ESE 5310: Digital Signal Processing

Lecture 1: January 18, 2024

Introduction and Overview

Lecture Outline

- Course Topics Overview
- Learning Objectives
- Course Structure
- Course Policies
- Course Content
- What is DSP?
- DSP Examples

Course Topics Overview

- Discrete-Time (DT) Signals
- Time-Domain Analysis of DT Systems
- Discrete Fourier Transform (DFT)
- Fast Fourier Transform (FFT)
- Discrete-Time Fourier Transform (DTFT)
- z-Transform
- Sampling of Continuous Time Signals
- Data Converters and Modulation
- Upsampling/Downsampling
- Discrete-Time Filter Design
- Special Topics

Learning Objectives

- Learn the fundamentals of digital signal processing
- Provide an understanding of discrete-time signals and systems and digital filters
- Enable you to apply DSP concepts to a wide range of fields
- Gain the ability to read the technical literature on DSP
- Apply the techniques learned in two projects focused on filter design

Learning Objectives

- In other words…

- Math, Math, Math*

*With MATLAB application for intuition

https://cets.seas.upenn.edu/software/matlab/

Course Structure

- TTh Lecture, 3:30-5:00pm in DRLB A8
- Website (http://www.seas.upenn.edu/~ese5310/)

Digital Signal Processing

Course: ESE5310

Units: 1.0 CU
Term: Spring 2024 (all times below are EST)
When: TTh 3:30-5pm
Where: DRLB A8
Instructor: Tania Khanna (Levine 262) (seas: taniak) (office hours: W 3:30-5pm (in person) and by appointment)
TA: Saurabh Parulekar (seas: psaurabh) OH: F time 2-5pm
TA: Noah Schwab (seas: nhschwab) OH: MW 9-11am Towne M70
TA: Meijun Tian (seas: meijunt) OH: TTh 10-12pm location TBD
Graders: Zeyu Gu, Manas Kulkarni

Prerequisites: ESE 3250/2240 or equivalent. Undergraduate students need permission of instructor. Roundup of topics you should be familiar with.

Quick Links: [Course Objectives] [Grading] [Policies] [Spring 2024 Calendar] [Reading] [Ed Discussion]

Catalog Level Description: This course covers the fundamentals of discrete-time signals and systems and digital filters. Specific topics include discrete-time Fourier transform (DTFT); Z-transforms; frequency response of linear discrete-time systems; sampling of continuous time signals, analog to digital conversion, sampling-rate conversion; basic discrete-time filter structures and types; finite impulse response (FIR) and infinite impulse response (IIR) filters; linear phase conditions; design of FIR and IIR filters; discrete Fourier transform (DFT) and the fast Fourier transform (FFT) algorithm. Applications in filtering and spectrum estimation, image filtering, adaptive filters, equalization.

Role and Objectives

Students will:

Course Structure

- TTh Lecture, 3:30-5:00pm in DRLB A8
- Website (http://www.seas.upenn.edu/~ese5310/)
  - Course calendar is used for all handouts (lectures slides, assignments)
  - Canvas/Gradescope used for lecture videos, assignment submission, and grades
  - Ed Discussion used for announcements and discussions

Course Structure

- Course Staff (complete info on course website)
- Instructor: Tania Khanna
  - In-person Office hours – Wednesday 3:30-5pm or by appointment
  - Email: taniak@seas.upenn.edu
    - Best way to reach me
- TAs: Saurabh Parulekar, Noah Schwab, Meijun Tian
  - Office Hours – see course website, some locations TBD
- Graders: Zeyu, Manas
- Ed discussions for questions

TA Introduction: Saurabh Parulekar

- 2nd year master's student in Electrical Engineering
- Took this class in Spring 2023
- Worked for Tesla as System Validation Engineer intern
- Was a TA for "Introduction to Python", so feel free to reach out if you have questions related to python language.
- Pennkey: psaurabh

TA Introduction: Noah Schwab

- Took this course in Spring 2022
- 2nd year EE Master's Student, Information and Decision Systems
- Main interests are DSP and ML
- Worked on Autopilot Algorithm Validation at Tesla
- Working at MIT Lincoln Lab after graduation
- Currently doing research on adapting generative AI models to image compression
- Hobbies: soccer, running, skiing

A picture of me at Yosemite Falls in May 2023

TA Introduction: Meijun Tian

- Took ESE 5310 in Spring 2023
- Second-year EE master's student
- Worked as a research assistant in the High-Speed Integrated Circuits Lab @ PSU
- Planning to pursue a Ph.D. focus on RFIC
- I like traveling and baking!

Course Structure

- Lectures
  - Statistically speaking, you will do better if you come to lecture
  - Better if interactive, **everyone** engaged
    - Asking and answering questions
    - Actively thinking about material
- Recitation
  - Videos from previous year posted
  - Work out example problems and review for exam
- Textbook
  - A. V. Oppenheim and R. W. Schafer (with J. R. Buck), Discrete-Time Signal Processing. 3rd. Edition, Prentice-Hall, 2010
  - Homework from text
  - Class will follow text structure… mostly

Course Structure - Assignments/Exams

- Homework – one week long (8 total) [25%]
  - Due Sundays/Tuesdays at midnight
  - Combination of book problems and Matlab problems
- Projects – two projects [30%]
  - Work individually or in team
  - Different DSP applications of filter/system design
- Midterm exam [20%]
- Final exam [25%]

Course Policies

See web page for full details

- Turn-in homework in Canvas/Gradescope
  - Single PDF
  - Anything handwritten/drawn must be clearly legible
  - Submit code, graphs, test results when specified
- Individual work unless specified
  - code, test simulations, analysis, writeups
  - May discuss strategies, but acknowledge help
- Late assignments
  - 1 day late = -15ps, 2 days late = -25pts

Course Content

- Introduction
- Discrete Time Signals & Systems
- Discrete Time Fourier Transform
- Z-Transform
- Inverse Z-Transform
- Sampling of Continuous Time Signals
- Frequency Domain of Discrete Time Series
- Downsampling/Upsampling
- Data Converters, Sigma Delta Modulation
- Frequency Response of LTI Systems
- Signal Flow Representation
- Basic Structures for IIR and FIR Systems
- Design of IIR and FIR Filters
- Butterworth, Chebyshev, and Elliptic Filters
- Filter Banks
- Adaptive Filters
- Computation of the Discrete Fourier Transform
- Fast Fourier Transform

