Introduction to Image Processing

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Topic 07 Image Transforms

1. Introduction

- In some cases, image **processing tasks** are best formulated by **transforming** the input images, carrying the specified task in a **transform** domain, and applying the **inverse transform** to return to the spatial domain.
- A particularly important class of 2-D linear transforms, can be expressed in the general form.

$$T(u, v) = \sum_{x=0}^{M-1} \sum_{y=0}^{N-1} f(x, y) r(x, y, u, v)$$

where f is the input image, r is called the **forward transformation** kernel, and the equation is evaluated for and u = 0, 1, ..., M-1 and v = 0, 1, ..., N-1.

1. Introduction

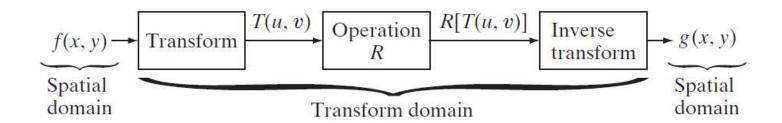
- T is called the **forward transform** of f.
- Given T we can recover f using the **inverse transform**.

$$f(x, y) = \sum_{u=0}^{M-1} \sum_{v=0}^{N-1} T(u, v) s(x, y, u, v)$$

- The equation is evaluated for and x=0, 1,..., M-1 and y=0, 1,..., N-1.
- And s is called the inverse transformation kernel.

1. Introduction

- By using these transforms, it is possible to express an image as a **combination** of a set **of basic signals**, known as the **basis functions**.
- The image output in the **transformed space** may be analyzed, interpreted, and further processed for implementing **diverse** image processing **tasks**.
- General approach for operating in the linear transform domain.



2. Discrete Fourier Transform

• Substituting the kernels below into the previous equations yields the **Discrete Fourier transform** pair

$$r(x, y, u, v) = e^{-j2\pi(ux+vy)/n}$$
$$s(x, y, u, v) = \frac{1}{n^2} e^{j2\pi(ux+vy)/n}$$
$$M = N = n$$

- The Discrete Fourier Transform was discussed in "Topic 04
- Filtering in the Frequency Domain".

2. Discrete Fourier Transform

The 2-D Discrete Fourier Transform and its inverse.

$$F(u, v) = \sum_{x=0}^{M-1} \sum_{y=0}^{N-1} f(x, y) e^{-j2\pi(ux/M + vy/N)}$$

$$u = 0, 1, 2, \dots, M-1 \text{ and } v = 0, 1, 2, \dots, N-1.$$

$$f(x, y) = \frac{1}{MN} \sum_{u=0}^{M-1} \sum_{v=0}^{N-1} F(u, v) e^{j2\pi(ux/M + vy/N)}$$

 $x = 0, 1, 2, ..., M-1 \text{ and } y = 0, 1, 2, ..., N-1.$

- DFT uses a set of complex exponential functions.
- Normally used for general spectral analysis applications.

- **Discrete Cosine Transform** (DCT) is the basis for many image and video **compression algorithms**, especially the baseline JPEG and MPEG standards for compression of **still** and **video** images respectively.
- It is obtained by using the following (equal) kernels:

$$r(x, y, u, v) = s(x, y, u, v)$$

$$= \alpha(u)\alpha(v)\cos\left[\frac{(2x+1)u\pi}{2n}\right]\cos\left[\frac{(2y+1)v\pi}{2n}\right]$$

$$\alpha(u) = \begin{cases} \sqrt{\frac{1}{n}} & \text{for } u = 0\\ \sqrt{\frac{2}{n}} & \text{for } u = 1, 2, \dots, n-1 \end{cases}$$

- DCT uses only (real-valued) cosine functions.
- It translates the correlated data to uncorrelated data.
- The DCT and inverse DCT can be computed using the DFT.
- Example: 8 basis functions of a 4-by-4 matrix.

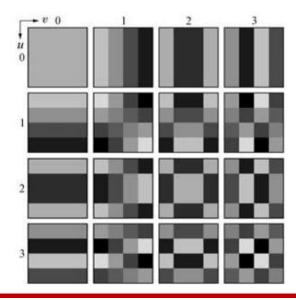
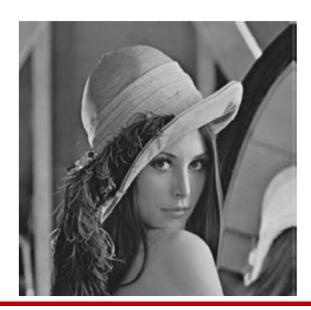
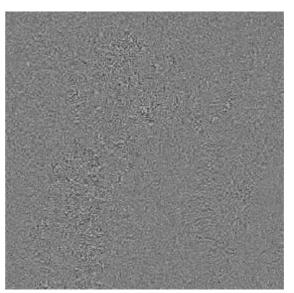


FIGURE 8.23 Discrete-cosine basis functions for n = 4. The origin of each block is at its top left.

- Example: approximations of the a monochrome image.
 - ➤ The result was obtained by dividing the original image into subimages of size 8 x 8 using the DCT, truncating 50% of the resulting coefficients, and taking the inverse transform of the truncated coefficient arrays.





- **Mean-square** reconstruction **error** is related directly to the **energy** or **information** packing properties of the transform employed.
- An image g(x, y) can be expressed as a function of its 2-D transform

$$g(x, y) = \sum_{u=0}^{n-1} \sum_{v=0}^{n-1} T(u, v) s(x, y, u, v)$$

$$x, y = 0, 1, 2, \dots, n-1$$

• The inverse kernel s can be viewed as defining a set of **basis functions** or **basis images**.

• This interpretation becomes clearer if we use the notation:

$$\mathbf{G} = \sum_{u=0}^{n-1} \sum_{v=0}^{n-1} T(u, v) \mathbf{S}_{uv}$$

$$\mathbf{S}_{uv} = \begin{bmatrix} s(0,0,u,v) & s(0,1,u,v) & \cdots & s(0,n-1,u,v) \\ s(1,0,u,v) & \vdots & \cdots & \vdots \\ \vdots & \vdots & \cdots & \vdots \\ \vdots & \vdots & \cdots & \vdots \\ s(n-1,0,u,v) & s(n-1,1,u,v) & \cdots & s(n-1,n-1,u,v) \end{bmatrix}$$

• **G** contains the pixels of the image and is defined as a linear combination of n^2 matrices of size $n \times n$ that is, S_{uv} , for u, v = 0, 1, 2,...,n-1.

