

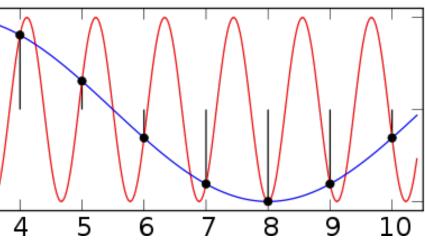
THE UNIVERSITY OF BRITISH COLUMBIA

CPSC 425: Computer Vision

2 3

Image Credit: https://en.wikibooks.org/wiki/Analog_and_Digital_Conversion/Nyquist_Sampling_Rate

unless otherwise stated slides are taken or adopted from **Bob Woodham, Jim Little** and **Fred Tung**)



Lecture 6: Sampling

Menu for Today

Topics:

- **Sampling** theory
- Nyquist rate

Readings:

- Today's Lecture: Szeliski 2.3, Forsyth & Ponce (2nd ed.) 4.5, 4.6

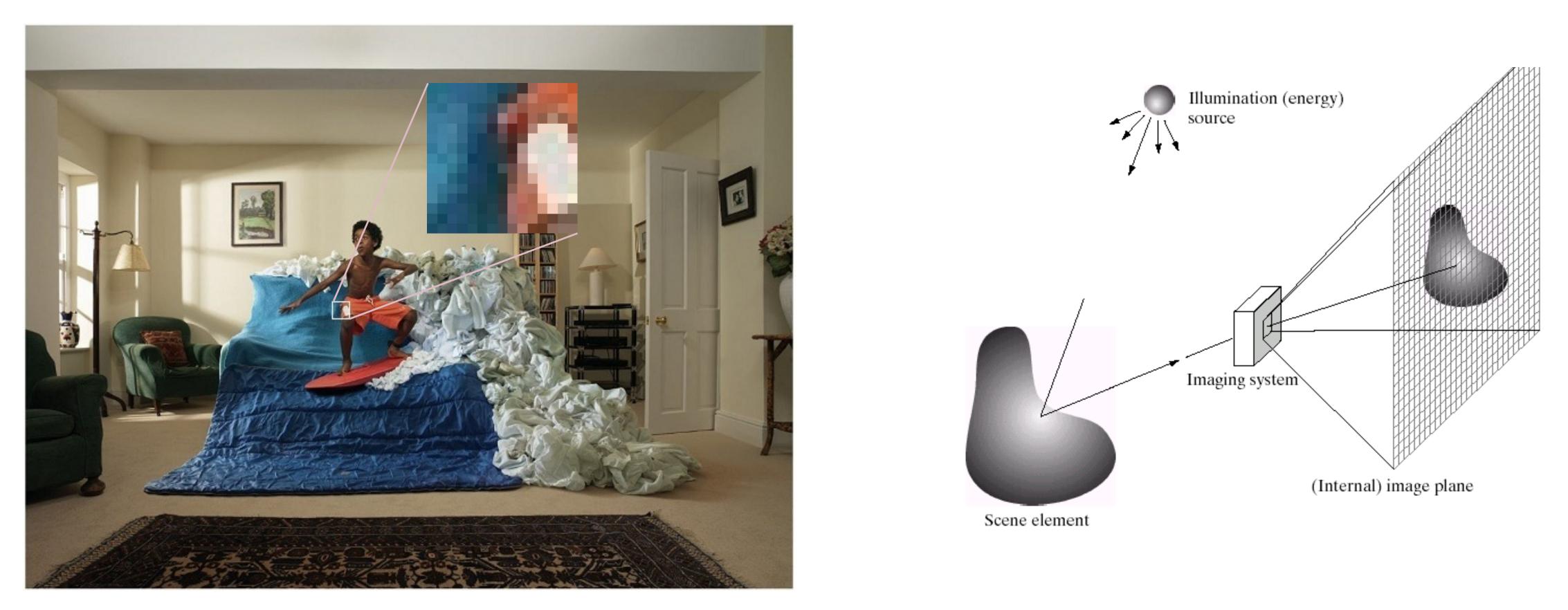
Reminders:

- Color Filter Arrays - **Image** encoding

— Assignment 1: Image Filtering and Hybrid Images due December 30th



Reminder



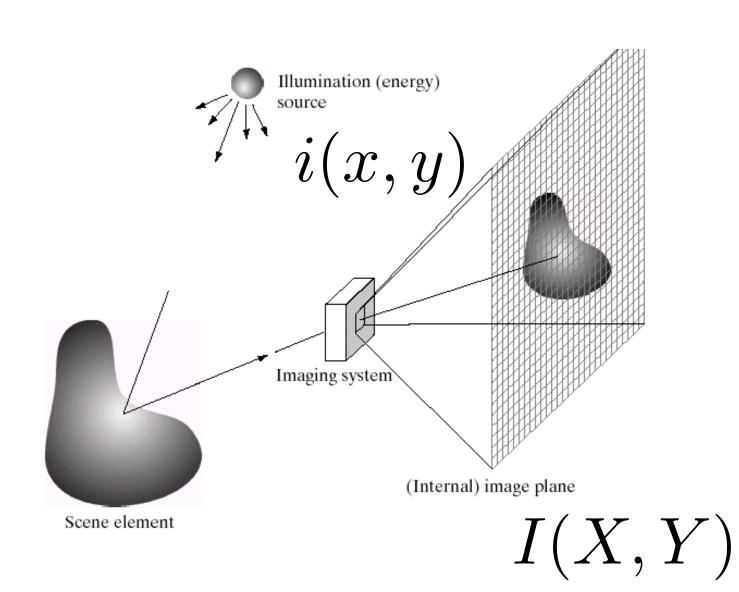
Images are a discrete, or sampled, representation of a continuous world

What is **Sampling**?



A continuous function $i(x, y, \lambda)$ is presented at the image sensor at each time instant

How do we convert this to a **digital signal** (array of numbers) $I(x, y, \lambda)$?

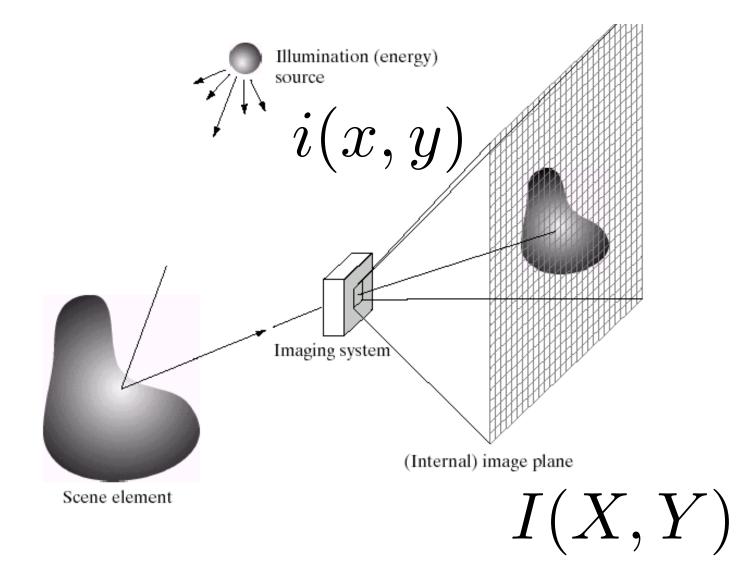


What is **Sampling**?



A continuous function $i(x, y, \lambda)$ is presented at the image sensor at each time instant

How can we **manipulate**, e.g., resample, this digital signal correctly?



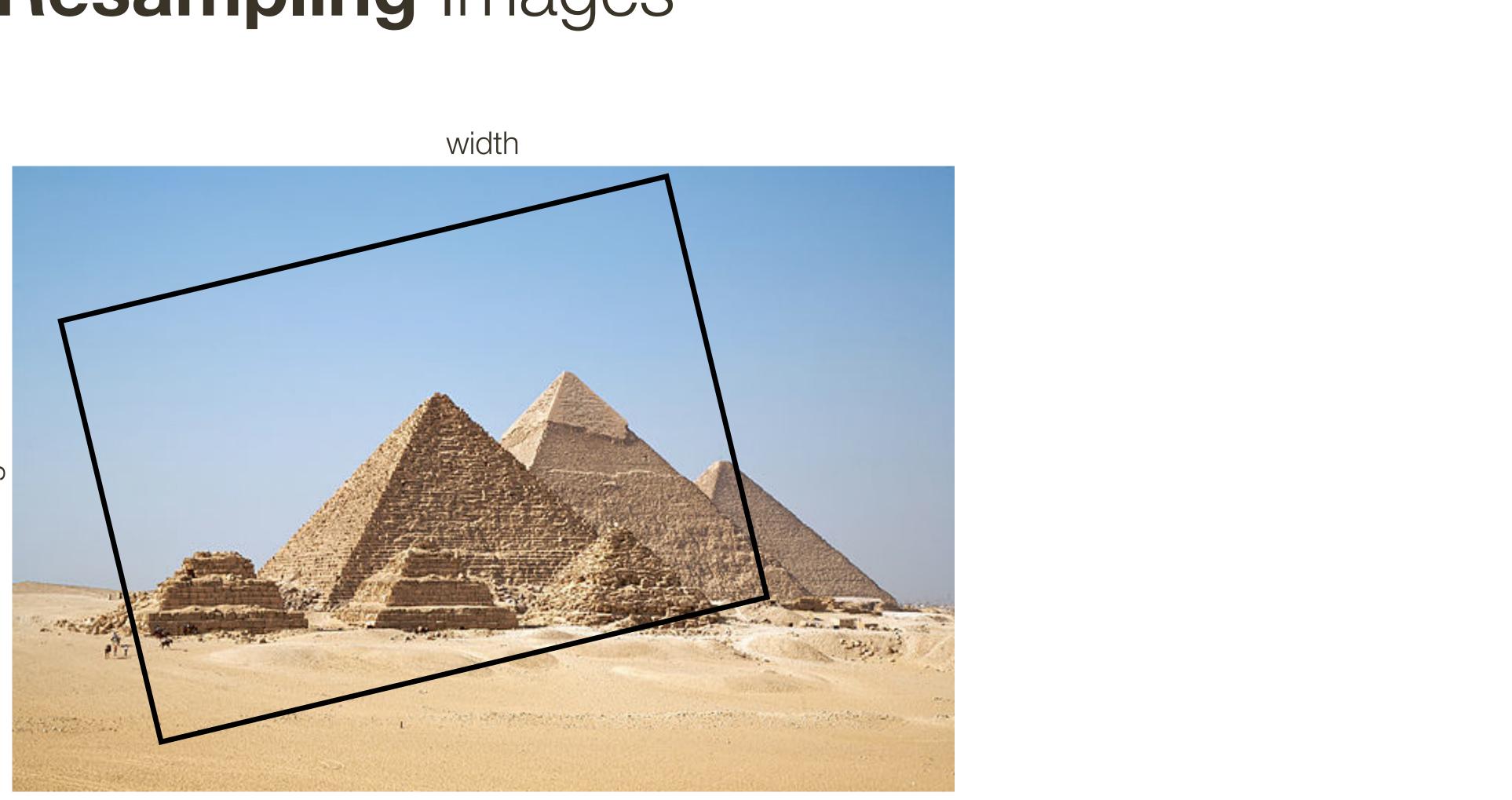
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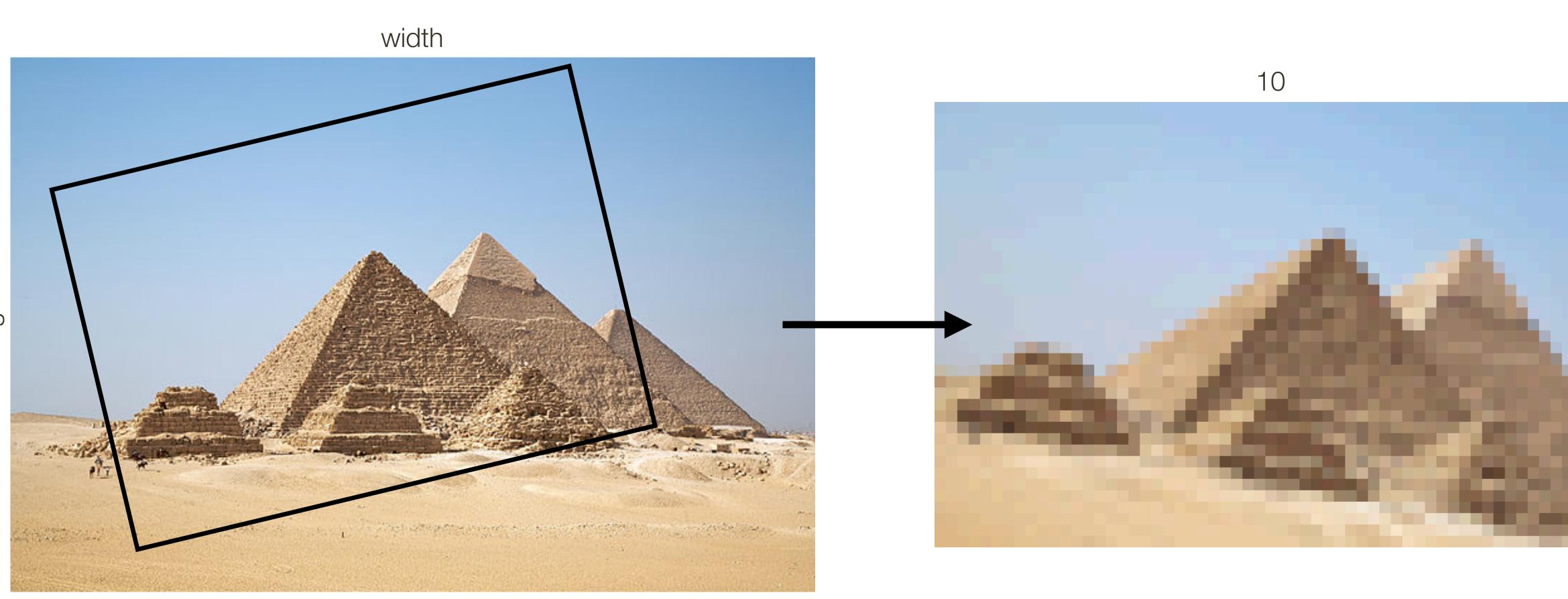
width



height

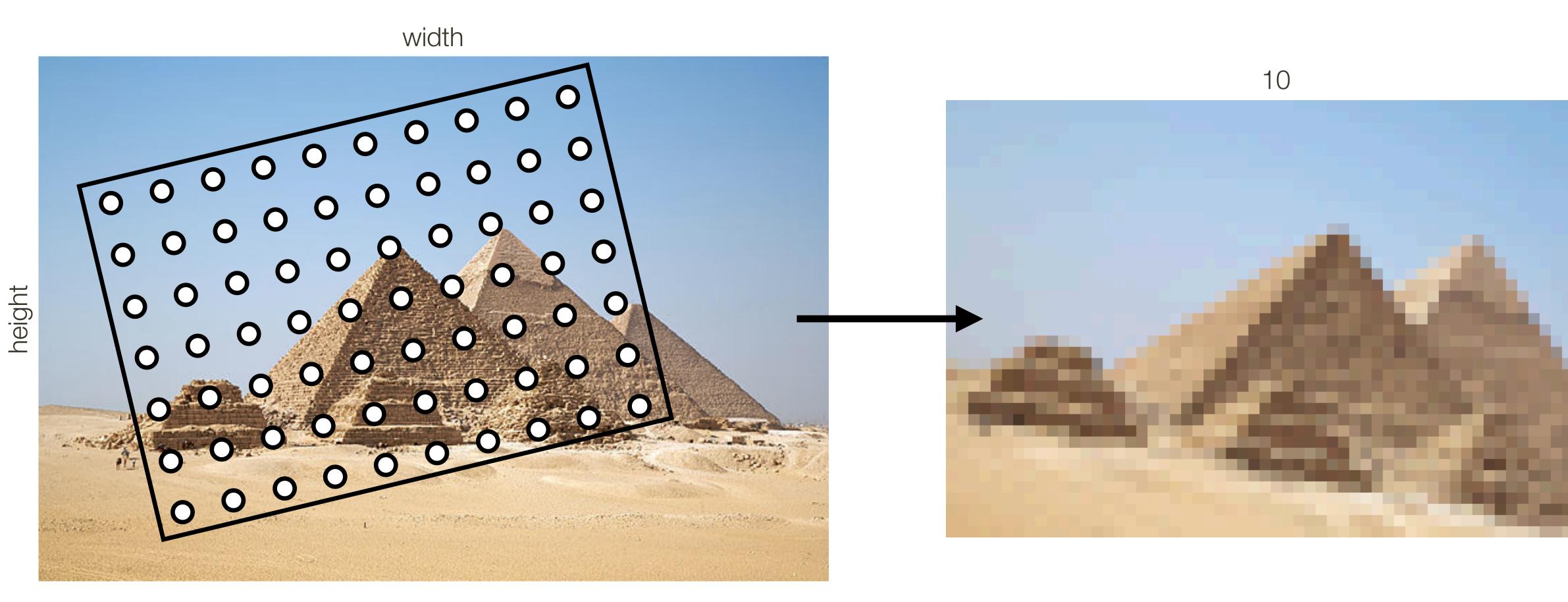






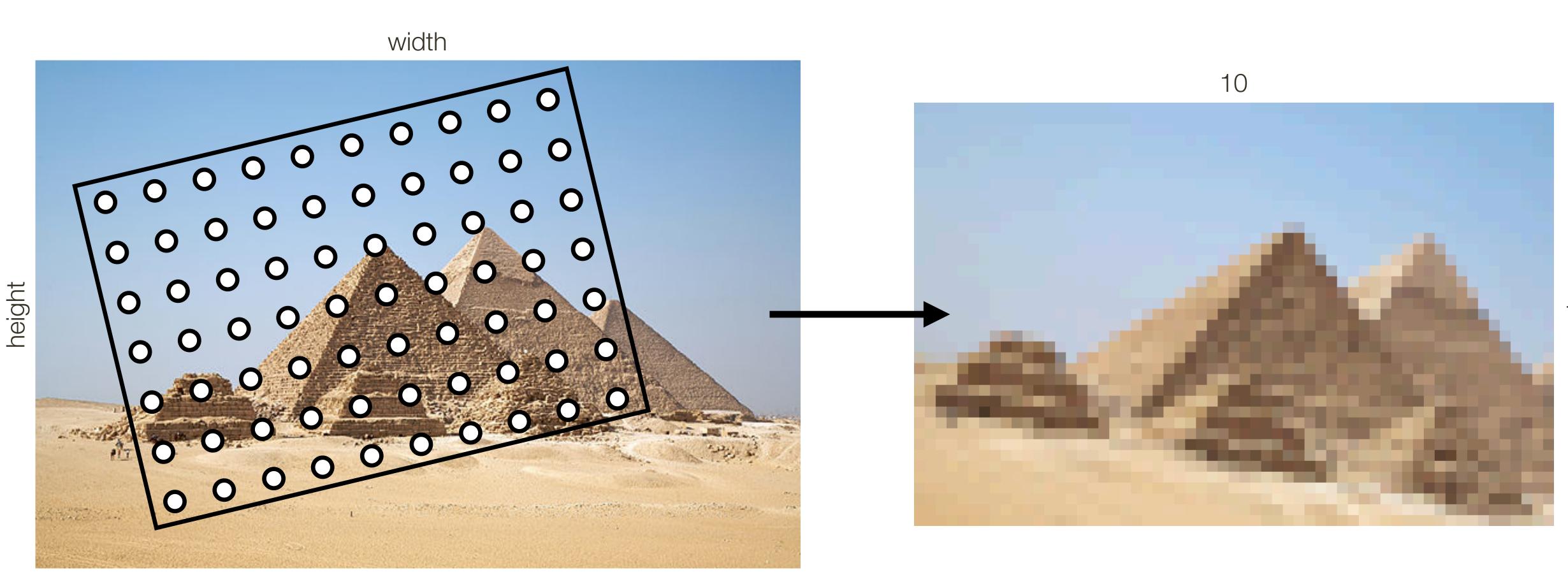








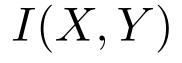




How do we correctly generate samples to resample or warp an image?

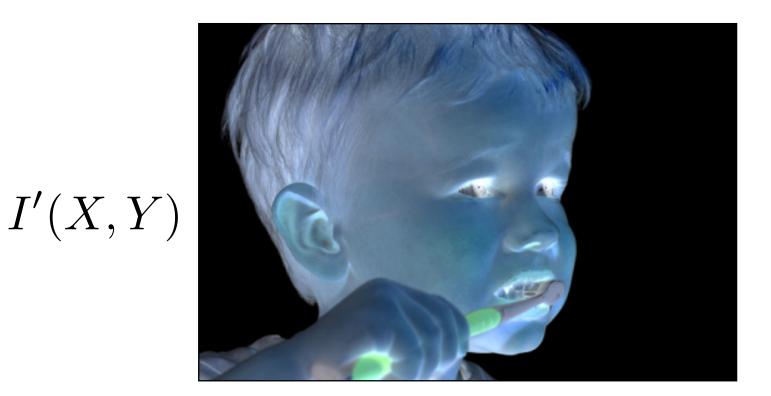


What types of transformations can we do?





Filtering



changes range of image function

I(X, Y)



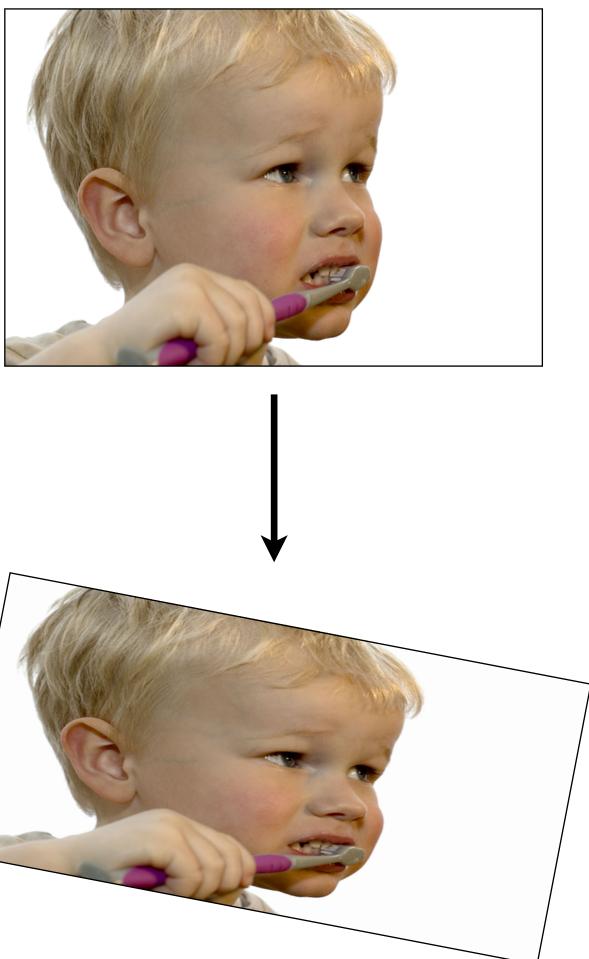
Warping



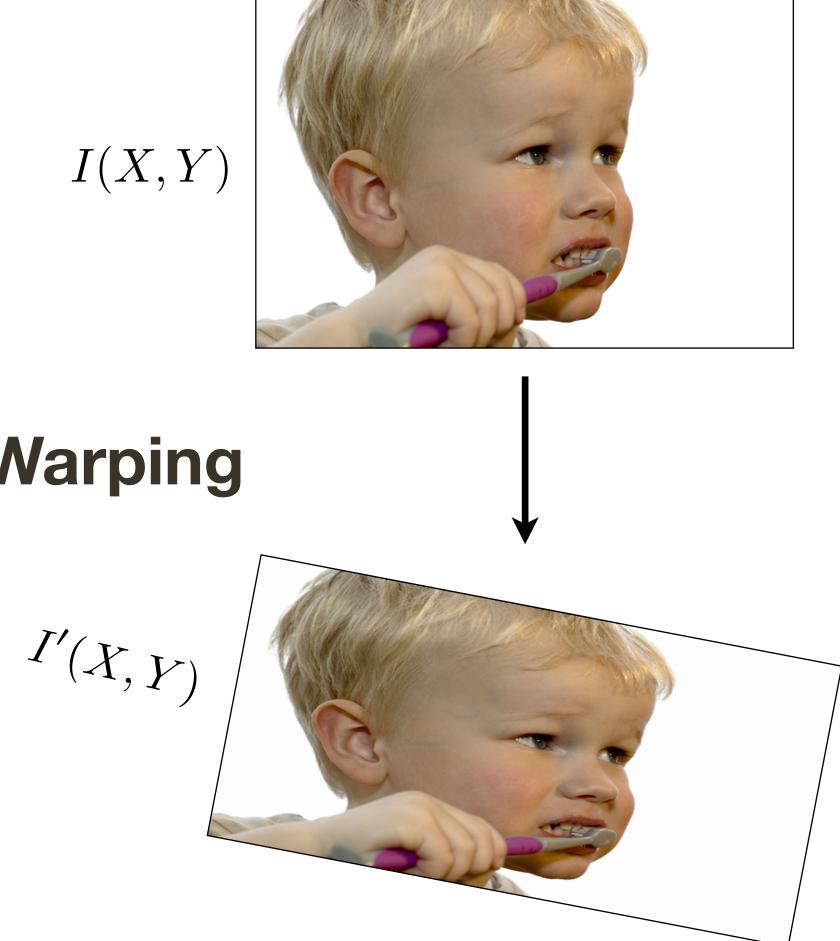
changes domain of image function

Slide Credit: Ioannis (Yannis) Gkioulekas (CMU)

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Warping

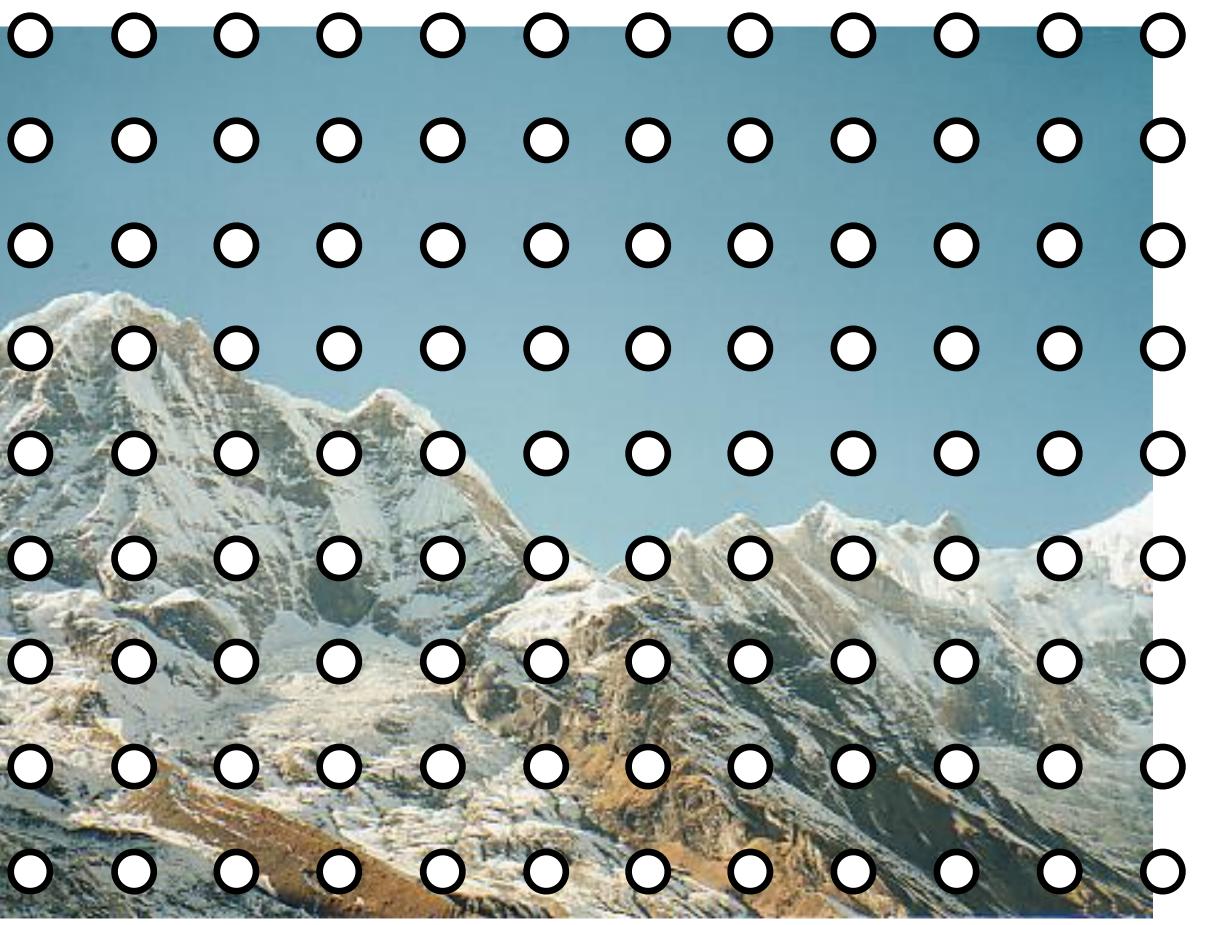


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Goal: Resample the image to get a lower resolution counterpart

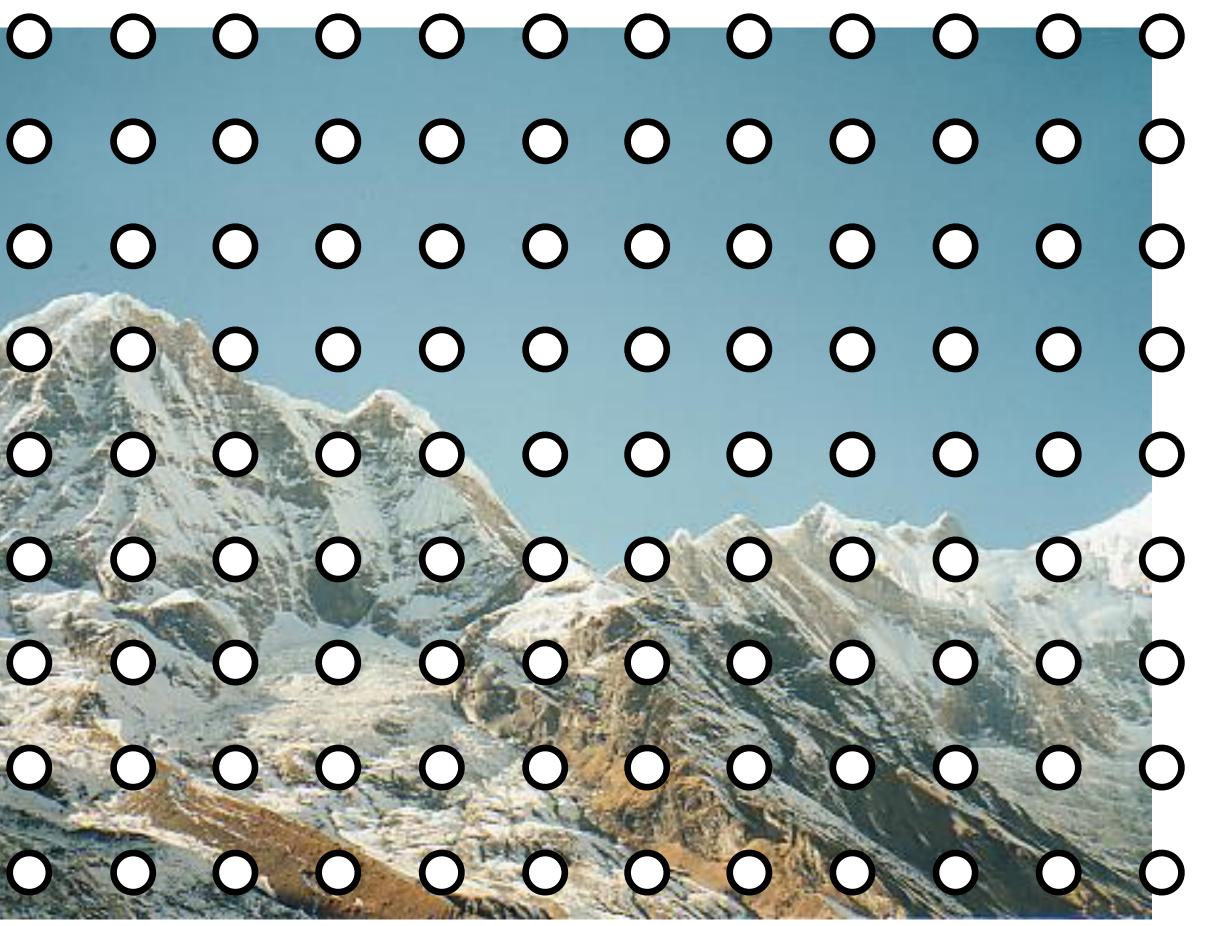
What is the simplest way to do this (e.g., produce image 1/5 of original size)?