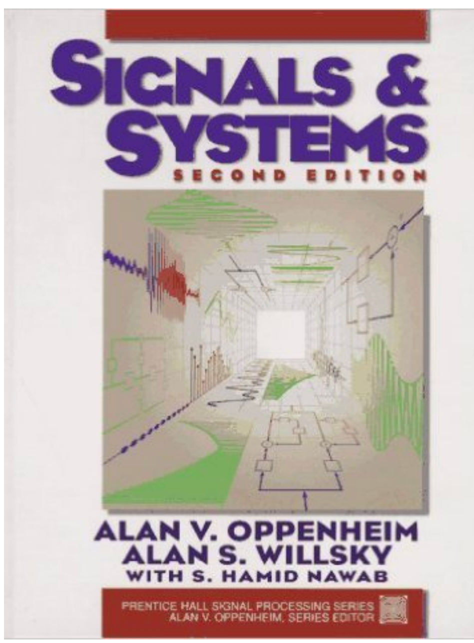
Course Content

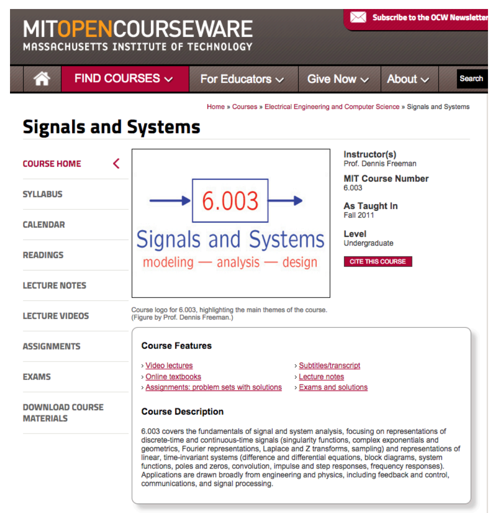
ESE5310 Spring 2024 Working Schedule

Wk	Lect.	Date		Lecture	Slides	Due	Reading
1	1	1/18	Th	Intro/Overview	[lec1]		review course webpage completely
2	2	1/23	T	Discrete Time Signals & Systems, Part 1			2.1-2.2
	3	1/25	Th	Discrete Time Signals & Systems, Part 2		HW0, Diagnostics Quiz	2.3-2.4
3	4	1/30	T	Discrete Time Fourier Transform			2.5-2.7
	5	2/1	Th	Z-Transform			3.0-3.1
		2/4	Su			HW 1	
4	6	2/6	T	Inverse Z-Transform			3.3
	7	2/8	Th	Sampling and Reconstruction			4.0-4.3
		2/11	Su			HW 2	
5	8	2/13	T	DT/CT Processing of CT/DT Signals, Impulse Invariance			4.4-4.5
	9	2/15	Th	Downsampling/Upsampling and Practical Interpolation			4.6-4.6.3
		2/18	Su			HW 3	
6	10	2/20	T	Non-Integer and Multi-rate Sampling			4.6.4-4.7.2
	11	2/22	Th	Polyphase Decomposition and Multi-rate Filter Banks			4.7
		2/25	Su			HW 4	
7	12	2/27	T	Data Converters and Noise Shaping			4.8-4.9
		2/29	Th	No Class			
8		3/5	T	SPRING BREAK -- no class			
		3/7	Th	SPRING BREAK -- no class			
		3/10	Su			HW 5	

Signals and Systems Review

- Diagnostic Quiz in Canvas
  - Complete by Thursday 1/25 for full credit
  - Review in HW 0
    - Complex math review
    - https://www.seas.upenn.edu/~ese5310/spring2024/knowledge_roundup.html
    - Matlab Tutorial





Signals and Systems Review

- Diagnostic Quiz in Canvas
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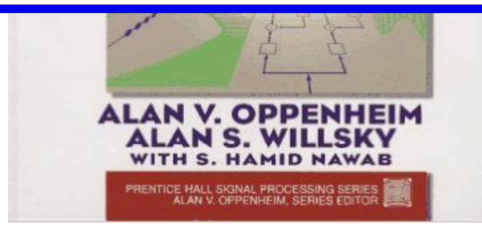
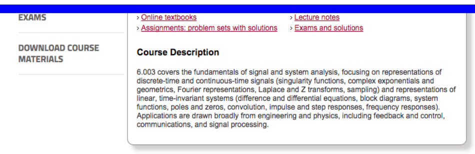
  - Re
  - lge_roundup.html

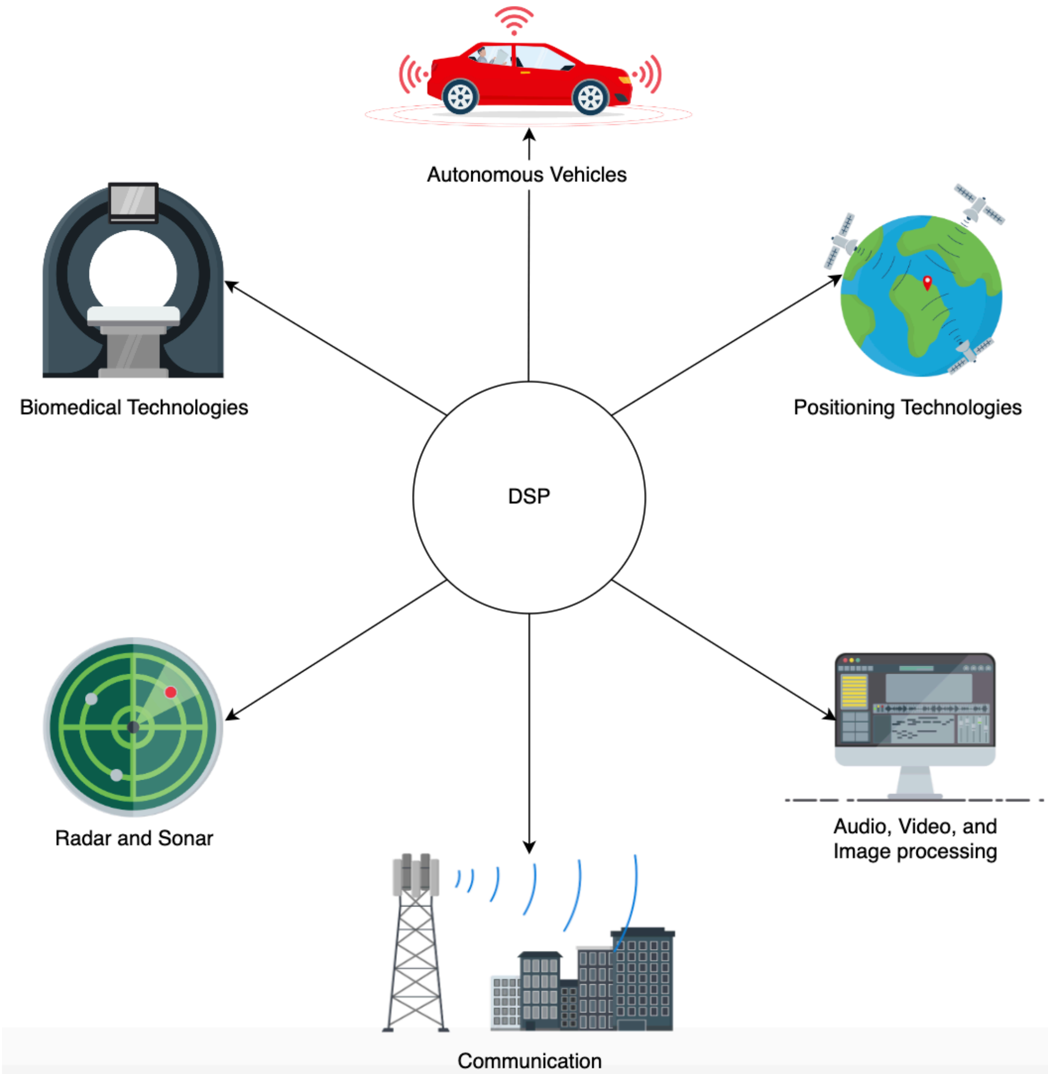
“I would strongly encourage students to go through the provided review material, regardless of their familiarity, as I remember it really helped prepare me for the first few weeks of the course.”