• We can **define** a transform coefficient **masking function** χ which is constructed to **eliminate** the **basis images** that make the smallest contribution to the total sum

$$\chi(u, v) = \begin{cases} 0 & \text{if } T(u, v) \text{ satisfies a specified truncation criterion} \\ 1 & \text{otherwise} \end{cases}$$

$$\hat{\mathbf{G}} = \sum_{u=0}^{n-1} \sum_{v=0}^{n-1} \chi(u, v) T(u, v) \mathbf{S}_{uv}$$

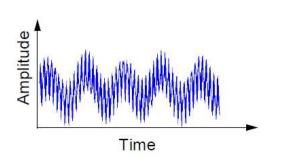
ullet The mean-square error between ${f G}$ and ${f \widehat{G}}$ approximation is

$$e_{ms} = E\left\{\|\mathbf{G} - \hat{\mathbf{G}}\|^{2}\right\} = \sum_{u=0}^{n-1} \sum_{v=0}^{n-1} \sigma_{T(u,v)}^{2} \left[1 - \chi(u,v)\right]$$

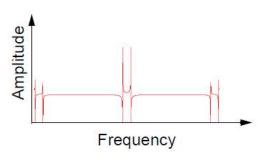
• The total mean-square approximation error thus is the sum of the variances of the discarded transform coefficients.

• Transformations that redistribute or pack the most information into the fewest coefficients provide the smallest reconstruction errors.

• Signal analysts already have at their disposal an impressive arsenal of tools. Perhaps the most well-known of these is **Fourier analysis**.







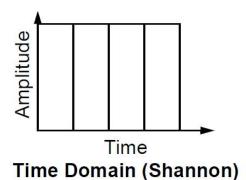
- Fourier analysis has a **serious drawback**. In transforming to the frequency domain, **time information is lost**.
- If a signal does not change much over time this drawback is not very important.

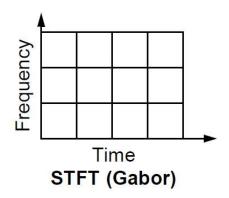
- However, most **interesting signals** contain numerous non-stationary or **transitory characteristics**.
- In an effort to **correct** this deficiency, Dennis Gabor (1946) adapted the Fourier transform to analyze only **a small section** of the signal at a **time**.
 - Short-Time Fourier Transform (STFT)

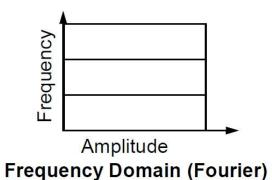


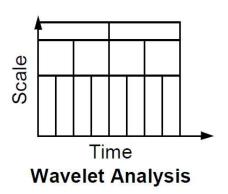
• Precision is determined by the size of the window.

• **Wavelet analysis** represents the next logical step: a windowing technique with variable-sized regions.

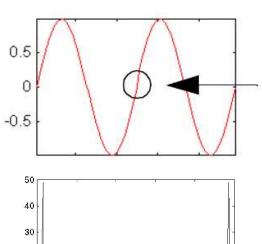




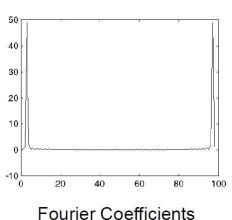


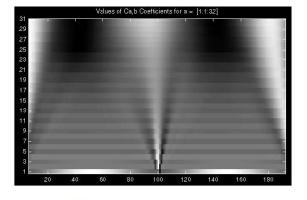


• One major **advantage** afforded by wavelets is the ability to perform **local analysis** — that is, to analyze a localized area of a larger signal.



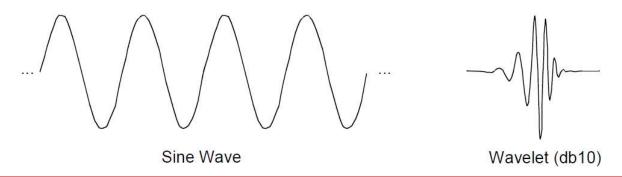
Sinusoid with a small discontinuity





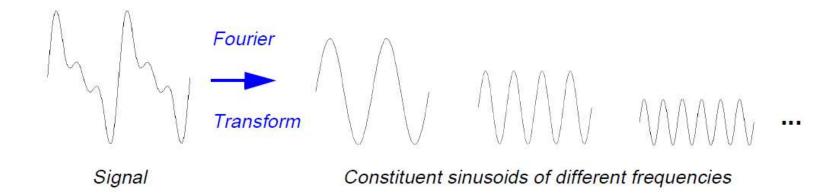
Wavelet Coefficients

- A wavelet is a waveform of effectively **limited duration** that has an **average** value of **zero**.
- Compare wavelets with sine waves, which are the basis of Fourier analysis:
 - Sinusoids do not have limited duration.
 - Sinusoids are smooth and predictable.
 - Wavelets tend to be irregular and asymmetric.



• Fourier analysis consists of breaking up a signal into sine waves of various frequencies.

$$F(\omega) = \int_{-\infty}^{\infty} f(t)e^{-j\omega t}dt$$



• The **Continuous Wavelet Transform** (CWT) of a continuous function f(x), relative to a real-valued wavelet, $\psi(x)$, is defined as

$$W_{\psi}(s,\tau) = \int_{-\infty}^{\infty} f(x)\psi_{s,\tau}(x) dx$$

$$\psi_{s,\tau}(x) = \frac{1}{\sqrt{s}}\psi\left(\frac{x-\tau}{s}\right)$$

where s and τ are called **scale** and **translation** parameters, respectively.

ullet The function f(x) can be obtained using the **inverse** Continuous Wavelet Transform,

$$f(x) = \frac{1}{C_{\psi}} \int_{0}^{\infty} \int_{-\infty}^{\infty} W_{\psi}(s, \tau) \frac{\psi_{s, \tau}(x)}{s^{2}} d\tau \, ds$$

$$C_{\psi} = \int_{-\infty}^{\infty} \frac{|\Psi(\mu)|^{2}}{|\mu|} d\mu$$

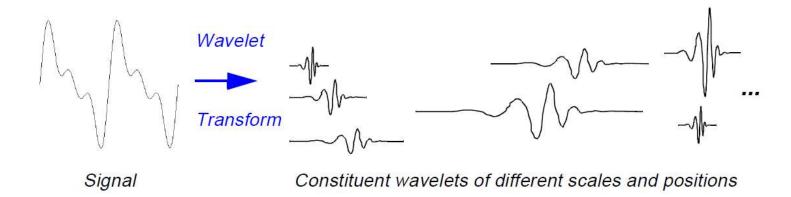
where $\Psi(\mu)$ is the Fourier transform of $\psi(x)$.

