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Objectives

Direct Forms:
Interpolation
Ratios of Integers

Conjugate Filters:
Two-Band
Design

Example:
Speech Waveform
Filterbank

On-Line Resources:
Signal Modeling
Multirate
Software

LECTURE 09: RESAMPLING

- Objectives:
  - Learn how to change the sample rate of a signal
  - Understand how this can be implemented using time domain interpolation (based on the Sampling Theorem)
  - Understand how this can be implemented efficiently using digital filters
  - Introduction to digital filter banks

A good reference textbook on these topics is:

> J.G. Proakis and D.G. Manolakis, *Digital Signal Processing: Principles, Algorithms, and Applications*, Prentice Hall, Upper Saddle River, New Jersey, USA, ISBN: 0-13-373762-4, 1996 (third edition).

The course textbook:

> X. Huang, A. Acero, and H.W. Hon, *Spoken Language Processing - A Guide to Theory, Algorithm, and System Development*, Prentice Hall, Upper Saddle River, New Jersey, USA, ISBN: 0-13-022616-5, 2001.

has a detailed explanation of filter banks (sections 5.6 and 5.7).

ECE 8463: FUNDAMENTALS OF SPEECH RECOGNITION

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Modern speech understanding systems merge interdisciplinary technologies from Signal Processing, Pattern Recognition, Natural Language, and Linguistics into a unified statistical framework. These systems, which have applications in a wide range of signal processing problems, represent a revolution in Digital Signal Processing (DSP). Once a field dominated by vector-oriented processors and linear algebra-based mathematics, the current generation of DSP-based systems rely on sophisticated statistical models implemented using a complex software paradigm. Such systems are now capable of understanding continuous speech input for vocabularies of hundreds of thousands of words in operational environments.

In this course, we will explore the core components of modern statistically-based speech recognition systems. We will view speech recognition problem in terms of three tasks: signal modeling, network searching, and language understanding. We will conclude our discussion with an overview of state-of-the-art systems, and a review of available resources to support further research and technology development.

Tar files containing a compilation of all the notes are available. However, these files are large and will require a substantial amount of time to download. A tar file of the html version of the notes is available here. These were generated using wget:

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wget -np -k -m http://www.isip.msstate.edu/publications/courses/ece_8463/lectures/current
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A pdf file containing the entire set of lecture notes is available here. These were generated using Adobe Acrobat.

Questions or comments about the material presented here can be directed to help@isip.msstate.edu.

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INTERPOLATION USING THE SAMPLING THEOREM

How do we change the sample frequency of a signal:

Method 1: Use the sampling theorem (Lecture No. 3)

Define F_s^1 as the original sample frequency, and F_s^2 as the new sample frequency. Recall our interpolation function, where $B = \frac{F_s^1}{2}$:

$$g(t) = \frac{\sin(2\pi Bt)}{2\pi Bt} .$$

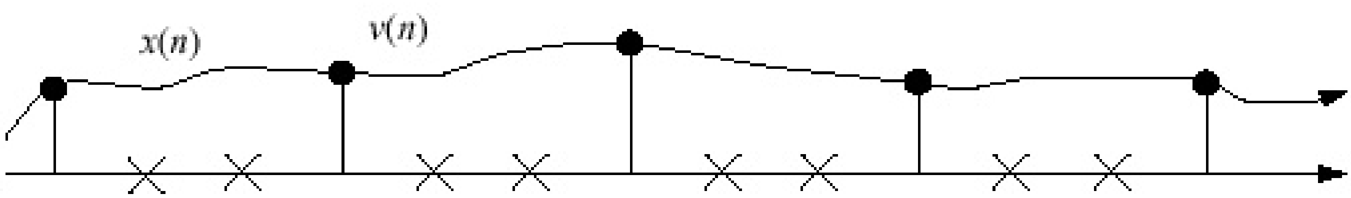
$x\left(\frac{m}{F_s^2}\right)$ may be expressed as:

$$x\left(\frac{m}{F_s^2}\right) = \sum_{n=-\infty}^{\infty} x\left(\frac{n}{F_s^1}\right) g\left(\frac{m}{F_s^2} - \frac{n}{F_s^1}\right) .$$

What are the disadvantages of this method?