What is DSP

DSP is Everywhere



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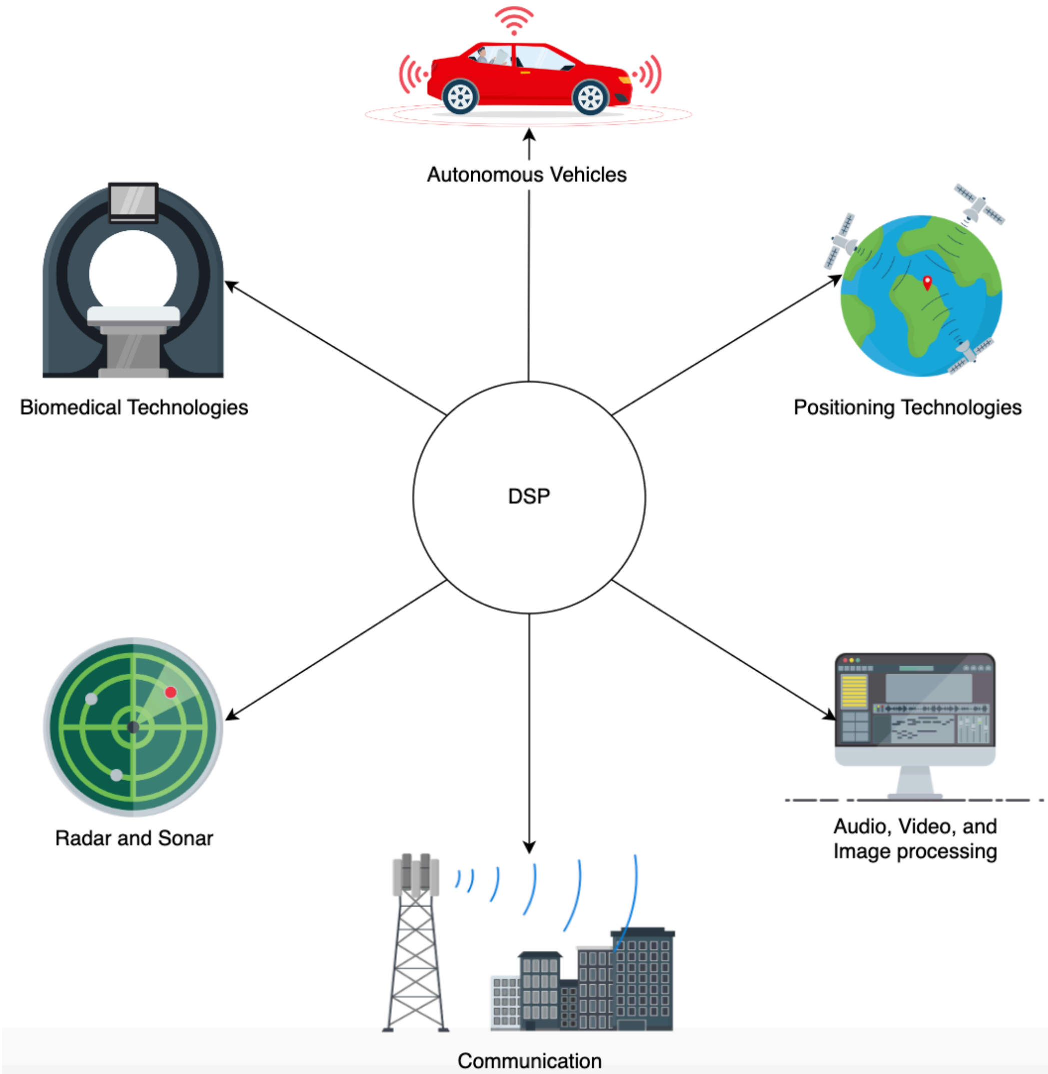
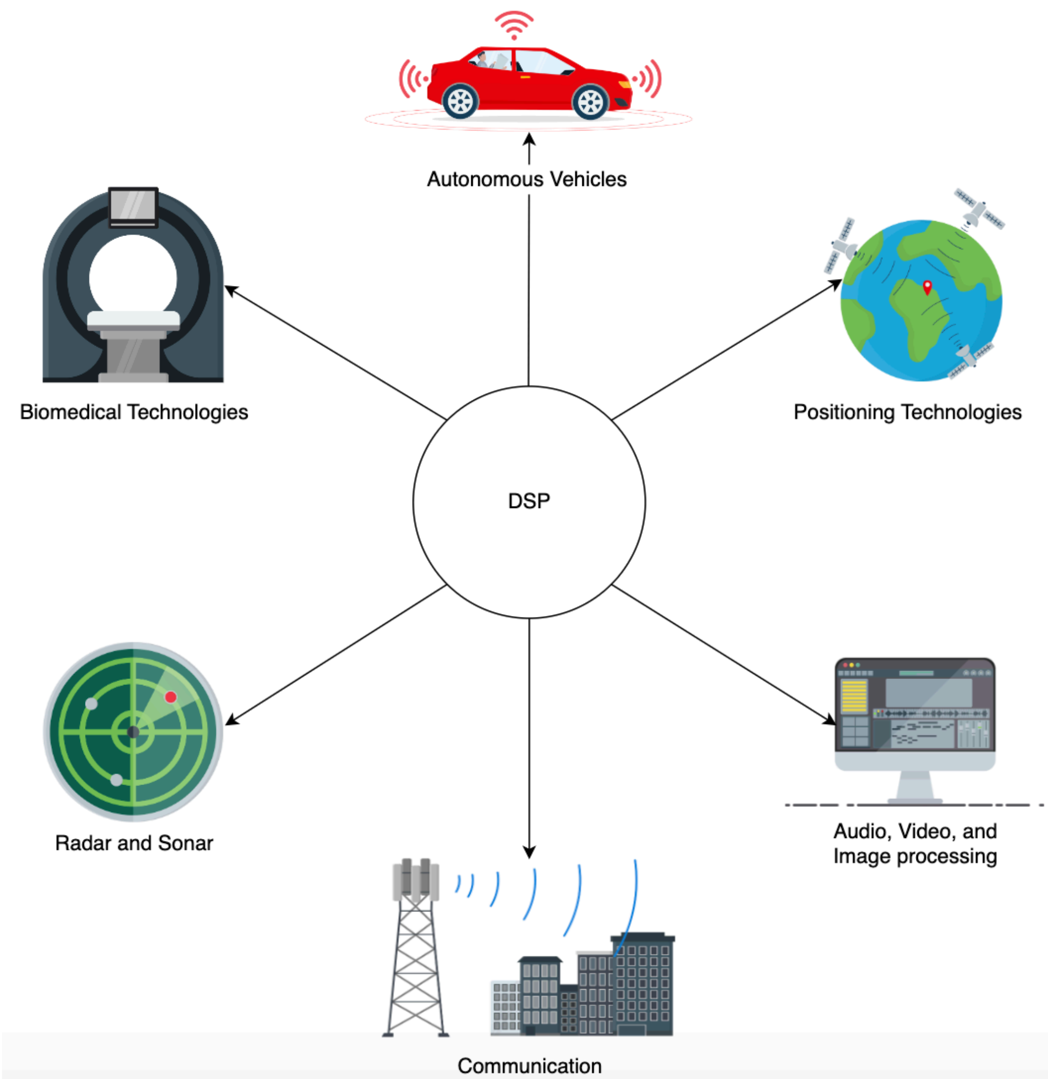