• The previous equations define a reversible transformation as long as the so-called admissibility criterion is satisfied,

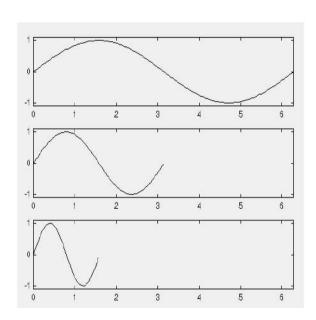
$$C_{\psi} < \infty$$

• Similarly, wavelet analysis is the breaking up of a signal into **shifted** and **scaled** versions of the original (or **mother**) wavelet.

$$C(scale, position) = \int_{-\infty}^{\infty} f(t) \psi(scale, position, t) dt$$



• Scaling a wavelet simply means **stretching** (or compressing) it.

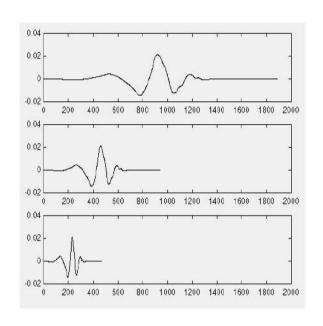


$$f(t) = \sin(t) \quad ; \quad a = 1$$

$$f(t) = \sin(2t) \quad ; \quad a = \frac{1}{2}$$

$$f(t) = \sin(4t) \quad ; \quad a = \frac{1}{4}$$

• The scale factor works exactly the same with wavelets.

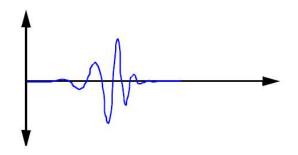


$$f(t) = \psi(t)$$
 ; $a = 1$

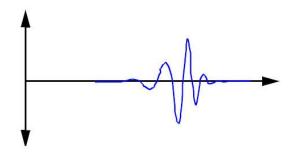
$$f(t) = \psi(2t)$$
 ; $a = \frac{1}{2}$

$$f(t) = \psi(4t) \quad ; \quad a = \frac{1}{4}$$

• Shifting a wavelet simply means delaying (or hastening) its onset.

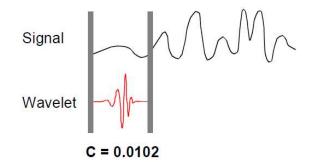


Wavelet function $\psi(t)$

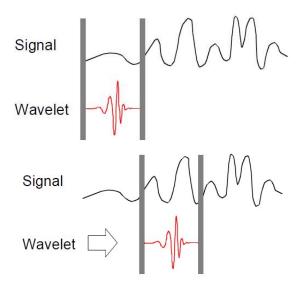


Shifted wavelet function $\psi(t-k)$

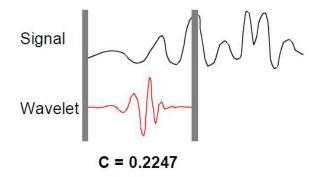
- The CWT is the sum over all time of the signal multiplied by scaled, shifted versions of the wavelet.
- Computing the CWT:
 - 1. Take a wavelet and **compare** it to a section at the start of the original signal.
 - 2. Calculate a number, C, that represents **how closely correlated** the wavelet is with this section of the signal.



- Computing the CWT:
 - 3. Shift the wavelet to the right and repeat steps 1 and 2 until you've covered the whole signal.

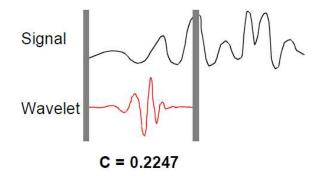


- Computing the CWT:
 - 4. Scale (stretch) the wavelet and repeat steps 1 through 3.



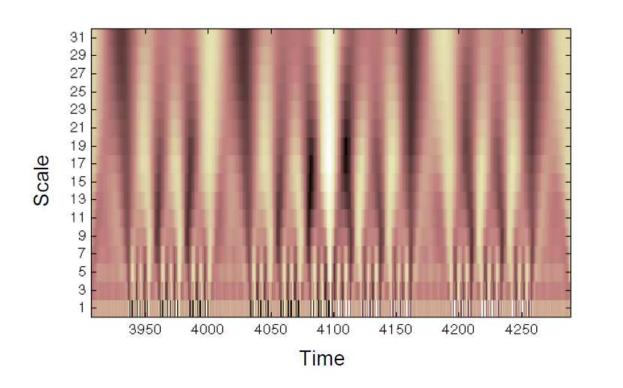
5. Repeat steps 1 through 4 for all scales.

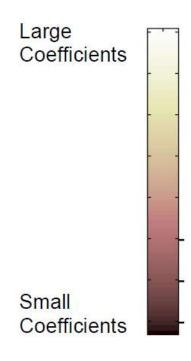
- Computing the CWT:
 - 4. Scale (stretch) the wavelet and repeat steps 1 through 3.



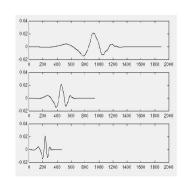
- 5. Repeat steps 1 through 4 for all scales.
- When you're done, you'll have the coefficients produced at different scales by different sections of the signal.

How to make sense of all these coefficients?





