# **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

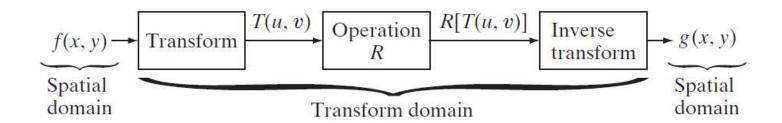
- T is called the **forward transform** of f.
- Given T we can recover f using the **inverse transform**.

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

- 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.

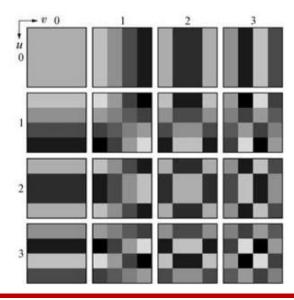
- **Discrete Cosine Transform** (DCT) is the basis for many image and video **compression algorithms**, especially the baseline JPEG and MPEG standards for compression of **still** and **video** images respectively.
- It is obtained by using the following (equal) kernels:

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

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

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

- 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.

- **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.

ullet We can **define** a transform coefficient **masking function**  $\chi$  which is constructed to **eliminate** the **basis images** that make the smallest contribution to the total sum

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

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

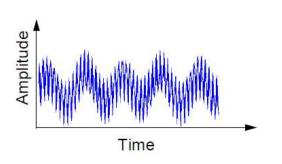
ullet The mean-square error between  $oldsymbol{G}$  and  $\widehat{oldsymbol{G}}$  approximation is

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

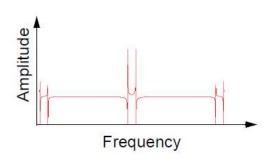
• The total mean-square approximation error thus is the sum of the variances of the discarded transform coefficients.

• 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**.

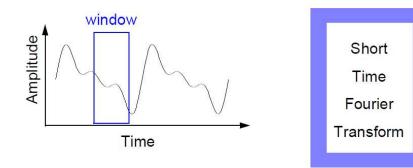


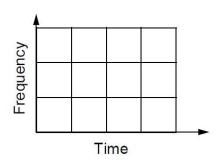




- Fourier analysis has a **serious drawback**. In transforming to the frequency domain, **time information is lost**.
- If a signal does not change much over time this drawback is not very important.

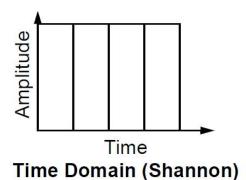
- However, most **interesting signals** contain numerous non-stationary or **transitory characteristics**.
- In an effort to **correct** this deficiency, Dennis Gabor (1946) adapted the Fourier transform to analyze only **a small section** of the signal at a **time**.
  - Short-Time Fourier Transform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

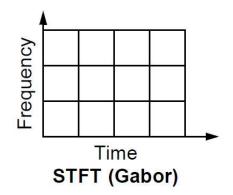


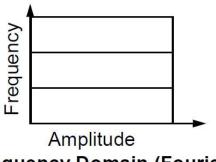


• Precision is determined by the size of the window.

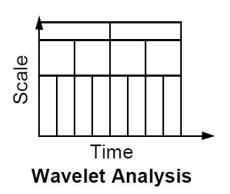
• Wavelet analysis represents the next logical step: a windowing technique with variable-sized regions.





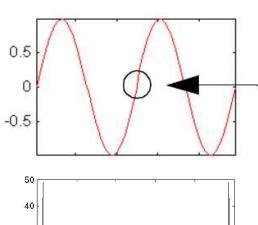


**Frequency Domain (Fourier)** 

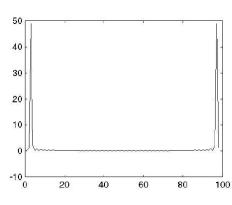


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• One major **advantage** afforded by wavelets is the ability to perform **local analysis** — that is, to analyze a localized area of a larger signal.



Sinusoid with a small discontinuity



**Fourier Coefficients** 

Values of Ca,b Coefficients for a = [1:1:32]

29

27

25

23

21

19

17

15

13

11

9

7

5

3

1

20

40

60

80

100

120

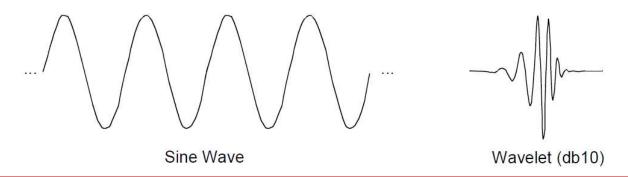
140

160

180

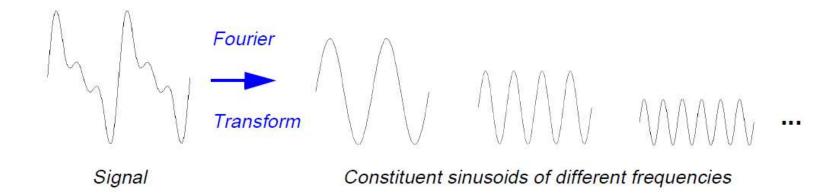
Wavelet Coefficients

- 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  $\tau$  are called **scale** and **translation** parameters, respectively.

