Toolbox**
lsim, filter, conv	diff, int
residue, residuez	dsolve, solve
impulse, step	laplace, ilaplace
impz, stepz, c2d	ztrans, iztrans
tf, ss, tf2ss, ss2tf, zpk	fourier, ifourier
butter, cheby1, cheby2	partfrac, expm
feedback, pid, bodeplot	dirac, heaviside
freqs, freqz, fft	kroneckerDelta
hamming, kaiser	rectangularPulse

Fig.15 – Conceptual and computational tools.

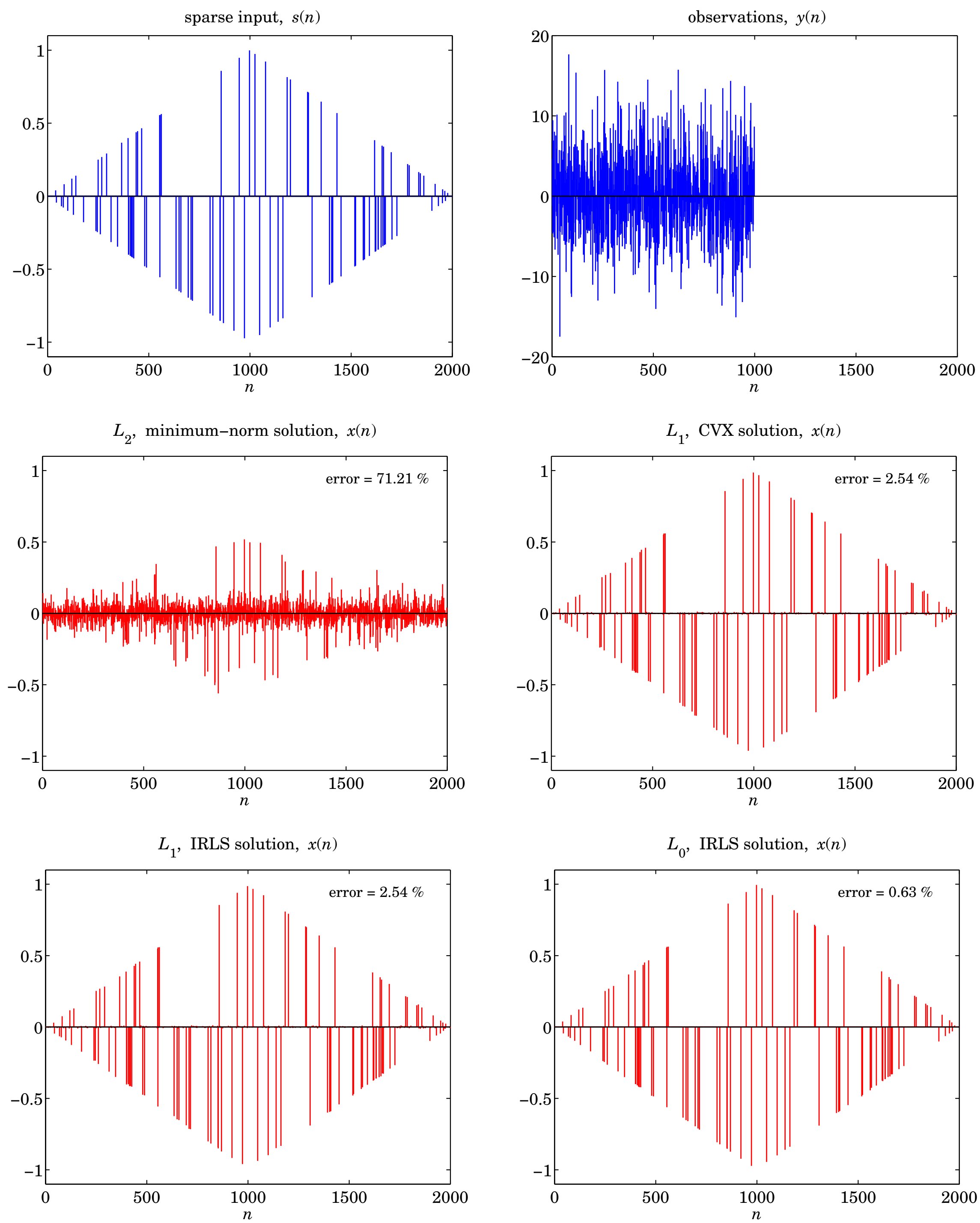


Fig.16 – Recovered signal based on the L_2, L_1, and L_0 criteria.