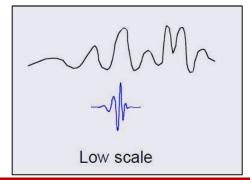
- There is a correspondence between wavelet scales and frequency:
 - ➤ Low scale → Compressed wavelet → Rapidly changing details → High frequency



$$f(t) = \psi(t)$$
 ; $a = 1$

$$f(t) = \psi(2t)$$
 ; $a = \frac{1}{2}$

$$f(t) = \psi(4t)$$
 ; $a = \frac{1}{4}$



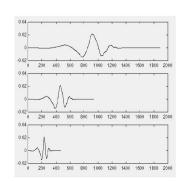
Signal

Wavelet



High scale

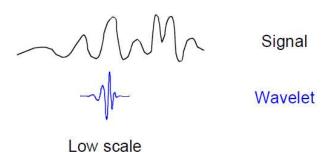
- There is a correspondence between wavelet scales and frequency:
 - ➤ High scale a → Stretched wavelet → Slowly changing, coarse features → Low frequency

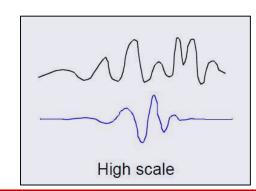


$$f(t) = \psi(t)$$
 ; $a = 1$

$$f(t) = \psi(2t)$$
 ; $a = \frac{1}{2}$

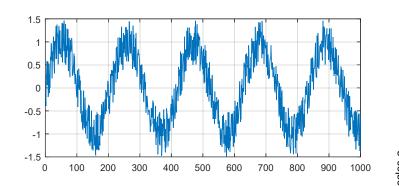
$$f(t) = \psi(4t)$$
 ; $a = \frac{1}{4}$

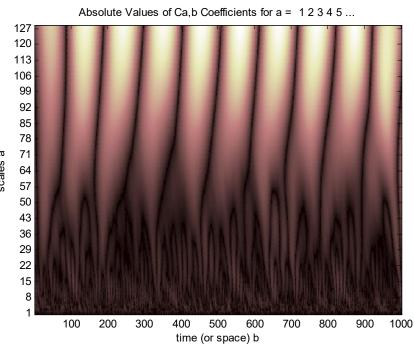




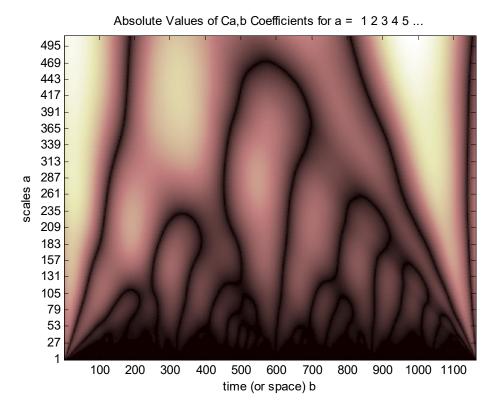
• There is a correspondence between wavelet scales and frequency:

```
load noissin;
c = cwt(noissin,1:128,'db4','plot');
```





• MATLAB: s32Lunar.m

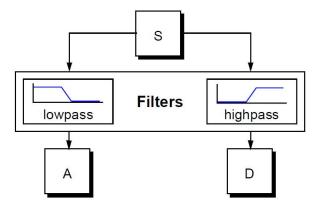


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- What is "continuous" about the Continuous Wavelet Transform (CWT) are the scales at which it operates.
- CWT can operate at **every scale**, from that of the original signal up to some maximum scale which you determine.
- The CWT is also continuous in terms of shifting.
- What if we choose only a **subset** of **scales** and **positions** at which to make our calculations?
 - ➤ It turns out, that if we choose scales and positions based on powers of two then our analysis will just as accurate.
- We obtain such an analysis from the **Discrete Wavelet Transform** (DWT).

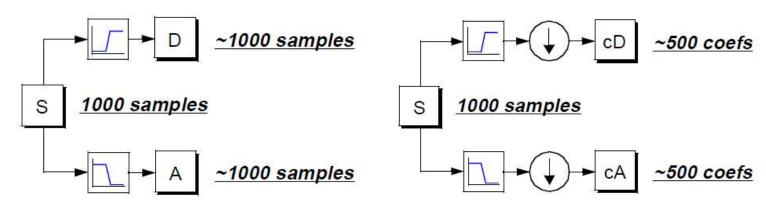
- An efficient way to implement this scheme **using filters** was developed in 1988 by **Mallat**.
- For many signals, the **low-frequency** content is the most important part. It is what gives the signal its **identity**.
- The **high-frequency** content, on the other hand, imparts **nuance**.
- It is for this reason that, in wavelet analysis, we often speak of **approximations** and **details**.

- The **approximations** are the high-scale, low-frequency components of the signal.
- The **details** are the low-scale, high-frequency components.
- The filtering process, at its most basic level, looks like this:



• If we actually perform this operation on a real digital signal, we wind up with **twice as much data** as we started with.

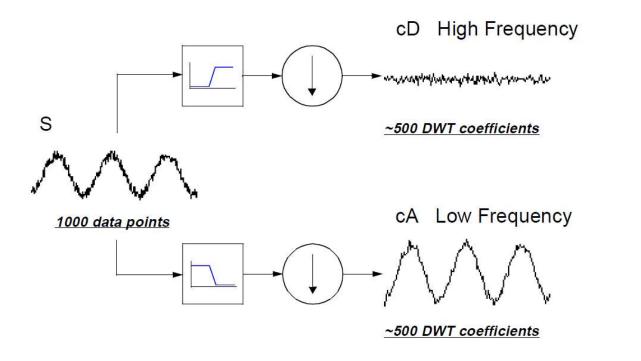
- To correct this problem, we introduce the notion of **downsampling**.
- This simply means **throwing away** every second data point.



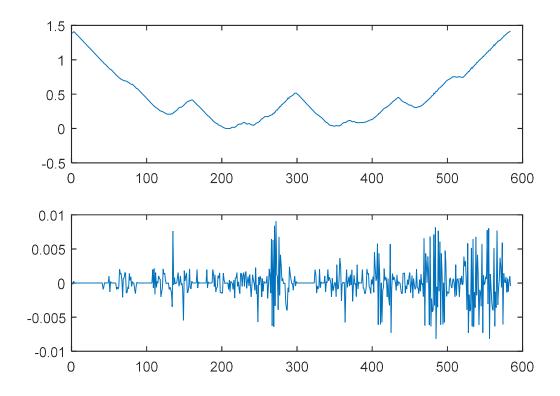
• The process on the right, which includes downsampling, produces **DWT coefficients**.

• To correct this problem, we introduce the notion of **downsampling**.