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

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

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

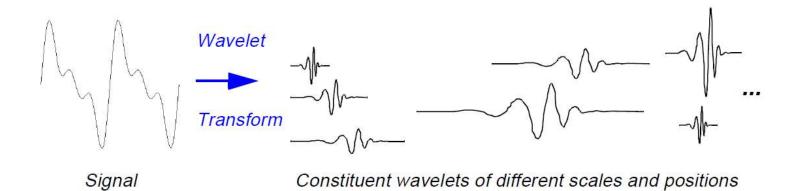
where  $\Psi(\mu)$  is the Fourier transform of  $\psi(x)$ .

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

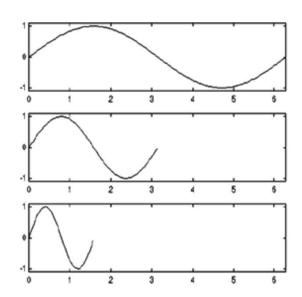
$$C_{\psi} < \infty$$

• 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.

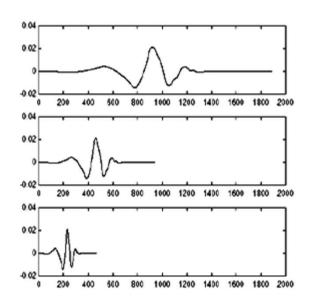


$$f(t) = \sin(t) \quad ; \quad a = 1$$

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

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

• The scale factor works exactly the same with wavelets.

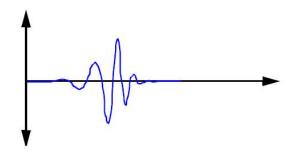


$$f(t) = \psi(t)$$
 ;  $a = 1$ 

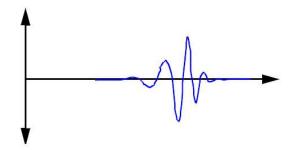
$$f(t) = \psi(2t)$$
 ;  $a = \frac{1}{2}$ 

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

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

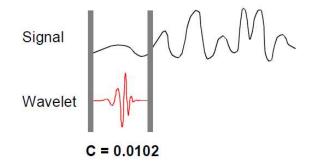


Wavelet function  $\psi(t)$ 

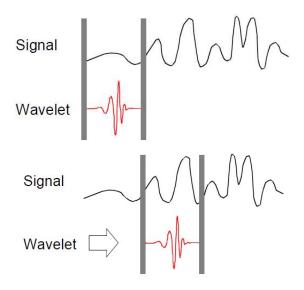


Shifted wavelet function  $\psi(t-k)$ 

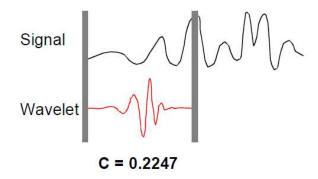
- The CWT is the sum over all time of the signal multiplied by scaled, shifted versions of the wavelet.
- Computing the CWT:
  - 1. Take a wavelet and **compare** it to a section at the start of the original signal.
  - 2. Calculate a number, C, that represents **how closely correlated** the wavelet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

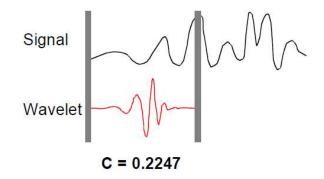


- Computing the CWT:
  - 4. Scale (stretch) the wavelet and repeat steps 1 through 3.



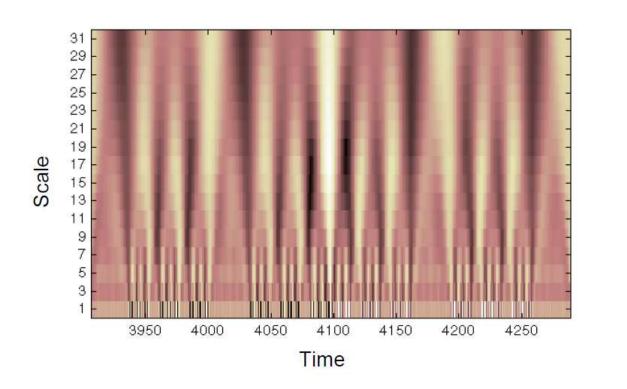
5. Repeat steps 1 through 4 for all scales.

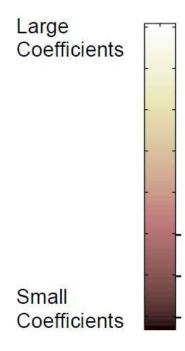
- Computing the CWT:
  - 4. Scale (stretch) the wavelet and repeat steps 1 through 3.



