
Image processing

© 2005, Denis Zorin

How computer images work?

Continuous real image

↓ *Digitization
(e.g. scanning)*

**square array of numbers
(abstract pixels)**

↓ *each physical pixel
covers an area*

display (physical pixels)

↓ *The eye blurs pixels into
continuous image*

perceived image

© 2005, Denis Zorin

What can go wrong?

Pipeline: sample - process - reconstruct

All kinds of artifacts can appear

- jaggies
- alias patterns
- moire patterns
- temporal aliasing (wheels going wrong way)

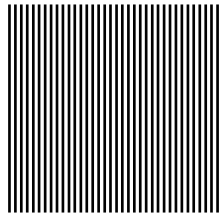
Question: how do we avoid all this?



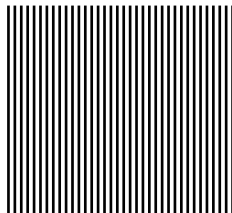
© McMillan '98

© 2005, Denis Zorin

Aliasing



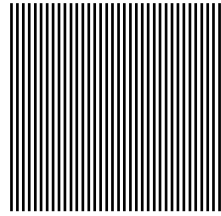
Slightly different
frequency



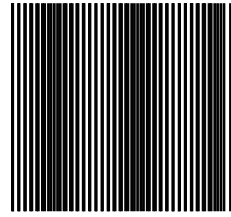
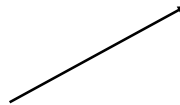
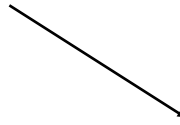
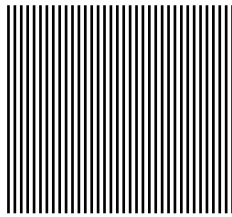
What can go wrong?

© 2005, Denis Zorin

Aliasing



Slightly different
frequency

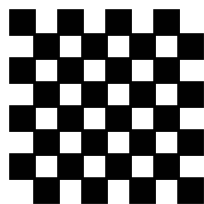


Lower frequency
appears

© 2005, Denis Zorin

Shrinking

Naïve 1.5x shrinking : drop 2 out of 3



What do we get?

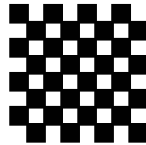
© 2005, Denis Zorin

Shrinking

Naïve 1.5x shrinking: drop 1 out of 3



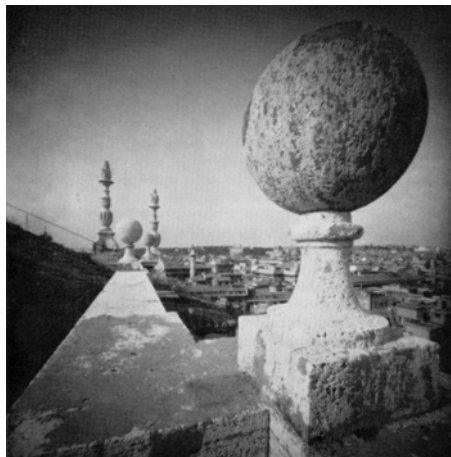
But we want



(impossible, not enough samples)

© 2005, Denis Zorin

Shrinking



original



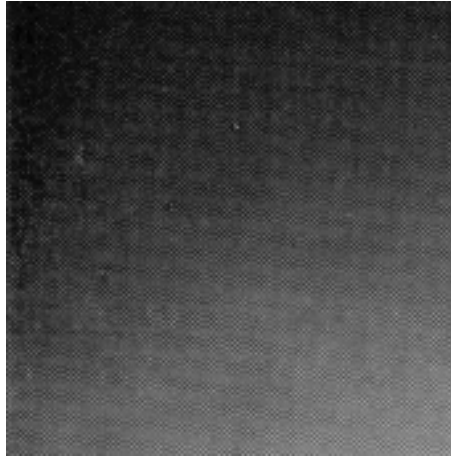
resized, nearest neighbor



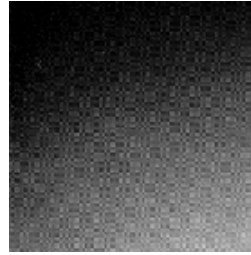
resized, 11-point filter

© 2005, Denis Zorin

Shrinking



original



resized, nearest neighbor



resized, 11-point filter

© 2005, Denis Zorin

Frequency analysis

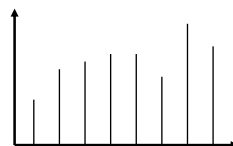
The key to fighting aliasing is to avoid frequencies we cannot represent