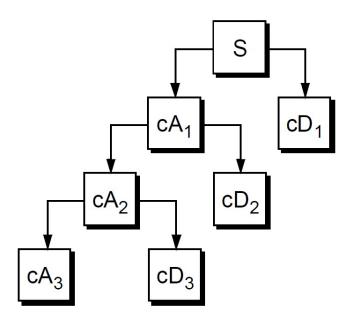
```
s = sin(20.*linspace(0,pi,1000)) + 0.5.*rand(1,1000);
[cA,cD] = dwt(s,'db2');
```



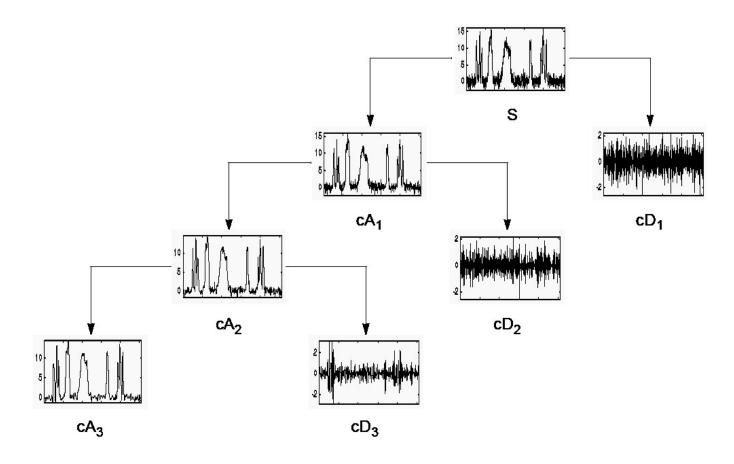
• MATLAB: s39ApproxDet.m



- The decomposition process can be iterated, with successive approximations being decomposed in turn, so that one signal is broken down into many lower-resolution components.
- This is called the wavelet decomposition tree.

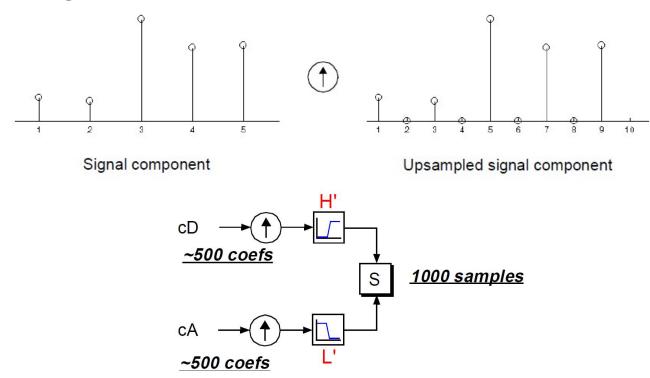


• The decomposition process can be iterated.



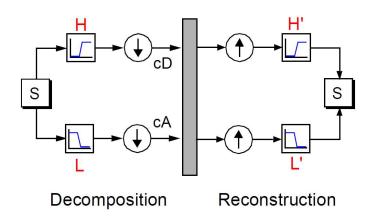
- We've learned how the discrete wavelet transform can be used to analyze, or decompose, signals and images.
- The other half of the story is how those components can be assembled back into the original signal with no loss of information.
- This process is called reconstruction, or synthesis.
- •The mathematical manipulation that effects synthesis is called the inverse discrete wavelet transform (IDWT).

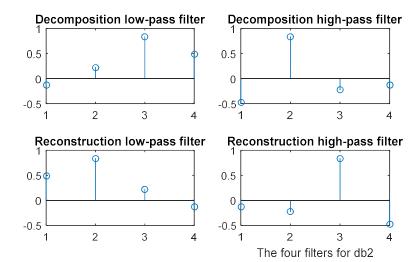
- Wavelet analysis involves filtering and downsampling.
- The wavelet reconstruction process consists of upsampling and filtering.



- Wavelet analysis involves filtering and downsampling.
- By carefully choosing the decomposition and reconstruction filters we can "cancel out" the effects of aliasing caused by the subsampling.

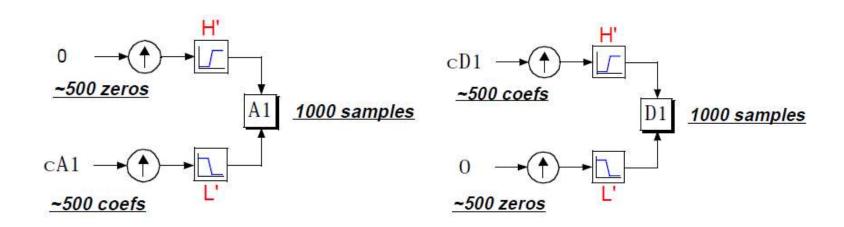
• Quadrature mirror filters:





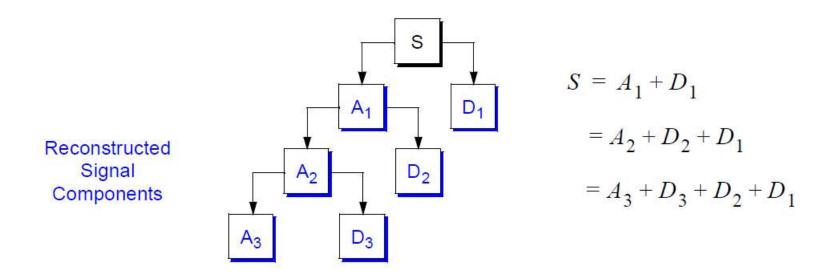
MATLAB: s44Filters.m

• It is also possible to reconstruct the approximation A and detail D themselves **from their coefficient vectors** cA and cD.



• And S = A1 + D1.

• Extending this technique to a multi-level analysis.



- The choice of filters not only determines whether perfect reconstruction is possible, it also determines the shape of the wavelet we use to perform the analysis.
- To construct a wavelet of some practical utility, you seldom start by drawing a waveform.
- Instead, it usually makes more sense to design the appropriate filters and then use them to create the waveform.