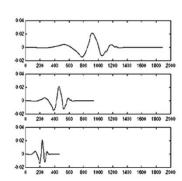
- 5. Repeat steps 1 through 4 for all scales.
- When you're done, you'll have the coefficients produced at different scales by different sections of the signal.

How to make sense of all these coefficients?





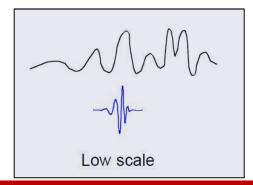
- There is a correspondence between wavelet scales and frequency:
  - ➤ Low scale → Compressed wavelet → Rapidly changing details → High frequency



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

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

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



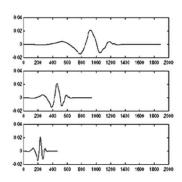
Signal

Wavelet



High scale

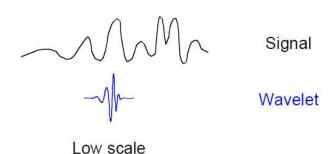
- There is a correspondence between wavelet scales and frequency:
  - ➤ High scale a → Stretched wavelet → Slowly changing, coarse features → Low frequency

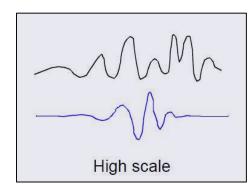


$$f(t) = \psi(t)$$
 ;  $a = 1$ 

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

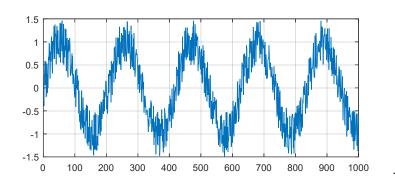
$$f(t) = \psi(4t)$$
 ;  $a = \frac{1}{4}$ 

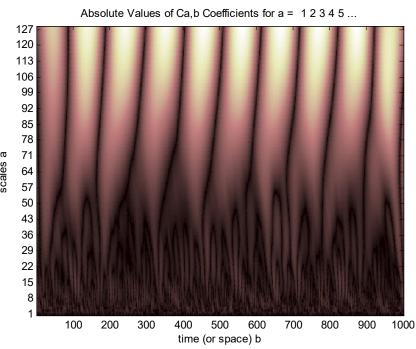




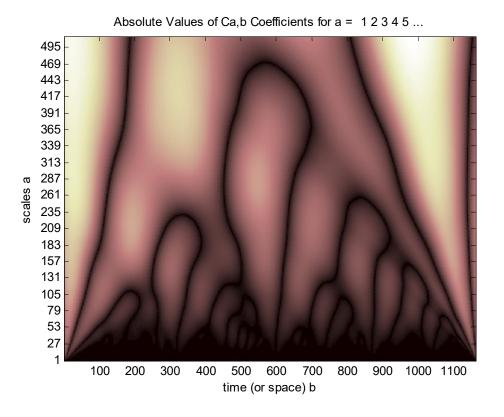
• There is a correspondence between wavelet scales and frequency:

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





• MATLAB: s32Lunar.m



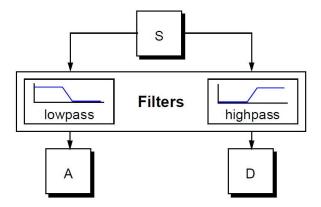
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- What is "continuous" about the Continuous Wavelet Transform (CWT) are the scales at which it operates.
- CWT can operate at **every scale**, from that of the original signal up to some maximum scale which you determine.
- The CWT is also **continuous** 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**.

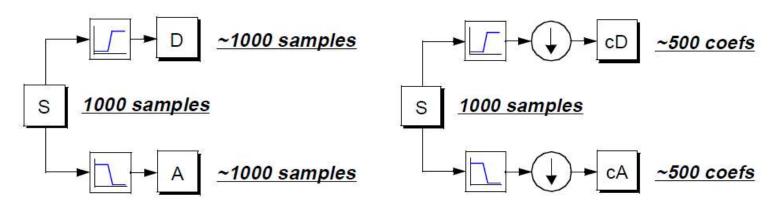
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- The **approximations** are the high-scale, low-frequency components of the signal.
- The **details** are the low-scale, high-frequency components.
- The filtering process, at its most basic level, looks like this:



• 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.

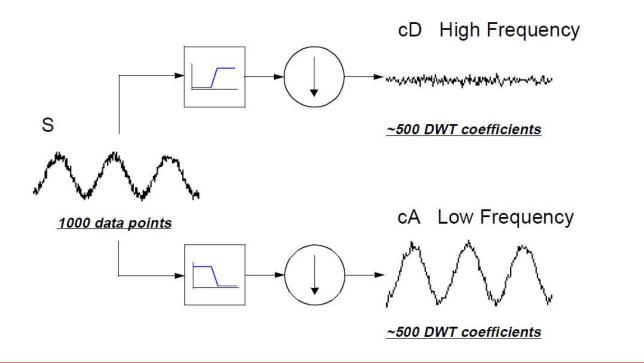


• The process on the right, which includes downsampling, produces **DWT coefficients**.