$$
\boxed{
\begin{aligned}
(L_0):&\quad J = \|\mathbf{b} - A\mathbf{x}\|_2^2 + \lambda\, \|\mathbf{x}\|_0 = \min \\
(L_1):&\quad J = \|\mathbf{b} - A\mathbf{x}\|_2^2 + \lambda\, \|\mathbf{x}\|_1 = \min \\
(L_2):&\quad J = \|\mathbf{b} - A\mathbf{x}\|_2^2 + \lambda\, \|\mathbf{x}\|_2^2 = \min
\end{aligned}
}
$$

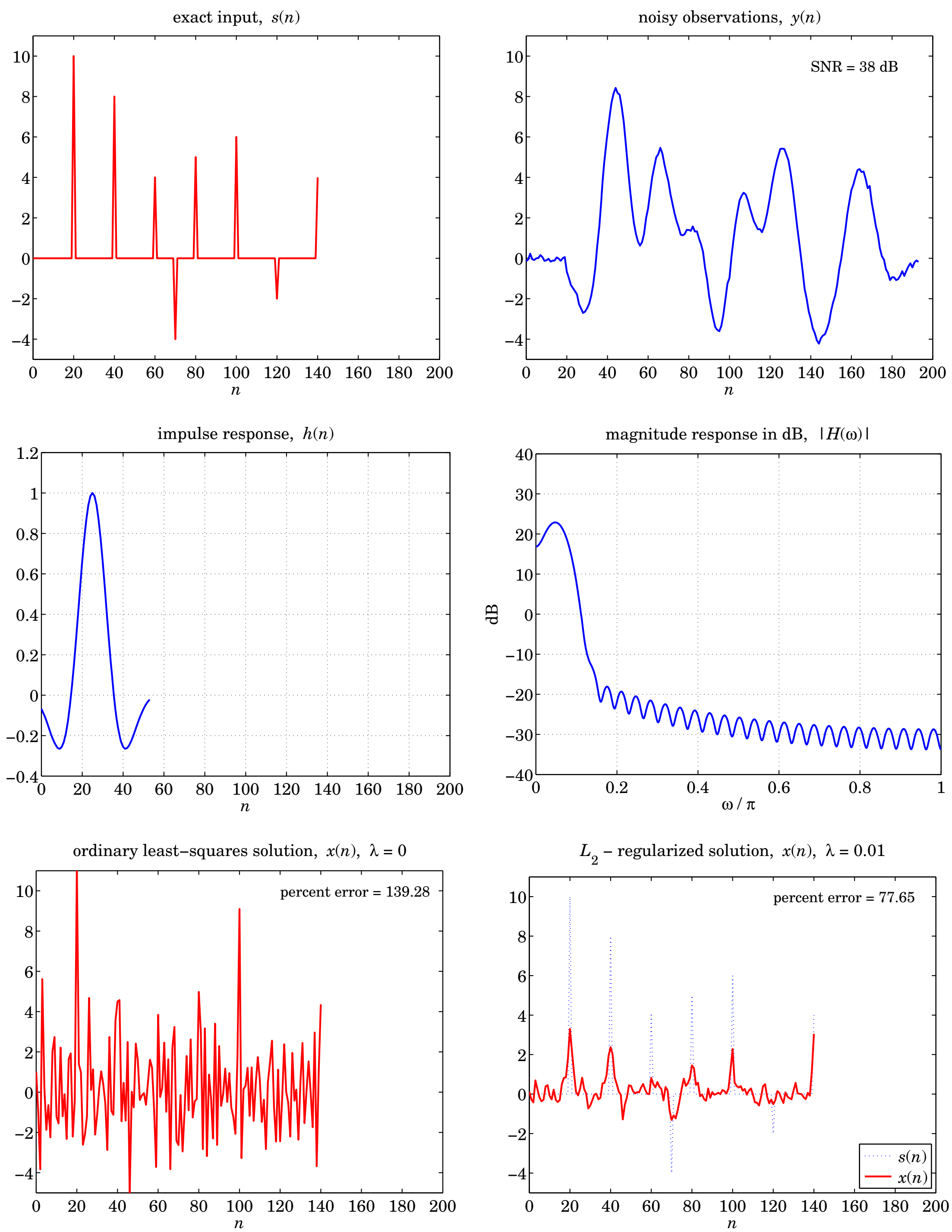


Fig.17a – Sparse deconvolution based on the L_2, L_1, and L_0 criteria.