Method 2: Downsampling a signal by dropping samples

Consider the signal $x(n)$. What is the spectrum of $v(n) = x(Ln)$?



Recall the frequency-scaling property:

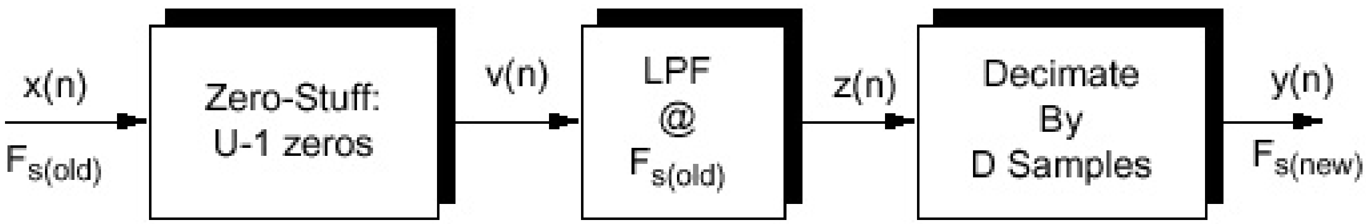
$$\begin{aligned} V(\omega) &= \sum_{m=-\infty}^{\infty} v(m) e^{-j\omega m} \\ &= \sum_{n=-\infty}^{\infty} x(Ln) e^{-j\omega n L} \\ &= X(\omega / L) \end{aligned}$$