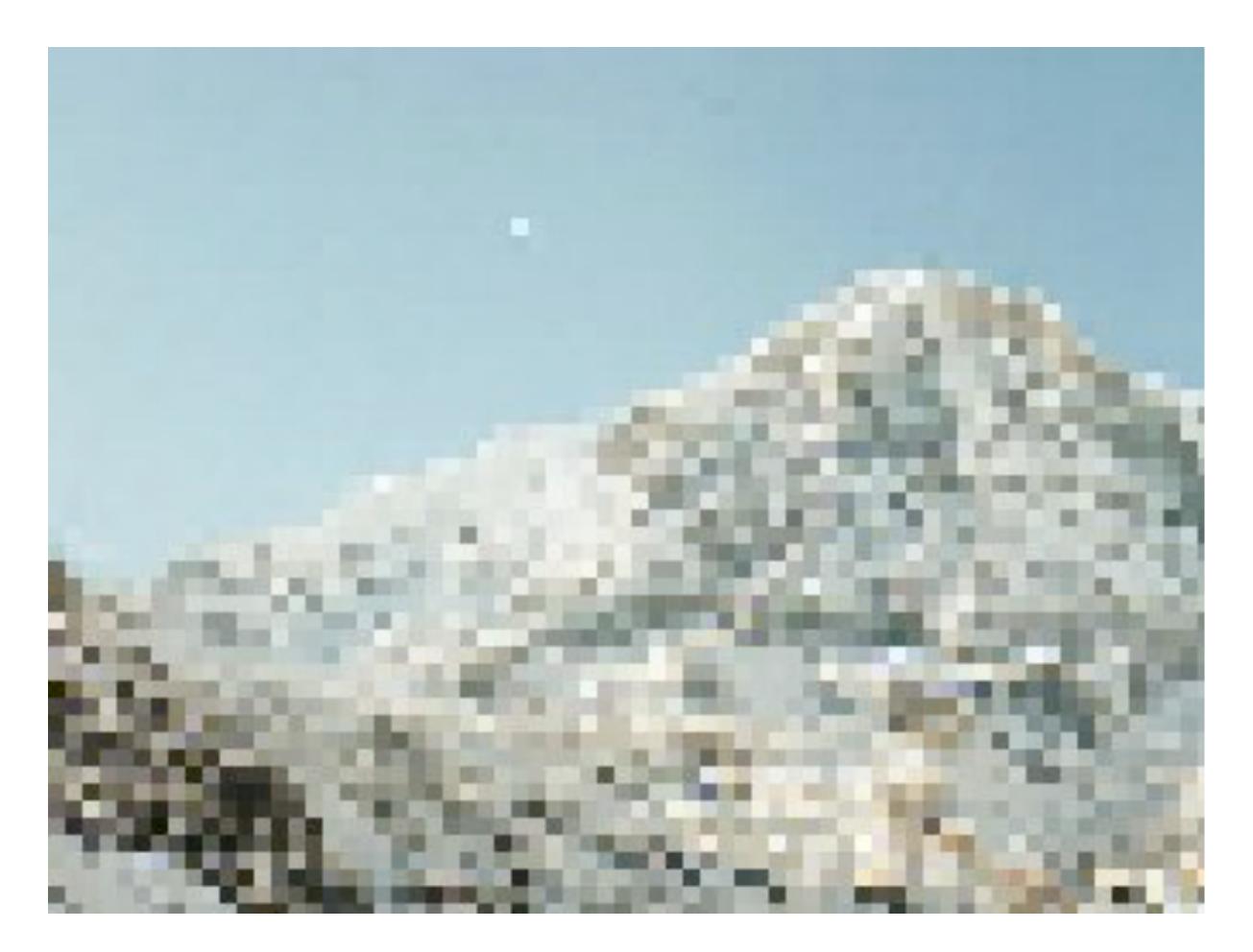
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Naive Method: Form new image by taking every n-th pixel of the original image

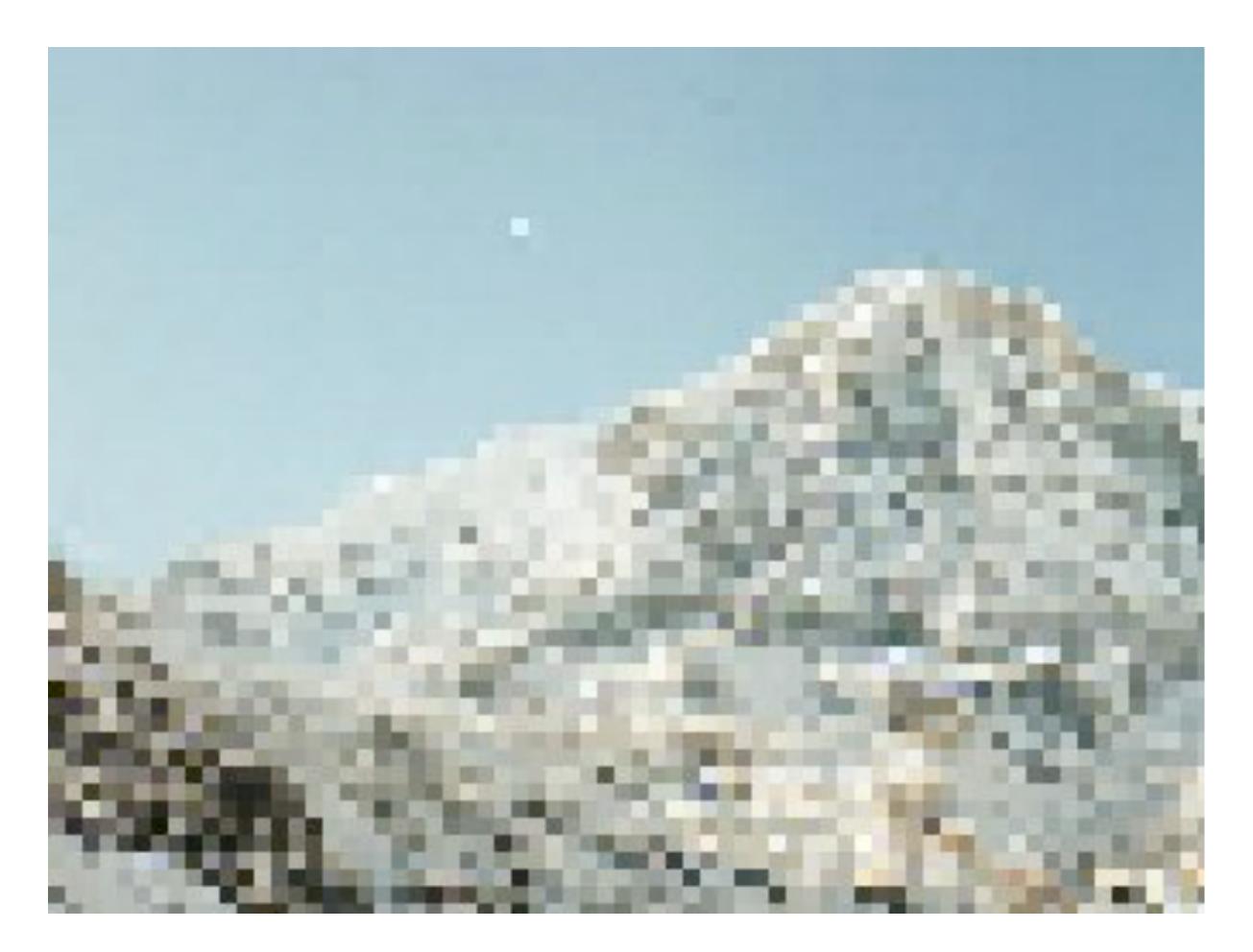




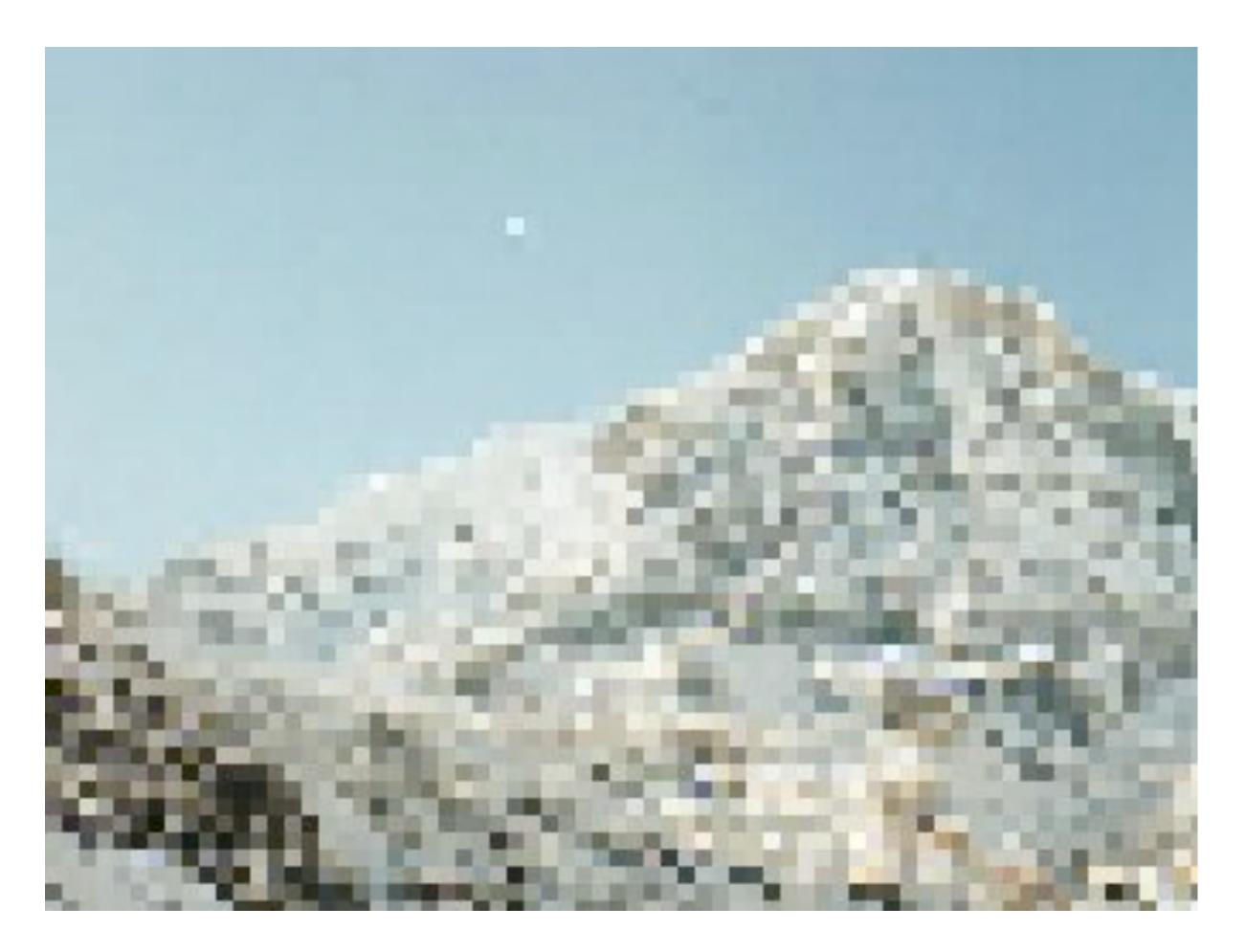
Sampling every 5-th pixel, while shifting rightwards one pixel at a time



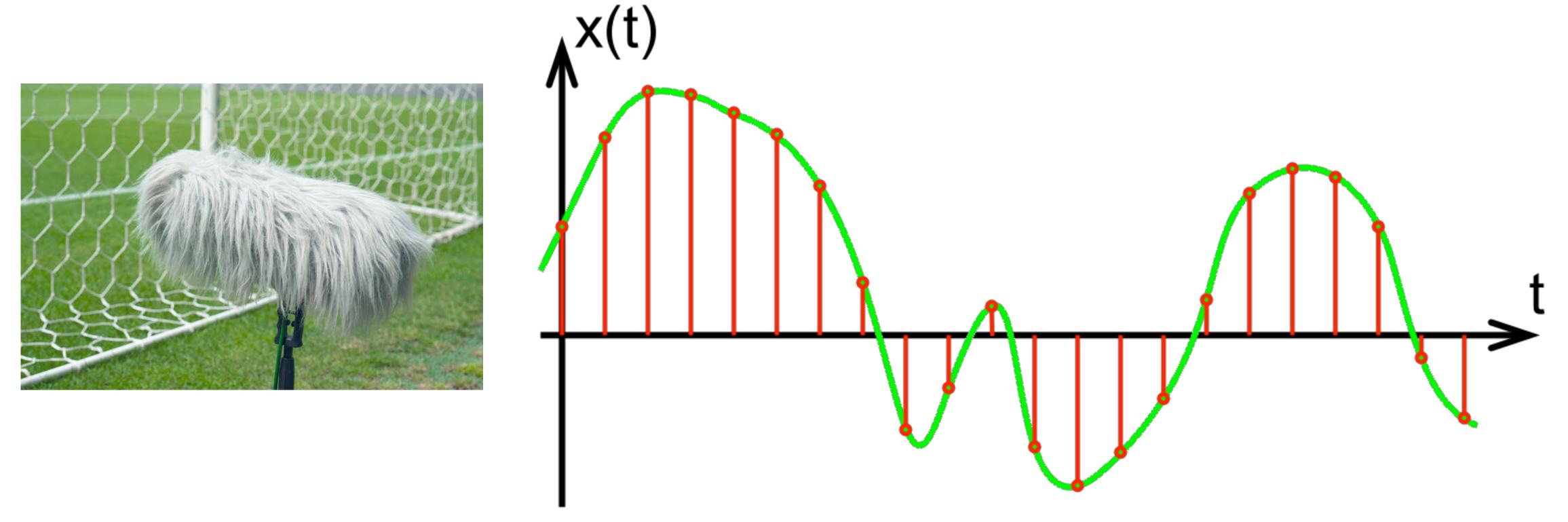
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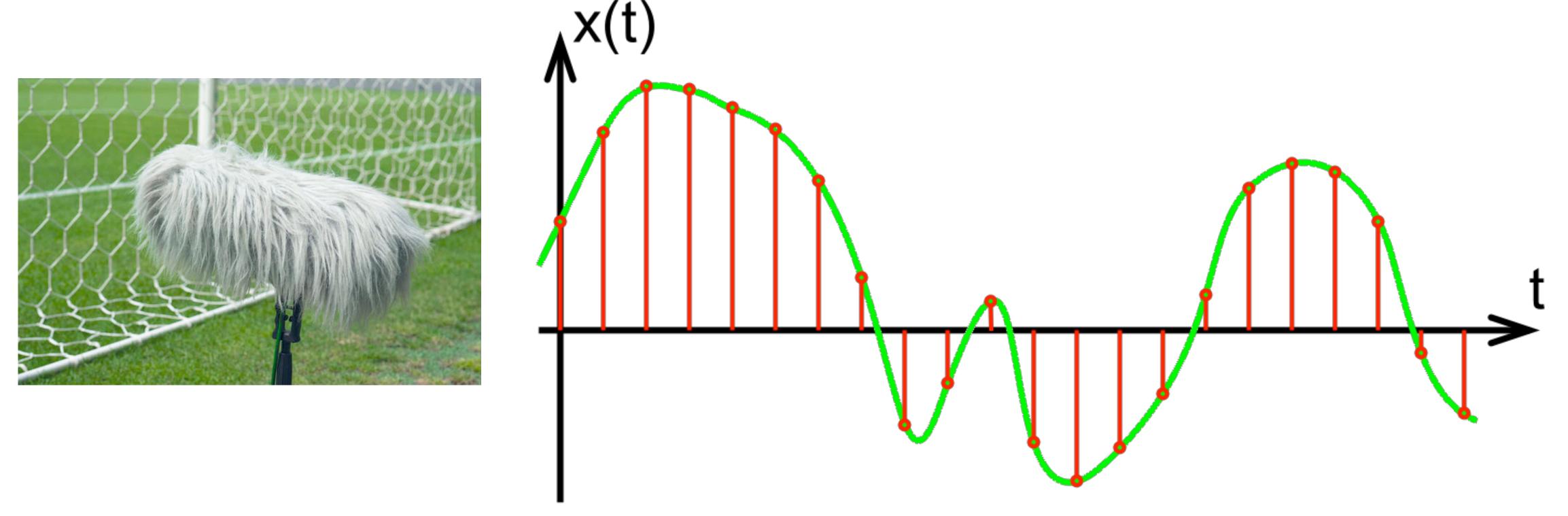
Sampling every 5-th pixel, while shifting rightwards one pixel at a time



What's wrong with this method?

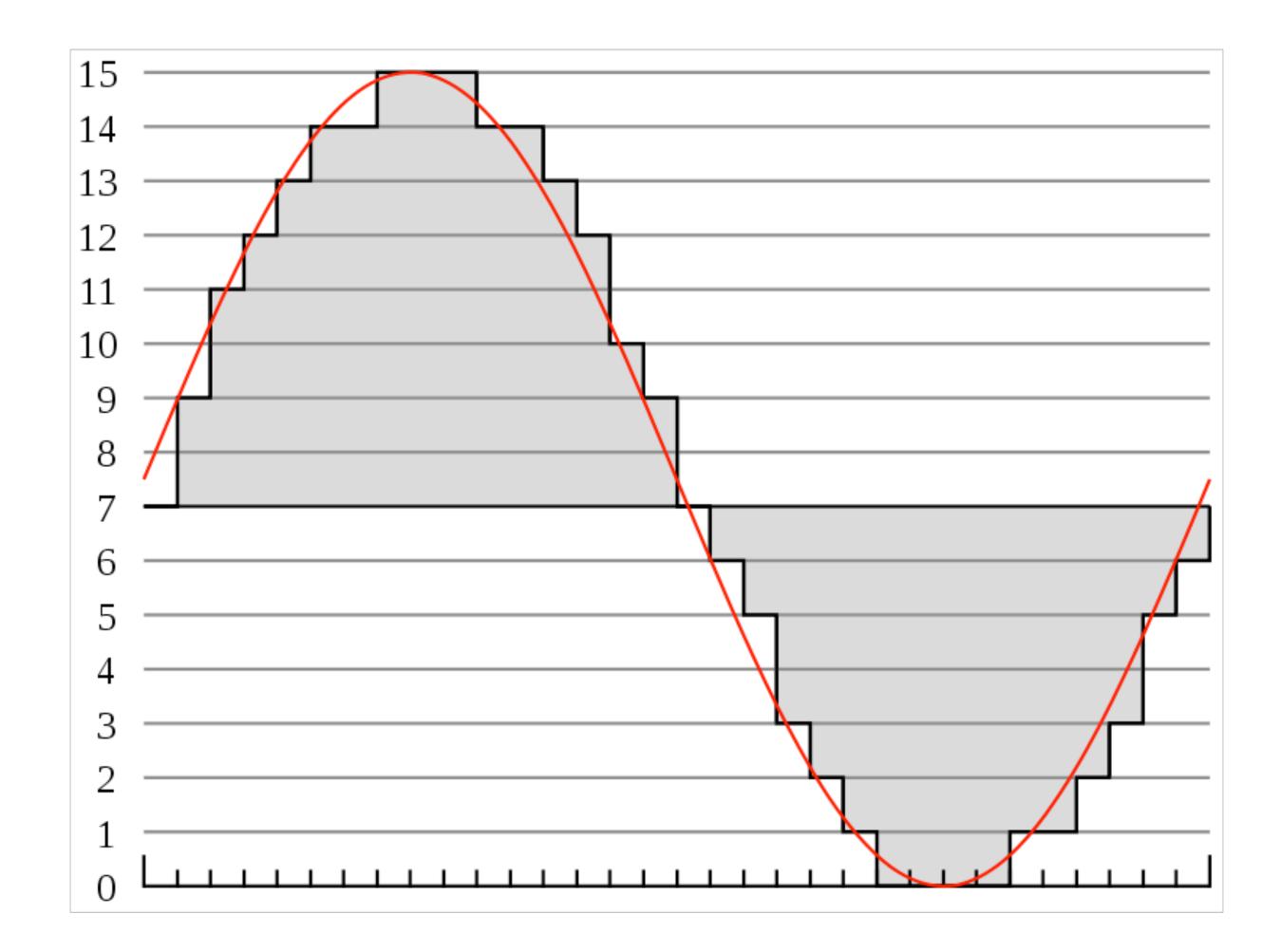


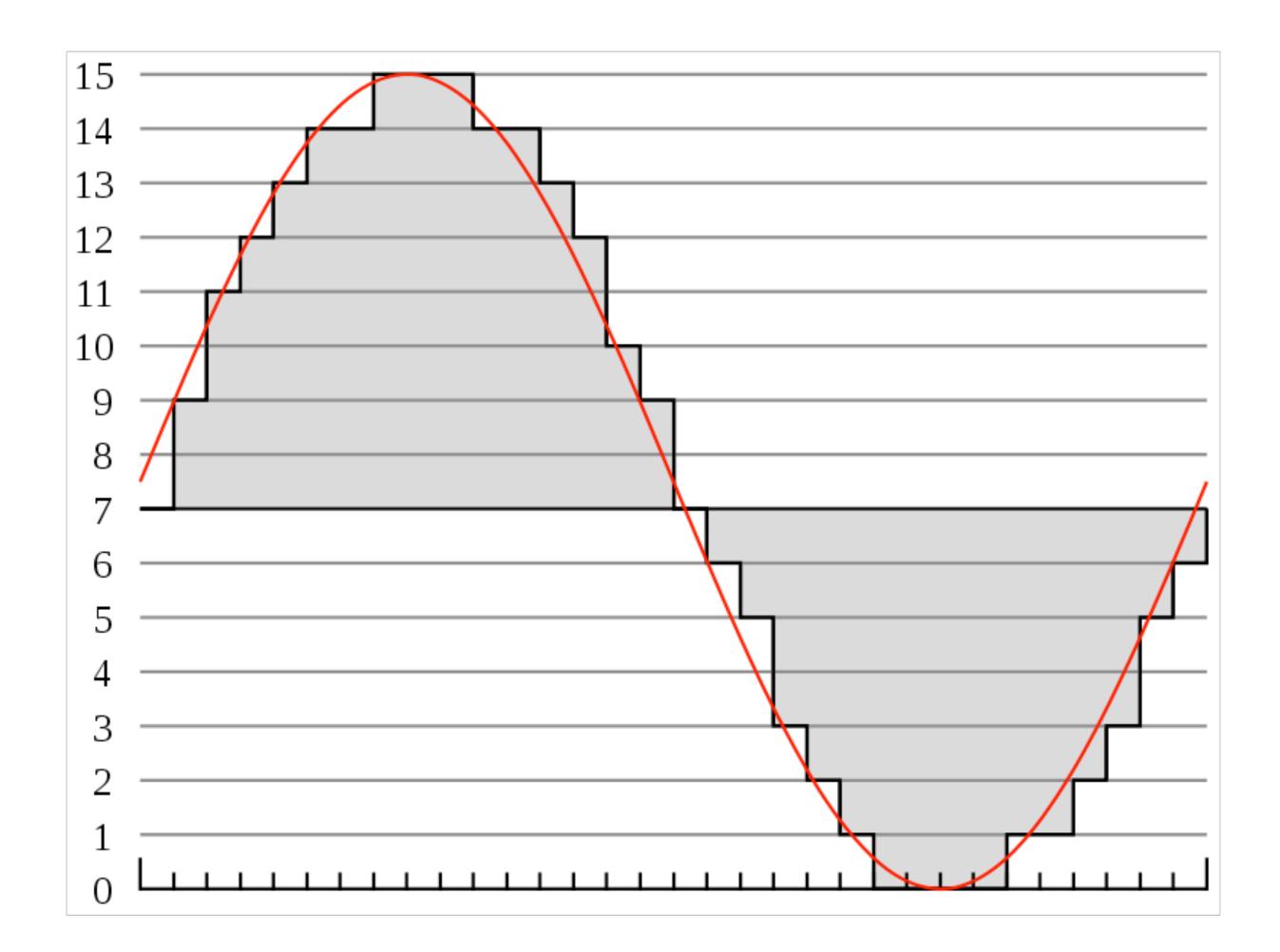
Question: What choice/parameters do we have when sampling audio signal?



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Sampling rate and bit depth, e.g., 44.1 kHz (samples/second), 16 bits/sample





Quantization noise / error is the difference between black and red curves

- Aliasing causes undesirable artifacts in audio reproduction

import scipy.io.wavfile **as** wavfile

rate, signal = wavfile.read("stevie.wav")

data=signal[0:(rate*10),:] # 10 seconds of audio

data_2=data[0:-1:2,:] # select every 2nd sample data_4=data[0:-1:4,:] # select every 4th sample data_8=data[0:-1:8,:] # select every 8th sample

wavfile.write('test2.wav', int(rate/2), data_2) wavfile.write('test4.wav', int(rate/4), data_4) wavfile.write('test8.wav', int(rate/8), data_8)

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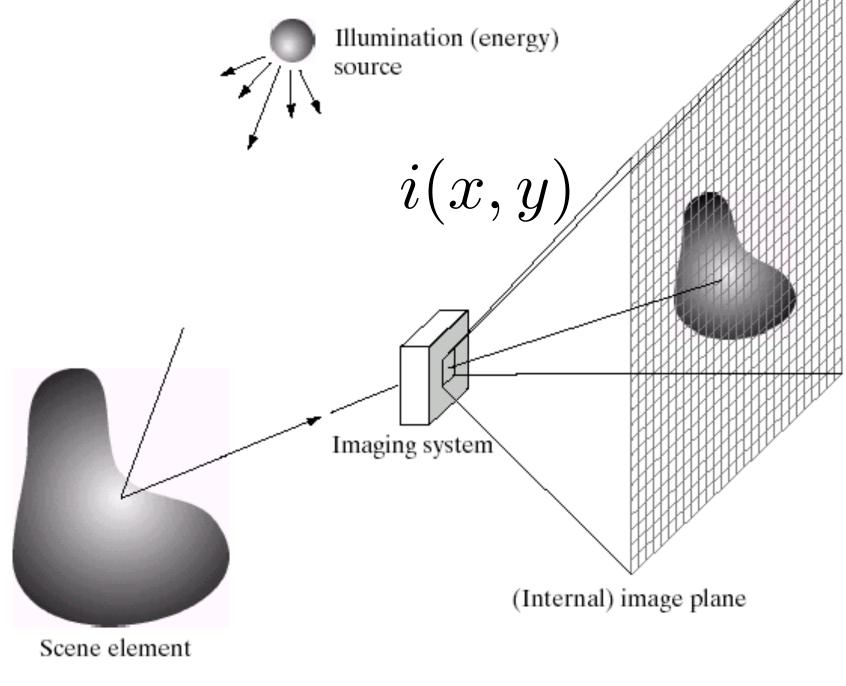
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Example: Image Sampling

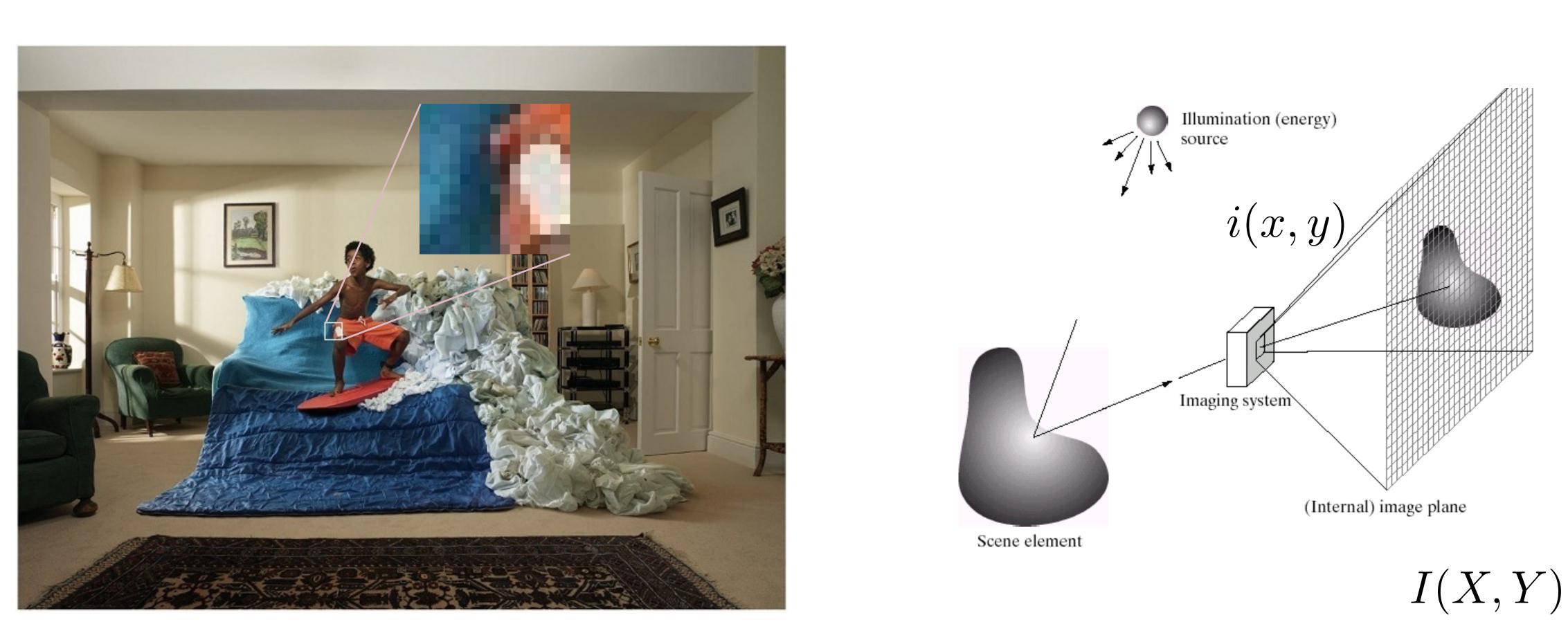




I(X, Y)

)

Example: Image Sampling



Sampling rate and bit depth (e.g., 8-bits)

)

Continuous Case: Observations

-i(x,y) is a real-valued function of real spatial variables, x and y

-i(x,y) is **bounded above and below**. That is $0 \le i(x,y) \le M$

for some maximum brightness ${\cal M}$

Continuous Case: Observations

-i(x,y) is a real-valued function of real spatial variables, x and y

-i(x,y) is bounded above and below. That is

for some maximum brightness M

-i(x,y) is **bounded in extent**. That is, i(x,y) is non-zero (i.e., strictly positive) over, at most, a bounded region

 $0 \leq i(x, y) \leq M$

Pixel Bit Rate

Recall: $0 \le i(x, y) \le M$

We divide the range [0, M] into a finite called **quantization**.