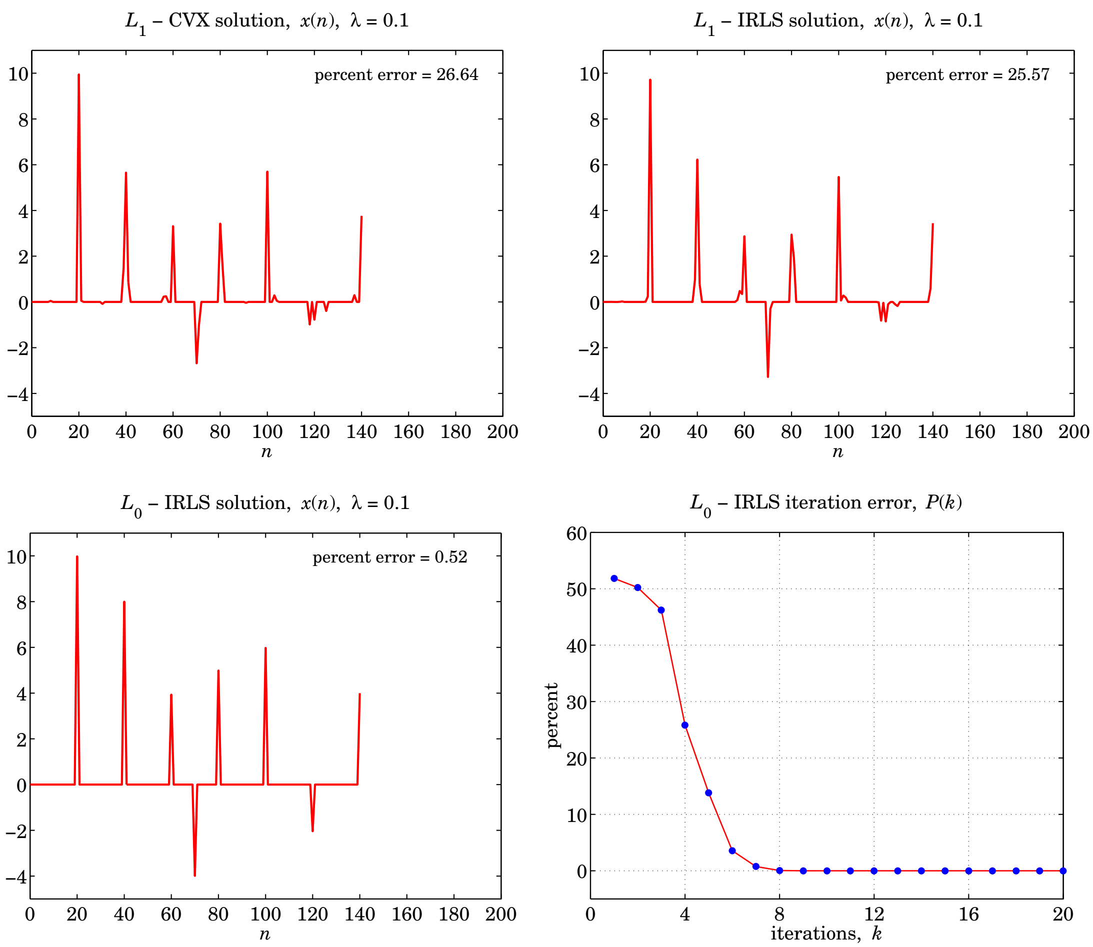