Big question:

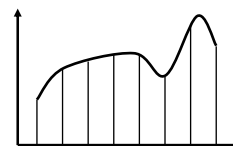
When can we reconstruct a continuous signal from samples?



original



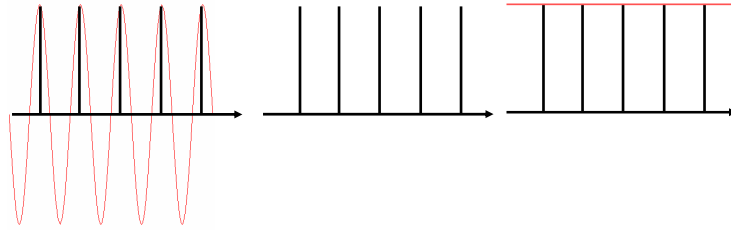
samples



reconstructed

© 2005, Denis Zorin

Frequency analysis



original

samples

reconstructed

Need enough samples, to be more precise,
sampling frequency should be twice the frequency
of the wave

© 2005, Denis Zorin

Frequency analysis

- decompose everything into waves
- mathematically convenient to use complex waves

$$A \cos(\omega t + \varphi)$$

amplitude \nearrow \nwarrow phase
frequency \uparrow

$$A e^{i(\omega t + \varphi)} = A(\cos(\omega t + \varphi) + i \sin(\omega t + \varphi))$$

© 2005, Denis Zorin

Fourier transform

Decomposes into freq. components

$$\text{FT}[x](\omega) = \int_{-\infty}^{\infty} x(t)e^{-i\omega t} dt$$

Inverse transform reconstructs the function

$$\text{IFT}[X](t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(\omega)e^{i\omega t} d\omega$$

© 2005, Denis Zorin

Examples of FT pairs

FT(delta function) = constant or wave

FT(constant or wave) = delta function

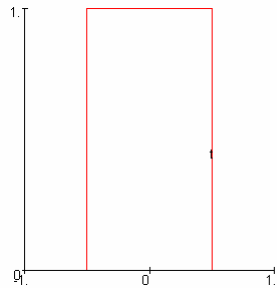
delta function (not quite a function):

for all $x \neq 0$, $\delta(x) = 0$

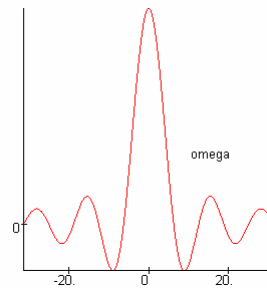
for all $\varepsilon > 0$, $\int_{-\varepsilon}^{+\varepsilon} \delta(x) dx = 1$

© 2005, Denis Zorin

Examples of FT pairs



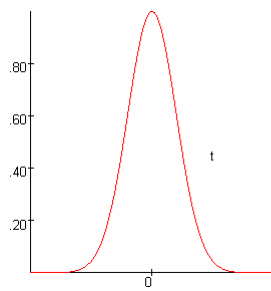
box



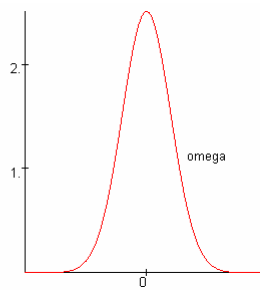
sinc

© 2005, Denis Zorin

Examples of FT pairs



Gaussian

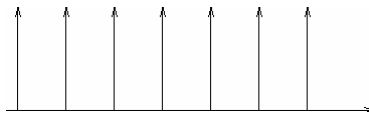


Gaussian

© 2005, Denis Zorin

Comb function

$$\text{comb}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT)$$



FT of a comb is also a comb

© 2005, Denis Zorin

Shannon theorem

(Rough idea, not mathematically precise formulation)

Almost any function or signal can be decomposed into a sum of sine waves with different frequencies:

$$f(t) = \sum_j a_j \exp i(\omega_j t + \varphi_j)$$

amplitude frequency phase

A function can be reconstructed from samples if the sampling frequency is at least twice the max. frequency in the FT of the function.