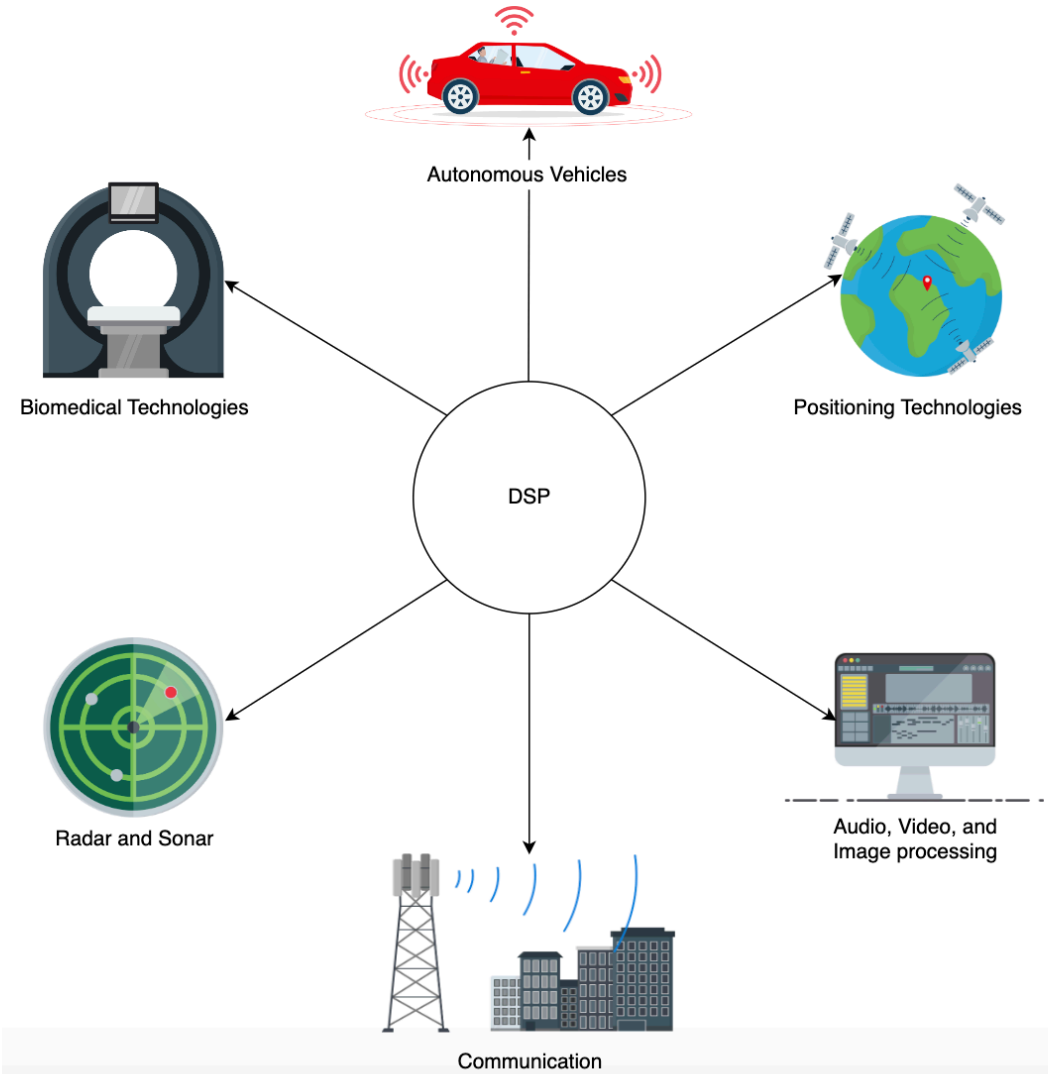
- Automotive
  - ABS, GPS, Active Noise Cancellation, Cruise Control, Parking, Driverless Cars

DSP is Everywhere



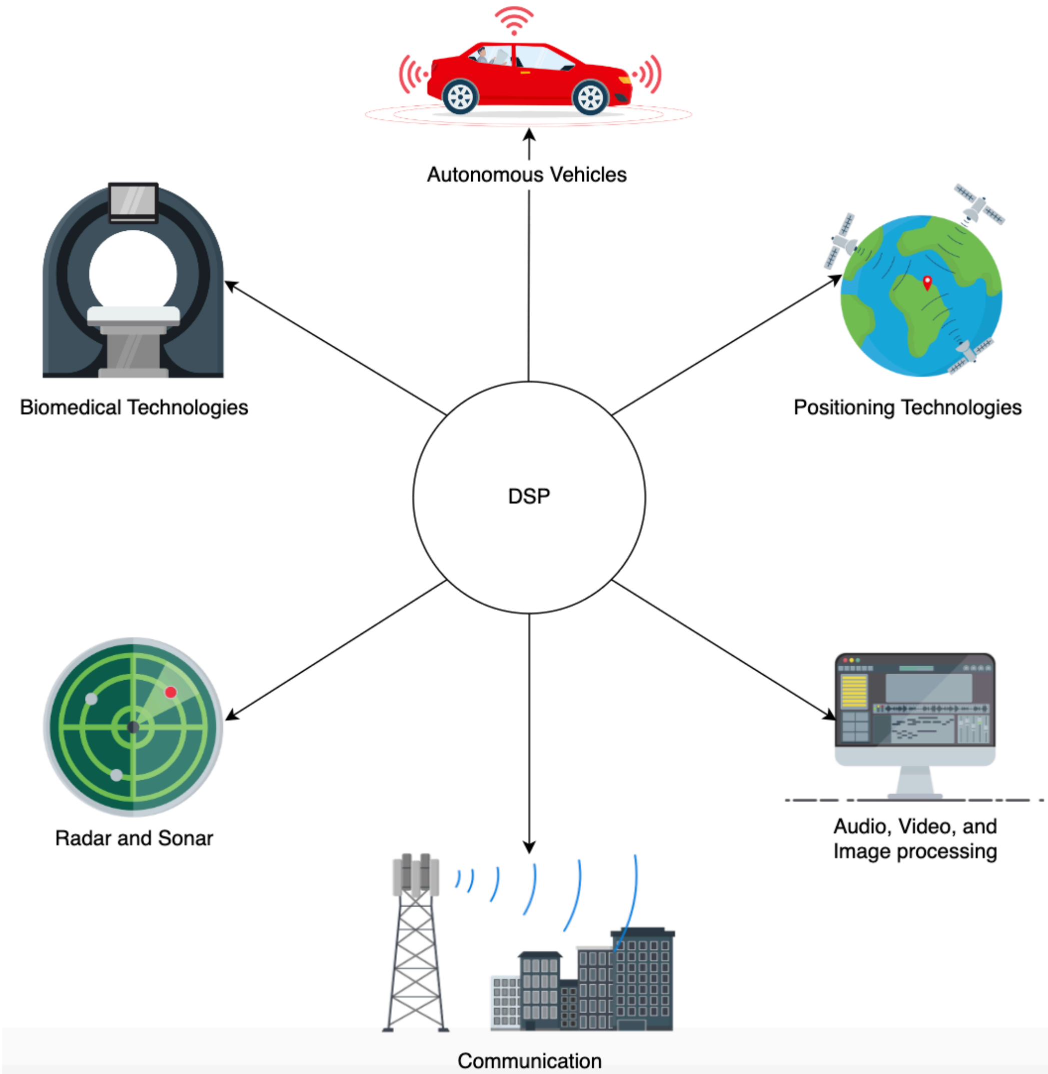
- Sound applications
  - Compression, enhancement, special effects, synthesis, recognition, echo cancellation
  - Cell phones, MP3 players, movies, dictation, text-to-speech
- Image and Video Applications
  - DVD, JPEG, Movie special effects, video conferencing