- Example:
 - > Starting from the reconstruction low-pass filter L'

```
[L, H, Lprime, Hprime] = wfilters('db2')

Lprime = [0.4830     0.8365     0.2241  -0.1294]
```

➤ If we reverse the order Lmult and then multiply every second sample by -1, we obtain the highpass filter H':

```
Hprime = [-0.1294 -0.2242 -0.8365 0.4830]
```

> Next, upsample Hprime by two, inserting zeros in alternate positions:

```
HU = \begin{bmatrix} -0.1294 & 0 & -0.2241 & 0 & 0.8365 & 0 & -0.4830 & 0 \end{bmatrix}
```

• Example:

Convolve the upsampled vector with the original lowpass filter:

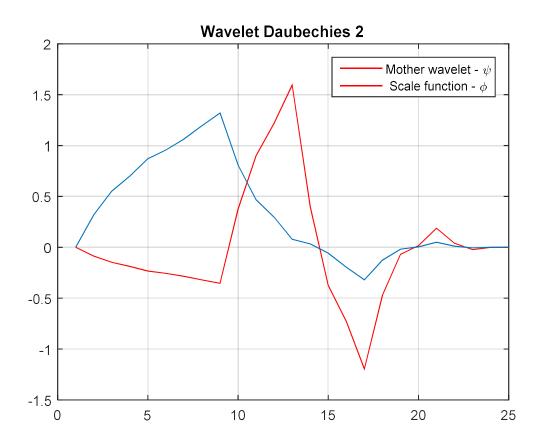
```
H2 = conv(HU, Lprime);
```

- ➤ If we iterate this process several more times, repeatedly upsampling and convolving the resultant vector with the four-element filter vector Lprime, a pattern begins to emerge.
- > Scale the final result by $(\sqrt{2})^i$, where i is the number of iterations.
- This result shows that the wavelet's shape is determined entirely by the coefficients of the reconstruction filters.

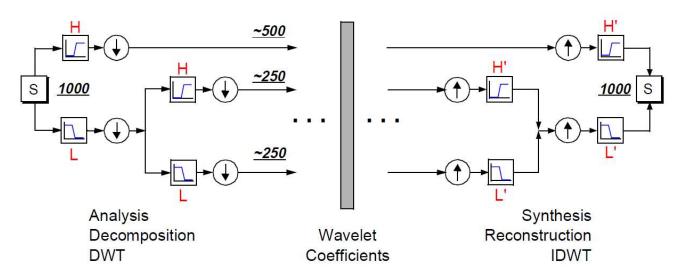
- This relationship has profound implications.
- It means that you cannot choose just any shape, call it a wavelet, and perform an analysis.
- At least, you can't choose an arbitrary wavelet waveform if you want to be able to reconstruct the original signal accurately.
- You are compelled to choose a shape determined by
- quadrature mirror decomposition filters.

- The **mother wavelet** function ψ is determined by the highpass filter, which also produces the **details** of the wavelet decomposition.
- There is an additional function associated with some but not all wavelets. This is the so-called **scaling** function, ϕ .
- It is determined by the lowpass quadrature mirror filters, and thus is associated with the **approximations** of the wavelet decomposition.
- Iteratively upsampling and convolving the lowpass filter produces a shape approximating the **scaling function**.

• MATLAB: s53WaveFilters.m

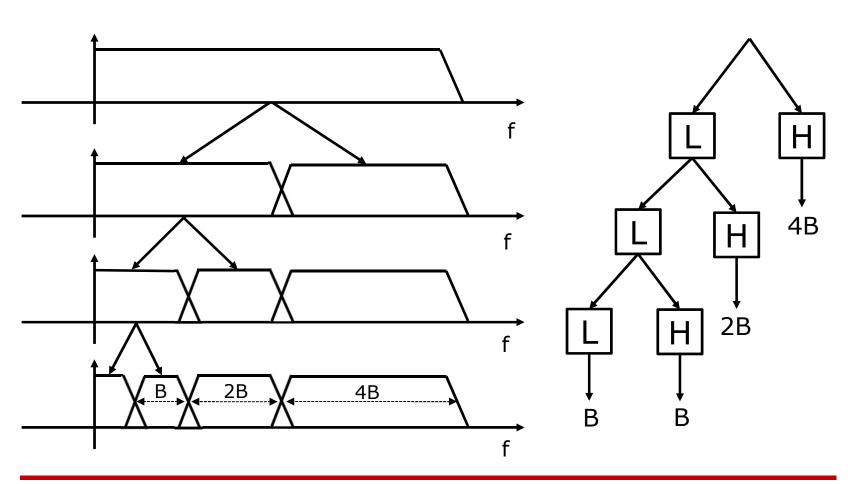


Multistep analysis-synthesis:

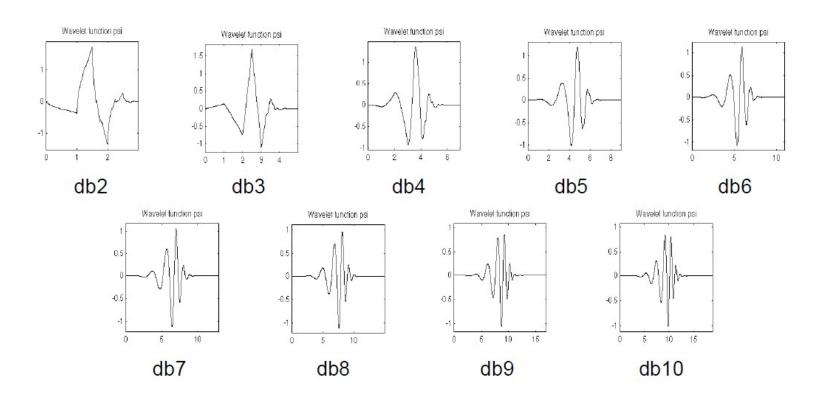


- Of course, there is no point breaking up a signal merely to have the satisfaction of immediately reconstructing it.
- We perform wavelet analysis because the coefficients thus obtained have many known uses, de-noising and compression being foremost among them.