© 2005, Denis Zorin

General idea

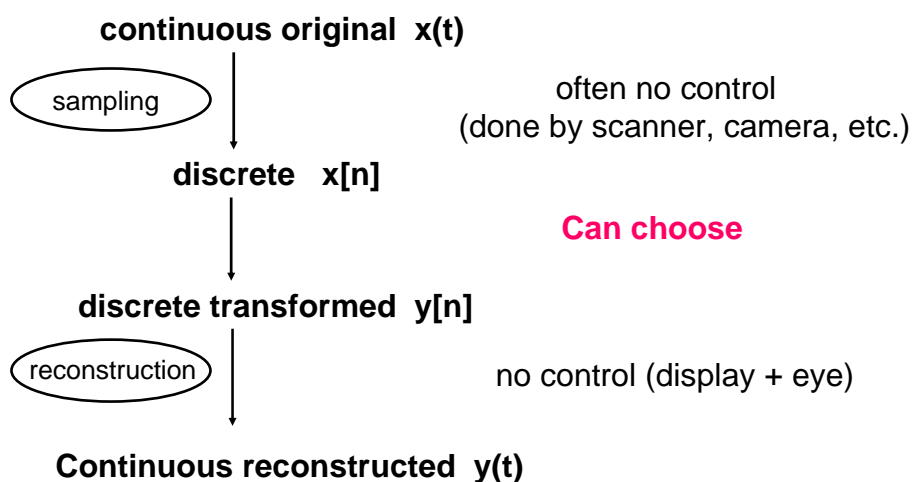
Keep the part of the signal (image) we can represent (low frequencies).

Eliminate the part of the signal (image) we cannot represent (high frequencies).

Smooth slowly changing signals have lower high frequency components, smoothing eliminates high frequencies,

© 2005, Denis Zorin

Image transformation



© 2005, Denis Zorin

Convolution

Fourier transform of a product of two functions $x(t)$ and $y(t)$ is a convolution of $X(\omega)=FT[x](\omega)$ and $Y(\omega)=FT[y](\omega)$, denoted $X(\omega)*Y(\omega)$. Conversely, Inverse Fourier transform of a product of $X(\omega)$ and $Y(\omega)$ is the convolution $x(t)*y(t)$.

Convolution is defined as follows:

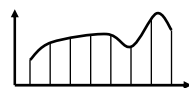
$$x(t) * y(t) = \int_{-\infty}^{\infty} x(s)y(t - s)ds$$

Convolution can be thought of as local averaging of x using y as a filter.

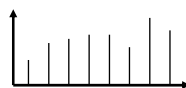
© 2005, Denis Zorin

Sampling and reconstruction

Sampling = multiplication by comb



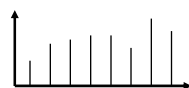
original



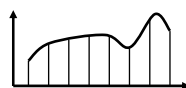
samples

$$\hat{x}(t) = \text{sinc}(t)x(t)$$

Reconstruction (ideal) = convolution with sinc



samples



reconstructed

$$x(t) = \text{sinc}(t) * \hat{x}(t)$$

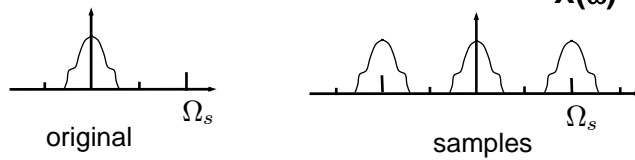
© 2005, Denis Zorin

Sampling and reconstruction

Frequency domain

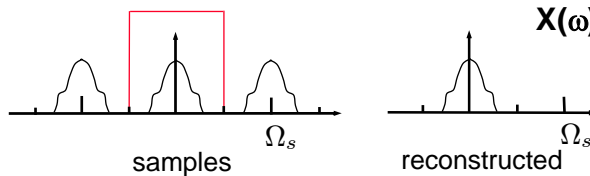
Sampling = convolution with comb
= replication and shift

$$\hat{X}(\omega) = \sum_n X(\omega - n\Omega_s)$$



Reconstruction = multiplication by box (get rid of extra copies)