DSP is Everywhere



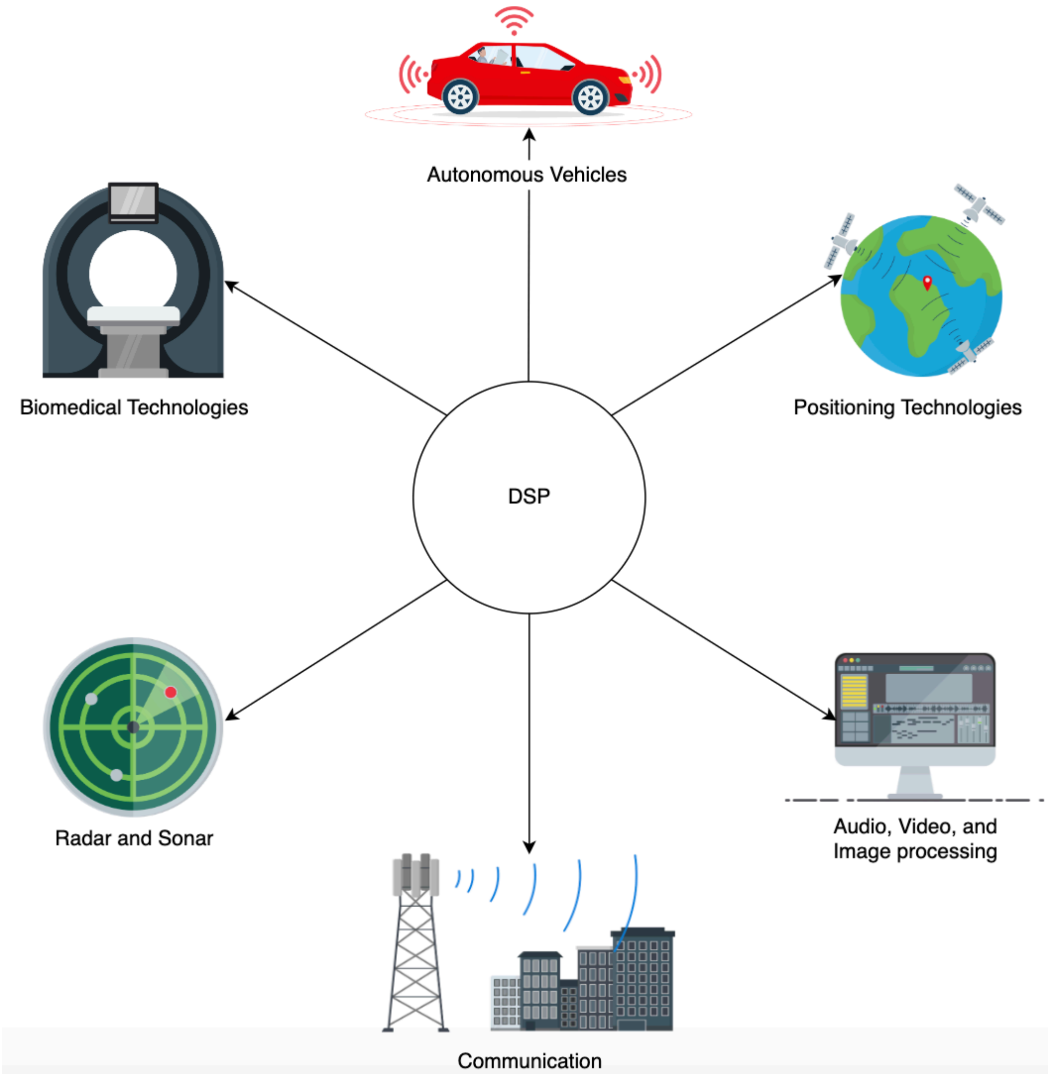
- Communication
  - Modulation, coding, detection, equalization, echo cancellation
  - Cell Phones, dial-up modem, DSL modem, Satellite Receiver

DSP is Everywhere



- Military
  - Radar, Sonar, Space photographs, remote sensing

DSP is Everywhere



- Medical
  - Magnetic Resonance, Tomography, Electrocardiogram, Biometric Monitoring

Signal Processing

- Humans are the most advanced signal processors
  - speech and pattern recognition, speech synthesis
- We encounter many types of signals in various applications
  - Electrical signals: voltage, current, magnetic and electric fields
  - Mechanical signals: velocity, force, displacement
  - Acoustic signals: sound, vibration
  - Other signals: pressure, temperature, biometrics
- Most real-world signals are analog
  - They are continuous in time and amplitude
  - Convert to voltage or currents using sensors and transducers

Signal Processing (con't)

- Analog circuits process these signals using
  - Resistors, Capacitors, Inductors, Amplifiers
- Analog signal processing examples
  - Audio processing in FM radios
  - High end stereo equipment
  - Video processing in traditional TV sets
  - Biopotential amplifiers

Limitations of Analog Signal Processing

- Accuracy limitations due to
  - Component tolerances
  - Undesired nonlinearities
- Limited repeatability due to
  - Tolerances
  - Changes in environmental conditions
    - Temperature
    - Vibration
- Sensitivity to electrical noise
- Limited dynamic range for voltage and currents
- Inflexibility to changes
- Difficulty of implementing certain operations
  - Nonlinear operations
  - Time-varying operations
- Difficulty of storing information