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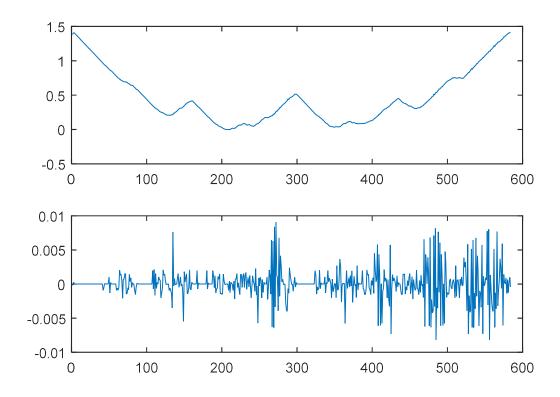
• To correct this problem, we introduce the notion of **downsampling**.

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

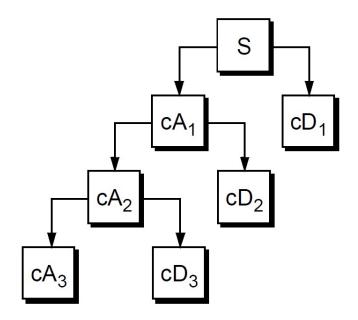


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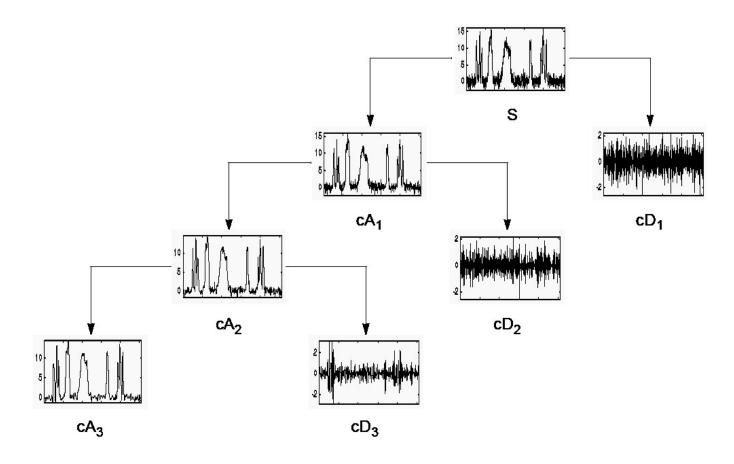
• MATLAB: s39ApproxDet.m



- 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.



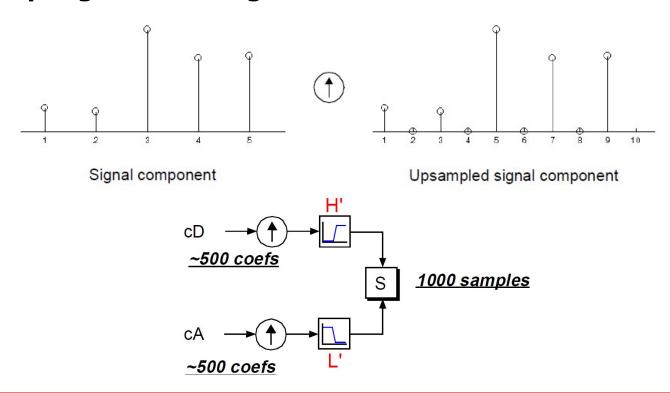
• The decomposition process can be iterated.



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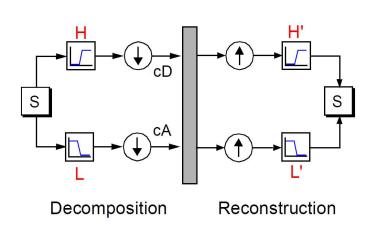
- We have learned how the discrete wavelet transform can be used to analyze, or **decompose**, signals and images.
- The other half of the story is how those components can be **assembled** back into the original signal with no loss of information.
- This process is called **reconstruction**, or **synthesis**.
- •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**.

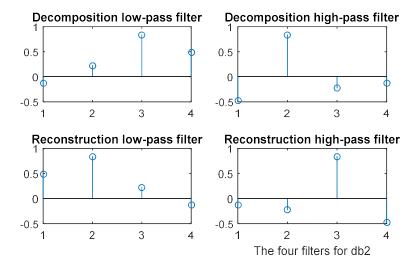


• By carefully choosing the decomposition and reconstruction filters we can "cancel out" the effects of aliasing caused by the subsampling.