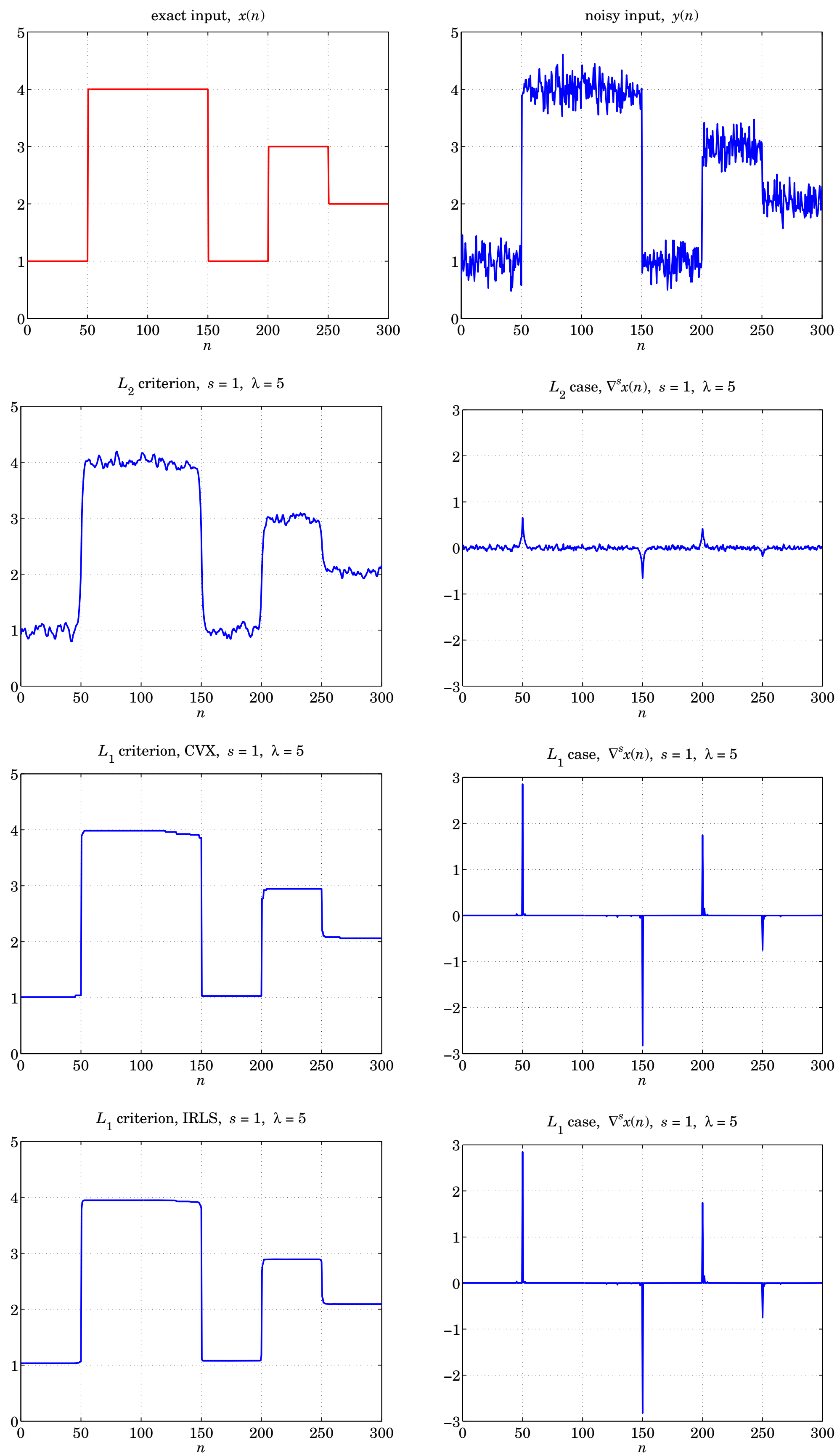