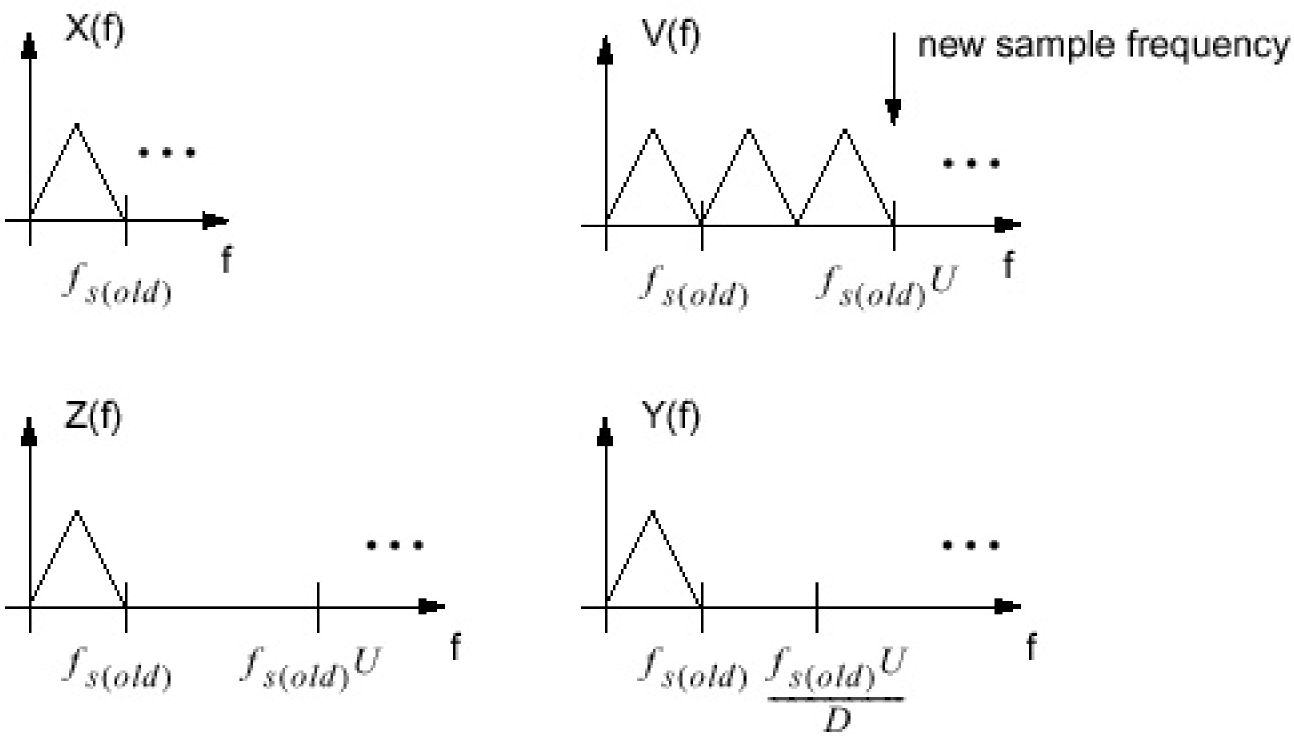
$$n = -\infty$$

$$= X(\omega / L)$$

INTERPOLATION AND DECIMATION
USING RATIOS OF INTEGERS

$$F_{s(new)} = F_{s(old)}\left(\frac{U}{D}\right)$$





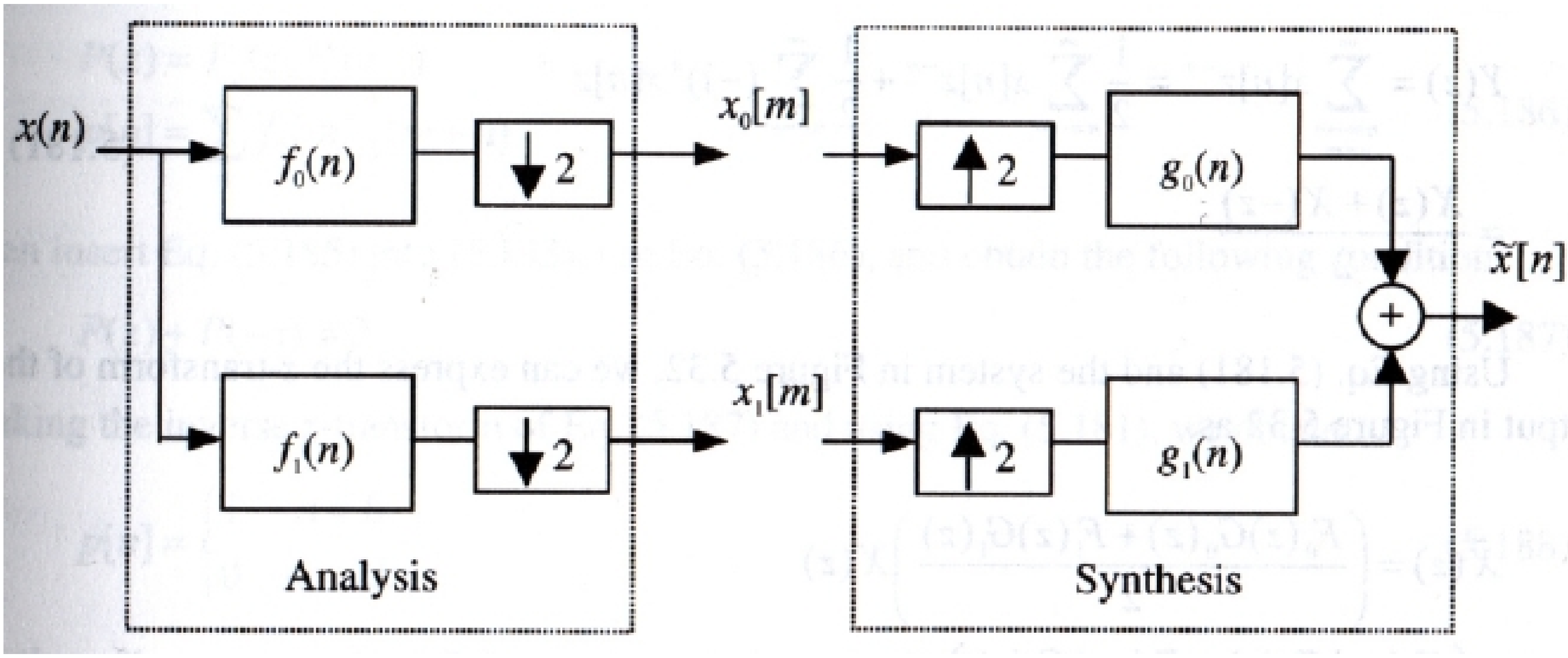
Note that the LPF is run at the decimation rate of D!