Digital Signal Processing

- Represent signals by a sequence of numbers
  - Sampling and quantization (or analog-to-digital conversion)
- Perform processing on these numbers with a digital processor
  - Digital signal processing
- Reconstruct analog signal from processed numbers
  - Reconstruction or digital-to-analog conversion

- Analog input → analog output
  - Eg. Digital recording music
- Analog input → digital output
  - Eg. Speech to text
- Digital input → analog output
  - Eg. Text to speech
- Digital input → digital output
  - Eg. Compression of a file on computer

Pros and Cons of Digital Signal Processing

- Pros
  - Accuracy can be controlled by choosing word length
  - Repeatable
  - Sensitivity to electrical noise is minimal
  - Dynamic range can be controlled using floating point numbers
  - Flexibility can be achieved with software implementations
  - Non-linear and time-varying operations are easier to implement
  - Digital storage is cheap
  - Digital information can be encrypted for security
  - Price/performance and reduced time-to-market

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- Cons
  - Sampling causes loss of information
  - Quantization and round-off errors
  - A/D and D/A requires mixed-signal hardware
  - Limited by speed of processors

DSP Examples

Example I: Audio Compression

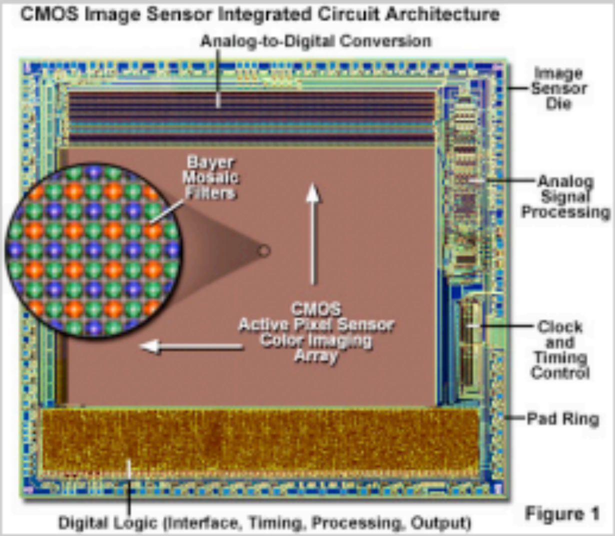
- Compress audio by 10x without perceptual loss of quality
- Sophisticated processing based on models of human perception
- 3MB files instead of 30MB
  - Entire industry changed in less than 10 years!

Historical Forms of Compression

- Morse code: dots (1 unit) dashes (3 units)
  - Code Length inversely proportional to frequency of character
    - E (12.7%) = . (1 unit) Q (0.1%) = --.- (10 units)

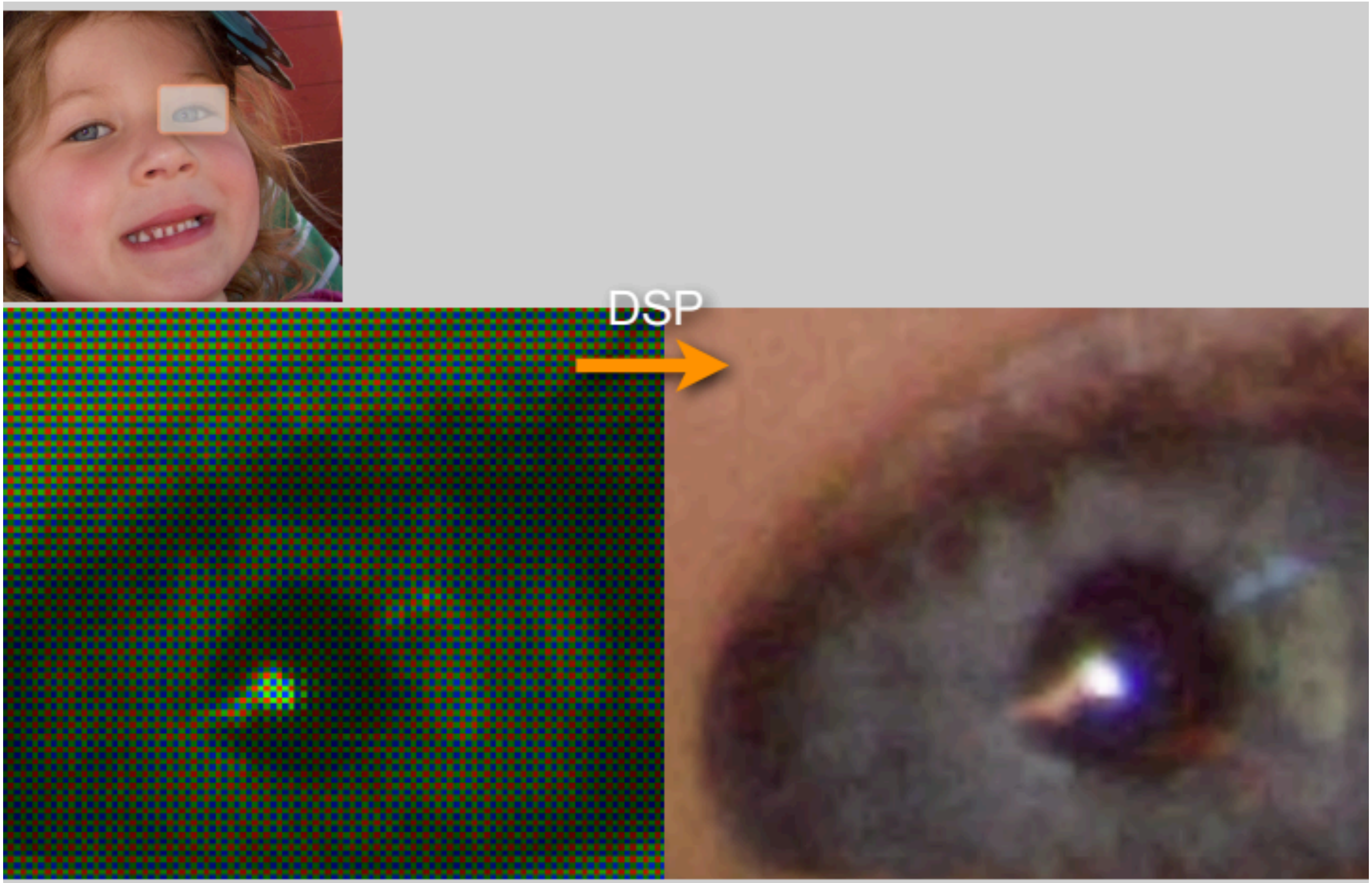
- “92 Code”
  - Used by Western-Union in 1859 to reduce BW on telegraph lines by numerical codes for frequently used phrases
    - 1 = wait a minute
    - 73 = Best Regards
    - 88 = Love and Kisses

Example II: Digital Imaging Camera



Focus/exposure Control → preprocessing → white-balancing → demosaic → Color transform → Post-processing → Compression