Fig.17b – Sparse deconvolution based on the L_2, L_1, and L_0 criteria.


exact input, $x(n)$
noisy input, $y(n)$
L_2 criterion, $s = 1$, $\lambda = 5$
L_2 case, $\nabla^s x(n)$, $s = 1$, $\lambda = 5$
L_1 criterion, CVX, $s = 1$, $\lambda = 5$
L_1 case, $\nabla^s x(n)$, $s = 1$, $\lambda = 5$
L_1 criterion, IRLS, $s = 1$, $\lambda = 5$
L_1 case, $\nabla^s x(n)$, $s = 1$, $\lambda = 5$
n
0 50 100 150 200 250 300
0 1 2 3 4 5
−3 −2 −1 0 1 2 3

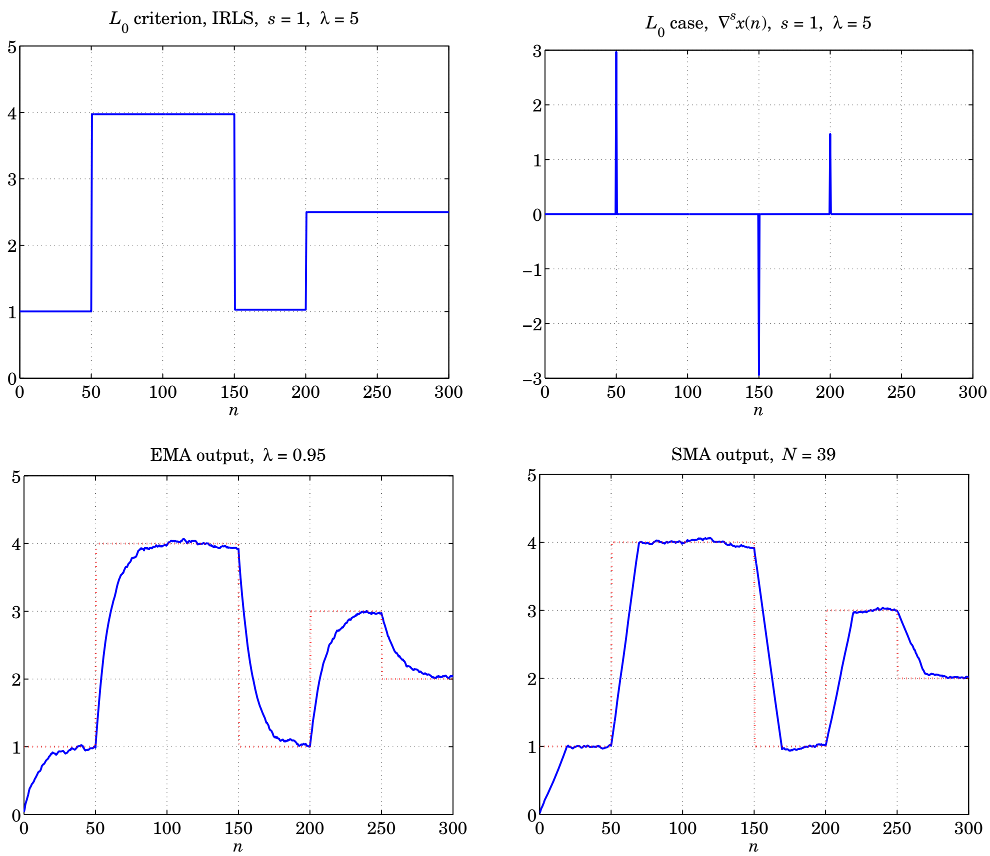


Fig.19a – Trend Extraction – Sparse Whittaker-Henderson Methods.

$$(L_2):\qquad J = \sum_{n=0}^{N-1} |y_n - x_n|^2 + \lambda \sum_{n=s}^{N-1} |\nabla^s x_n|^2 = ||\mathbf{y} - \mathbf{x}||_2^2 + \lambda\, ||D_s\mathbf{x}||_2^2 = \min$$

$$(L_1):\qquad J = \sum_{n=0}^{N-1} |y_n - x_n|^2 + \lambda \sum_{n=s}^{N-1} |\nabla^s x_n|^1 = ||\mathbf{y} - \mathbf{x}||_2^2 + \lambda\, ||D_s\mathbf{x}||_1 = \min$$

$$(L_0):\qquad J = \sum_{n=0}^{N-1} |y_n - x_n|^2 + \lambda \sum_{n=s}^{N-1} |\nabla^s x_n|^0 = ||\mathbf{y} - \mathbf{x}||_2^2 + \lambda\, ||D_s\mathbf{x}||_0 = \min$$