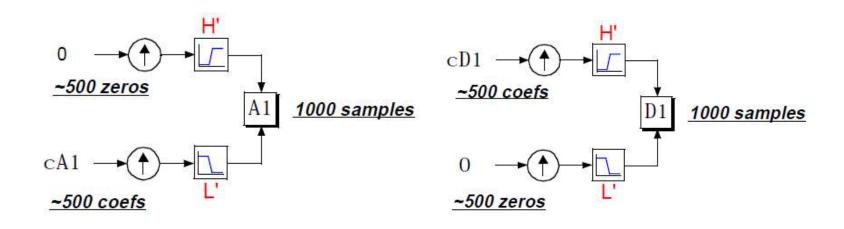
## • Quadrature mirror filters:



• MATLAB: s44Filters.m

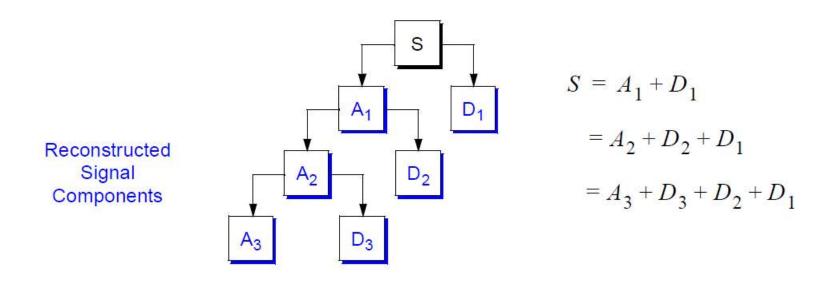


• It is also possible to reconstruct the approximation A and detail D themselves **from their coefficient vectors** cA and cD.