The values are called **grey-levels**.

Suppose *n* bits-per-pixel are available evenly spaced intervals.

Typically n = 8 resulting in grey-levels in the range [0, 255]

We divide the range [0, M] into a finite number of equivalence classes. This is

Suppose n bits-per-pixel are available. One can divide the range [0, M] into

Sampling Theory (informal)

Question: When is I(X, Y) an exact characterization of i(x, y)?

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Question (modified): When can we reconstruct i(x, y) exactly from I(X, Y)?

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Intuition: Reconstruction involves some kind of interpolation

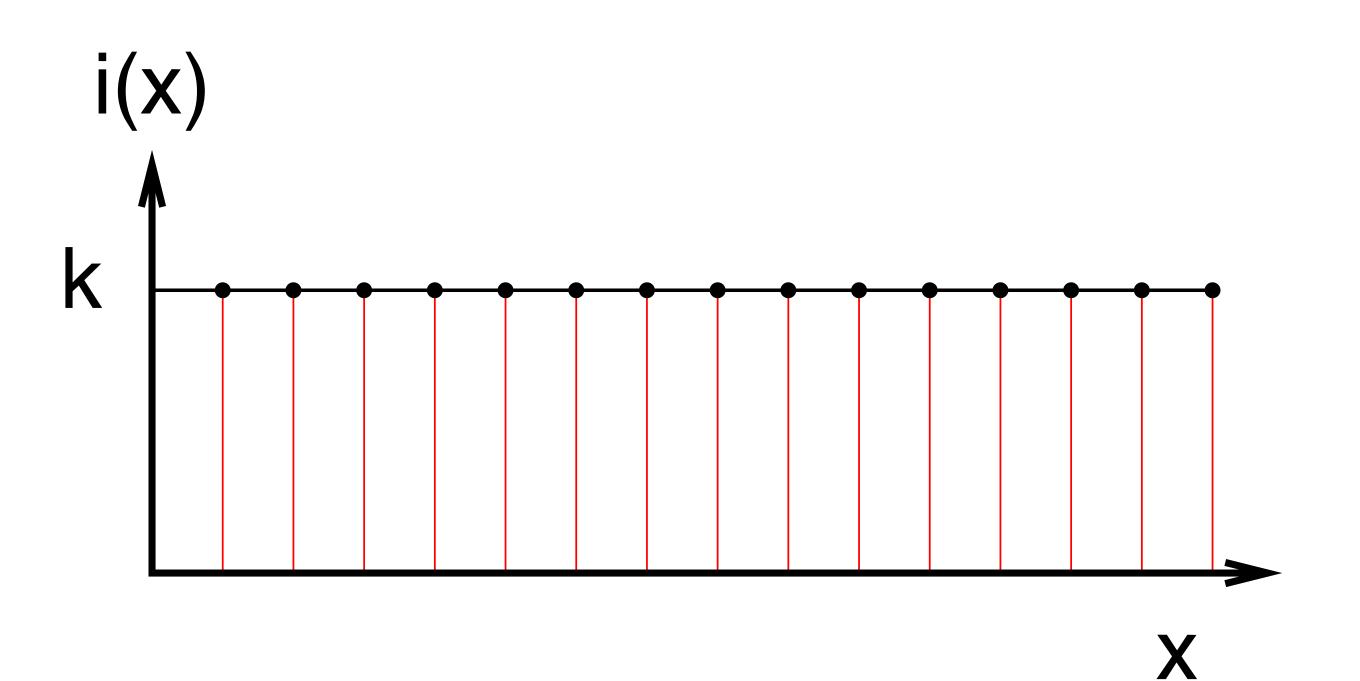
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Intuition: Reconstruction involves some kind of **interpolation**

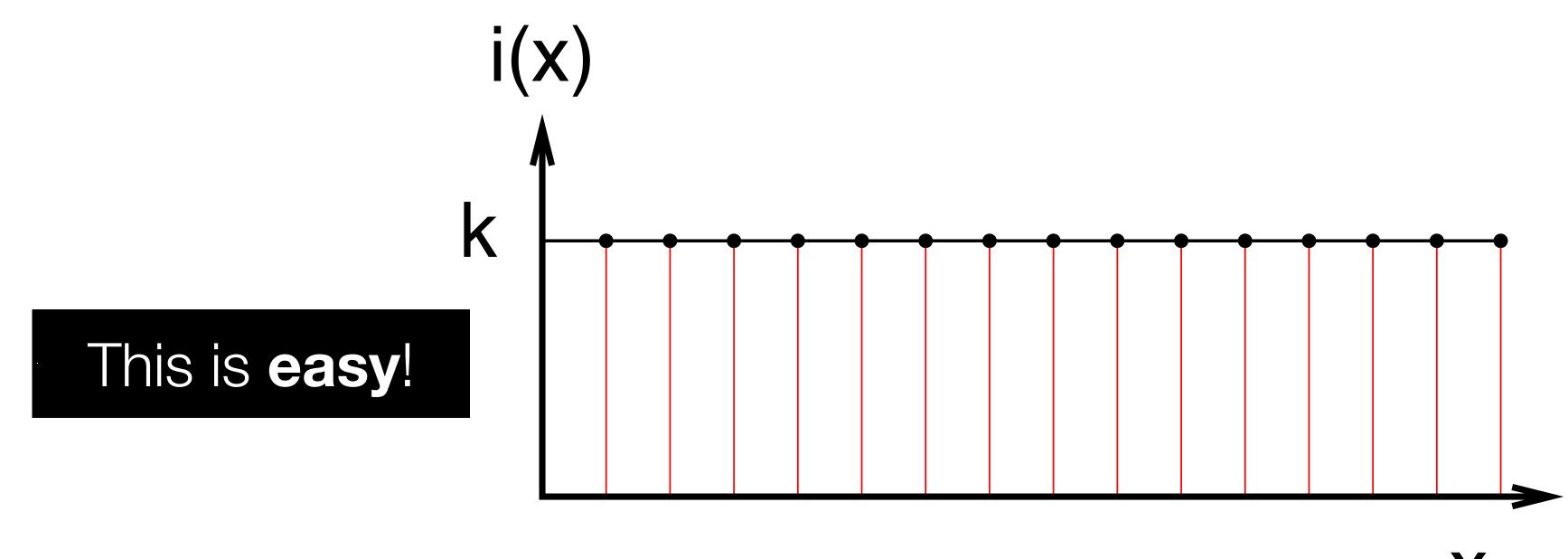
Heuristic: When in doubt, consider simple cases

Case 0: Suppose i(x, y) = k (with k being one of our gray levels)



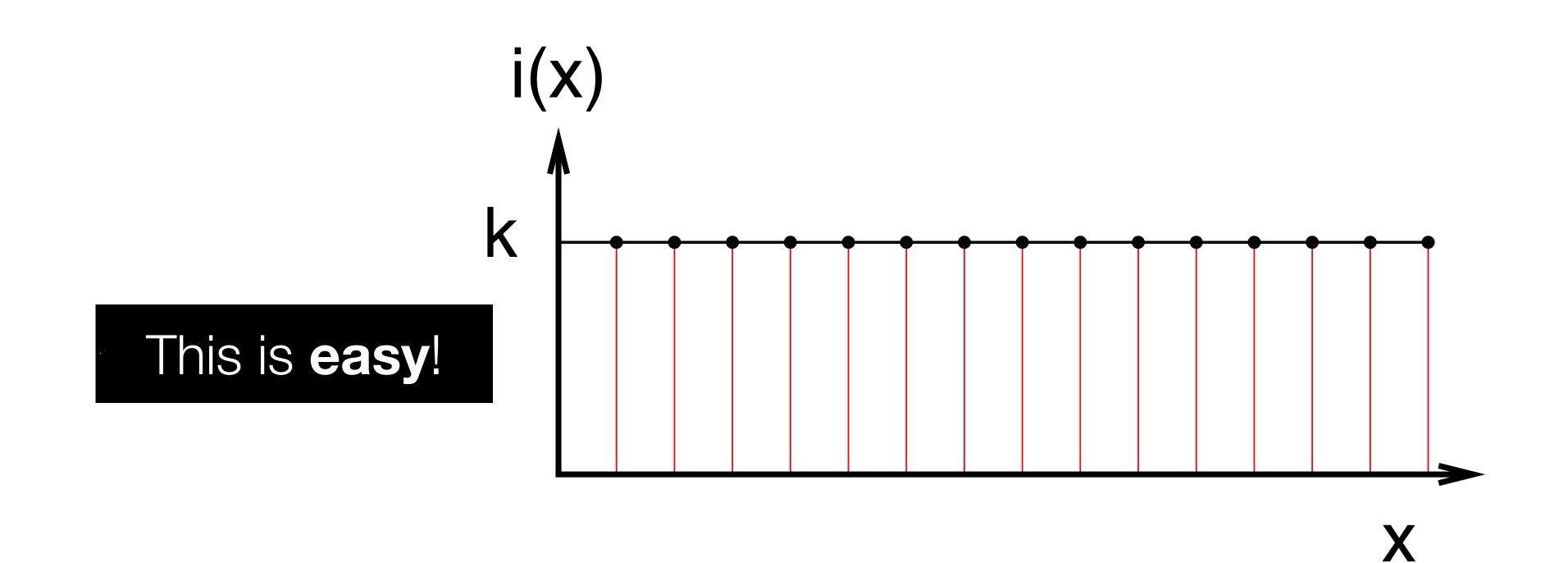
Note: we use equidistant sampling at integer values for convenience, in general, sampling doesn't need to be equidistant

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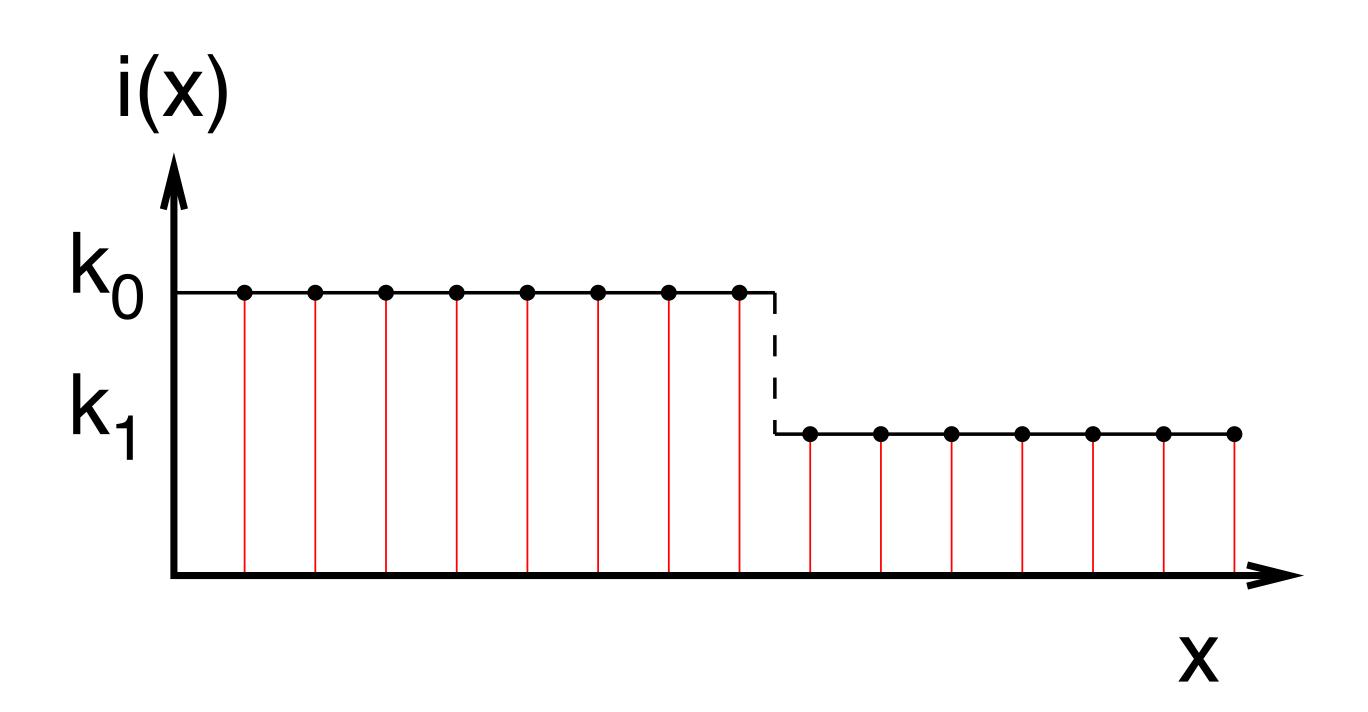


X

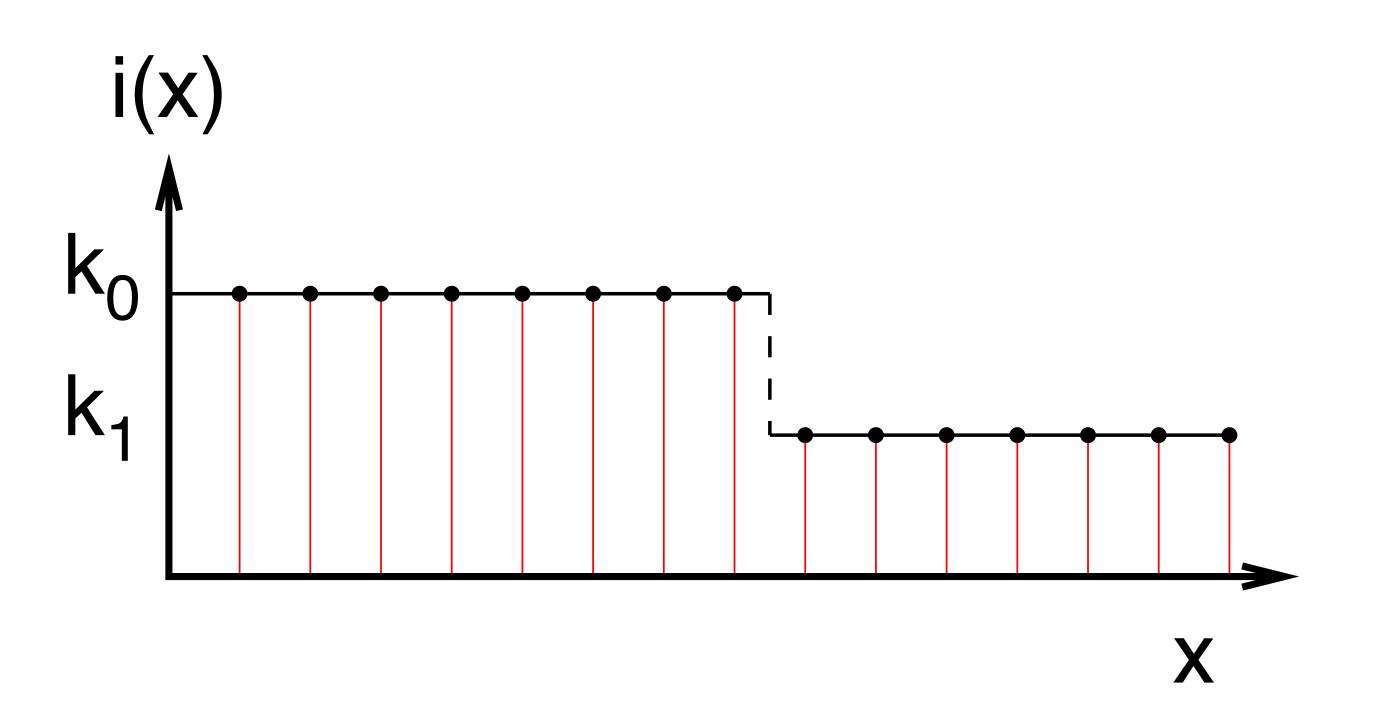
Case 0: Suppose i(x, y) = k (with k being one of our gray levels)



I(X,Y) = k. Any standard interpolation function would give i(x,y) = k for noninteger x and y (irrespective of how coarse the sampling is)



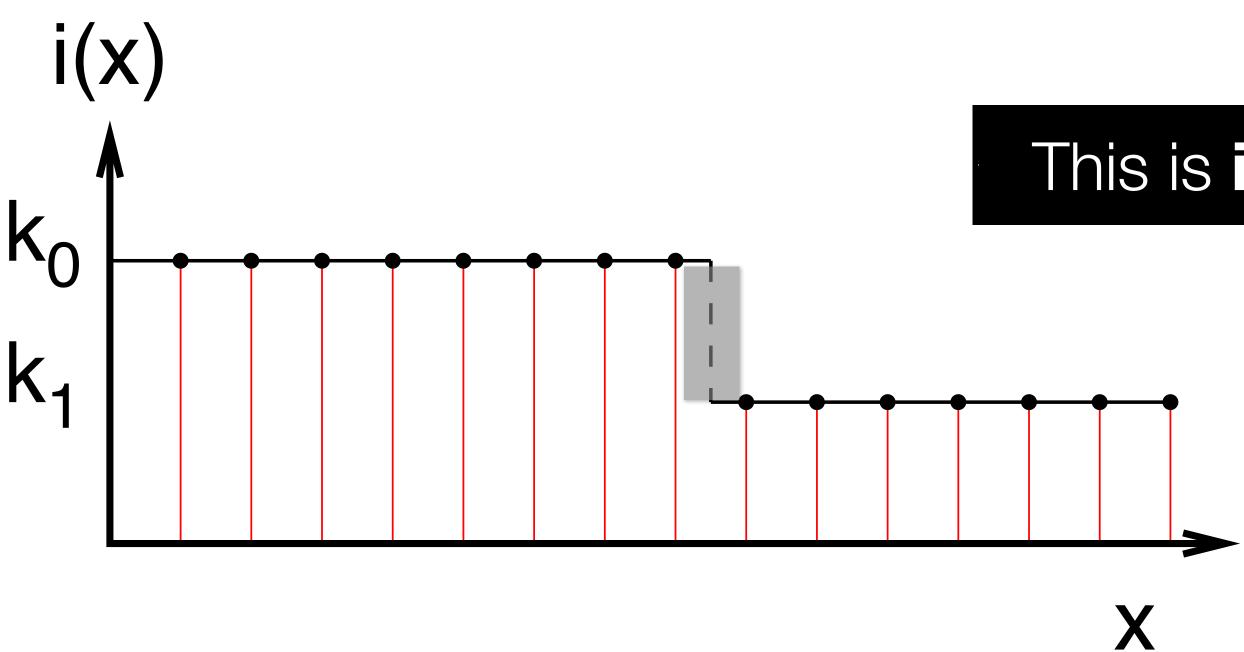
Case 1: Suppose i(x, y) has a discontinuity not falling precisely at integer x, y



We cannot reconstruct i(x, y) exactly because we can never know exactly where the discontinuity lies

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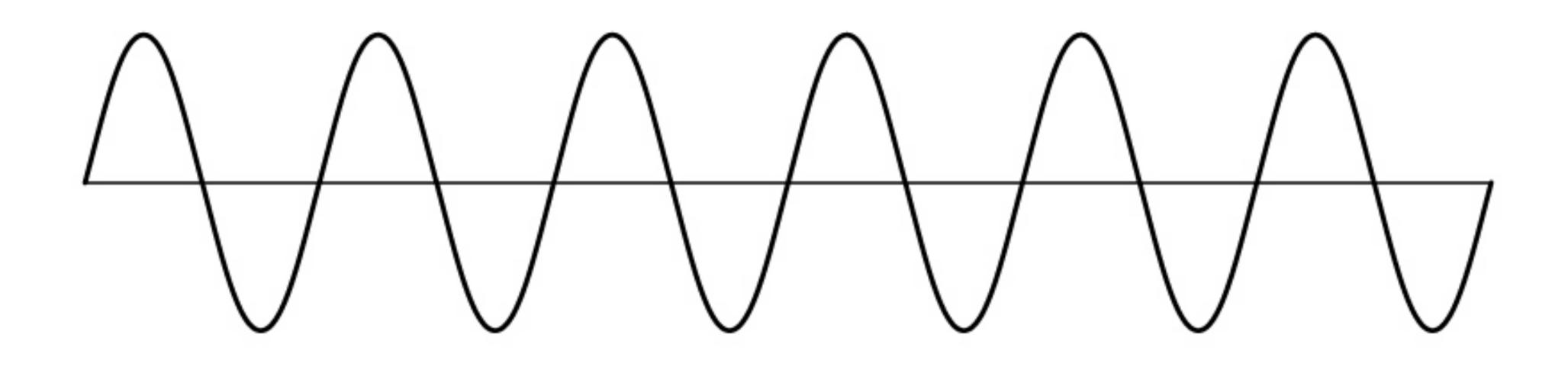
This is **impossible**!



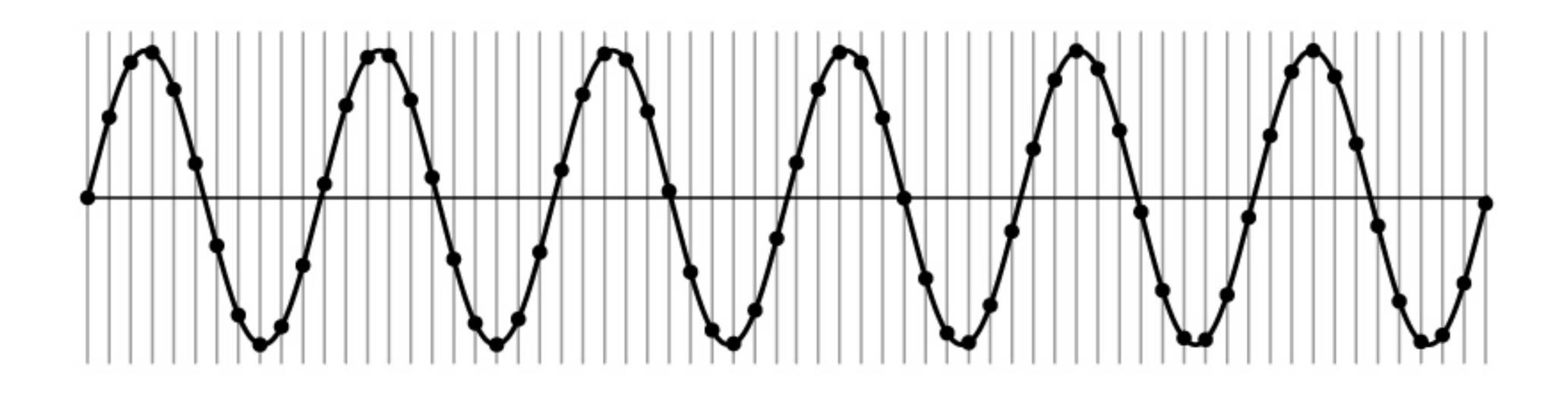
Question: How do we close the gap between "easy" and "impossible?"

Next, we build intuition based on informal argument

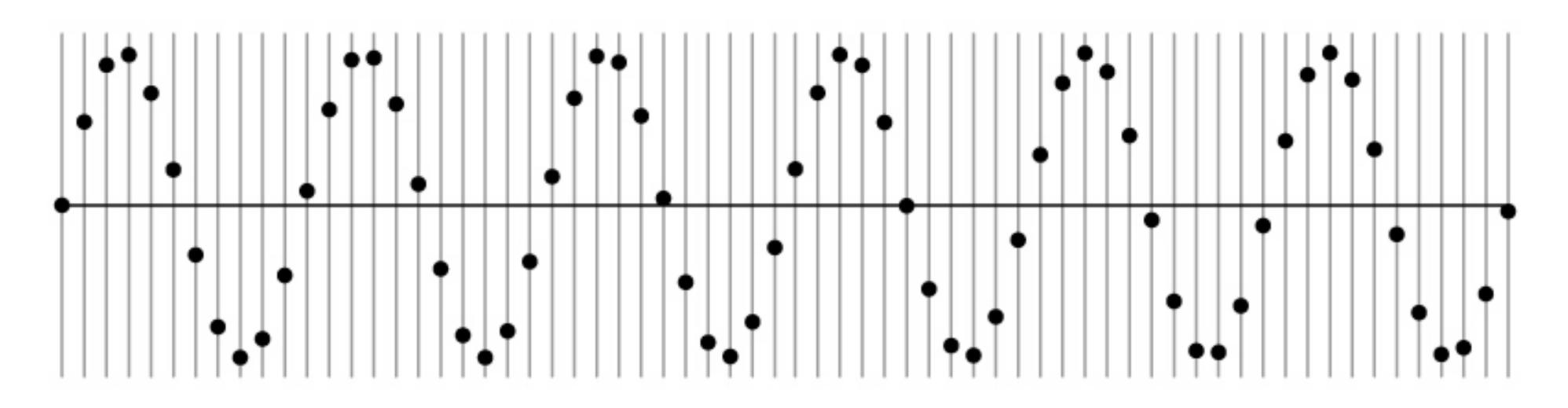
How do we discretize the signal?



How do we discretize the signal?

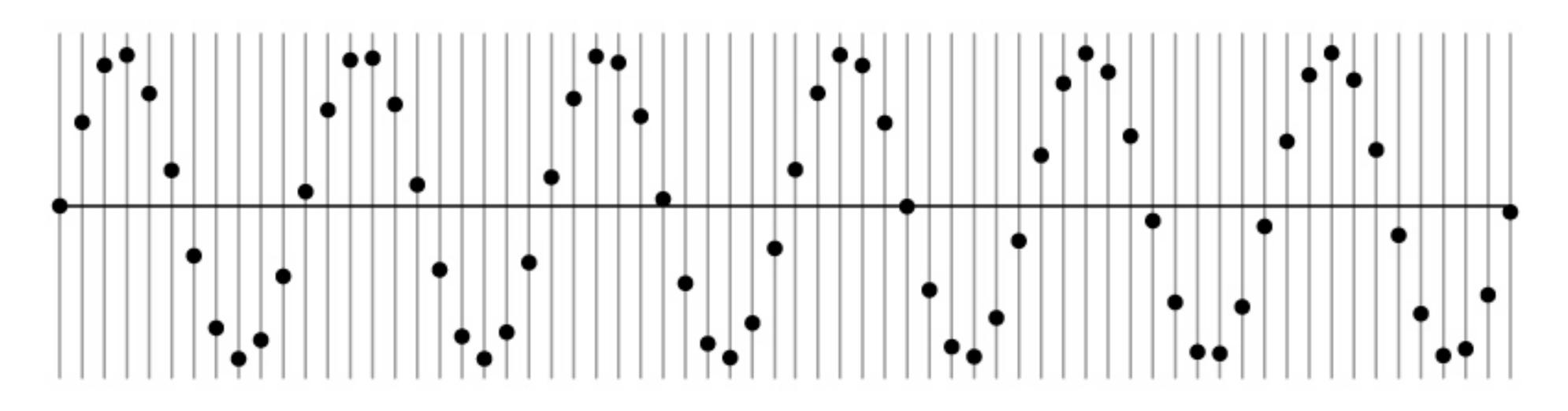


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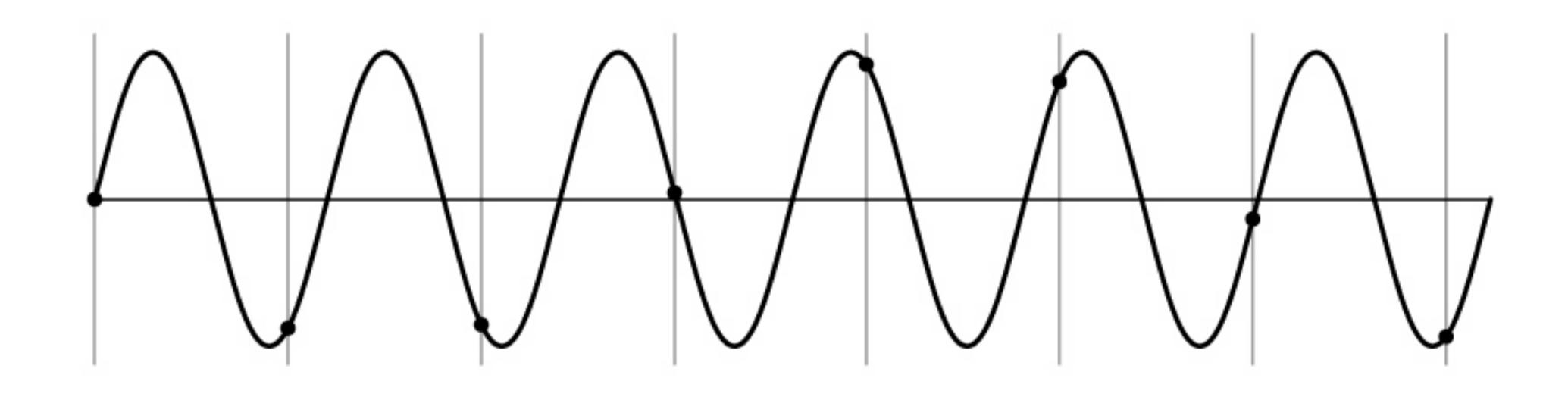
How many samples should I take? Can I take as many samples as I want?