Questions:

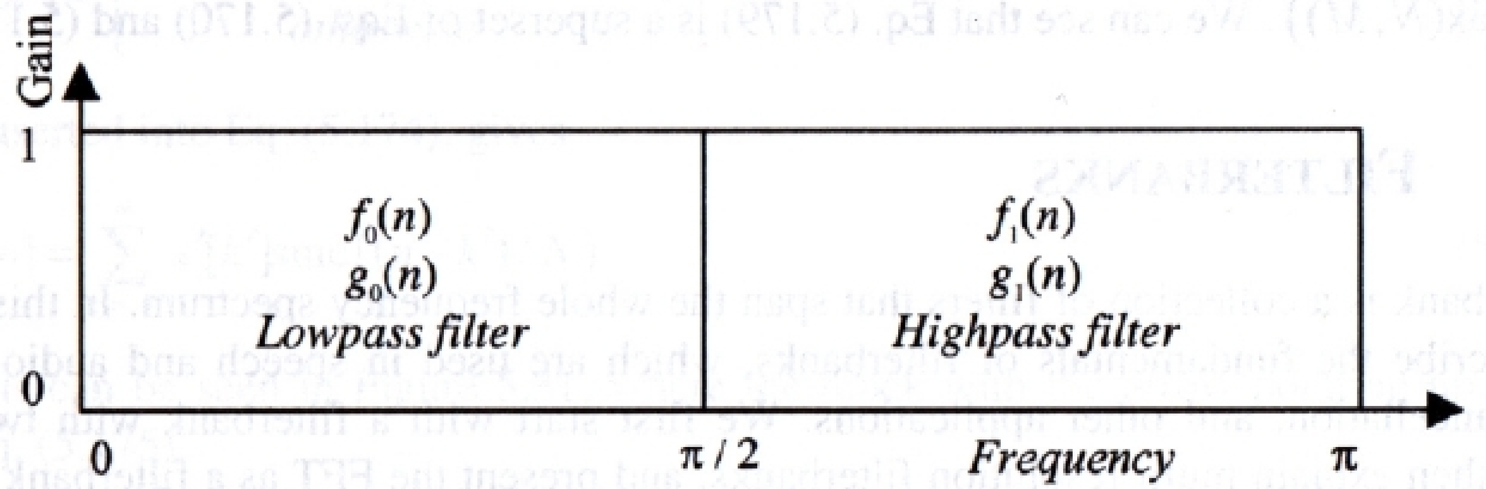
- Under what conditions will this introduce no distortion?
- How do we implement this efficiently?
- How should we convert from 8 kHz to 6.4 kHz?
- What about the infamous 44.1 kHz CD sample frequency?

DIGITAL FILTERBANKS BASED ON RESAMPLING

Consider the two-band filterbank shown below:



where f(n) and g(n) are complementary low pass and high pass filters:



- To achieve perfect reconstruction of x(n), we need ideal filters, which are not realizable.
- Is it possible to build a filterbank that has perfect reconstruction?
- Why might such a filterbank be useful for speech recognition?