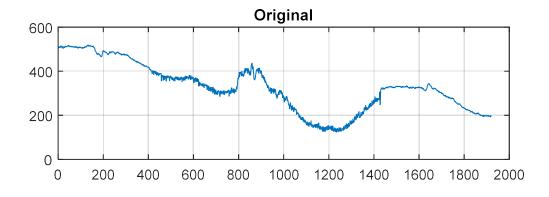
• Multistep analysis-synthesis:



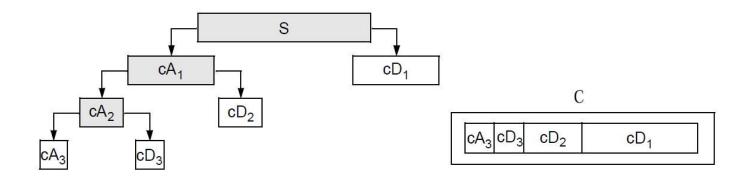
• Examples of wavelets: nine members of the Daubechies family.



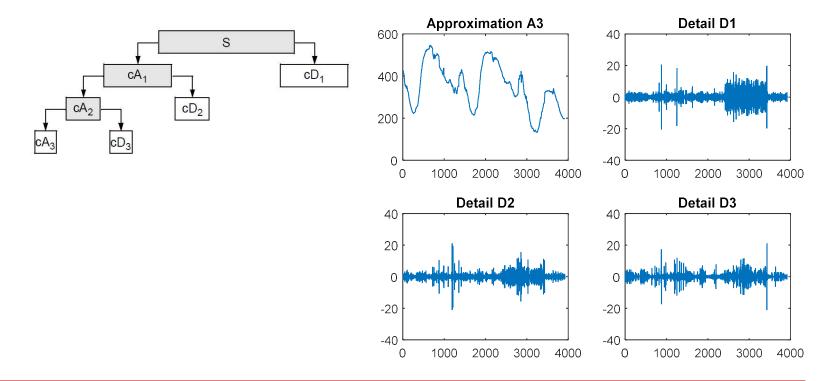
- Example of One-Dimensional Analysis:
 - ➤ This example involves a real-world signal electrical consumption measured over the course of three days.
 - This signal is particularly interesting because of noise introduced when a defect developed in the monitoring equipment as the measurements were being made.
 - Wavelet analysis effectively removes the noise.



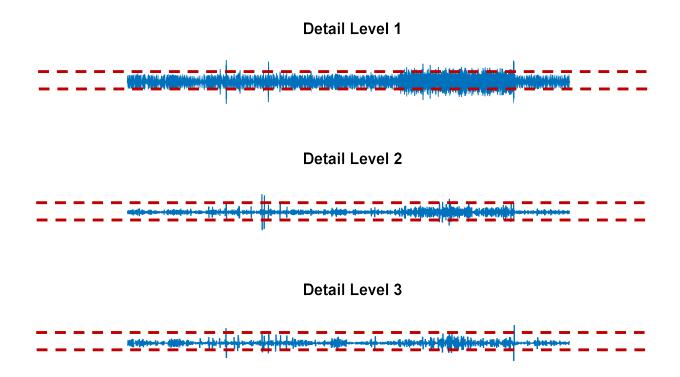
- Example of One-Dimensional Analysis:
 - ➤ Level 3 decomposition of the signal (again using the 'db3' wavelet.



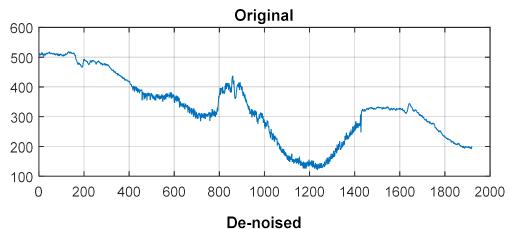
- Example of One-Dimensional Analysis:
 - ➤ Level 3 decomposition of the signal (again using the 'db3' wavelet.

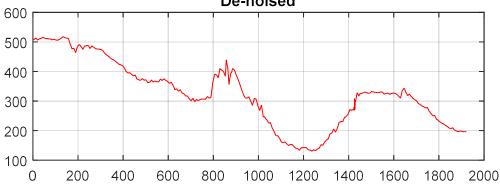


- Example of One-Dimensional Analysis:
 - > Setting a threshold

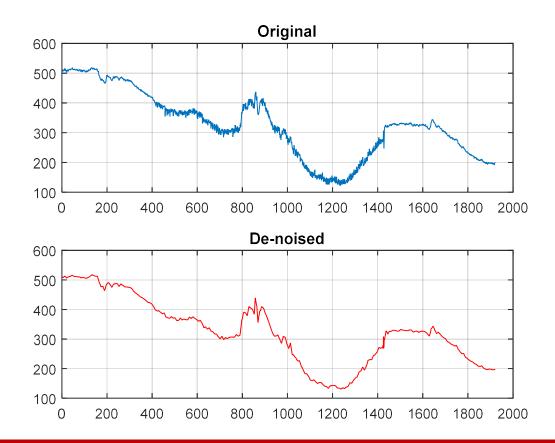


- Example of One-Dimensional Analysis:
 - > Reconstruction

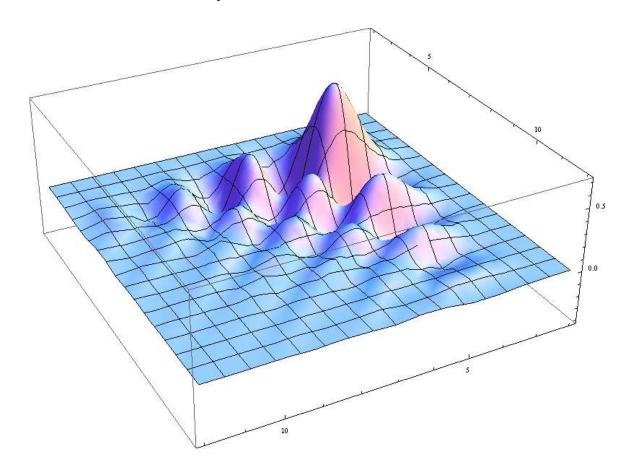




• MATLAB: s61Denoise1D.m



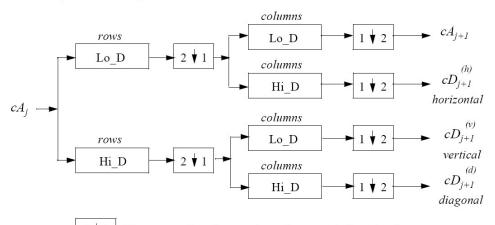
• Two-Dimensional Analysis



• Two-Dimensional Analysis

Two-Dimensional DWT

Decomposition step



Where: 2 ♥ 1 Downsa

Downsample columns: keep the even indexed columns.