How do we discretize the signal?



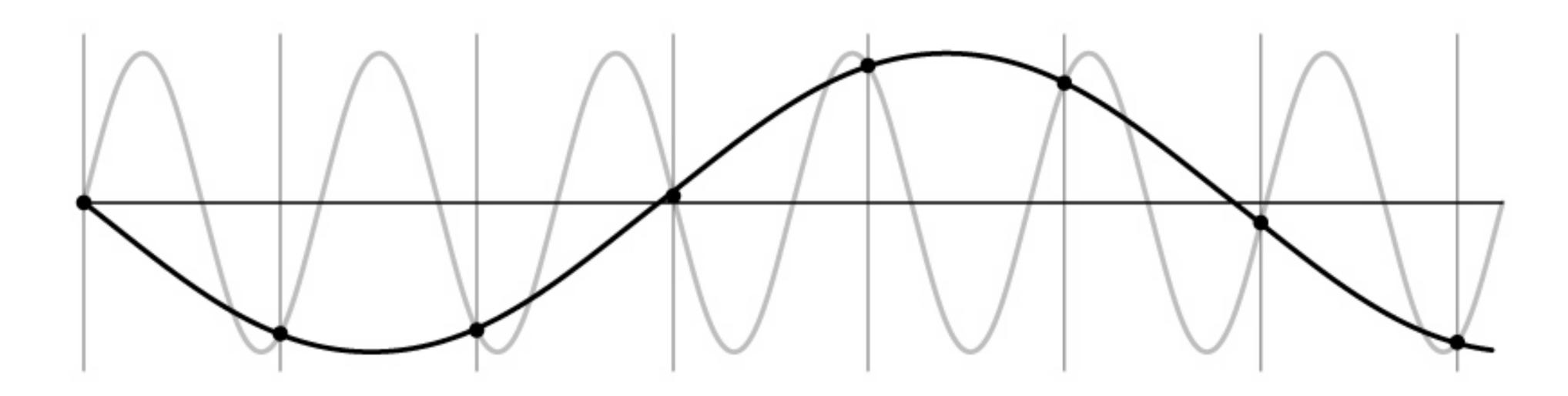
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How do we discretize the signal?



Signal can be confused with one at lower frequency

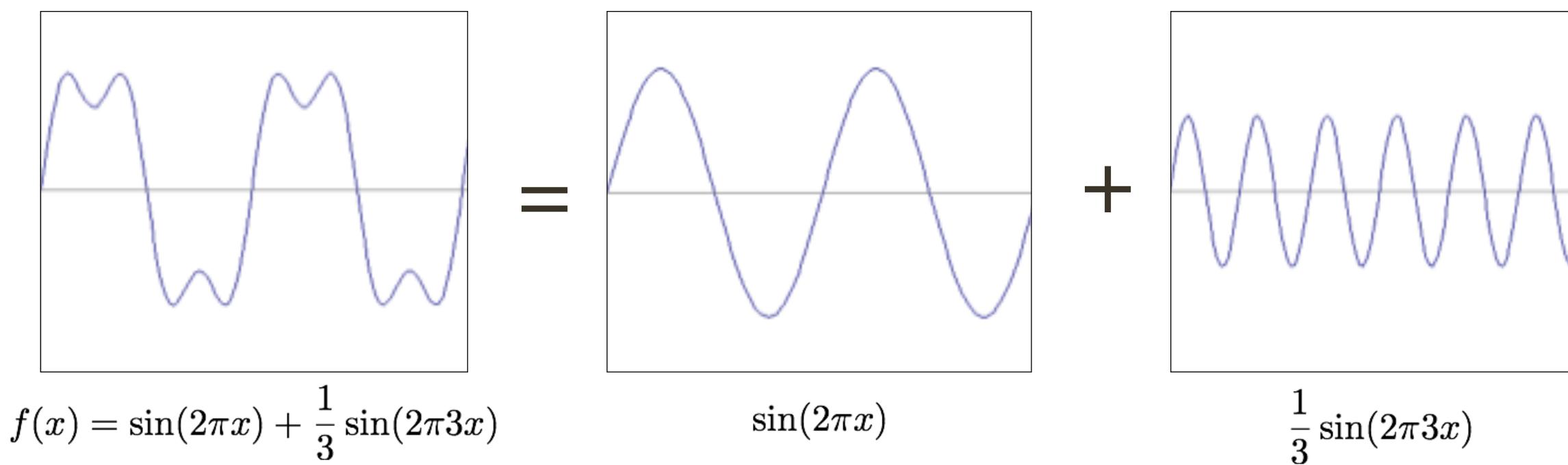
How do we discretize the signal?



Signal can be confused with one at lower frequency This is called "Aliasing"

Recall: Fourier Representation

Any signal can be written as a sum of sinusoidal functions







Nyquist Sampling Theorem

To avoid aliasing a signal must be sampled at twice the maximum frequency:

where f_s is the sampling frequency, and f_{max} is the maximum frequency present in the signal

Futhermore, Nyquist's theorem states that a signal is **exactly recoverable** from its **samples** if sampled at the **Nyquist rate** (or higher)

Note: that a signal must be **bandlimited** for this to apply (i.e., it has a maximum frequency)

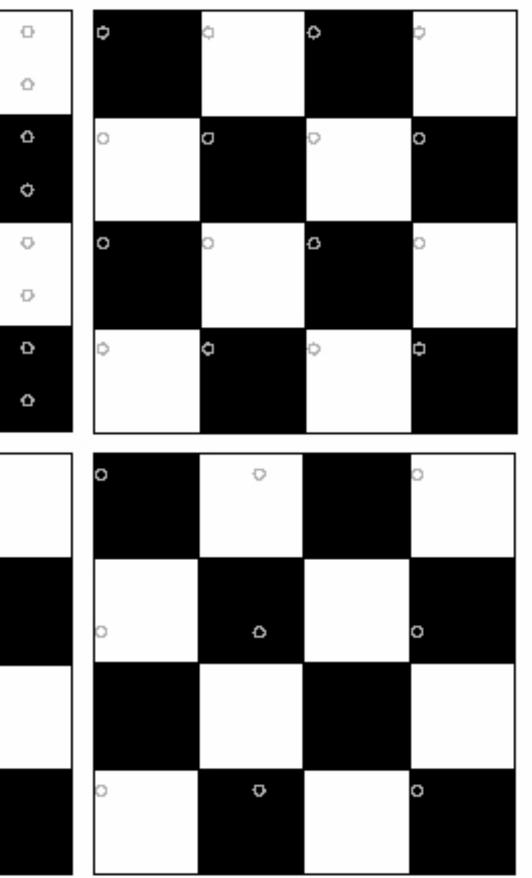
 $f_s > 2 \times f_{max}$

- between samples
- "rate of change" means derivative
- the formal concept is **bandlimited signal**
- "bandlimit" and "constraint on derivative" are linked
- Think of music
- bandlimited if it has some maximum temporal frequency
- the upper limit of human hearing is about 20 kHz
- Think of imaging systems. Resolving power is measured in
- "line pairs per mm" (for a bar test pattern)
- "cycles per mm" (for a sine wave test pattern)
- An image is bandlimited if it has some maximum spatial frequency

Exact reconstruction requires constraint on the rate at which i(x,y) can change

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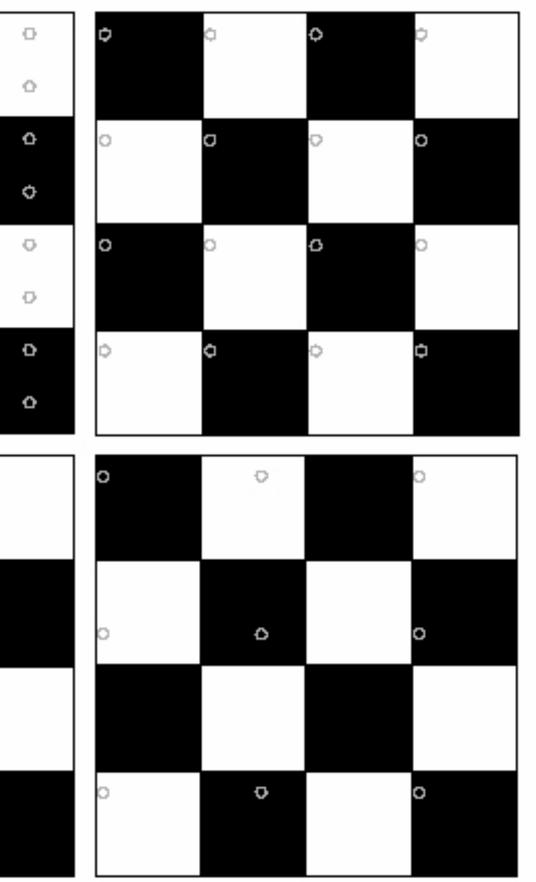
Forsyth & Ponce (2nd ed.) Figure 4.7





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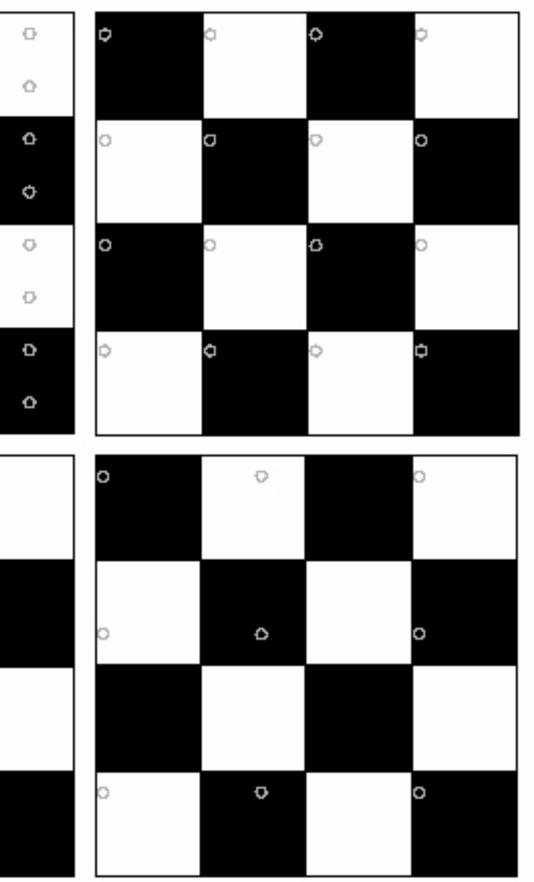
Forsyth & Ponce (2nd ed.) Figure 4.7





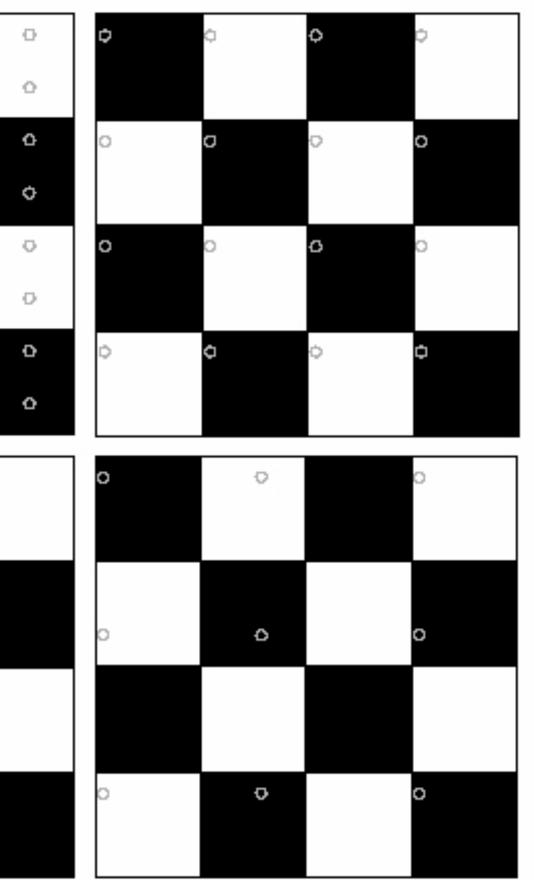
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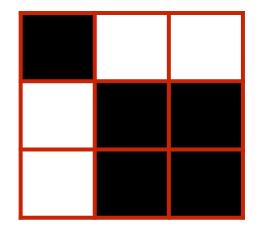
Forsyth & Ponce (2nd ed.) Figure 4.7



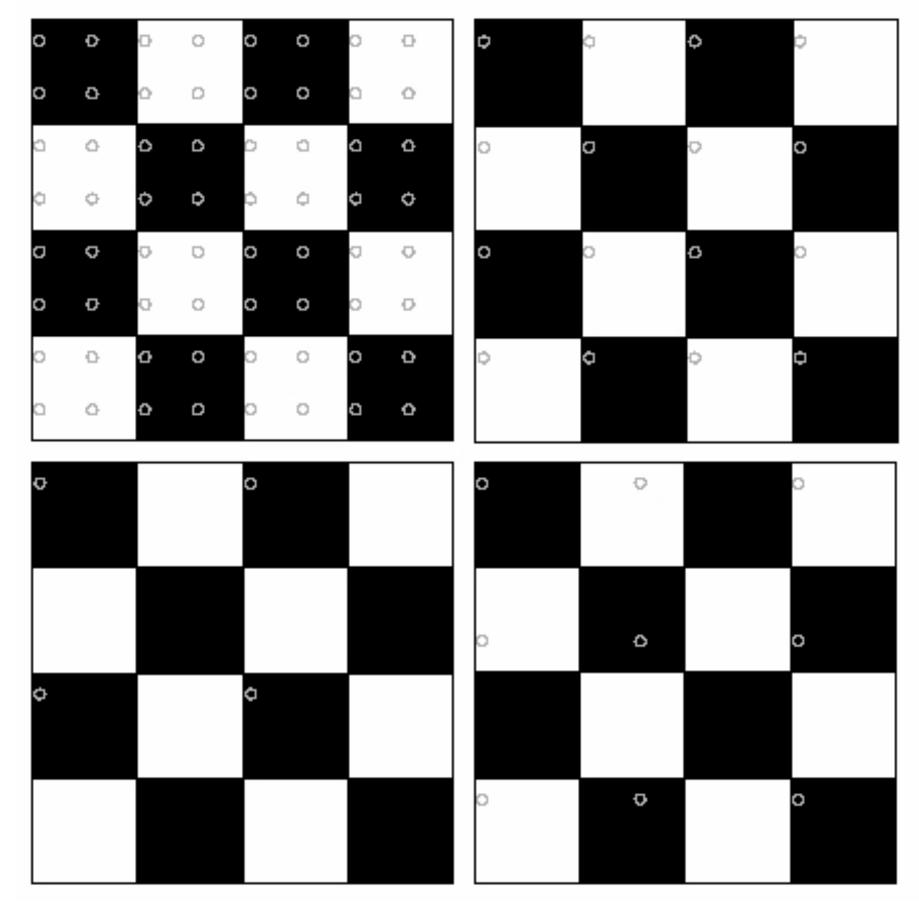


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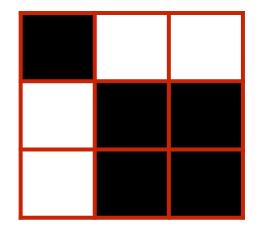






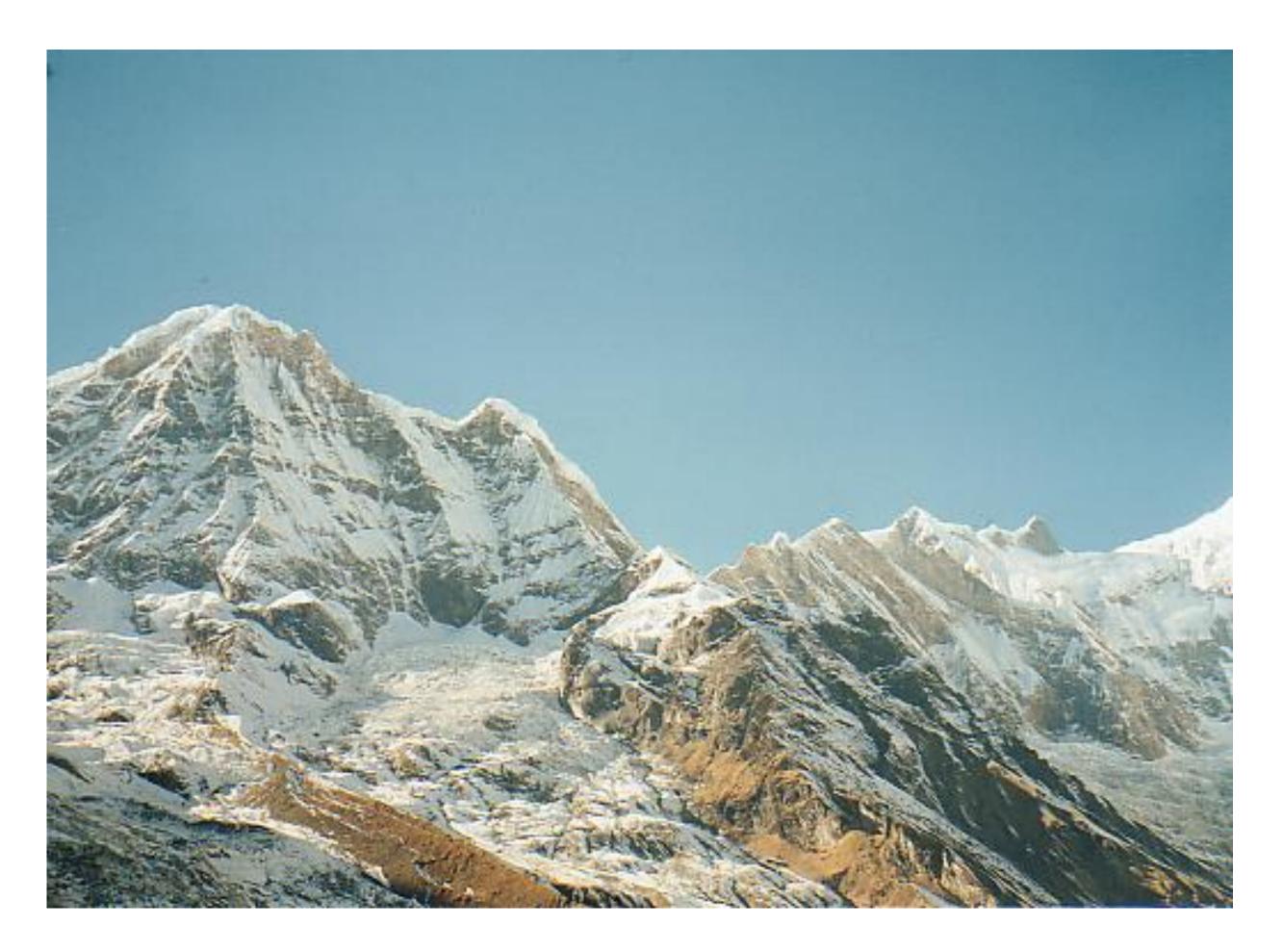


Forsyth & Ponce (2nd ed.) Figure 4.7





Goal: Resample the image to get a lower resolution counterpart

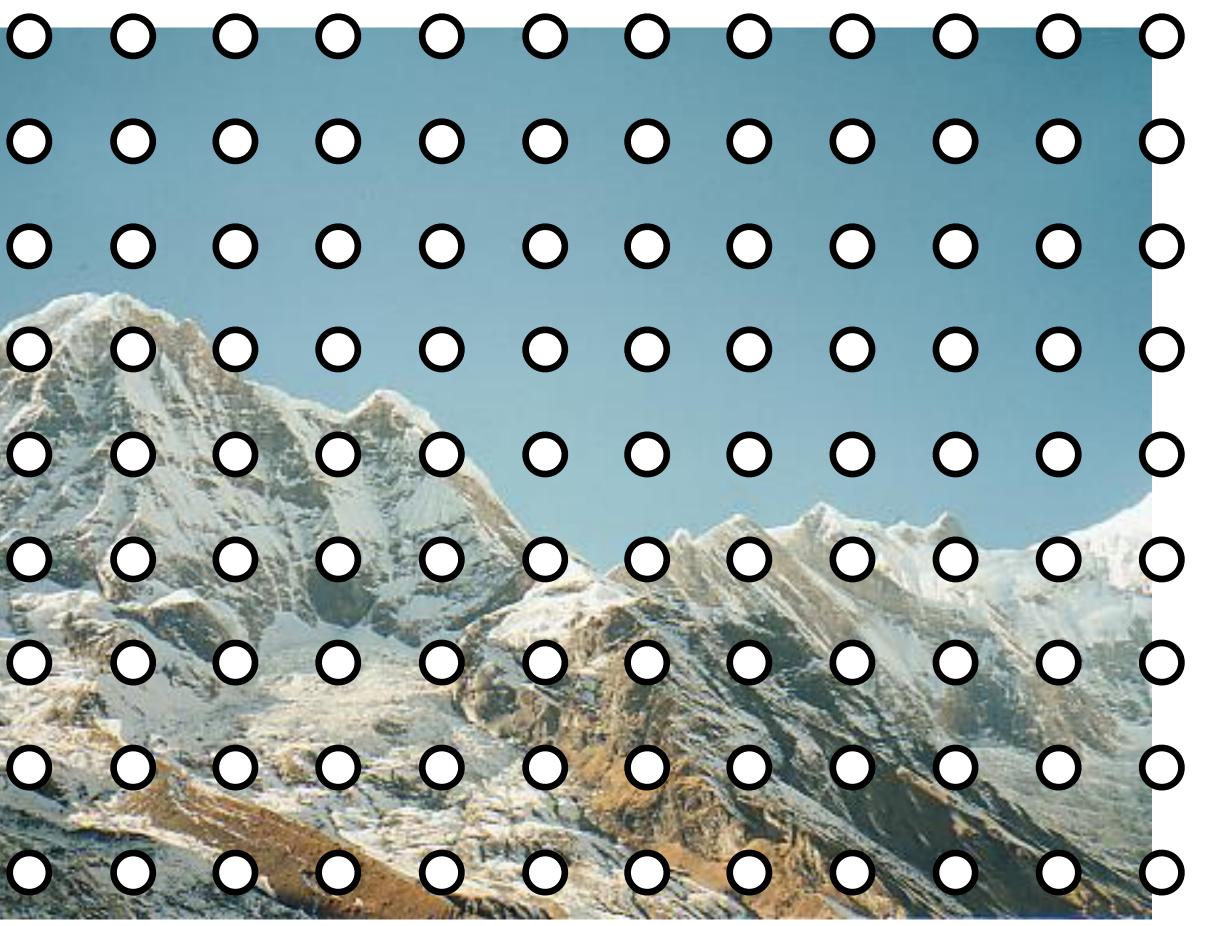


Naive Method: Form new image by taking every n-th pixel of the original image



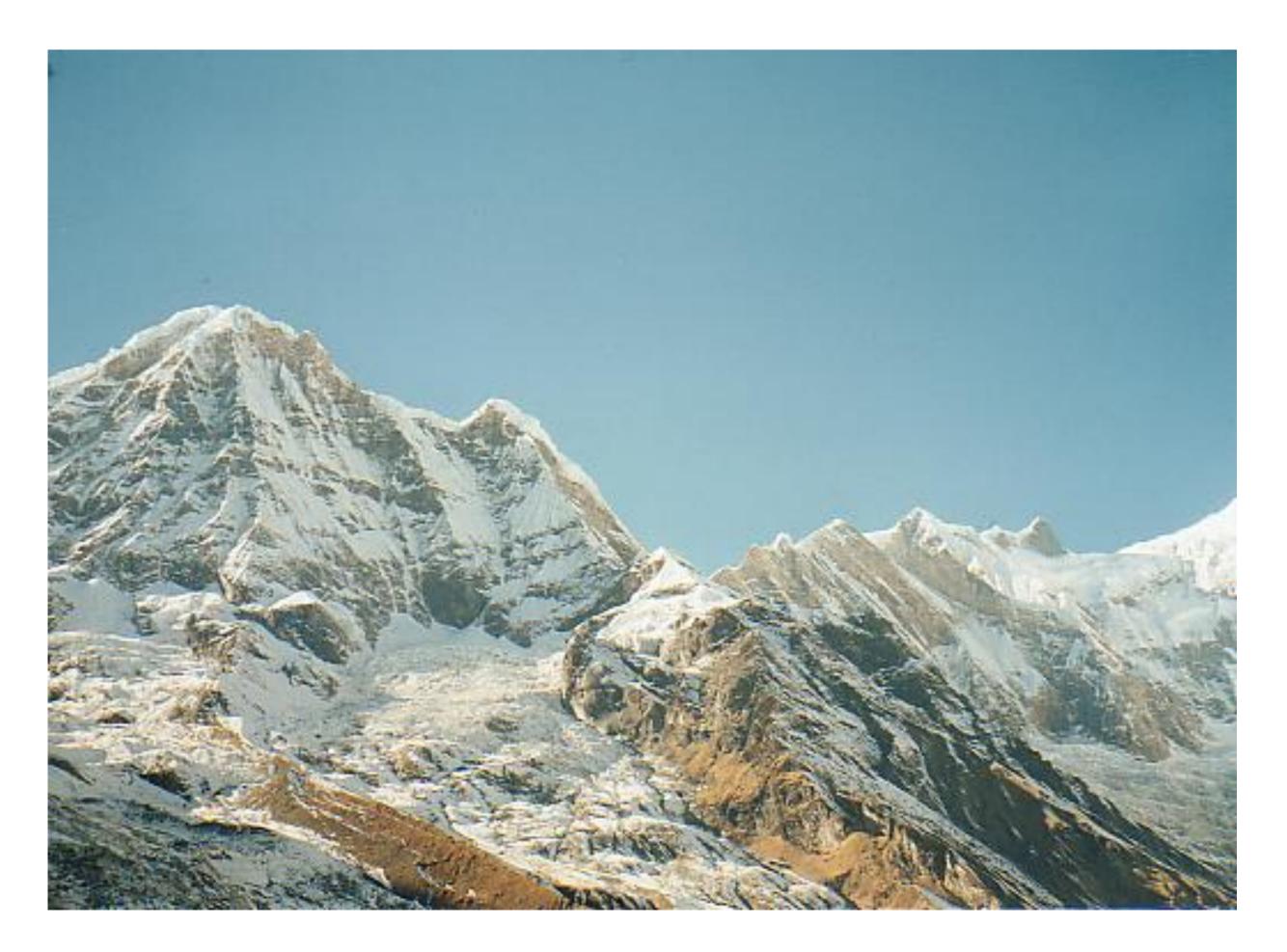
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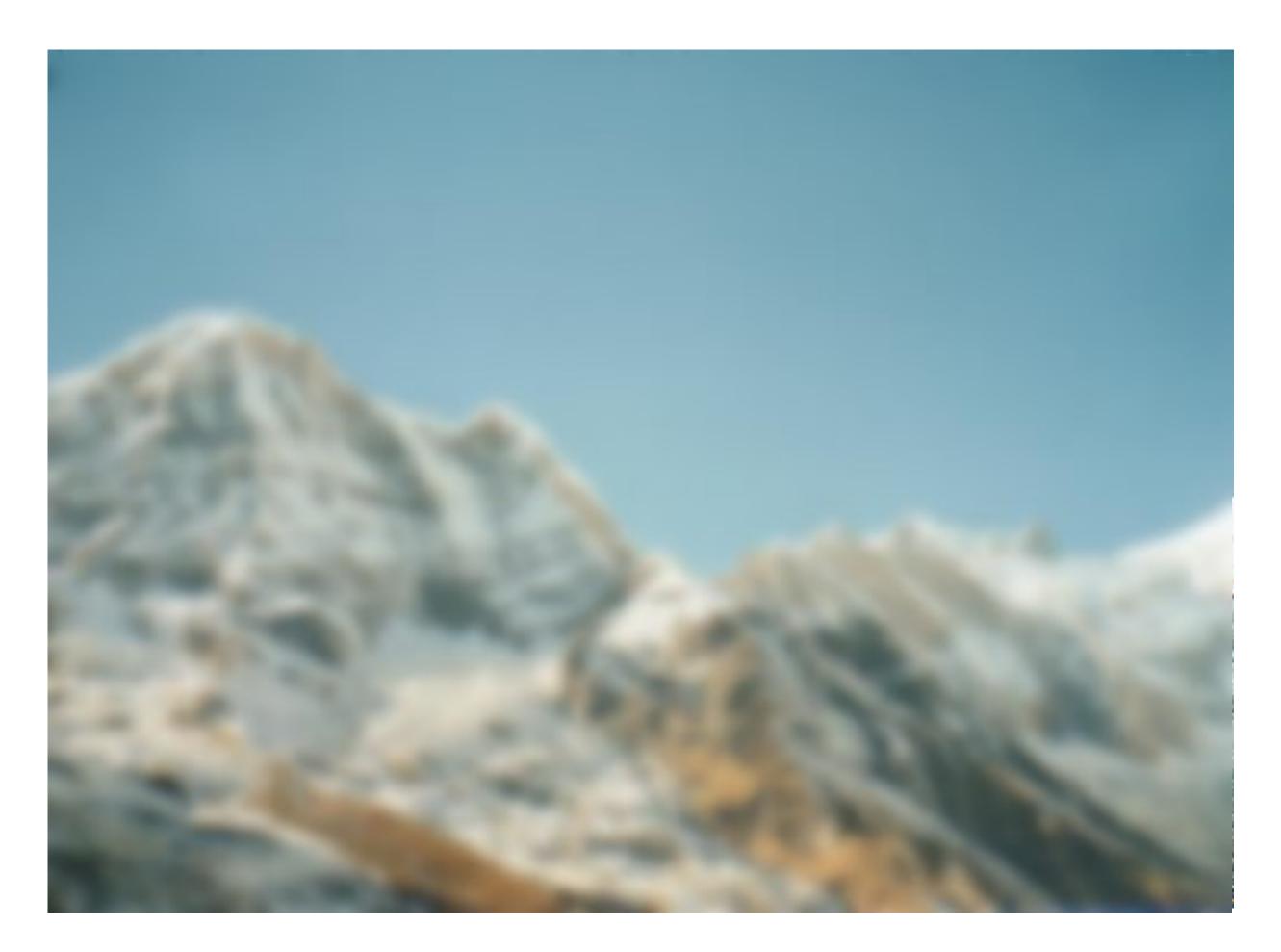