DESIGN OF CONJUGATE QUADRATURE MIRROR FILTERS

If we specify $f_1(n)$, $g_0(n)$, and $g_1(n)$ as a function of $f_0(n)$, we can derive a compact design procedure:

1. Design a (2L - 1) tap half-band linear phase low-pass filter p(n) (use Parks-McClellan or Kaiser Window approach).

2. Factor $P(z) = F_o(z)\, F_0(z^{-1})$ by finding roots.

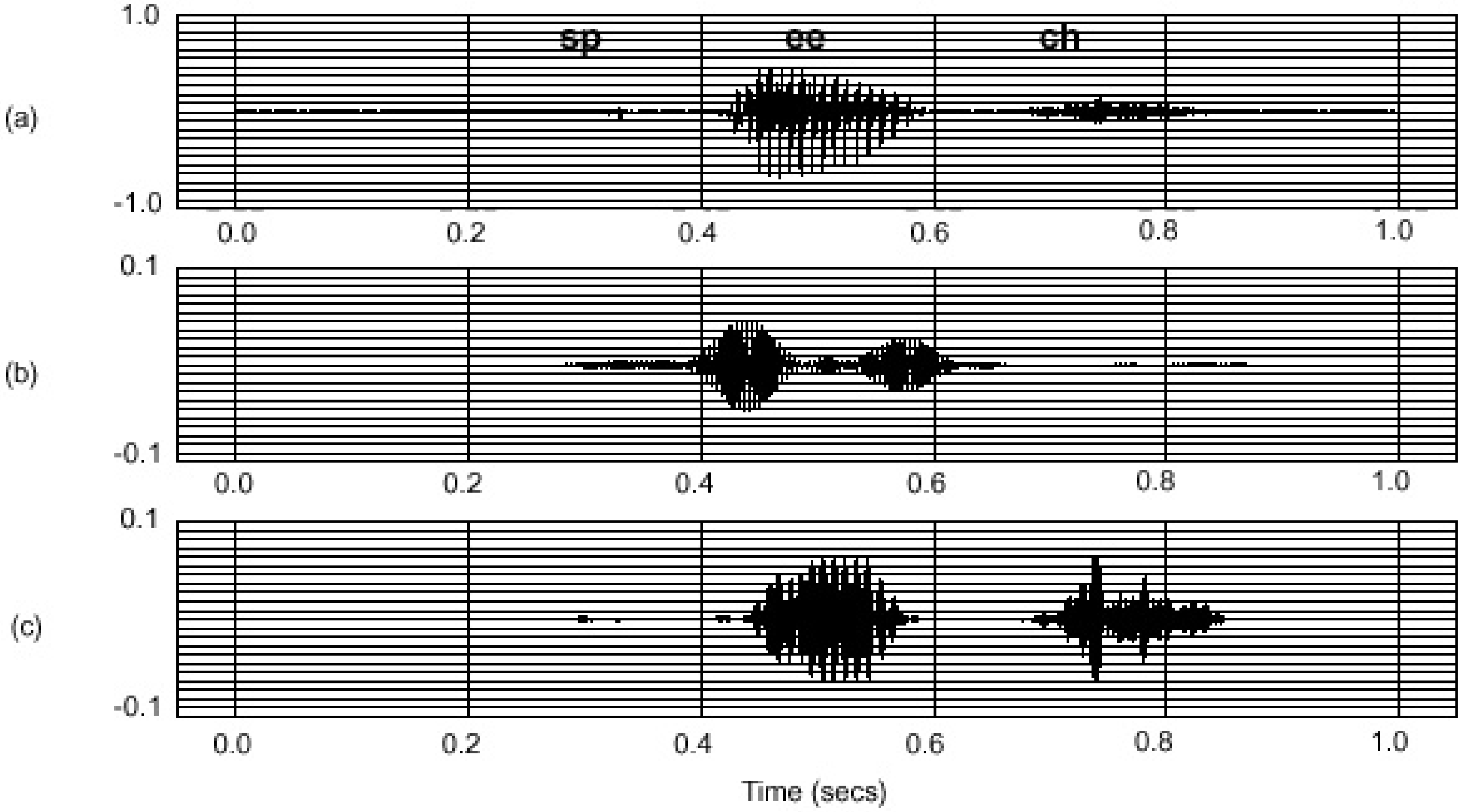
3. Compute the remaining filter impulse responses as follows:

   $f_1(n) = (-1)^n f_0(L-1-n)$
   $g_0(n) = f_0(L-1-n)$
   $g_1(n) = f_1(L-1-n)$

What are the advantages of this approach?