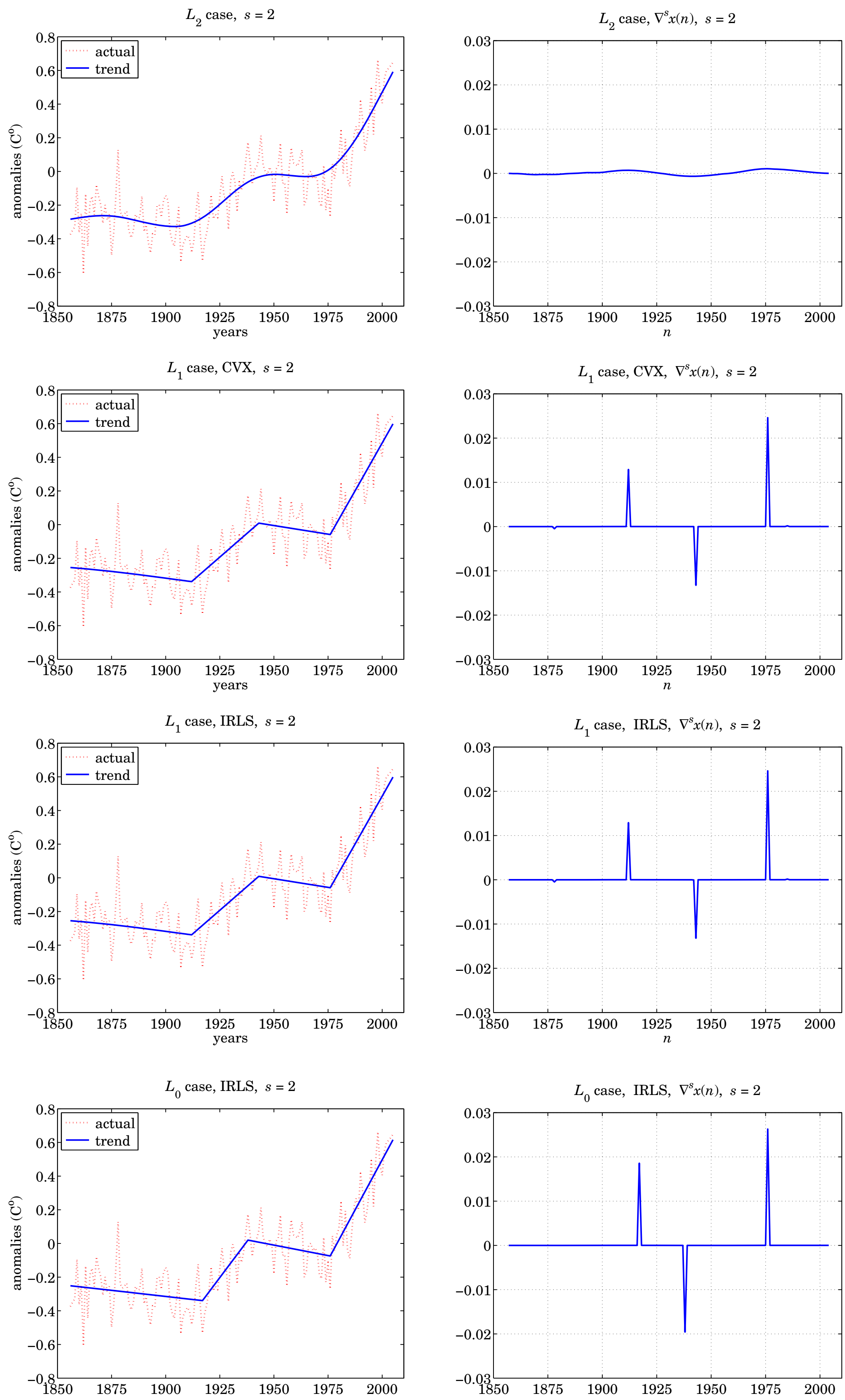


Fig.19b – Trend Extraction – Sparse Whittaker-Henderson Methods.