With correct sigma value for a Gaussian, no information is lost



Improved Method: First blur the image (with low-pass) then take n-th pixel

With correct sigma value for a Gaussian, no information is lost

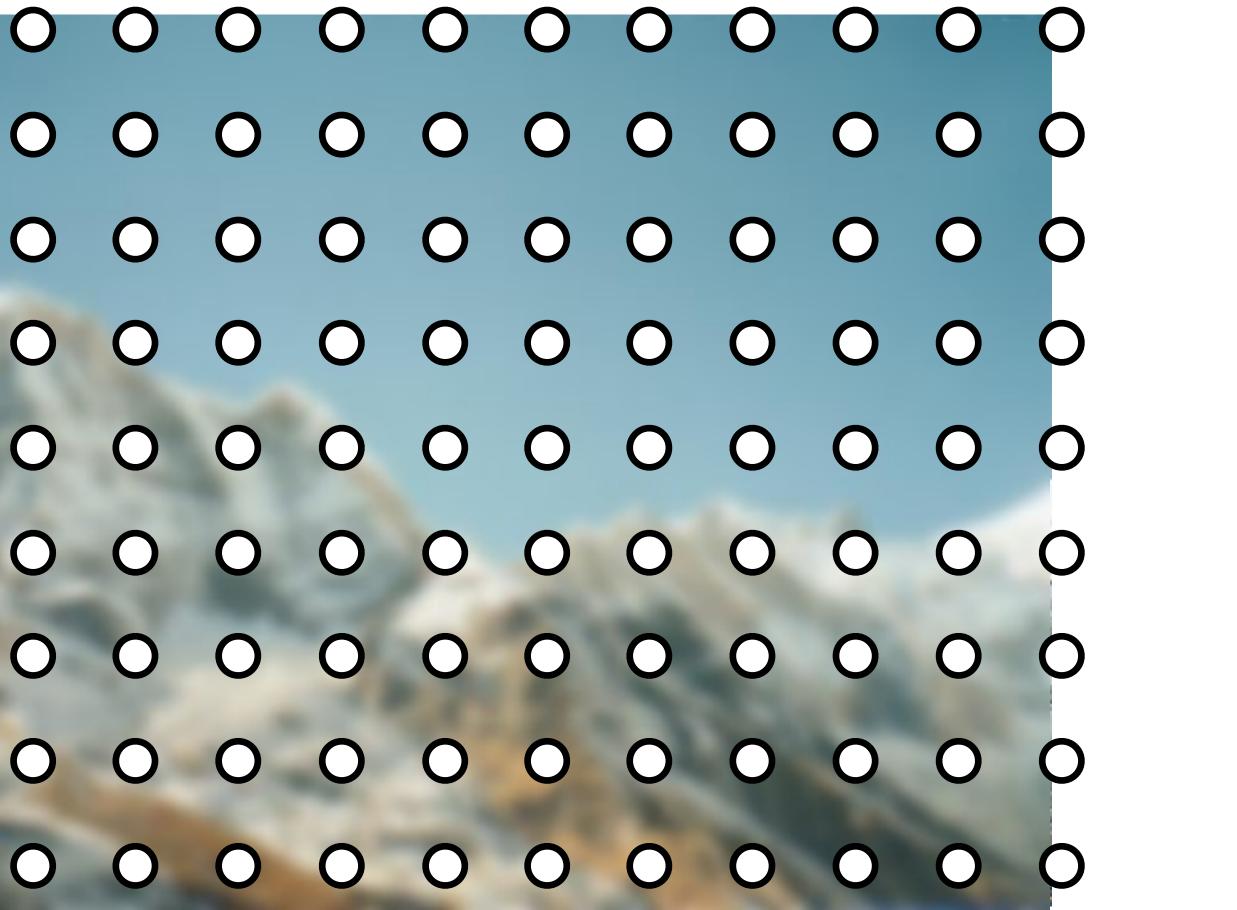


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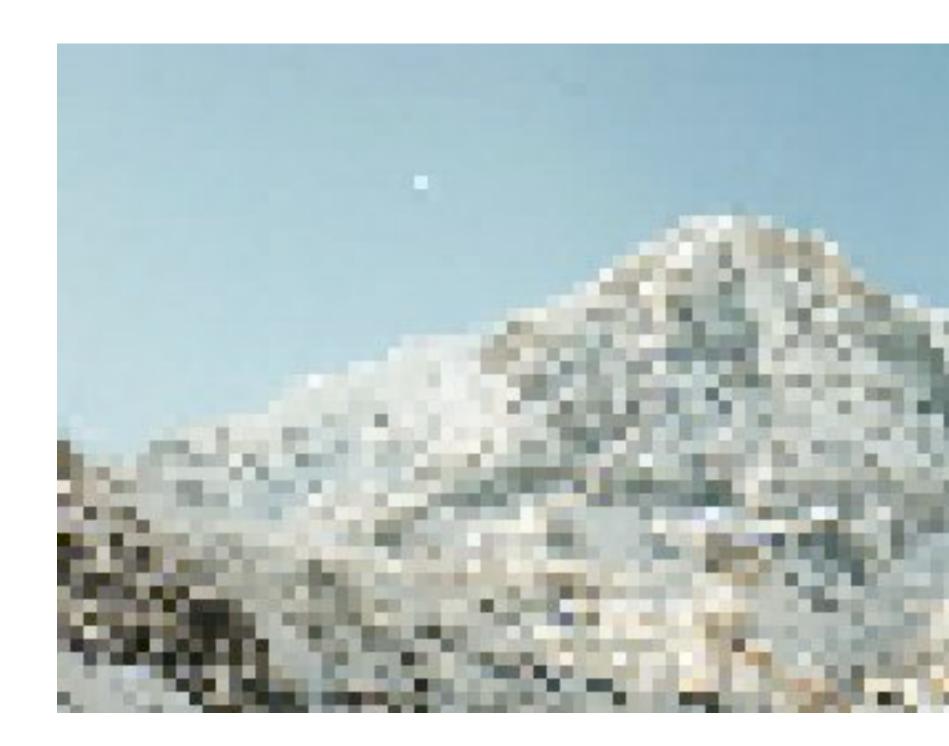
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With correct sigma value for a Gaussian, no information is lost



Aliasing Example

Sampling every 5th pixel with and without low-pass blur



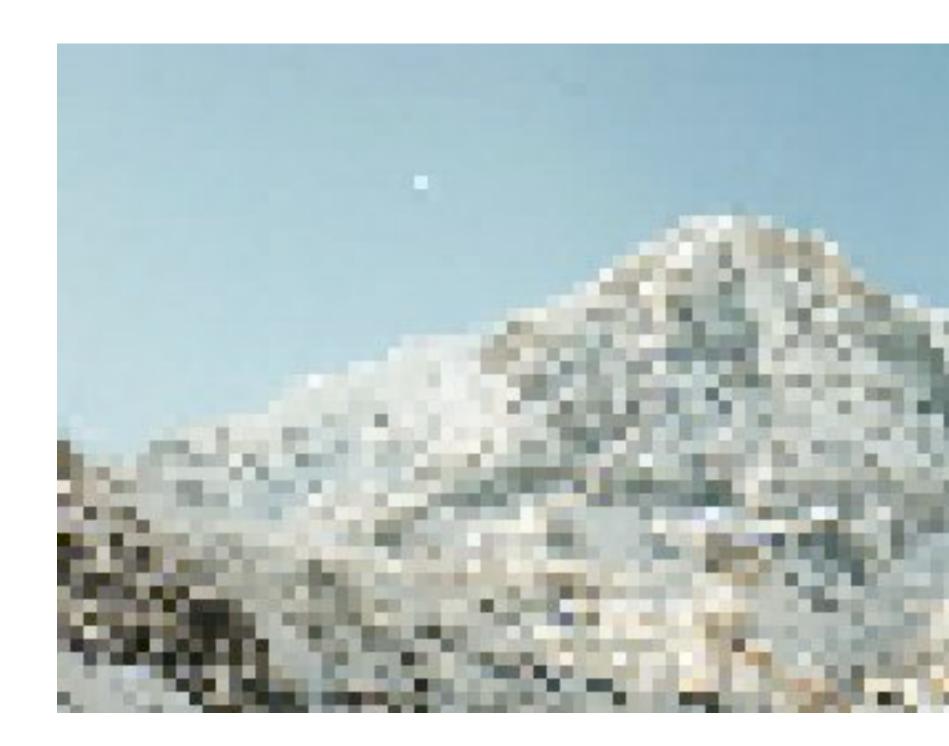
No filtering



Gaussian Blur $\sigma = 3.0$

Aliasing Example

Sampling every 5th pixel with and without low-pass blur



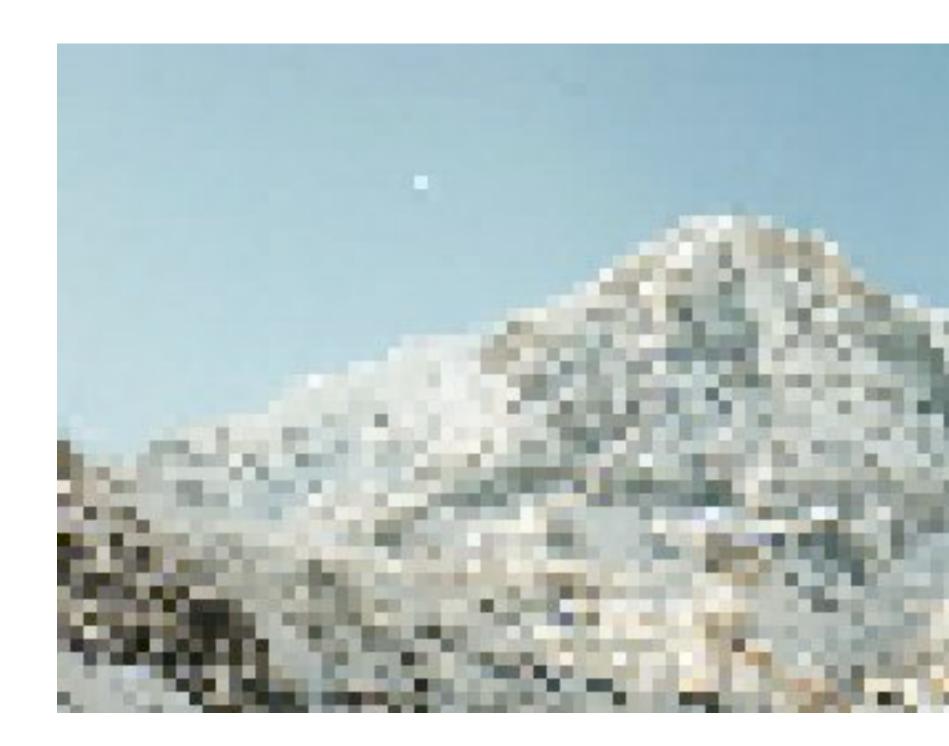
No filtering



Gaussian Blur $\sigma = 3.0$

Aliasing Example

Sampling every 5th pixel with and without low-pass blur



No filtering

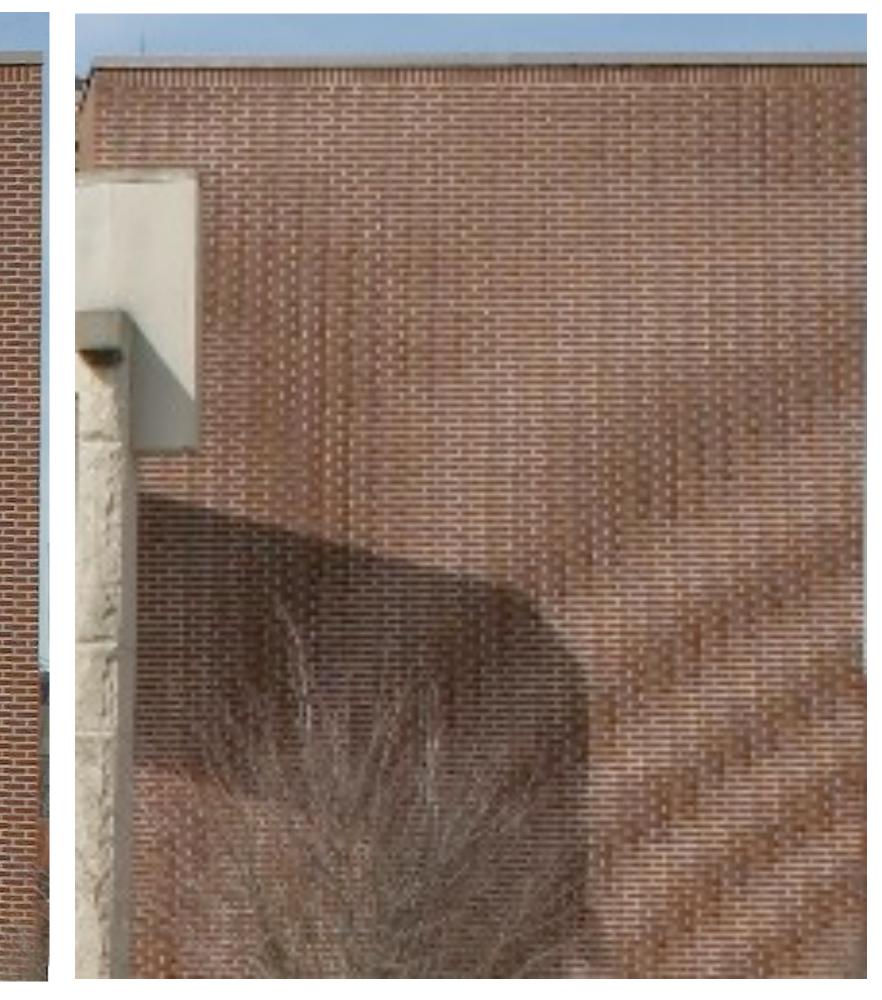


Gaussian Blur $\sigma = 3.0$

Image Sampling and Aliasing



 $f_s > 2 \times f_{max}$



 $f_s < 2 \times f_{max}$

Aliasing in Photographs

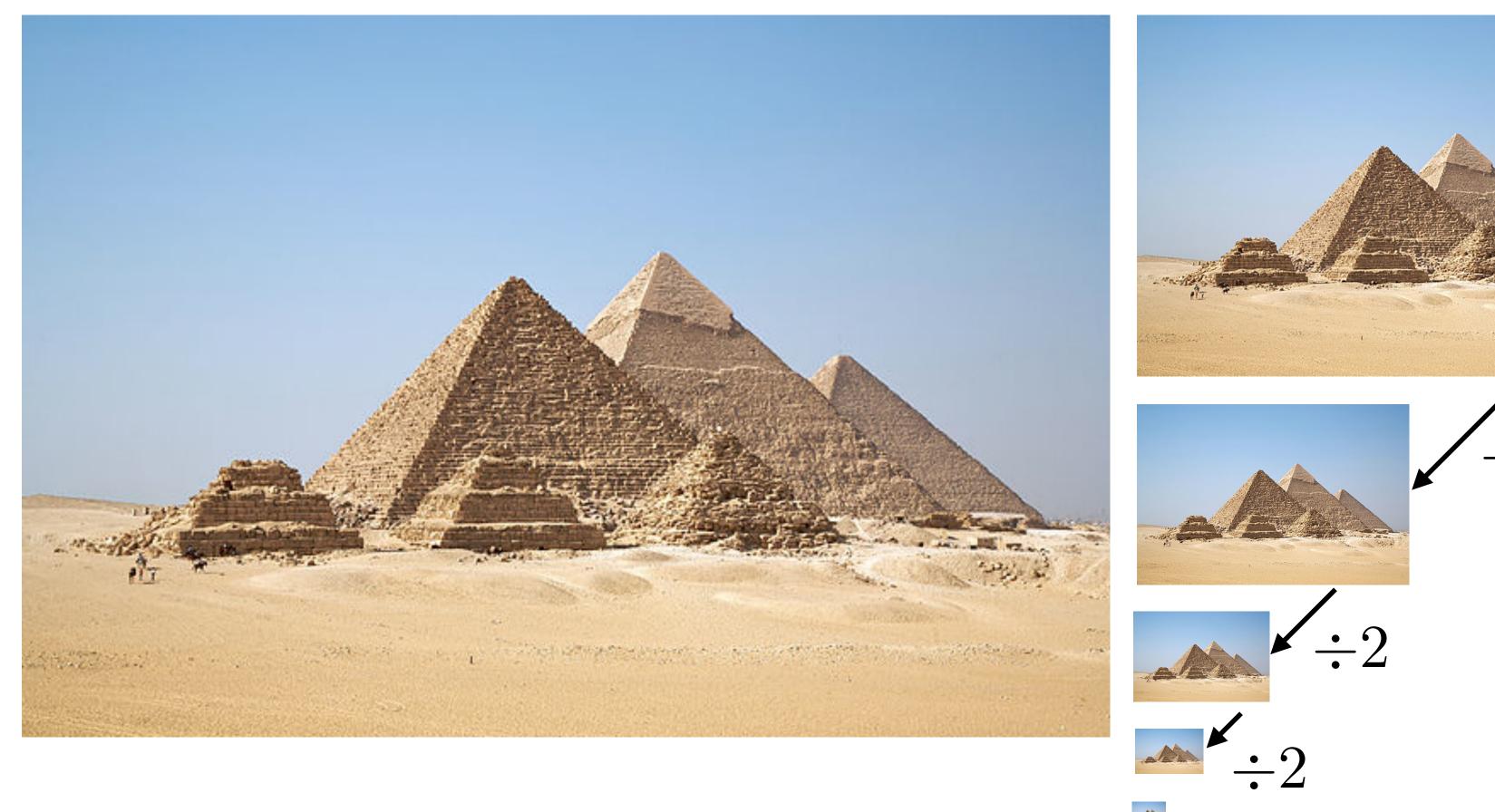
This is also known as "moire"







Image Pyramids



Used in Graphics (Mip-map) and Vision (for **multi-scale** processing)

Alter





Blur with a Gaussian kernel, then select every 2nd pixel

 $I_s(x,y) = I(x,y) * g_{\sigma}(x,y)$

Often approximations to the Gaussian kernel are used, e.g., $\frac{1}{16} \begin{bmatrix} 1 & 4 & 6 & 4 & 1 \end{bmatrix}$

[Assignment 2]







blur



 $I_s(x,y) = I(x,y) * g_\sigma(x,y)$

Often approximations to the Gaussian kernel are used, e.g., $\frac{1}{16} \begin{bmatrix} 1 & 4 & 6 & 4 & 1 \end{bmatrix}$







blur

G2

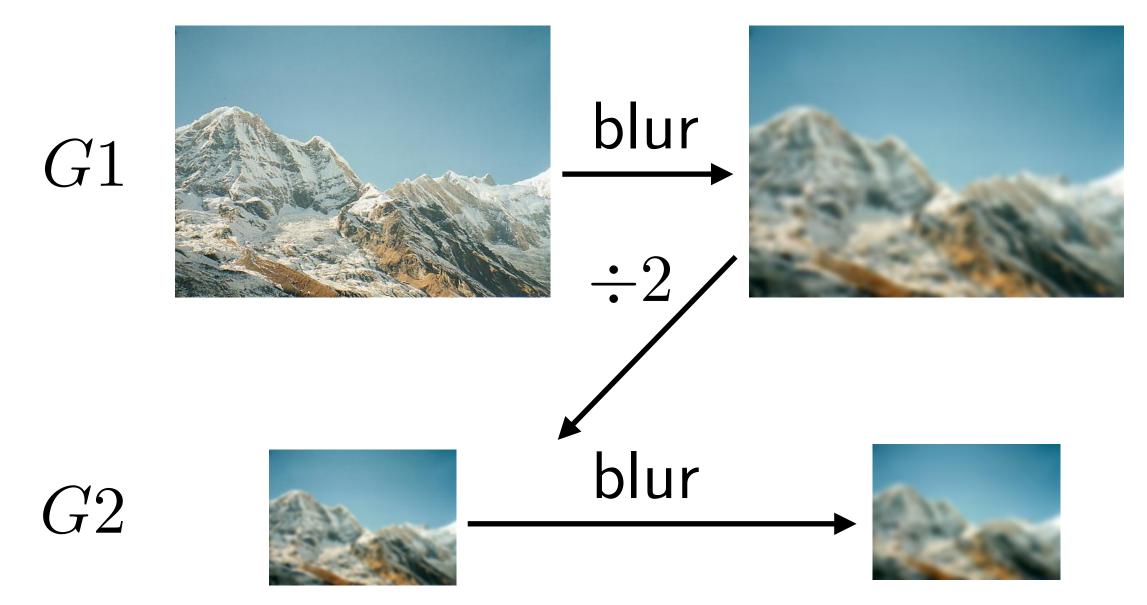




 $I_s(x,y) = I(x,y) * g_\sigma(x,y)$

Often approximations to the Gaussian kernel are used, e.g., $\frac{1}{16} \begin{bmatrix} 1 & 4 & 6 & 4 & 1 \end{bmatrix}$



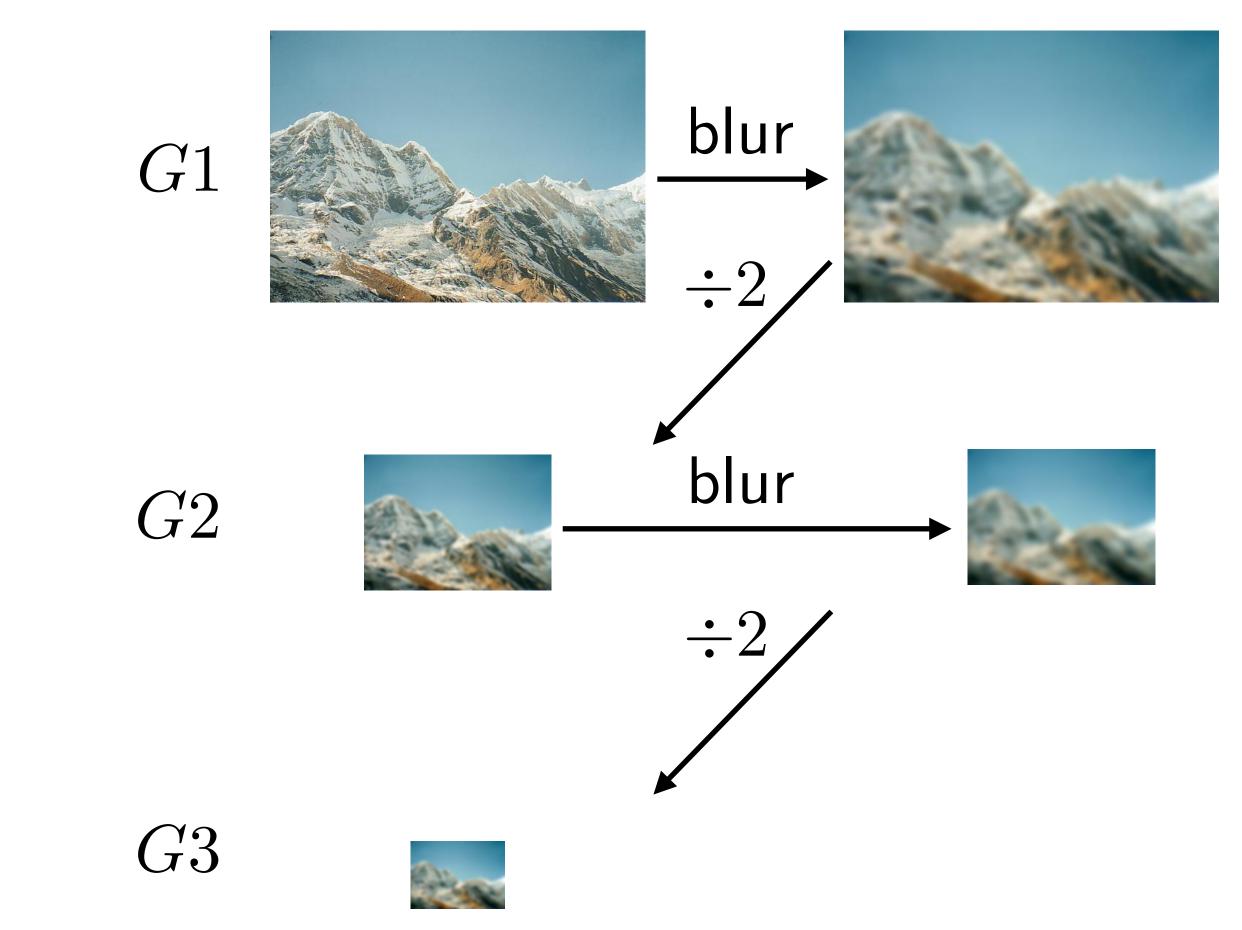


 $I_s(x,y) = I(x,y) * g_{\sigma}(x,y)$

Often approximations to the Gaussian kernel are used, e.g., $\frac{1}{16} \begin{bmatrix} 1 & 4 & 6 & 4 & 1 \end{bmatrix}$

[Assignment 2]



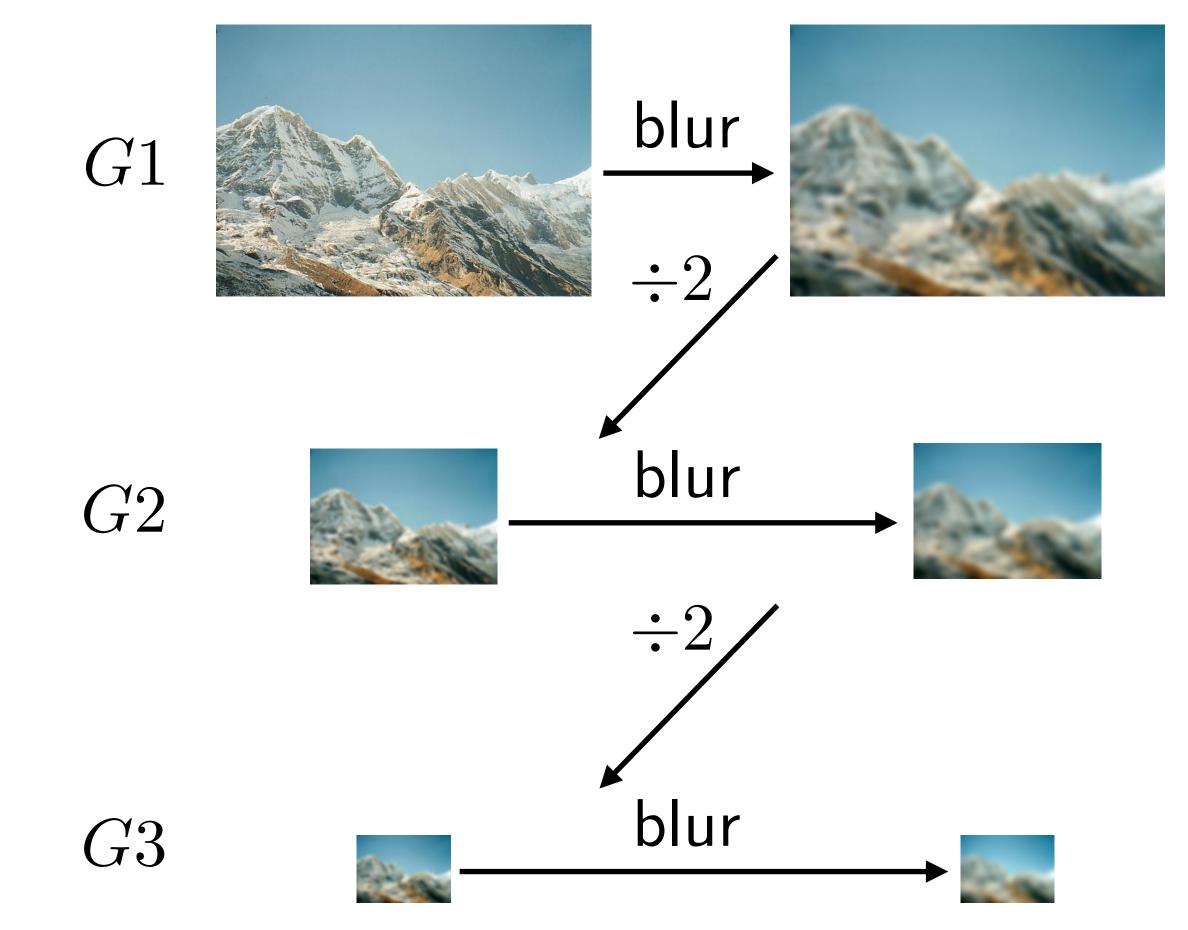


 $I_s(x,y) = I(x,y) * g_{\sigma}(x,y)$

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[Assignment 2]