There are several popular approaches to implementing such digital filterbanks. This general area of research is known as **multirate signal processing**. What are the merits of a frequency domain approach?

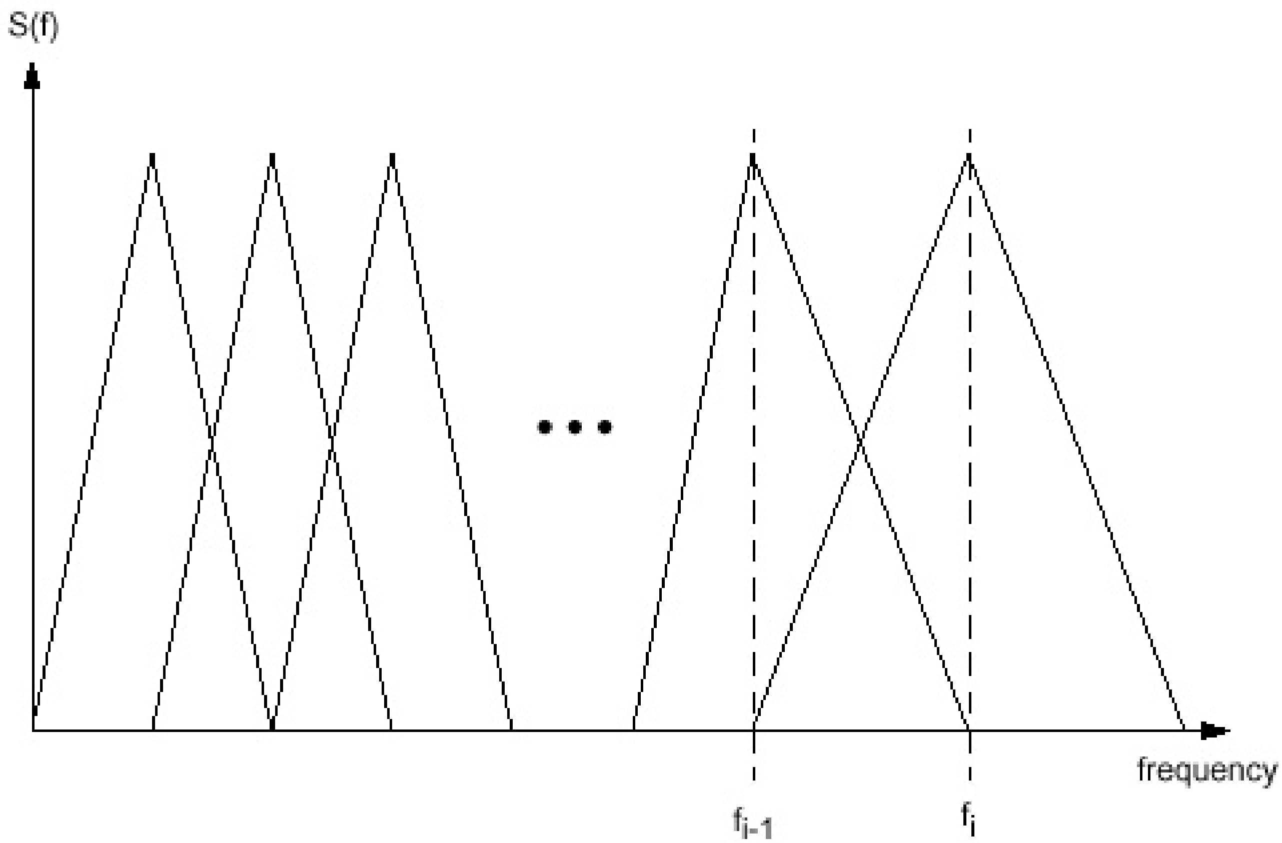
EXAMPLE OUTPUT FROM A DIGITAL FILTER BANK