1 ♥ 2 Downsample rows: keep the even indexed rows.

x Convolve with filter X the rows of the entry.

Convolve with fil

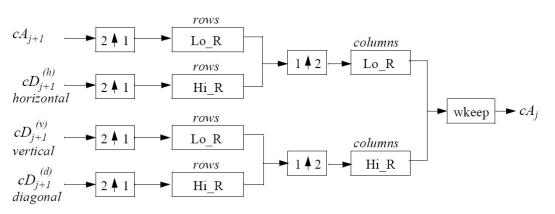
Convolve with filter X the columns of the entry.

Initialization $CA_0 = s$ for the decomposition initialization.

• Two-Dimensional Analysis

Two-Dimensional IDWT

Reconstruction step



Where:

□ ↓ 1 Upsample columns: insert zeros at odd-indexed columns.

1 \(\) 2 Upsample rows: insert zeros at odd-indexed rows.

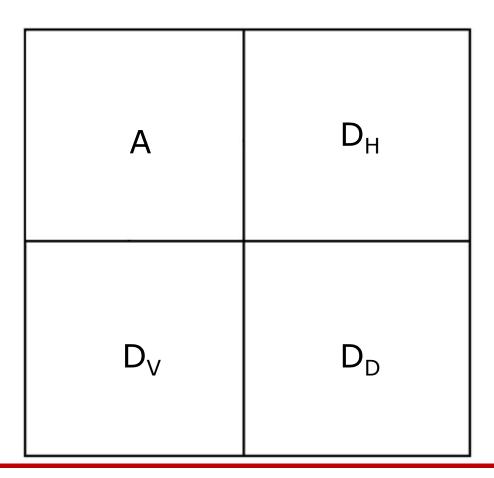
X Convolve with filter X the rows of the entry.

columns

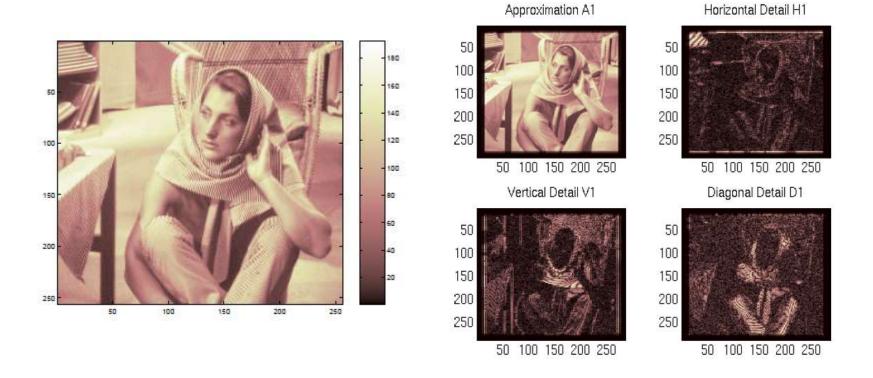
rows

X Convolve with filter X the columns of the entry.

• Two-Dimensional Analysis (one-step decomposition)



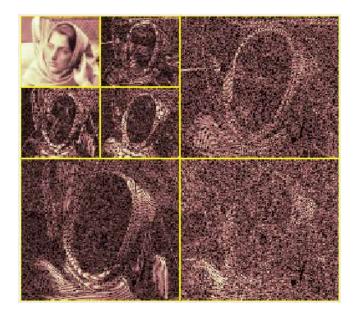
• Two-Dimensional Analysis (one-step decomposition)



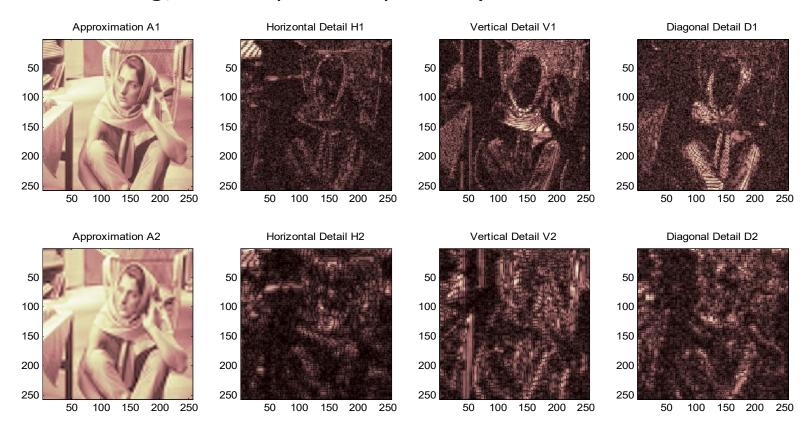
• Two-Dimensional Analysis (multiple-level decomposition)

A2	D2 _H	D1 _H
D2 _V	D2 _D	
D1 _V		D1 _D

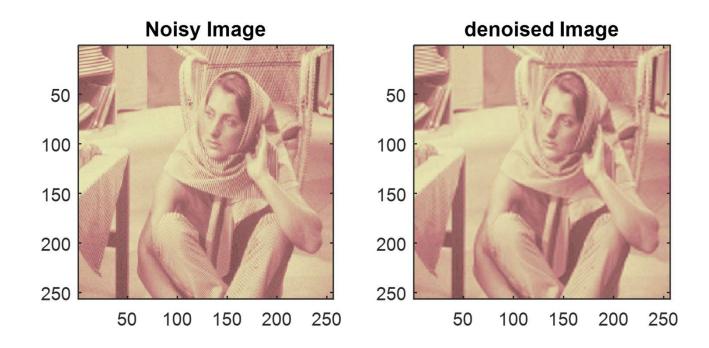
• Two-Dimensional Analysis (multiple-level decomposition)



• Example of Two-Dimensional Analysis (Extension to Image Denoising, Two-step Decomposition)



• Example of Two-Dimensional Analysis (Extension to Image Denoising, Two-step Decomposition): s72Denoise2Da.m



• Example of Two-Dimensional Analysis (Extension to Image Denoising, Two-step Decomposition): s73Denoise2Db.m





• Example of Two-Dimensional Analysis (Extension to Image Denoising, Two-step Decomposition): s74Denoise2Dc.m