• 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 Lprime 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 = [-0.1294 \ 0 \ -0.2241 \ 0 \ 0.8365 \ 0 \ -0.4830 \ 0]
```

## • Example:

Convolve the upsampled vector with the original lowpass filter:

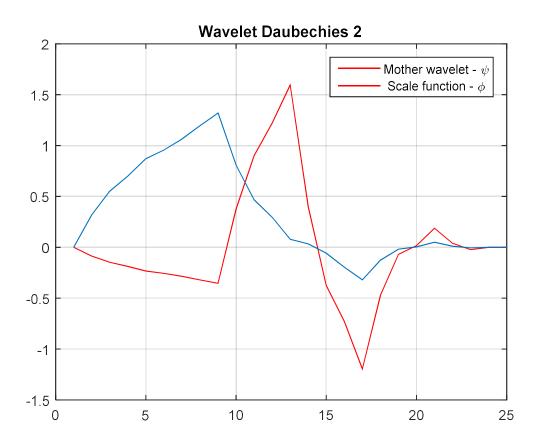
```
H2 = conv(HU, Lprime);
```

- ➤ 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.

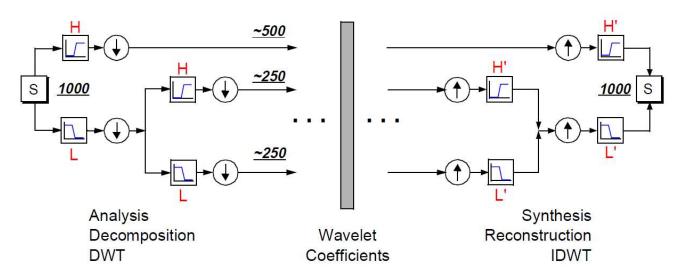
- This relationship has profound implications.
- It means that you cannot choose just any shape, call it a wavelet, and perform an analysis.
- At least, you can't choose an arbitrary wavelet waveform if you want to be able to reconstruct the original signal accurately.
- You are compelled to choose a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

- The **mother wavelet** function  $\psi$  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**,  $\phi$ .
- 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 reconstruction lowpass filter produces a shape approximating the **scaling function**.

• MATLAB: s53WaveFilters.m

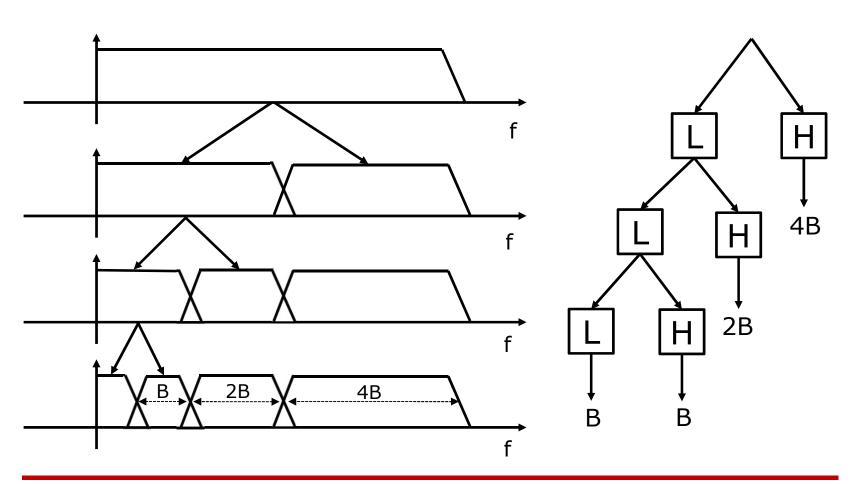


Multistep analysis-synthesis:

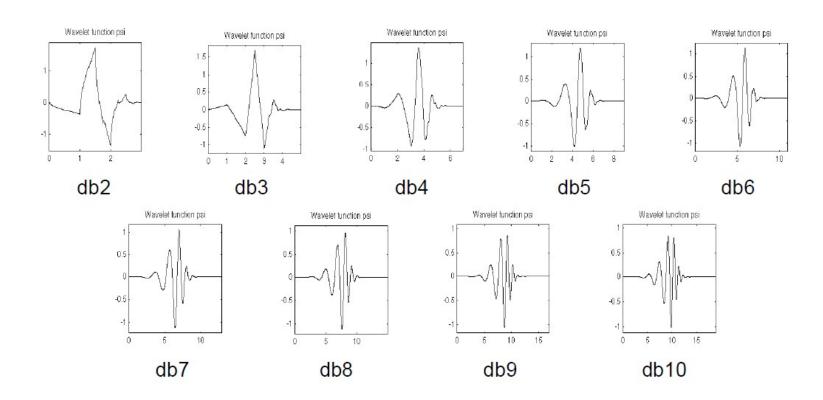


- Of course, there is no point breaking up a signal merely to have the satisfaction of immediately reconstructing it.
- We perform wavelet analysis because the coefficients thus obtained have many known uses, de-noising and compression being foremost among them.