- Digital filter bank outputs for a speech signal shown in (a), consisting of the word *speech*. In (b), the output from a filter with a center frequency of 250 Hz and a bandwidth of 100 Hz is shown. In (c), the output from a filter centered at 2500 Hz is shown.

- Note that the amplitude of the output for each filter varies depending on the nature of the sound. The final *ch* sound, for example, is mainly composed of high frequency information.

A DIGITAL FILTERBANK



- Note that an FFT yields frequency samples at $(k/N)f_s$.
- Oversampling provides a smoother estimate of the envelope of the spectrum.
- Other efficient techniques exist for different frequency scales (e.g., bilinear transform).

Index of /publications/journals/ieee_proceedings/1993/signal_modeling

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Multirate Signal Processing Group

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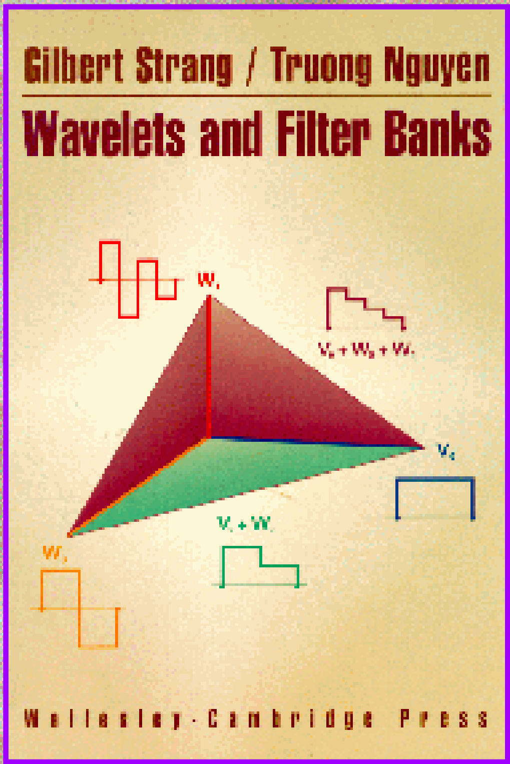

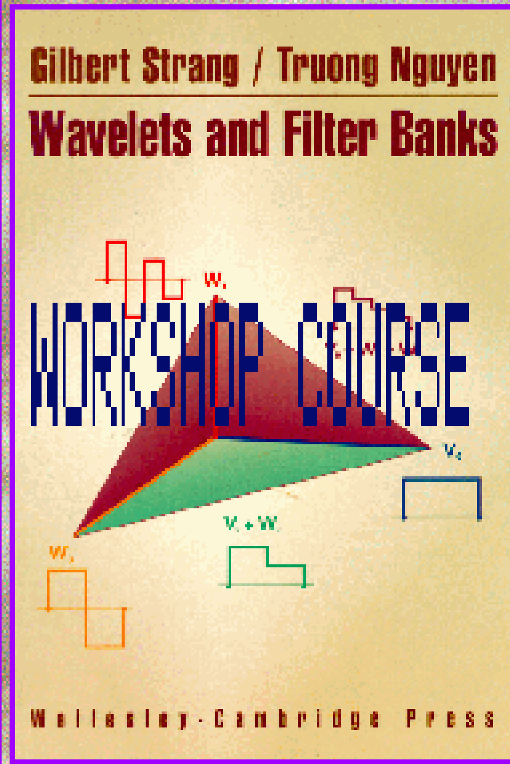

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This homepage will be updated when the new results are available, so stay tuned.

Last updated on Feb. 24, 1997 by T. Tran.