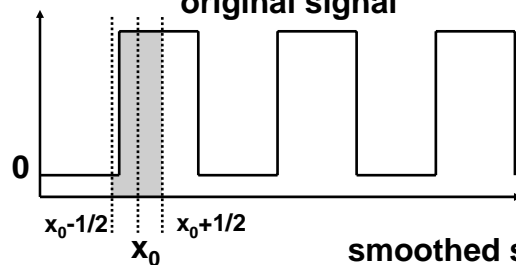
$$X(\omega) = \sum_n \text{box}(\omega) \hat{X}(\omega)$$



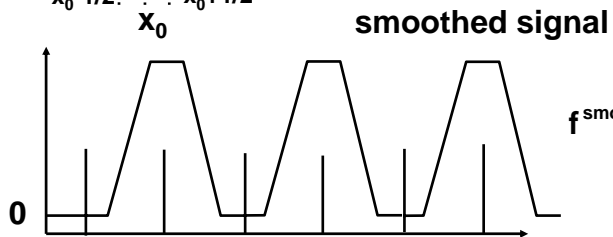
© 2005, Denis Zorin

Continuous smoothing

Smoothing can be done by local integration:
original signal



slide the integration interval along the x-axis

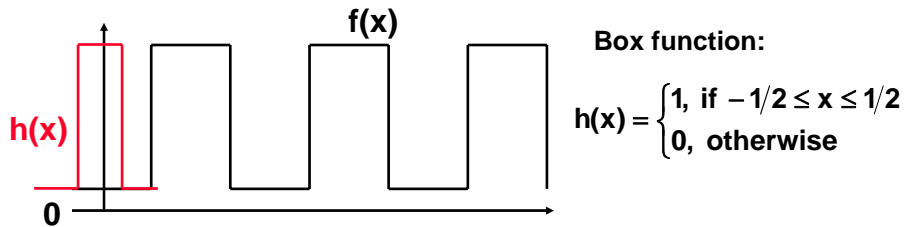


$$f^{\text{smoothed}}(x_0) = \int_{x_0-1/2}^{x_0+1/2} f(x) dx$$

© 2005, Denis Zorin

Continuous smoothing

Another way to look at the same integral:



$$f^{\text{smoothed}}(x_0) = \int_{x_0-1/2}^{x_0+1/2} f(x) dx = \int_{-\infty}^{\infty} h(x_0 - x) f(x) dx$$

This is convolution with $h(x)$!
 Can replace box with some other bump-like function $h(x)$, resulting in better smoothing.

© 2005, Denis Zorin

Filter quality

Why some filters $h(t)$ are better than others?

Recall that we want to eliminate high frequency components, and keep low-frequency components.

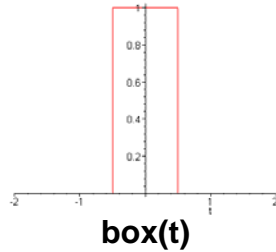
We can evaluate the filter by looking what it does to sine waves of different frequencies (frequency response). For an ideal filter,

$$\int_{-\infty}^{\infty} h(x_0 - x) \sin(\omega x) dx$$

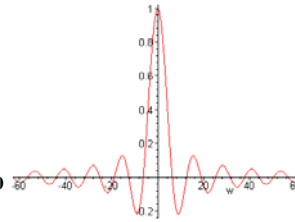
should be zero for frequencies $>$ sampling freq./2
 and 1 for frequencies $<$ sampling freq./2

© 2005, Denis Zorin

Box filter response



$$\int_{-\infty}^{\infty} \text{box}(x_0 - x) \sin \omega x dx = \frac{\sin \omega/2}{\omega/2} \sin \omega x_0$$



We have got the same wave back, but with different amplitude. The higher the frequency, the lower the amplitude. But the amplitude decreases as $1/\omega$, which is not very fast.

© 2005, Denis Zorin

Ideal filter

A function called sinc has the ideal response, that is, for low frequencies 1 for high frequencies zero:

$$\text{sinc}(x) = \frac{\sin \pi x}{\pi x}$$

$$\int_{-\infty}^{\infty} \text{sinc}(x_0 - x) \sin \omega x dx = \sin \omega x_0 \quad \text{if } \omega < \pi$$

$$\int_{-\infty}^{\infty} \text{sinc}(x_0 - x) \sin \omega x dx = 0 \quad \text{if } \omega > \pi$$

Why not use it? It is not zero anywhere, except a few points -- we would have to do infinite summations!

© 2005, Denis Zorin