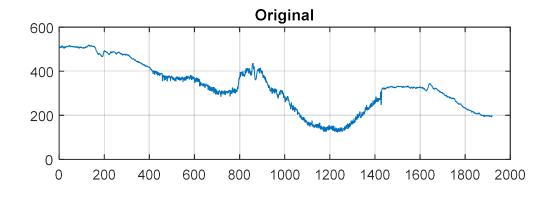
• Multistep analysis-synthesis:



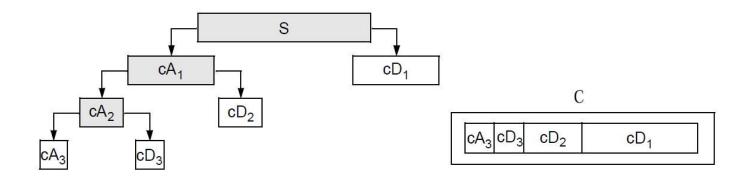
• Examples of wavelets: nine members of the Daubechies family.



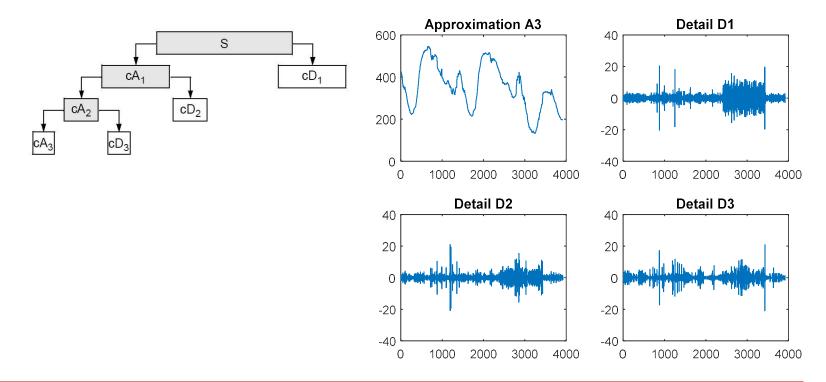
- Example of One-Dimensional Analysis:
  - ➤ This example involves a real-world signal electrical consumption measured over the course of three days.
  - This signal is particularly interesting because of noise introduced when a defect developed in the monitoring equipment as the measurements were being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



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

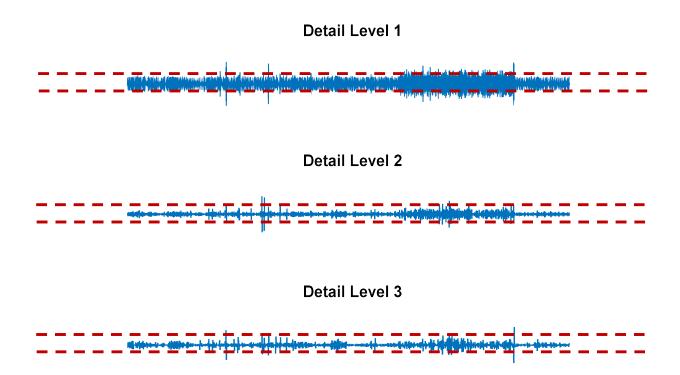


- Example of One-Dimensional Analysis:
  - ➤ Level 3 decomposition of the signal (using the 'db3' wavelet).



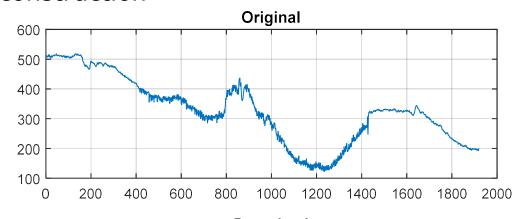
7/4/2017

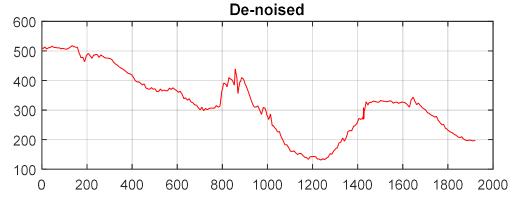
- Example of One-Dimensional Analysis:
  - > Setting a threshold



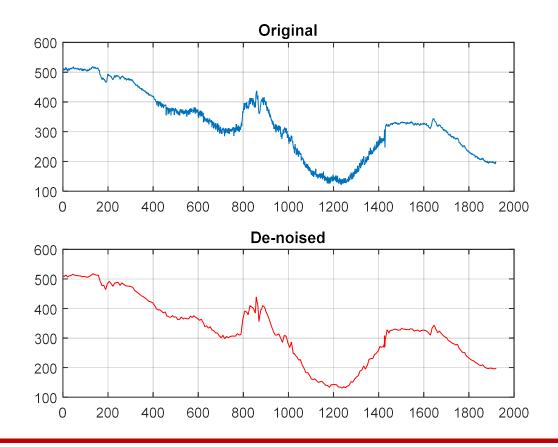
7/4/2017

- Example of One-Dimensional Analysis:
  - > Reconstruction



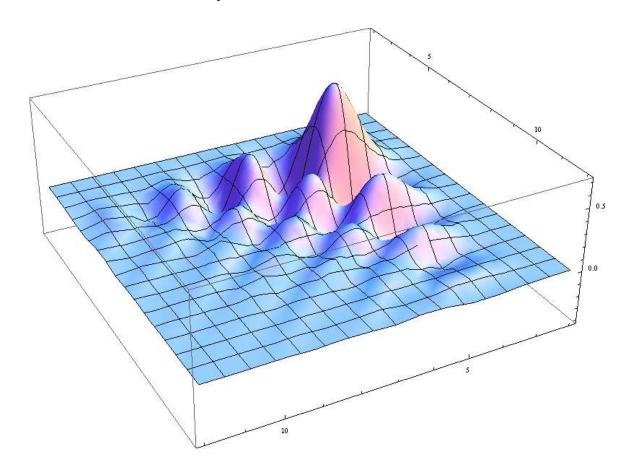


• MATLAB: s61Denoise1D.m



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• Two-Dimensional Analysis

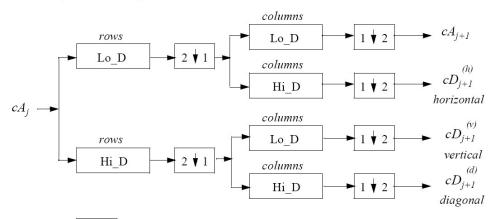


• Two-Dimensional Analysis

#### **Two-Dimensional DWT**

#### **Decomposition step**

rows