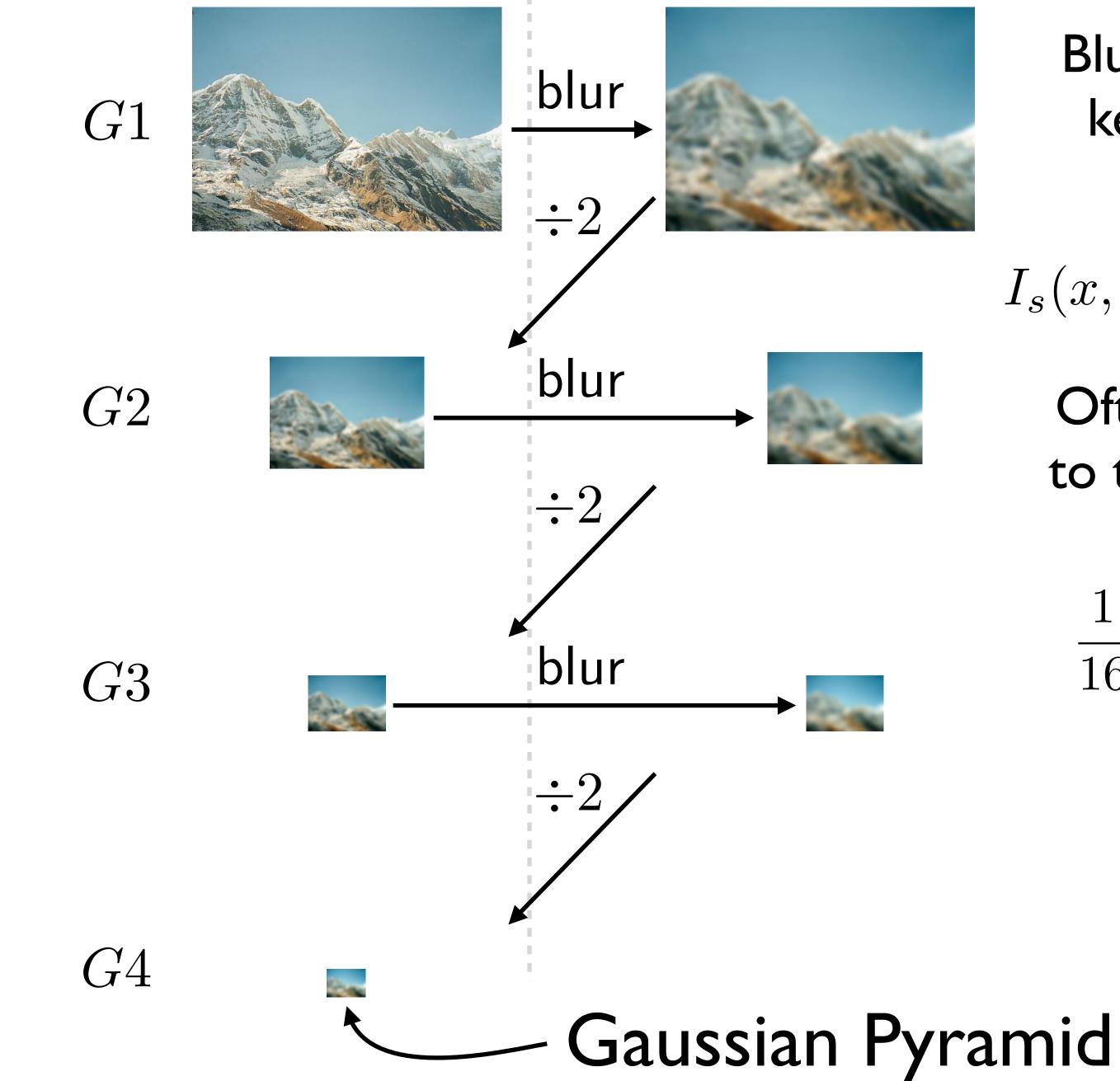




 $I_s(x,y) = I(x,y) * g_\sigma(x,y)$

Often approximations to the Gaussian kernel are used, e.g., $\frac{1}{16} \begin{bmatrix} 1 & 4 & 6 & 4 & 1 \end{bmatrix}$

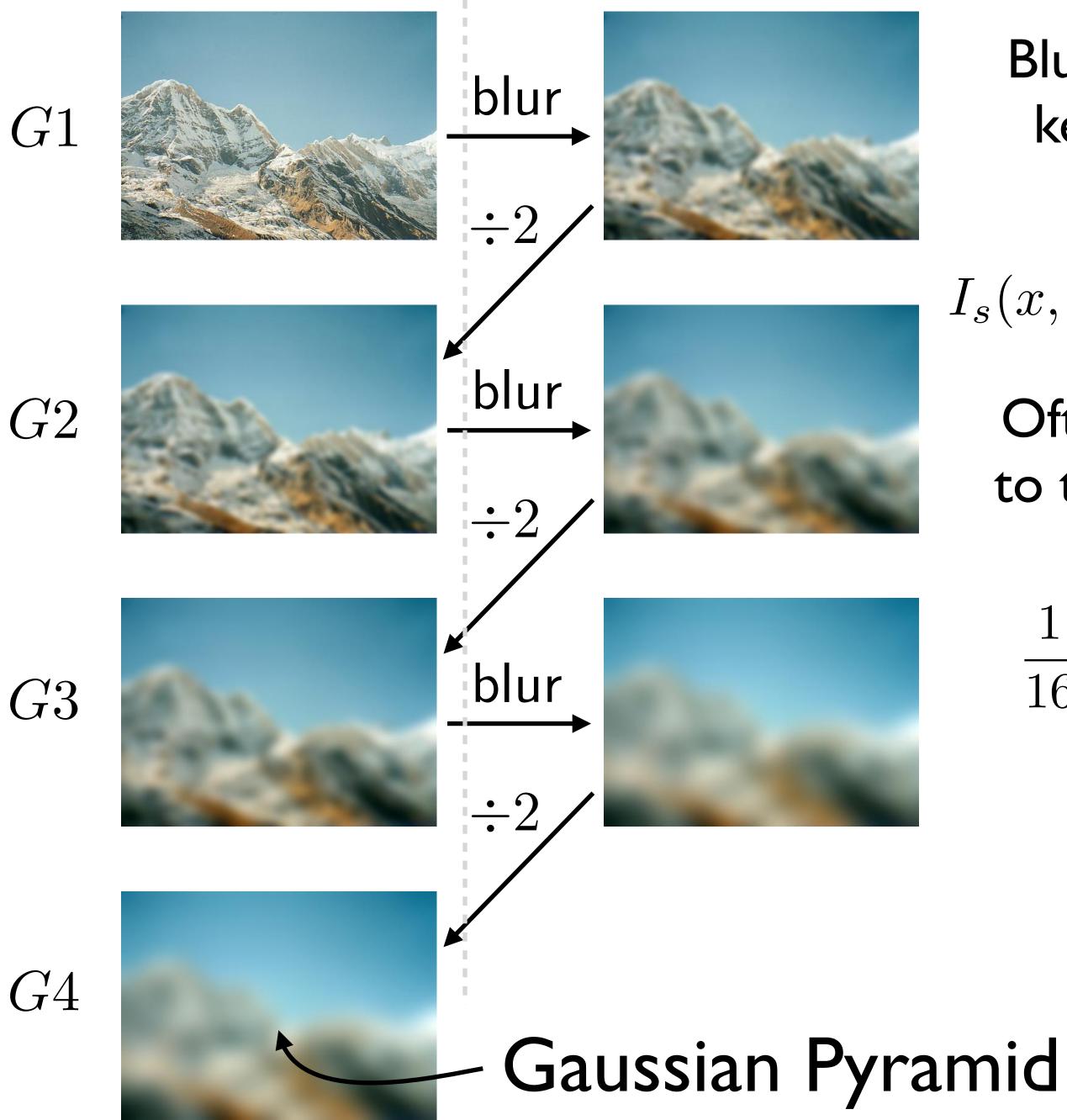




 $I_s(x,y) = I(x,y) * g_{\sigma}(x,y)$

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 $I_s(x,y) = I(x,y) * g_\sigma(x,y)$

Often approximations to the Gaussian kernel are used, e.g., $\frac{1}{16} \begin{bmatrix} 1 & 4 & 6 & 4 & 1 \end{bmatrix}$



Question: For a bandlimited signal, we greater than the Nyquist rate)

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Answer: Nothing bad happens! Samples are redundant and there are wasted bits

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Question: For a bandlimited signal, what if you **undersample** (i.e., sample at less than the Nyquist rate)

Question: For a bandlimited signal, what if you oversample (i.e., sample at

greater than the Nyquist rate)

bits

less than the Nyquist rate)

there aren't). There are artifacts (i.e., things that shouldn't be there are)

Question: For a bandlimited signal, what if you oversample (i.e., sample at

Answer: Nothing bad happens! Samples are redundant and there are wasted

Question: For a bandlimited signal, what if you **undersample** (i.e., sample at

Answer: Two bad things happen! Things are missing (i.e., things that should be

How to Prevent Aliasing?

1. Reduce the maximum frequency, by low pass filtering i.e., Smoothing before sampling.

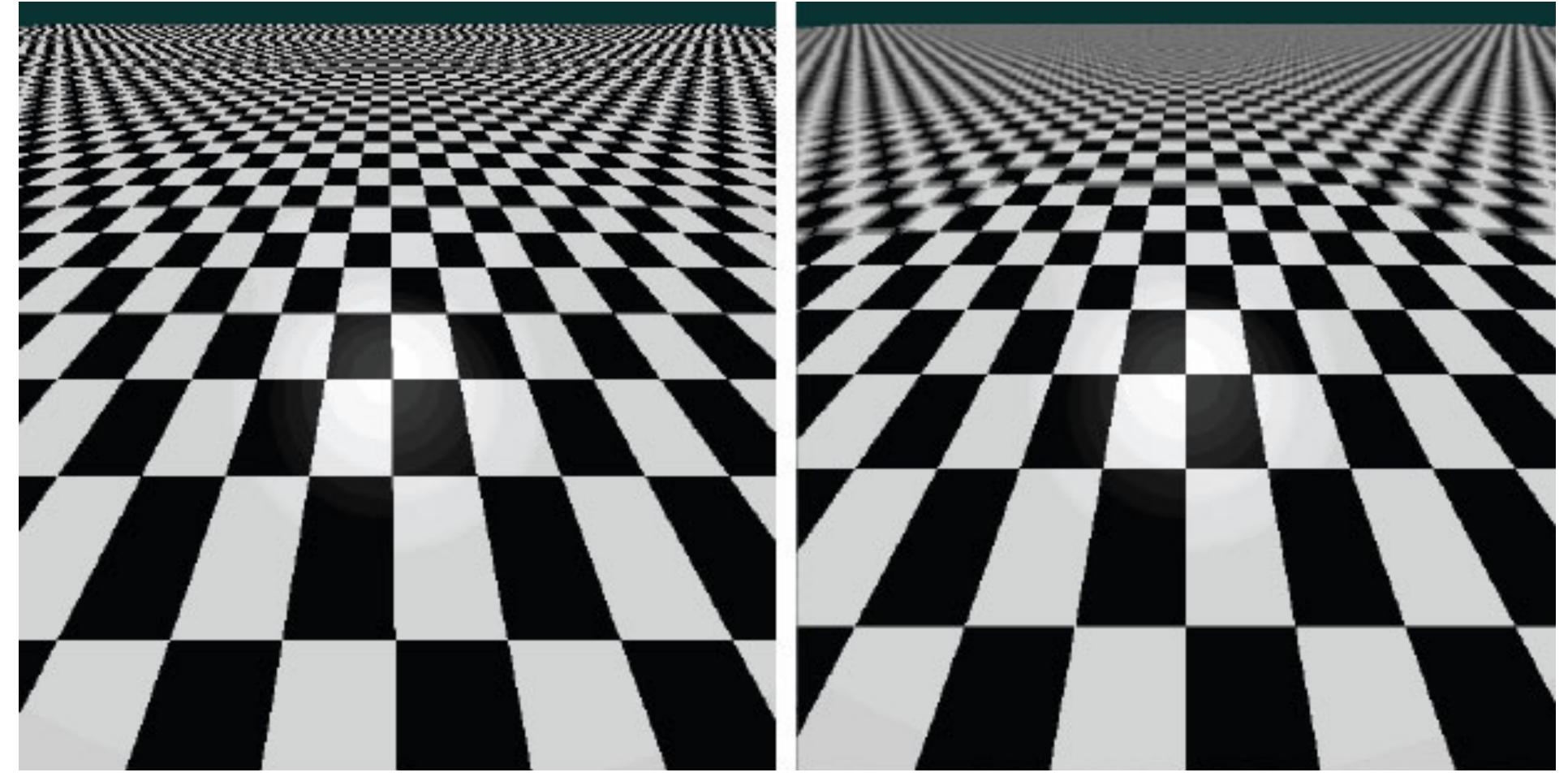
How to Prevent Aliasing?

1. **Reduce the maximum frequenc** before sampling.

2. **Sample more frequently** i.e., oversampling — sample more than you think you need and average (i.e., area sampling)

1. Reduce the maximum frequency, by low pass filtering i.e., Smoothing

Aliasing



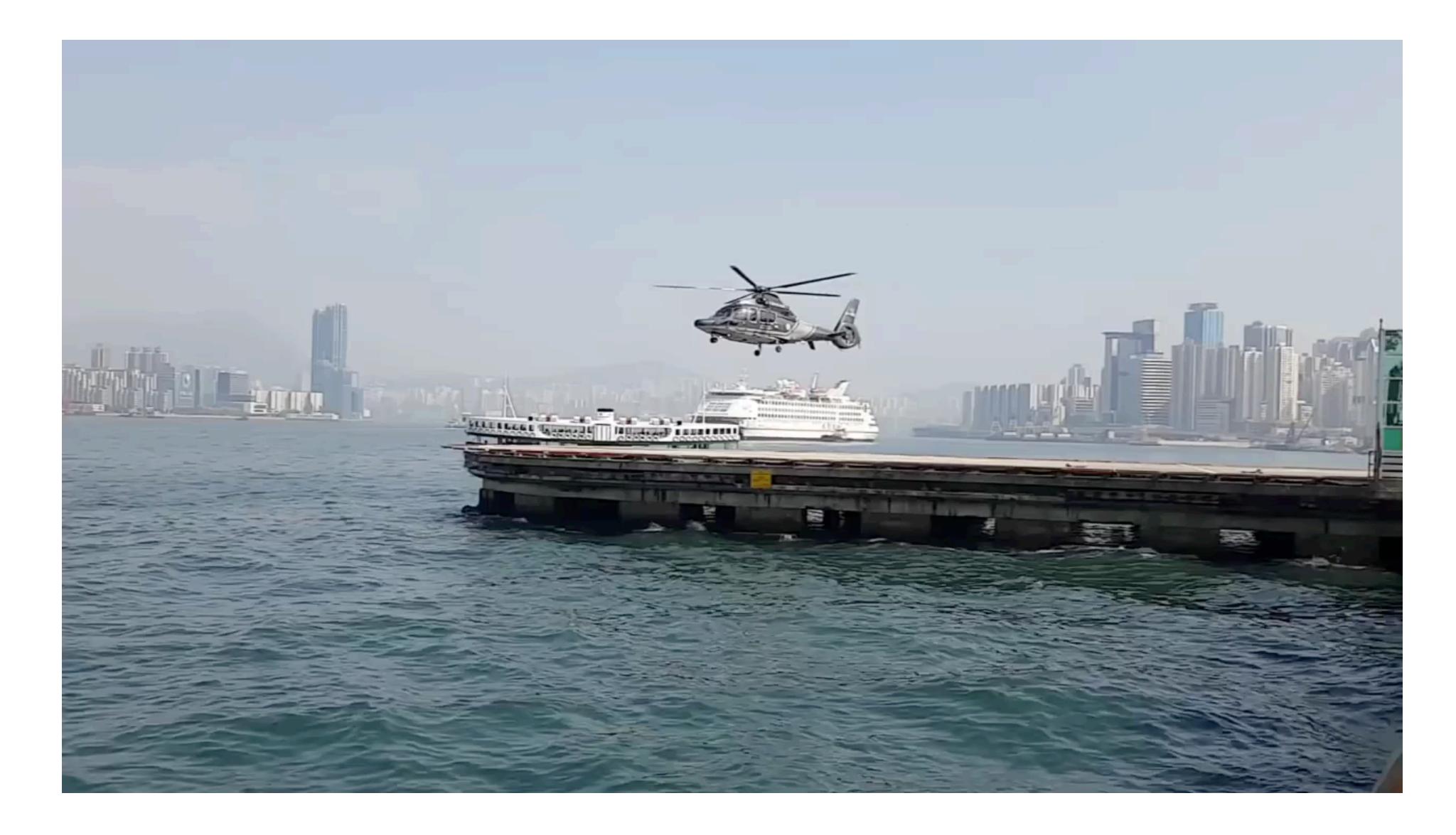
aliasing artifacts

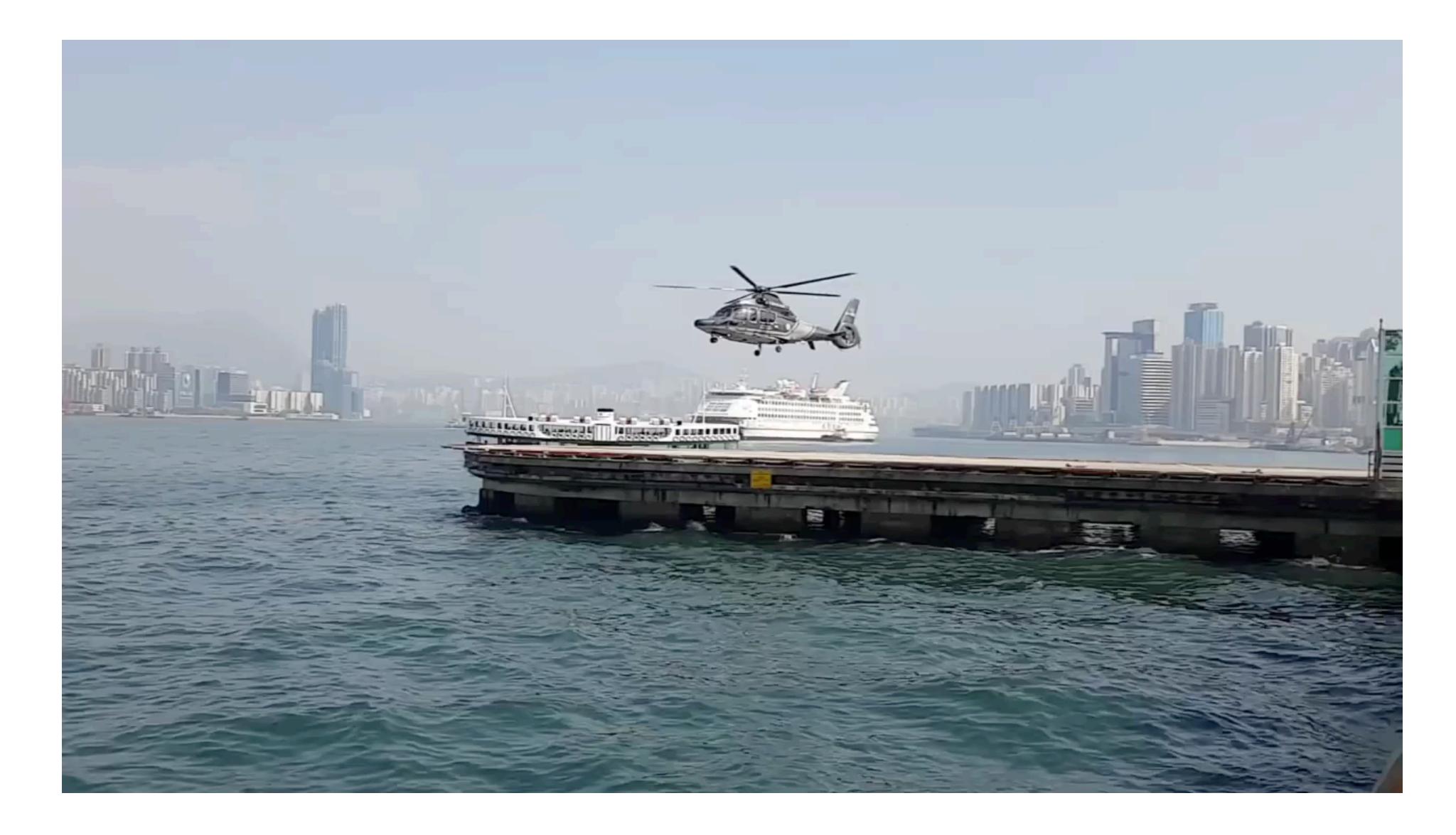
anti-aliasing by oversampling





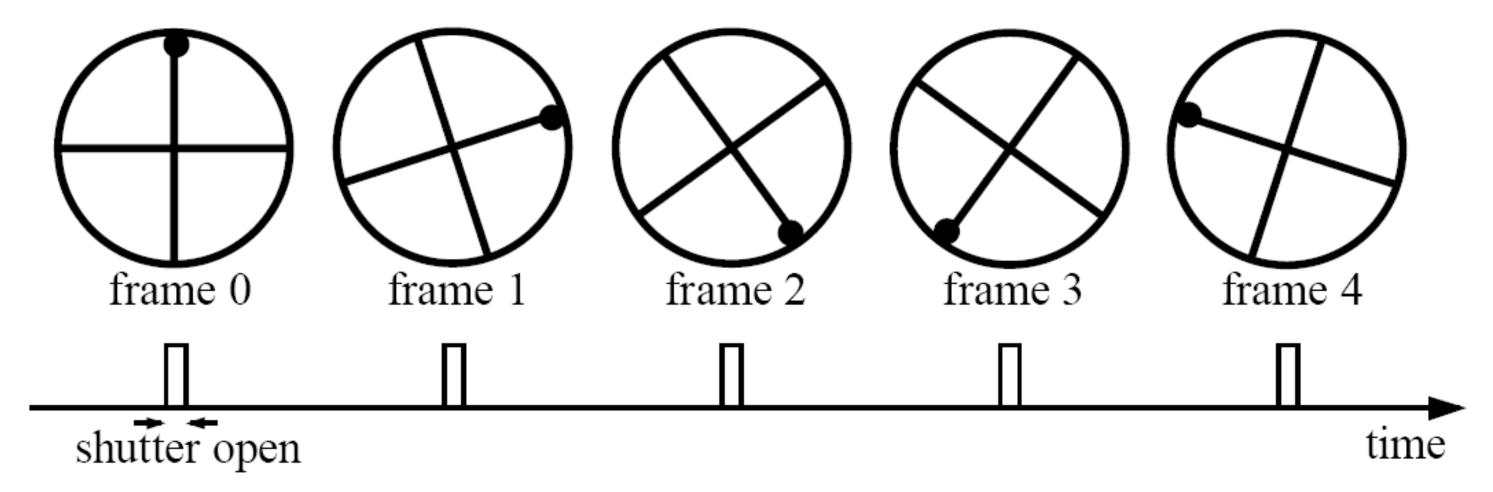






Mark wheel with dot so we can see what's happening.

time = 1/30 sec. for video, 1/24 sec. for film):



(counterclockwise)

- Imagine a spoked wheel moving to the right (rotating clockwise).
- If camera shutter is only open for a fraction of a frame time (frame

Without dot, wheel appears to be rotating slowly backwards!

Sometimes **undersampling** is unavor "things missing" and "artifacts."

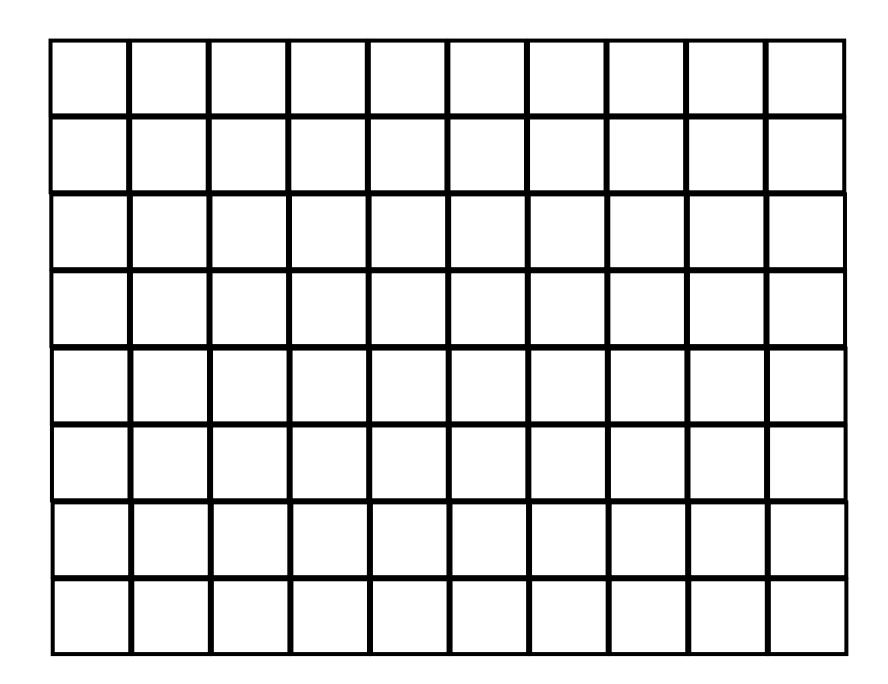
Medical imaging: usually try to maximize information content, tolerate some artifacts

Computer graphics: usually try to minimize artifacts, tolerate some information missing

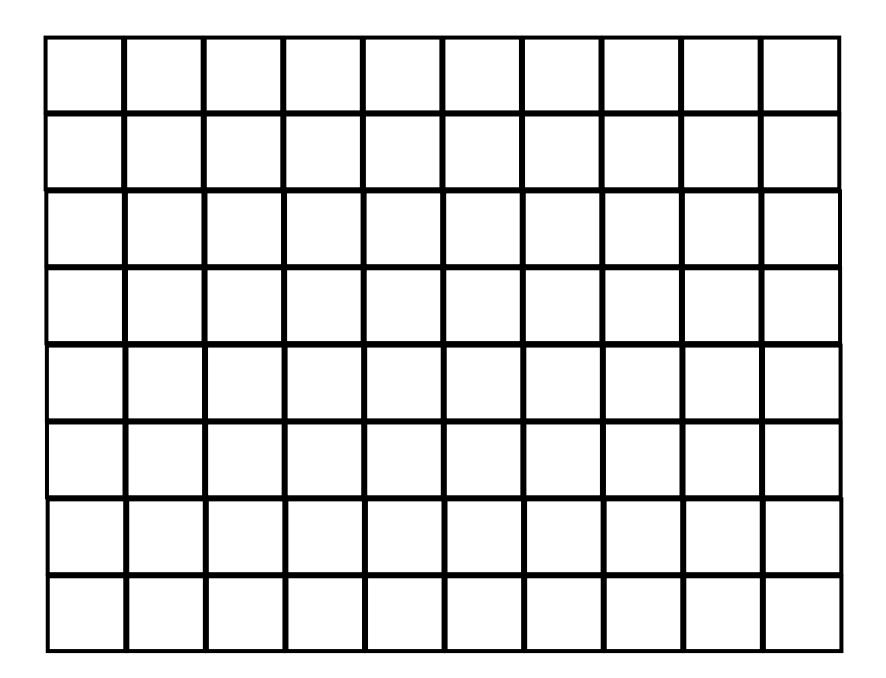
Sometimes undersampling is unavoidable, and there is a trade-off between

Example

Sensor Resolution: 10 x 8



Sensor Resolution: 10 x 8



Example

Sensor Resolution: 10 x 8

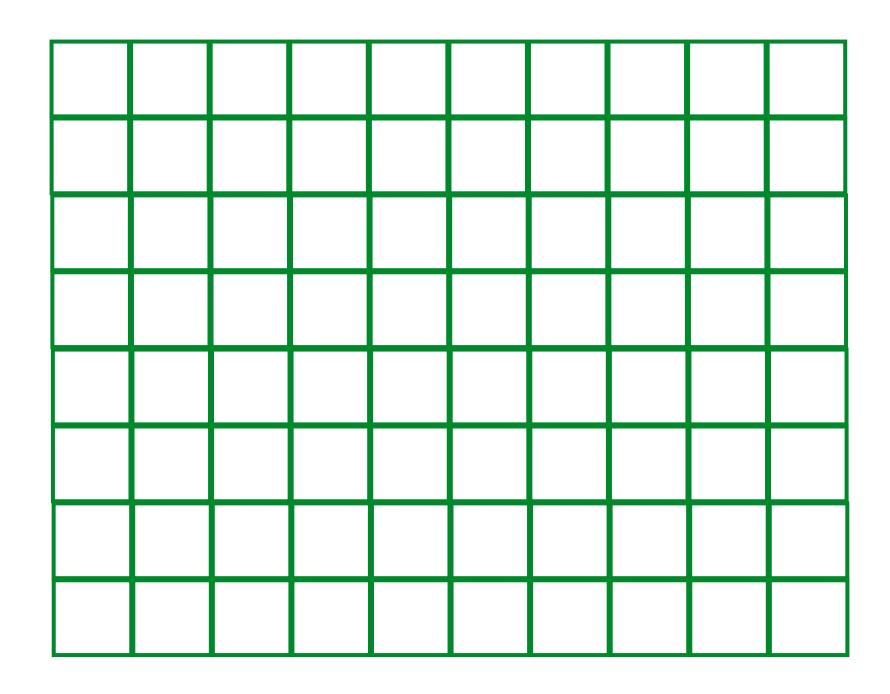


Image Resolution: 10 x 8

Sensor Resolution: 10 x 8

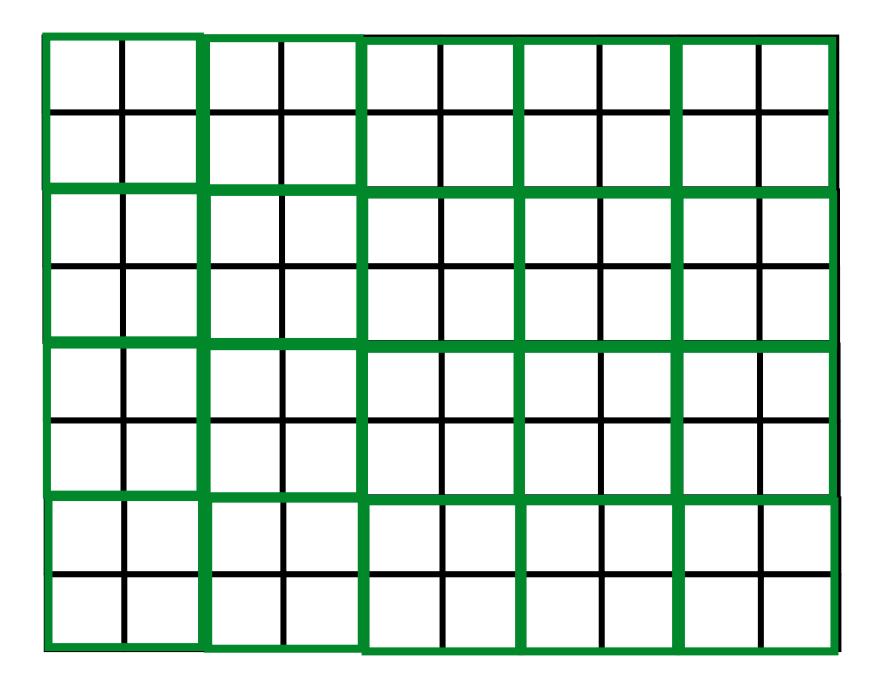
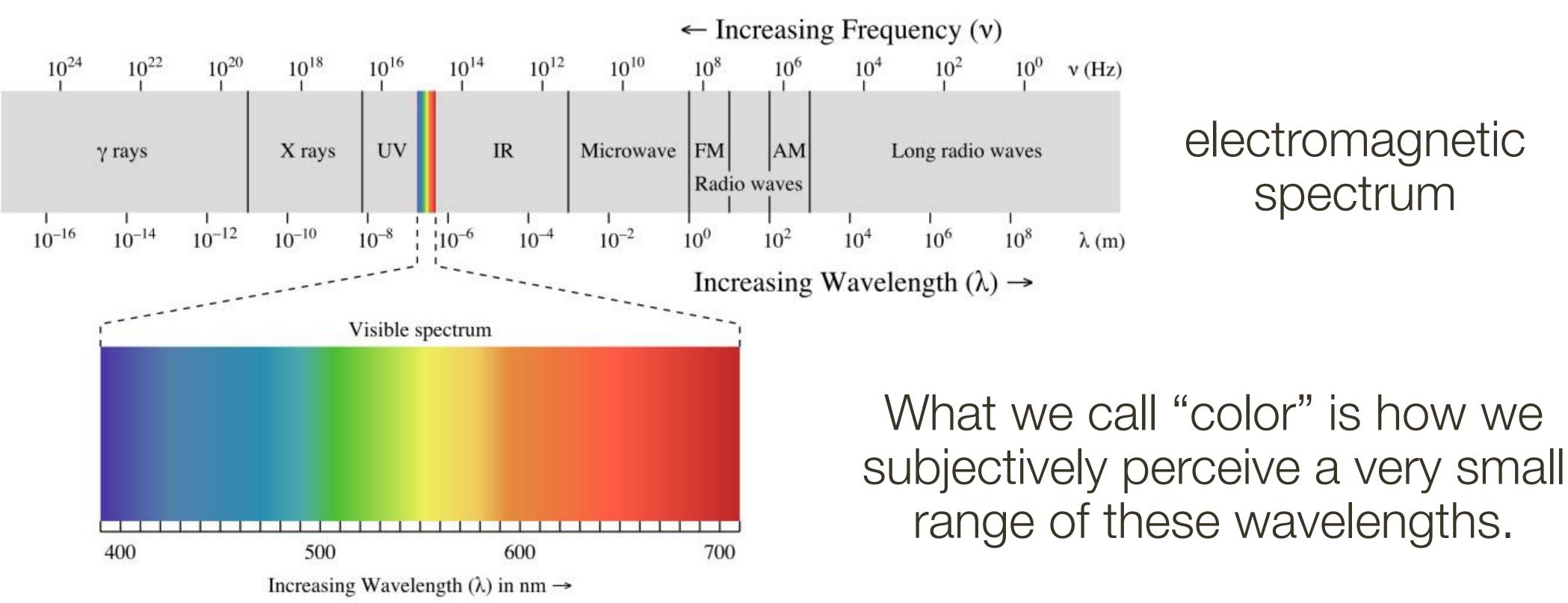