Example II: Digital Imaging Camera



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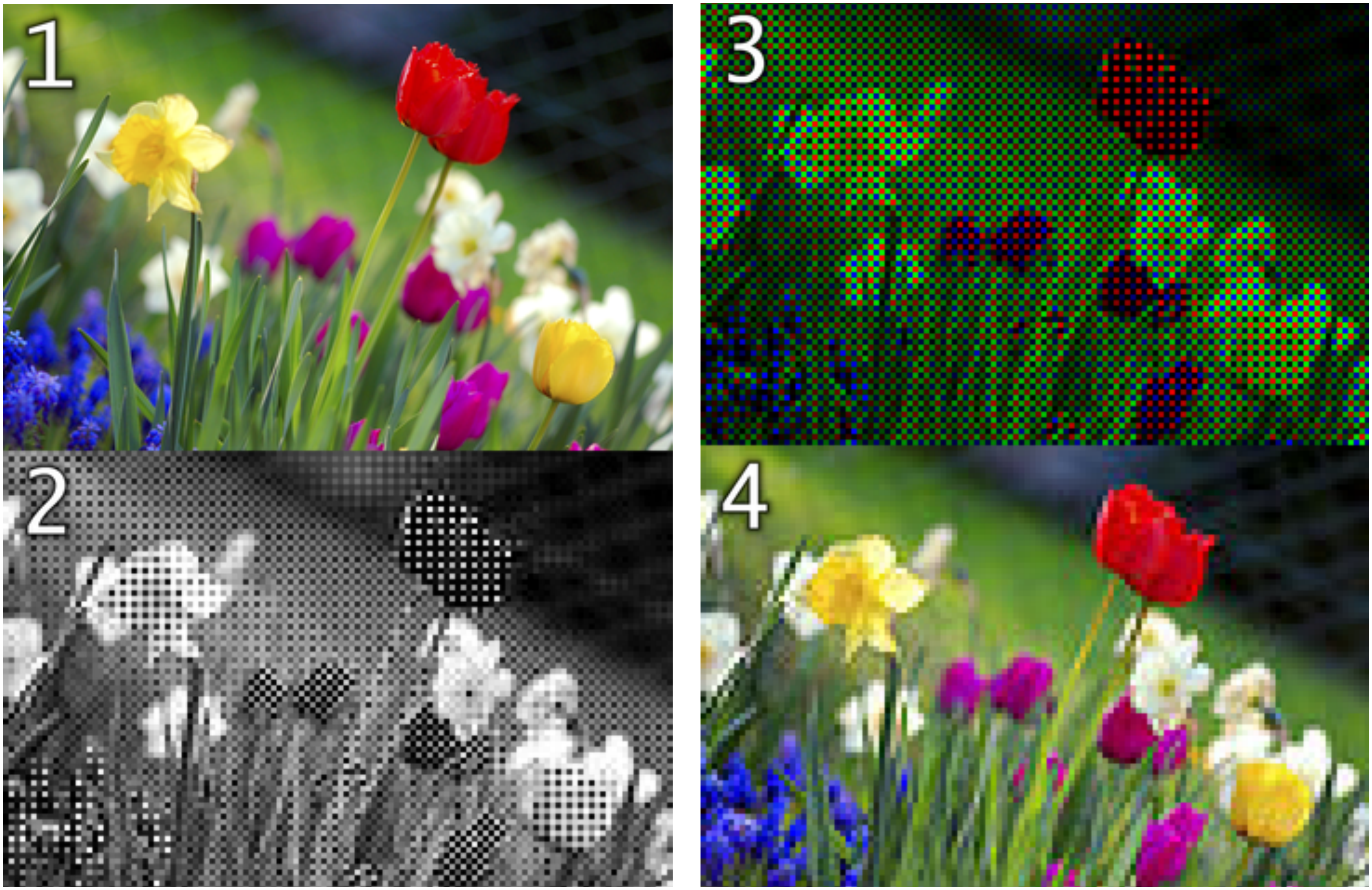

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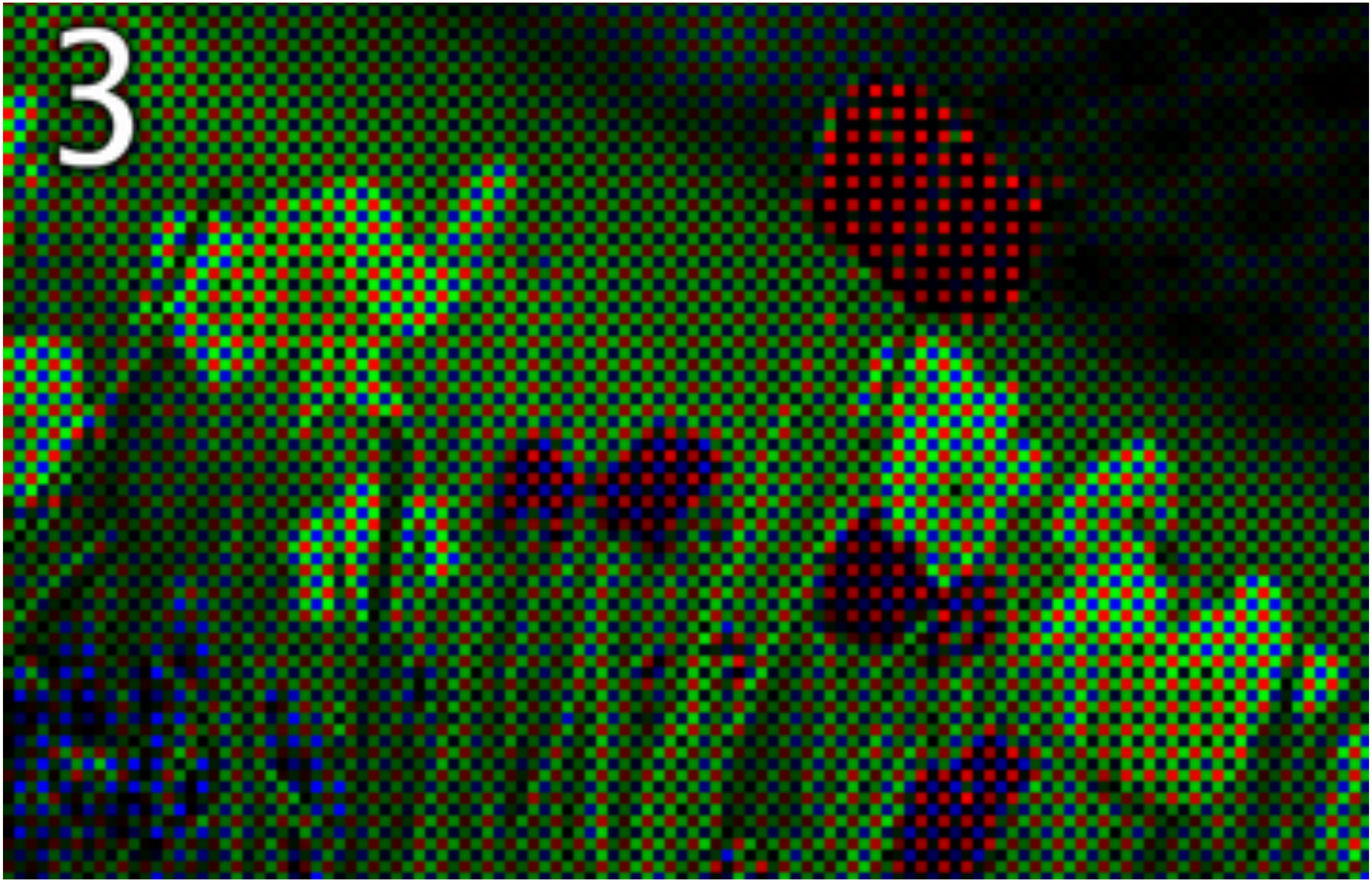