Where:  $2 \nmid 1$  Downsample columns: keep the even indexed columns.

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

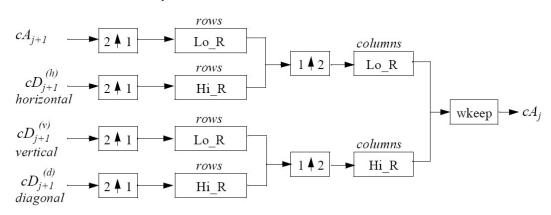
X Convolve with filter X the rows of the entry.

**Initialization**  $CA_0 = s$  for the decomposition initialization.

• Two-Dimensional Analysis

#### **Two-Dimensional IDWT**

#### Reconstruction step



Where:

2 ▲ 1 Upsample columns: insert zeros at odd-indexed columns.

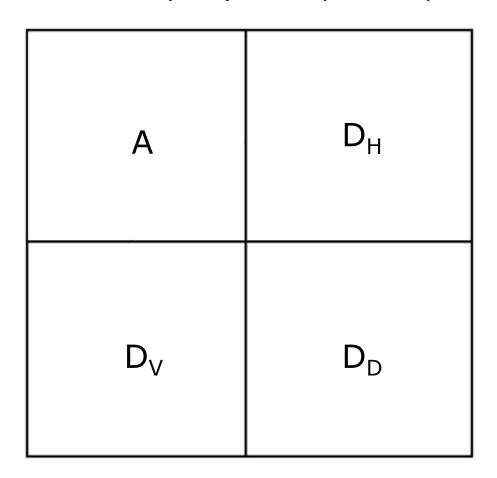
1 \( \) 2 Upsample rows: insert zeros at odd-indexed rows.

X Convolve with filter X the rows of the entry.

columns

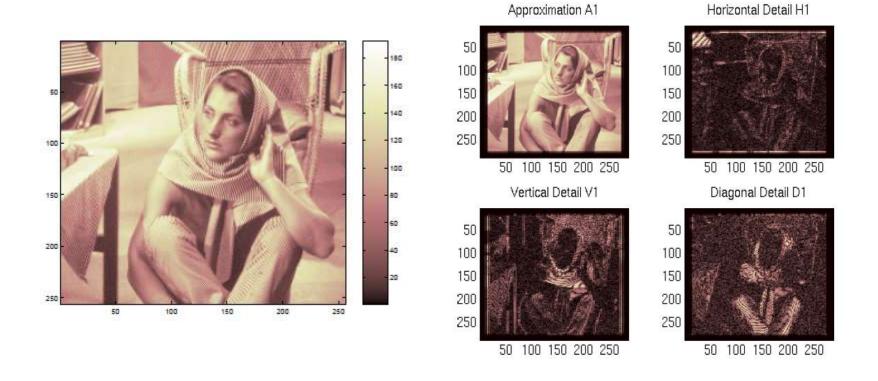
X Convolve with filter X the columns of the entry.

• Two-Dimensional Analysis (one-step decomposition)



7/4/2017

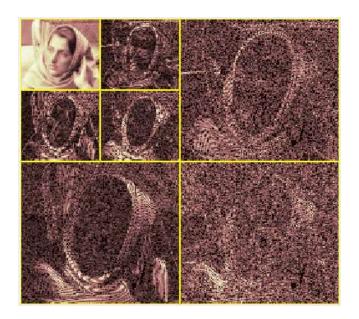
• Two-Dimensional Analysis (one-step decomposition)



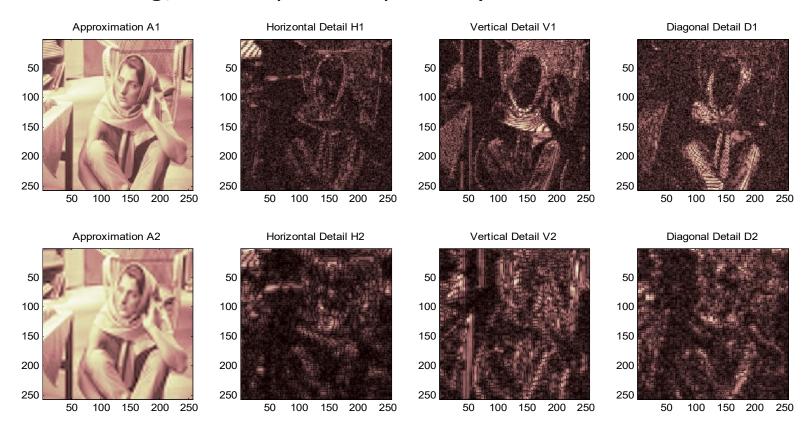
• Two-Dimensional Analysis (multiple-level decomposition)

A2	D2 <sub>H</sub>	D1 <sub>H</sub>
D2 <sub>V</sub>	D2 <sub>D</sub>	
D1 <sub>V</sub>		D1 <sub>D</sub>

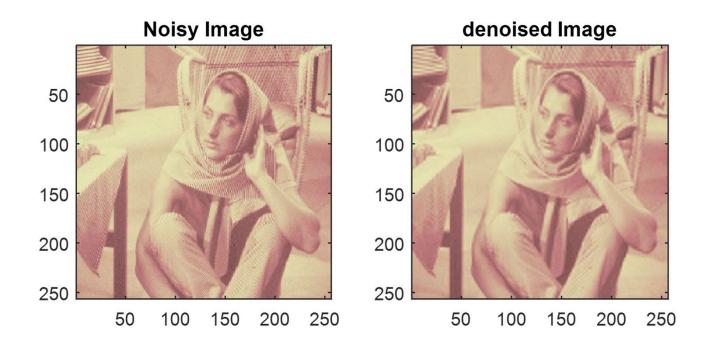
• Two-Dimensional Analysis (multiple-level decomposition)



• 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

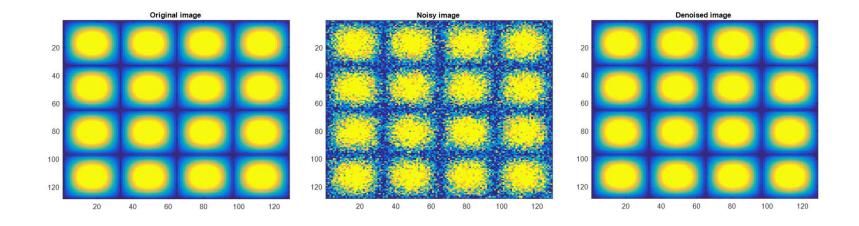


• Example of Two-Dimensional Analysis (Extension to Image Denoising, Two-step Decomposition): s73Denoise2Db.m





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



# 5. Further Reading