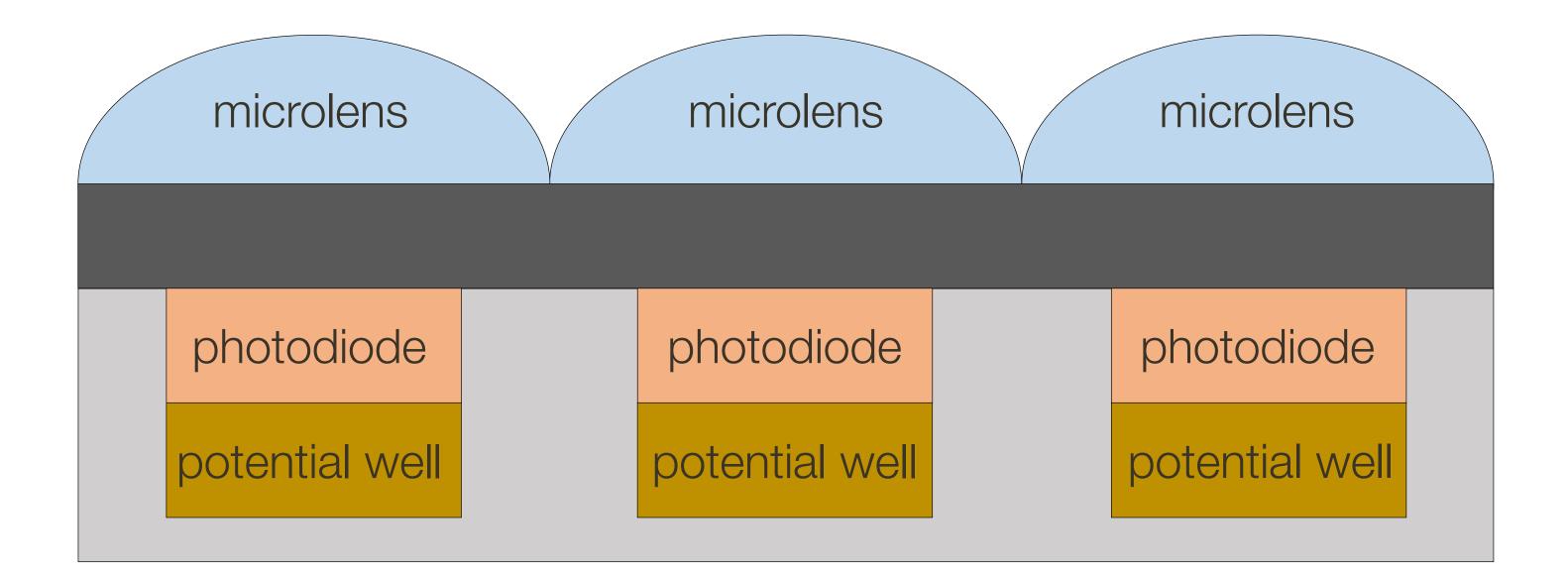
Image Resolution: 5 x 4

Color is an Artifact of Human Perception

"Color" is **not** an objective physical property of light (electromagnetic radiation). Instead, light is characterized by its wavelength.

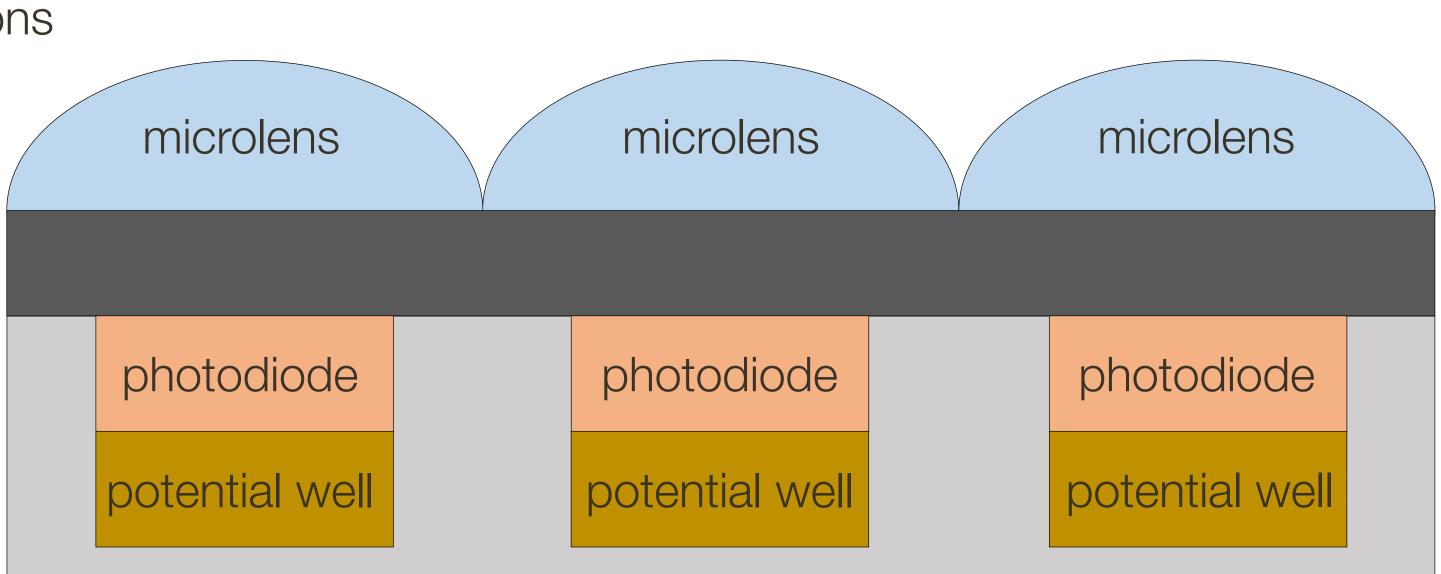






In addition to a camera lens,

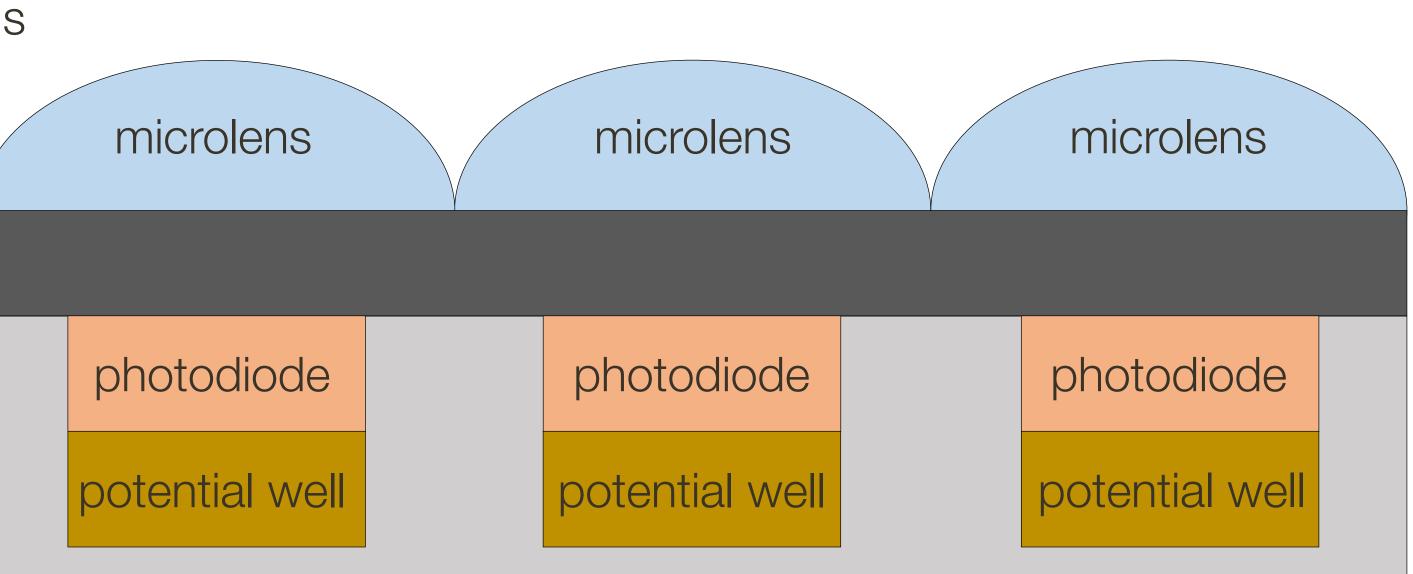
each pixel has a microns



In addition to a camera lens,

each pixel has a microns

Photodiode: converts photons to electrons

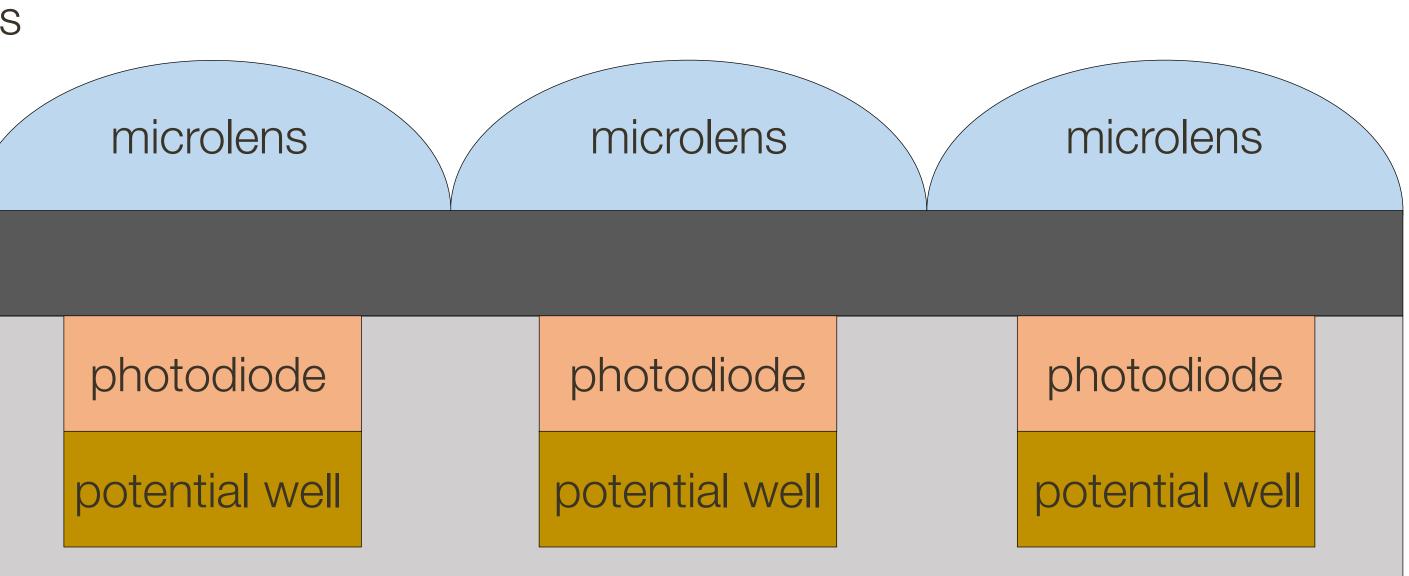


In addition to a camera lens,

each pixel has a microns

Photodiode: converts photons to electrons

Electrons stored in the potential well, until they are read off

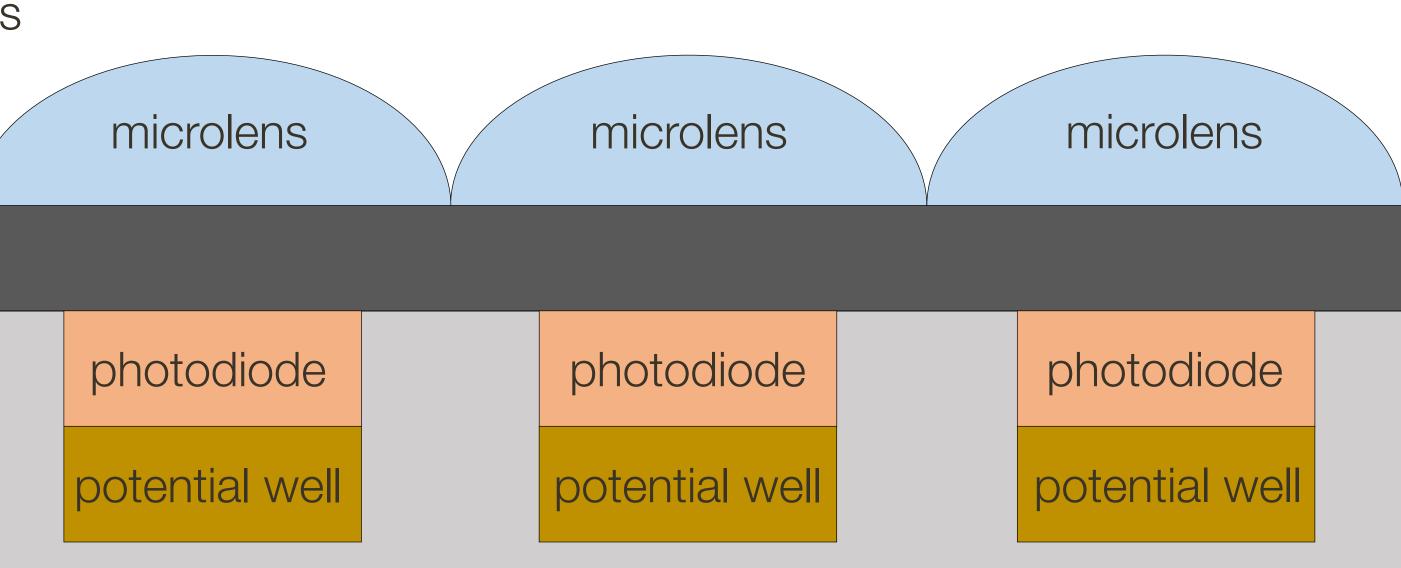


In addition to a camera lens,

each pixel has a microns

Photodiode: converts photons to electrons

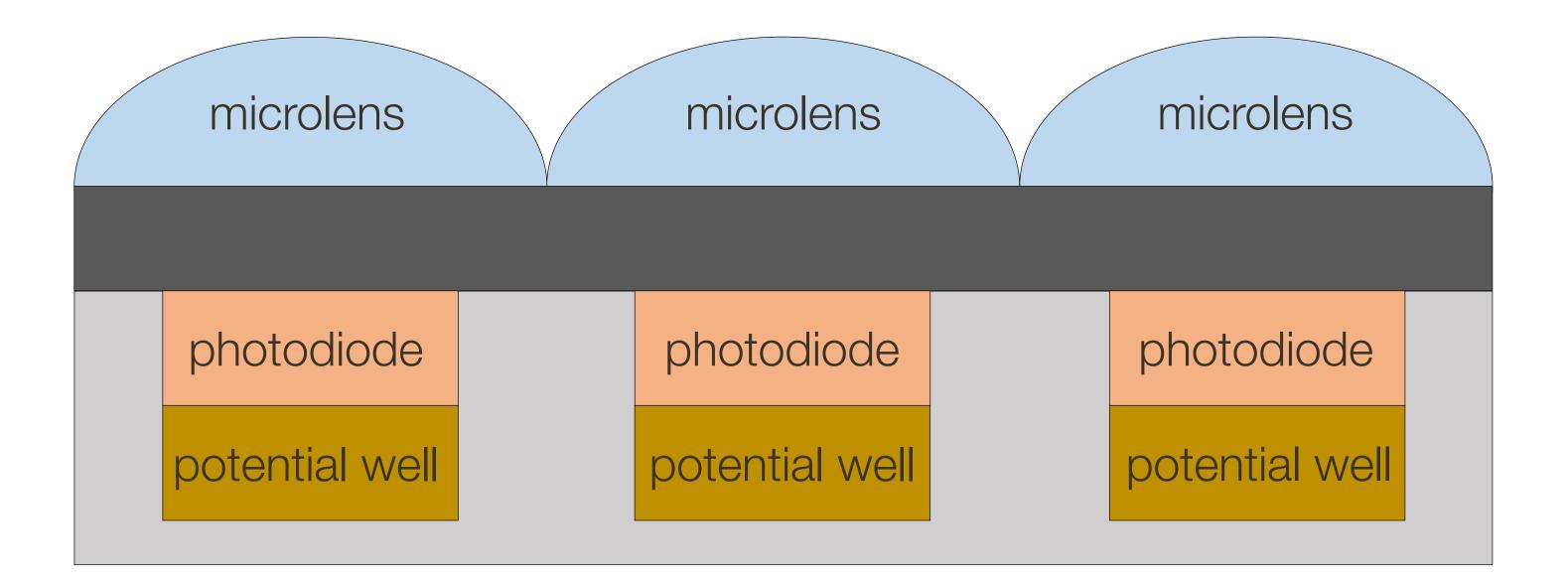
Electrons stored in the potential well, until they are read off

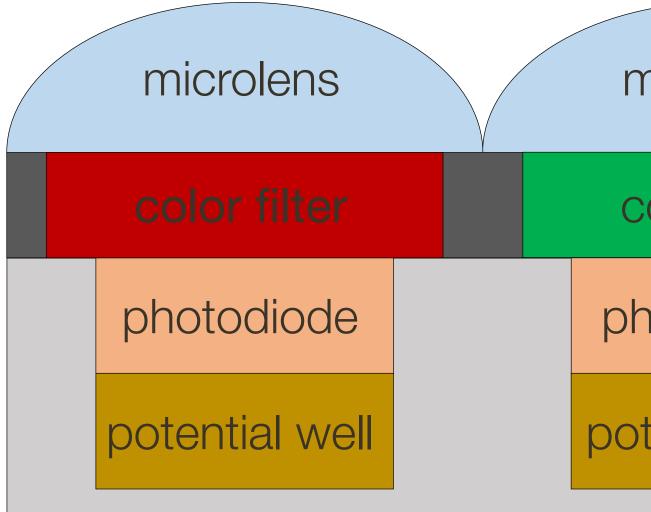


Quantum efficiency: fraction of photons being "detected" through this process (human eye QE: 20%, film cameras QE: 10%, CCD QE: 80%)



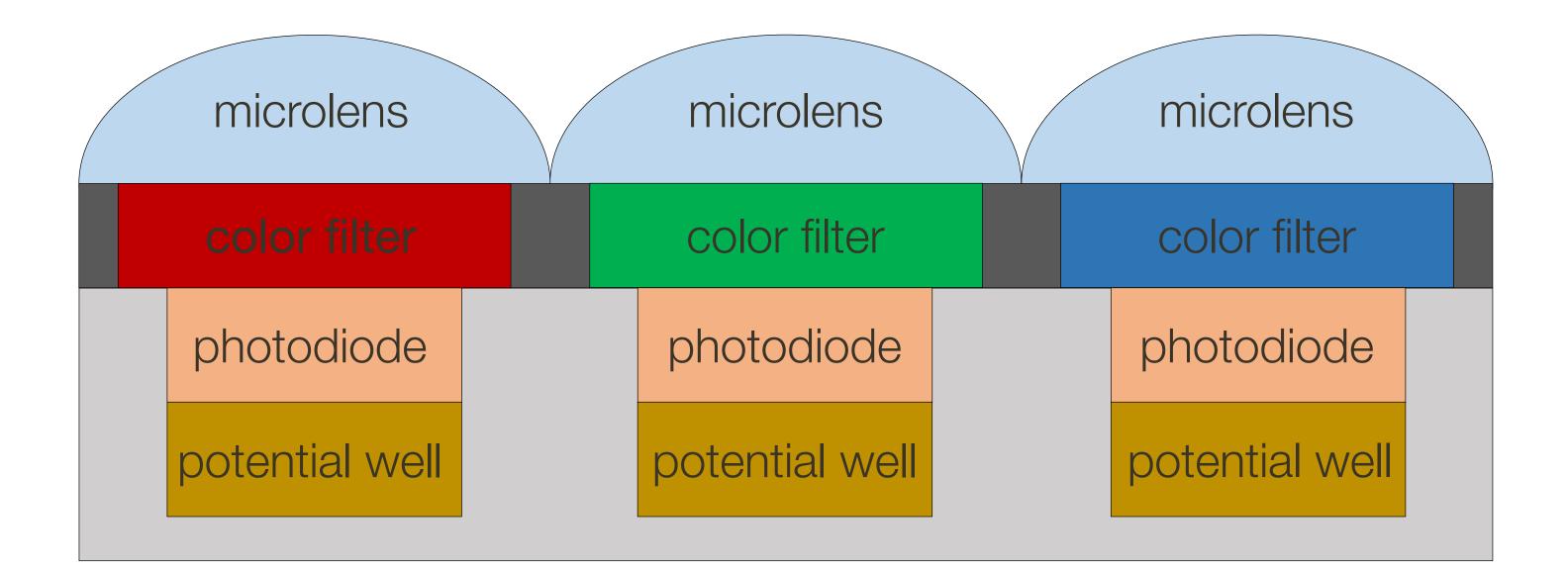
Issue: Color Filter Array (SFA) by itself has no way of distinguishing wavelengths of light, just ability to record the amount of light incident on an element





nicrolens		microlens	
color filter		color filter	
notodiode		photodiode	
otential well		potential well	

Implication: Only certain wavelengths of light are recorded at a given pixel



Two design choices:

- What spectral sensitivity functions $f(\lambda)$ to use for each color filter?
- How to spatially arrange ("mosaic") different color filters?

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- What spectral sensitivity functions $f(\lambda)$ to use for each color filter?
- How to spatially arrange ("mosaic") different color filters?

0.35

0.3

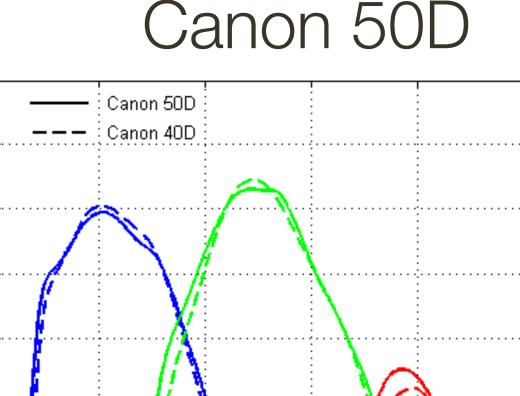
Dixel Quantum B Dixel Quantum B 0.2

0.1

0.05

4000

Efficiency



Generally do not match human sensitivity

 $f(\lambda)$ **Slide Credit**: Ioannis (Yannis) Gkioulekas (CMU)

5000

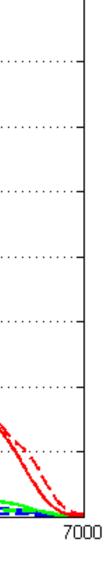
4500

6000

5500

Wavelength (A)

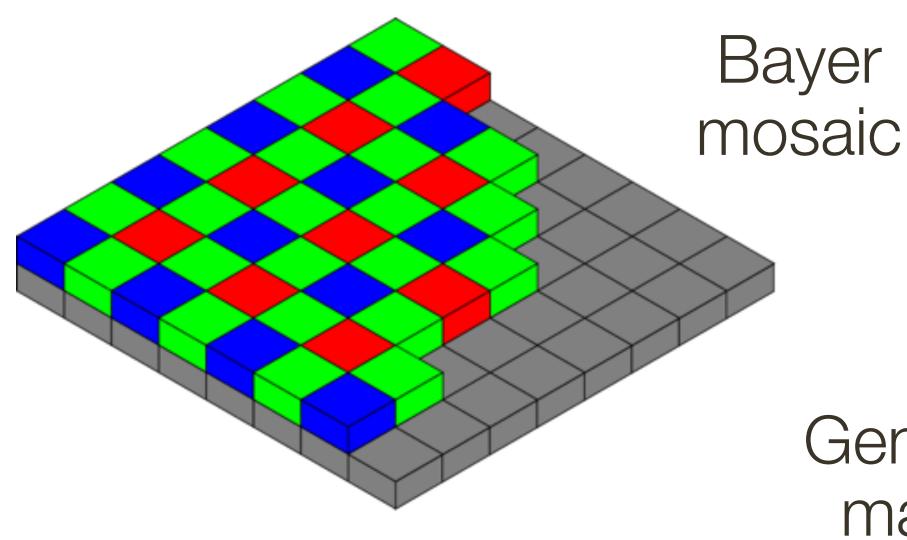
6500



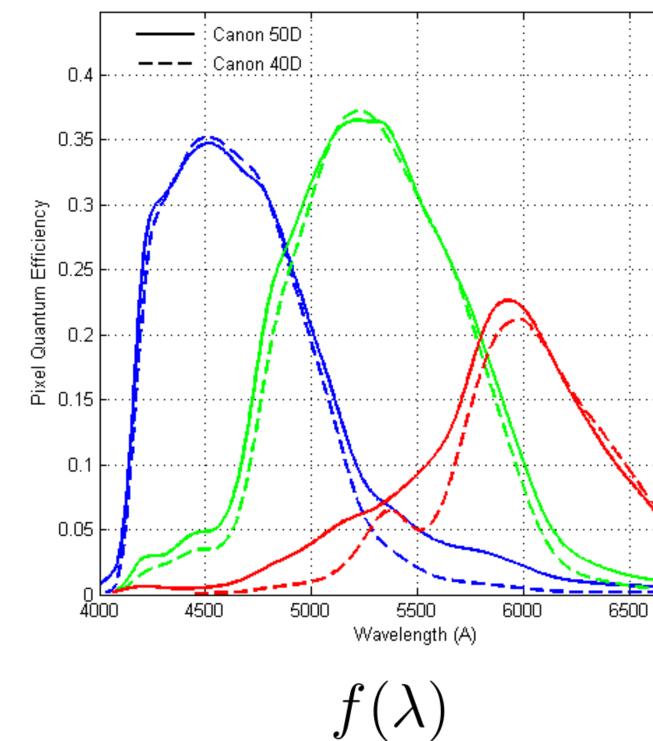


Two design choices:

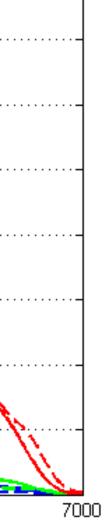
- What spectral sensitivity functions $f(\lambda)$ to use for each color filter?
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Canon 50D



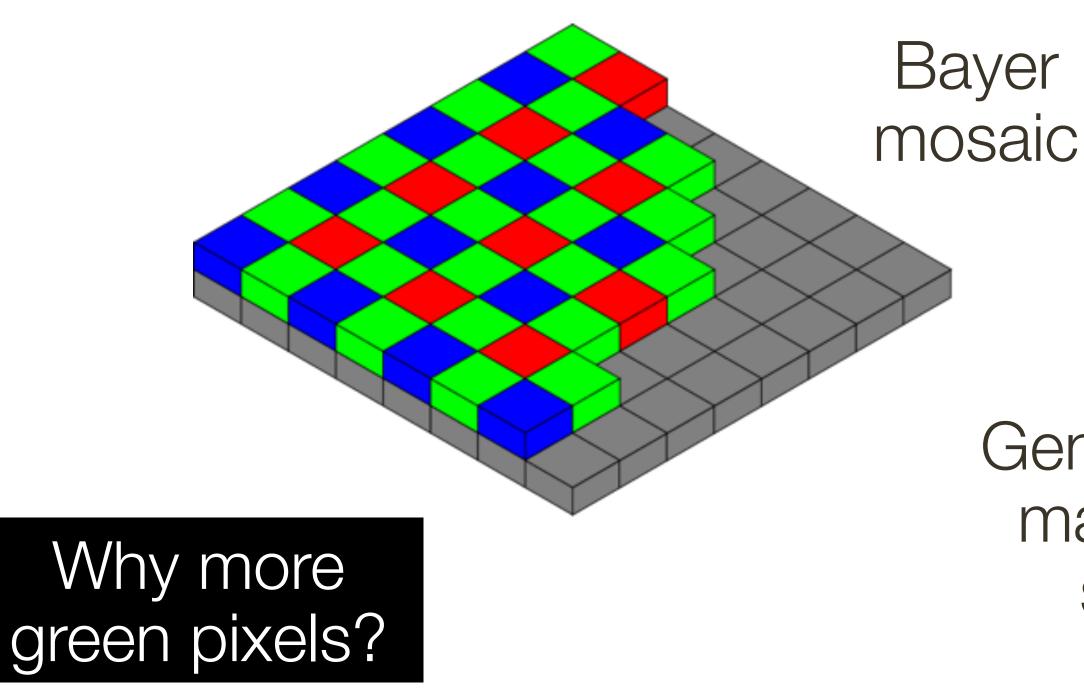
Generally do not match human sensitivity



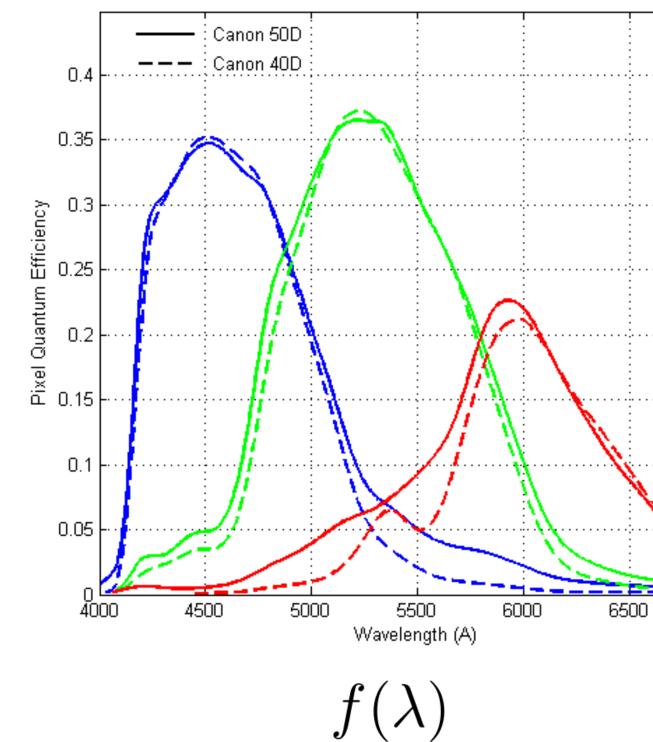


Two design choices:

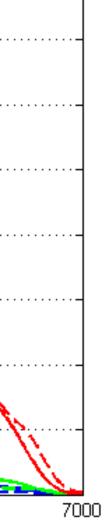
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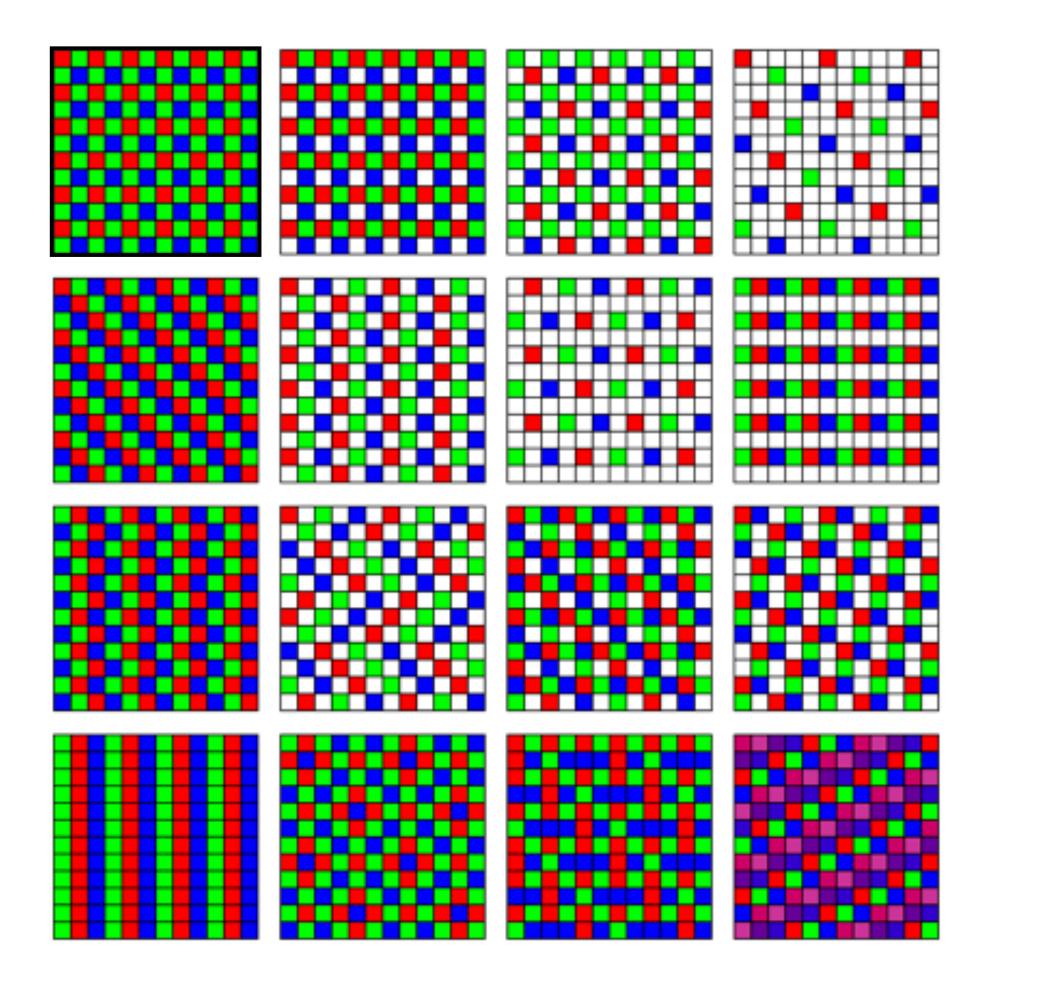
Generally do not match human sensitivity



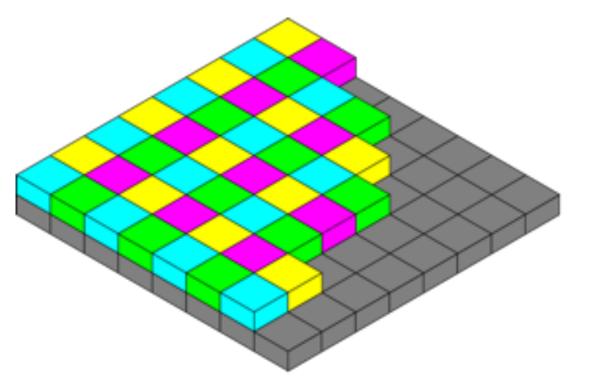


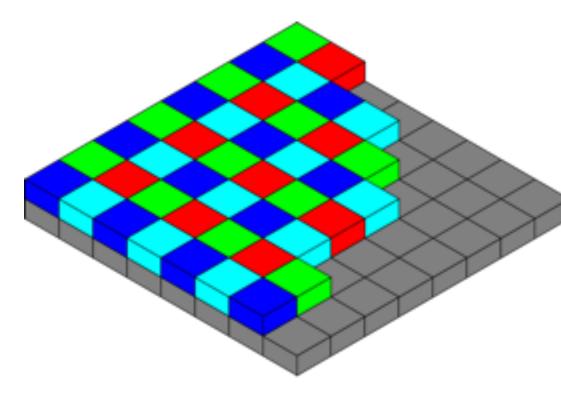
Different Color Filter Arrays (CFAs)

Finding the "best" CFA mosaic is an active research area.









CYGM Canon IXUS, Powershot

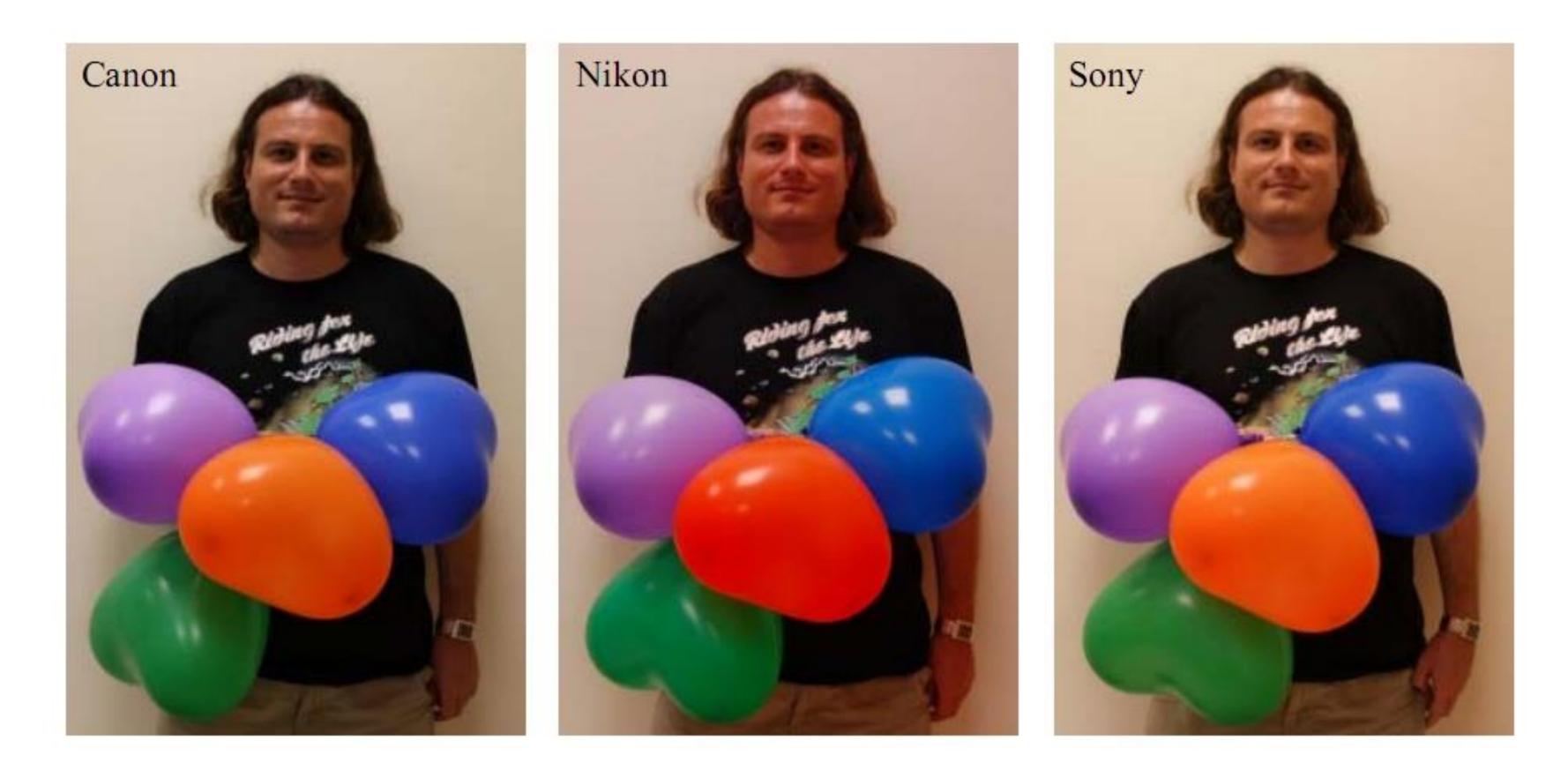
RGBE Sony Cyber-shot

How would you go about designing your own CFA? What criteria would you consider?



Many **Different Spectral Sensitivity** Functions

Each camera has its more or less unique, and most of the time secret, SSF



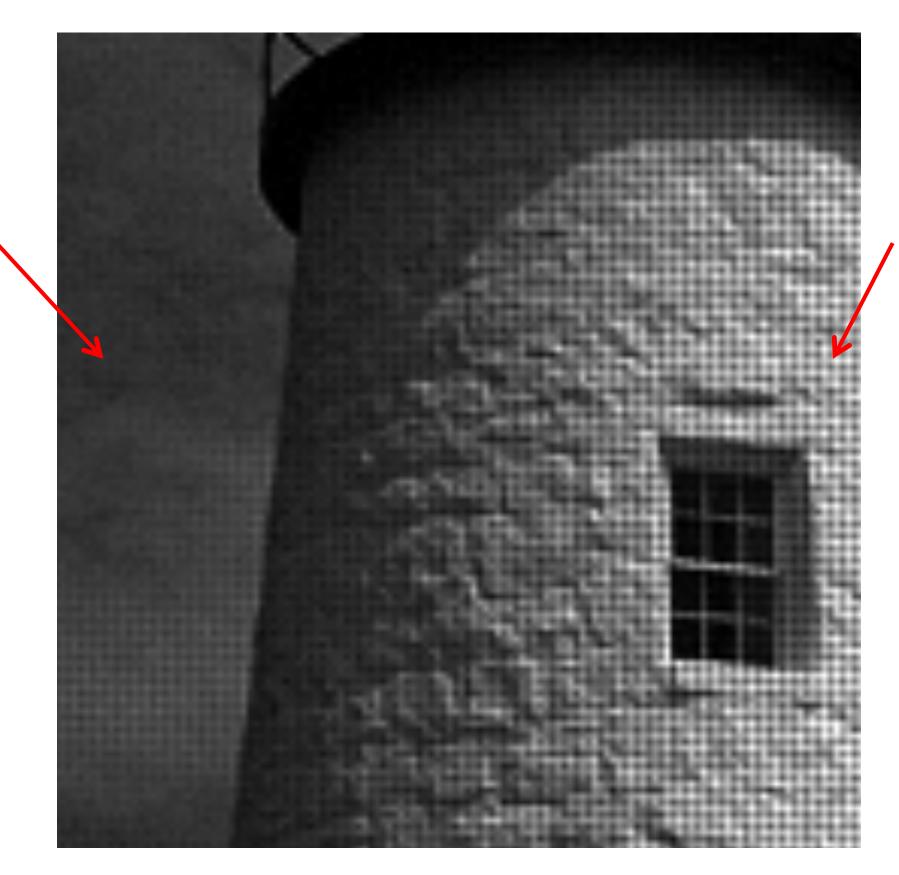
Same scene captured using 3 different cameras with identical settings

RAW Bayer Image

After all of this, what does an image look like?



lots of noise



mosaicking artifacts

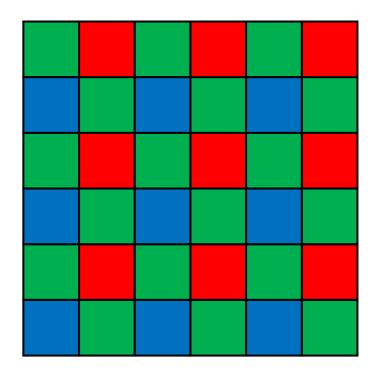
 Kind of disappointing We call this the RAW image

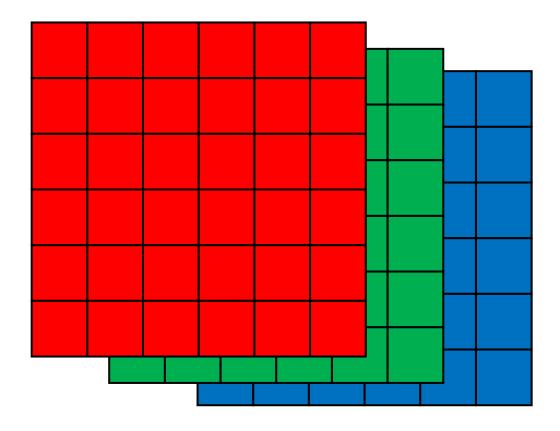




CFA Demosicing

Produce full RGB image from mosaiced sensor output

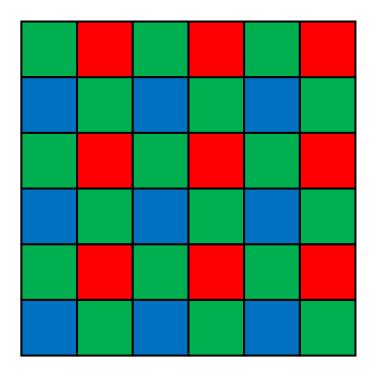




Any ideas on how to do this?

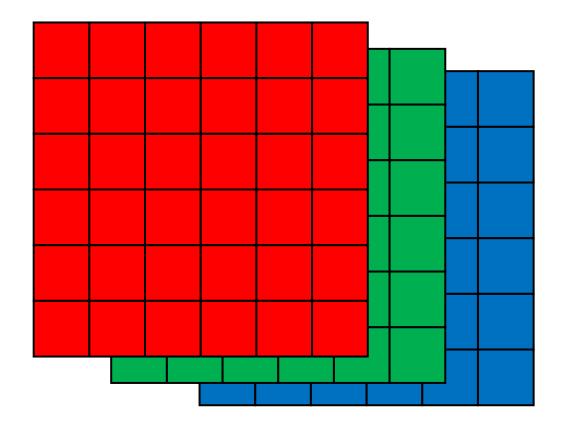
CFA **Demosicing**

Produce full RGB image from mosaiced sensor output



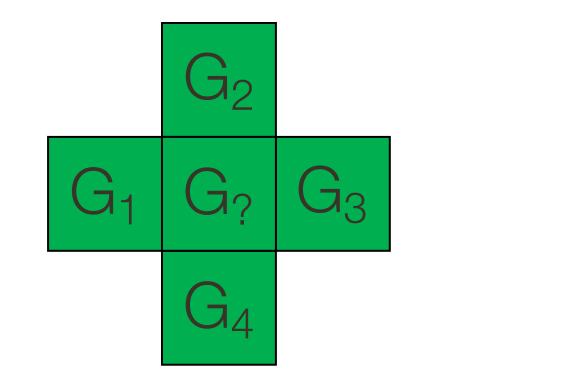
Interpolate from neighbors:

- Bilinear interpolation (needs 4 neighbors)
- Bicubic interpolation (needs more neighbors, may overblur)
- Edge-aware interpolation (e.g., Bilateral)



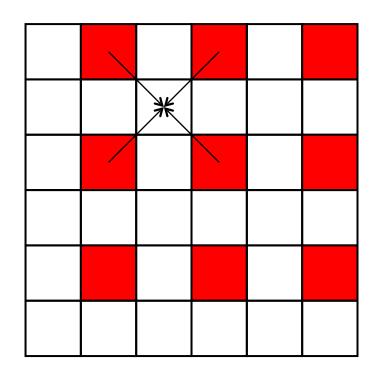
Demosaicing by Bilinear Interpolation

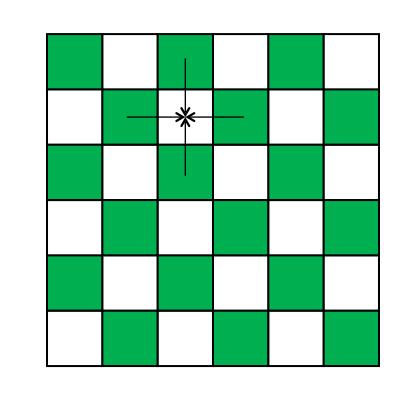
Bilinear interpolation: Simply average your 4 neighbors.

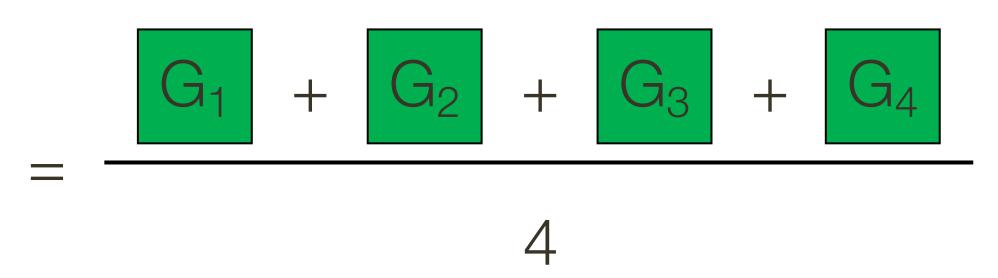


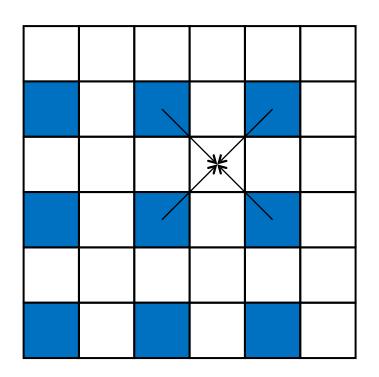


Neighborhood changes for different channels:



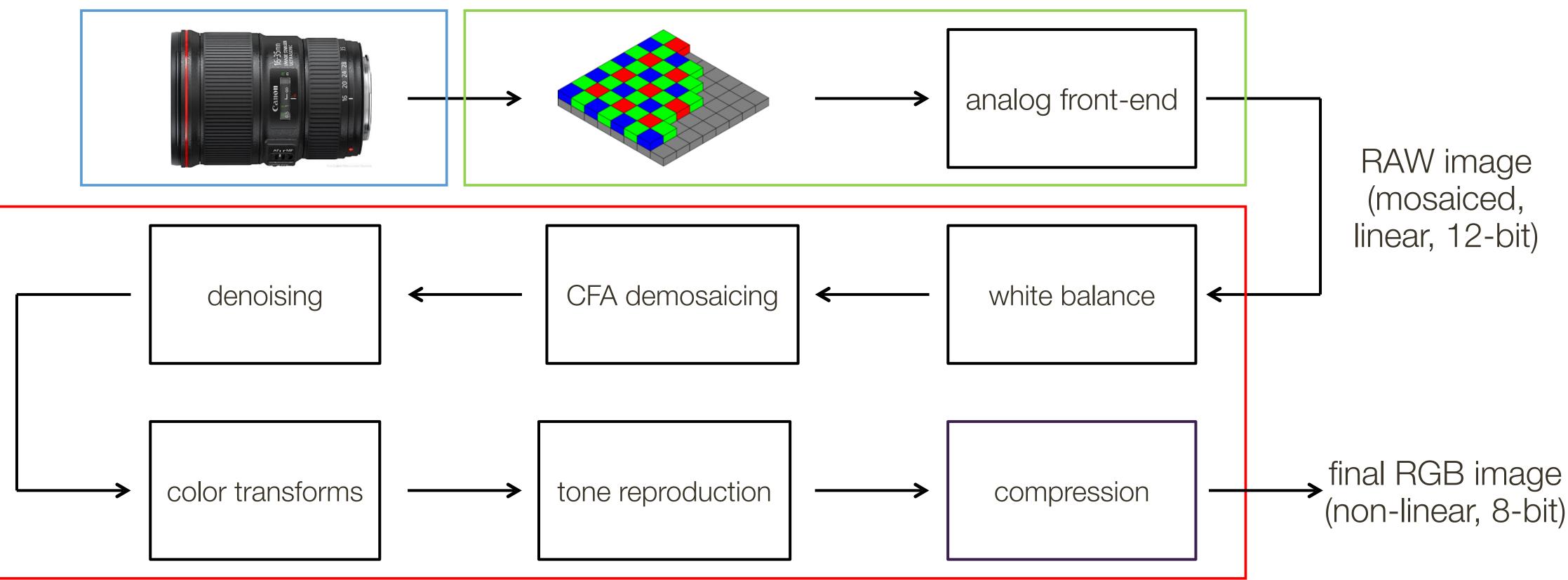






(in camera) Image Processing Pipeline

The sequence of image processing operations applied by the camera's image signal processor (ISP) to convert a RAW image into a "conventional" image.



(in camera) White balance

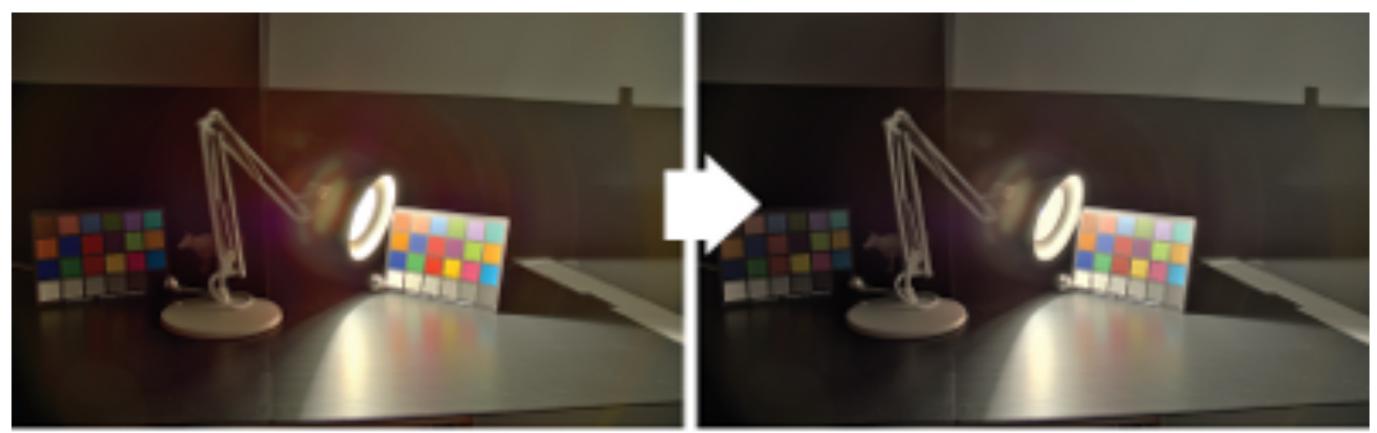


(in camera) White balance



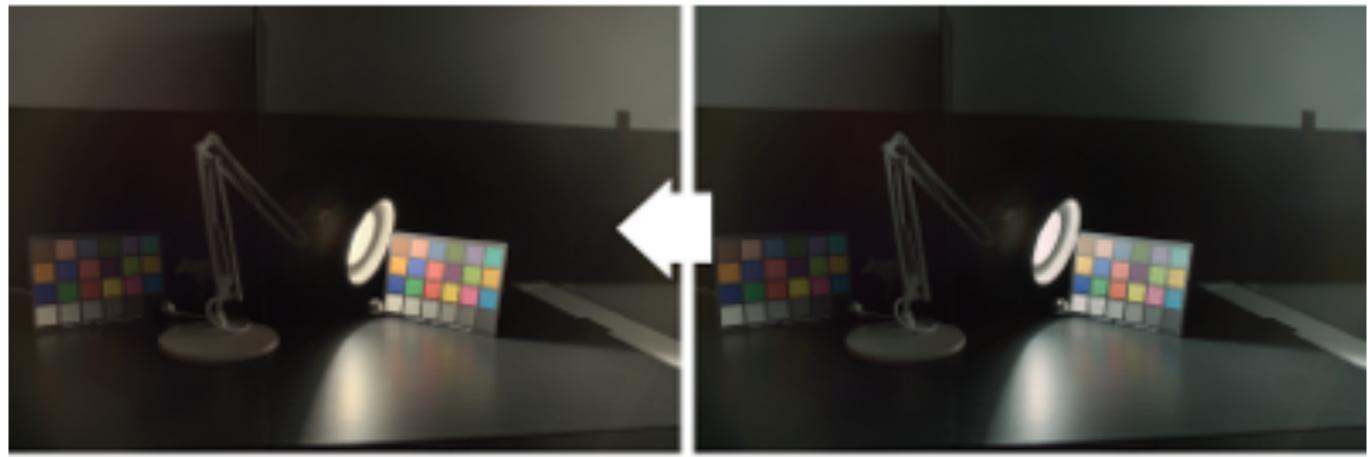
- **R**: 200 **R-correction**: + 55
- **G**: 255 \rightarrow **G-correction**: + 0
- **B-correction**: + 65 **B**: 190

(in camera) **Tone** reproduction



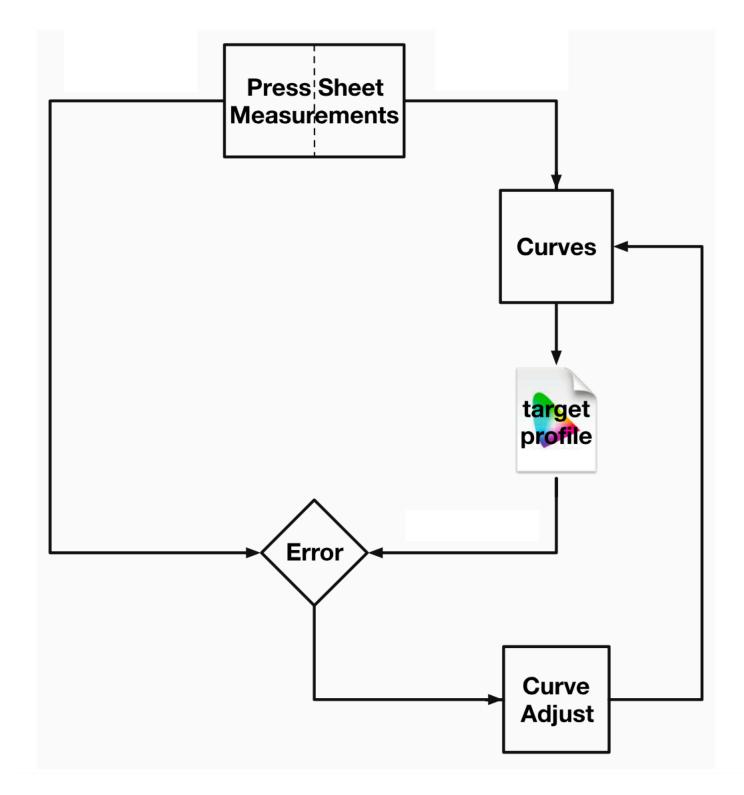
Tonemapped with Li et al. 2005

Corrected saturation reduced



Corrected saturation enhanced

Tonemapped with Reinhard et al. 2012



Summary

"Color" is **not** an objective physical property of light (electromagnetic radiation). Instead, light is characterized by its wavelength.

Color Filter Arrays (CFAs) allow capturing of mosaiced color information; the layout of the mosaic is called **Bayer** pattern.

Demosaicing is the process of taking the RAW image and interpolating missing color pixels per channel