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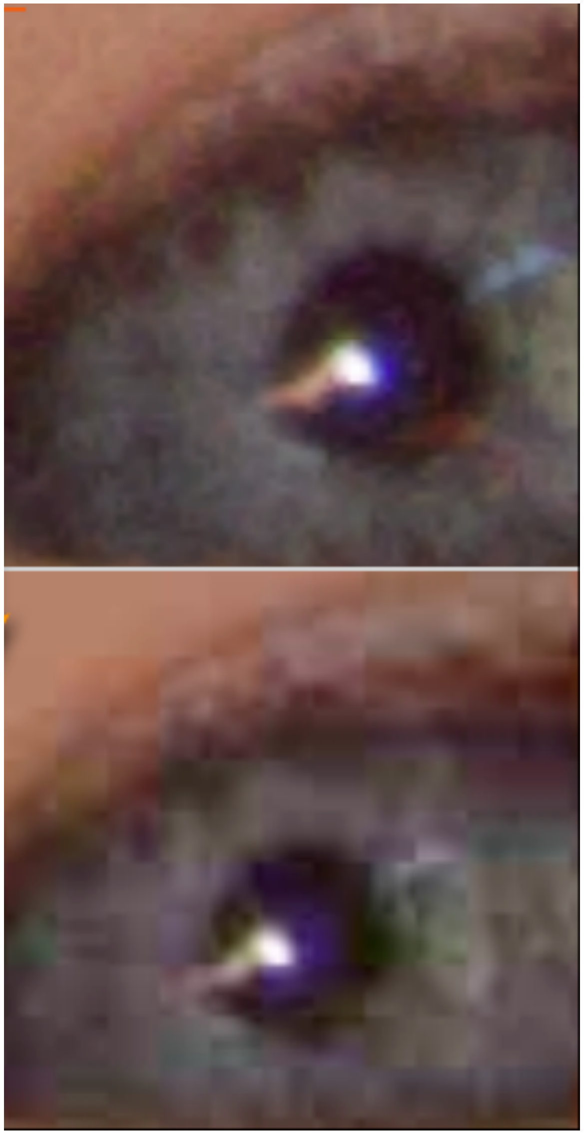


- Compression of 40x without perceptual loss of quality.
- Example of slight over compression: difference enables 60x compression!

Computational Photography

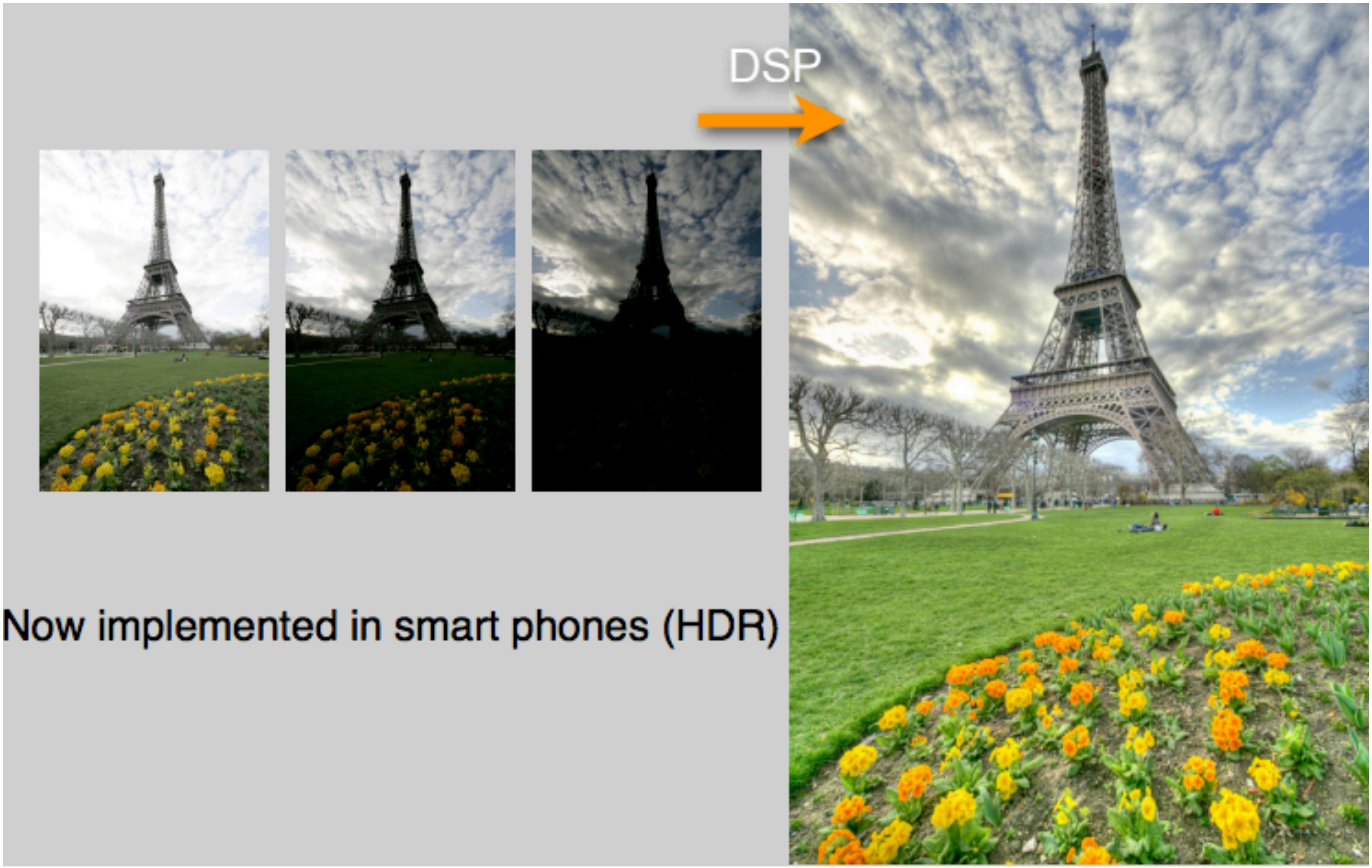


Image Processing

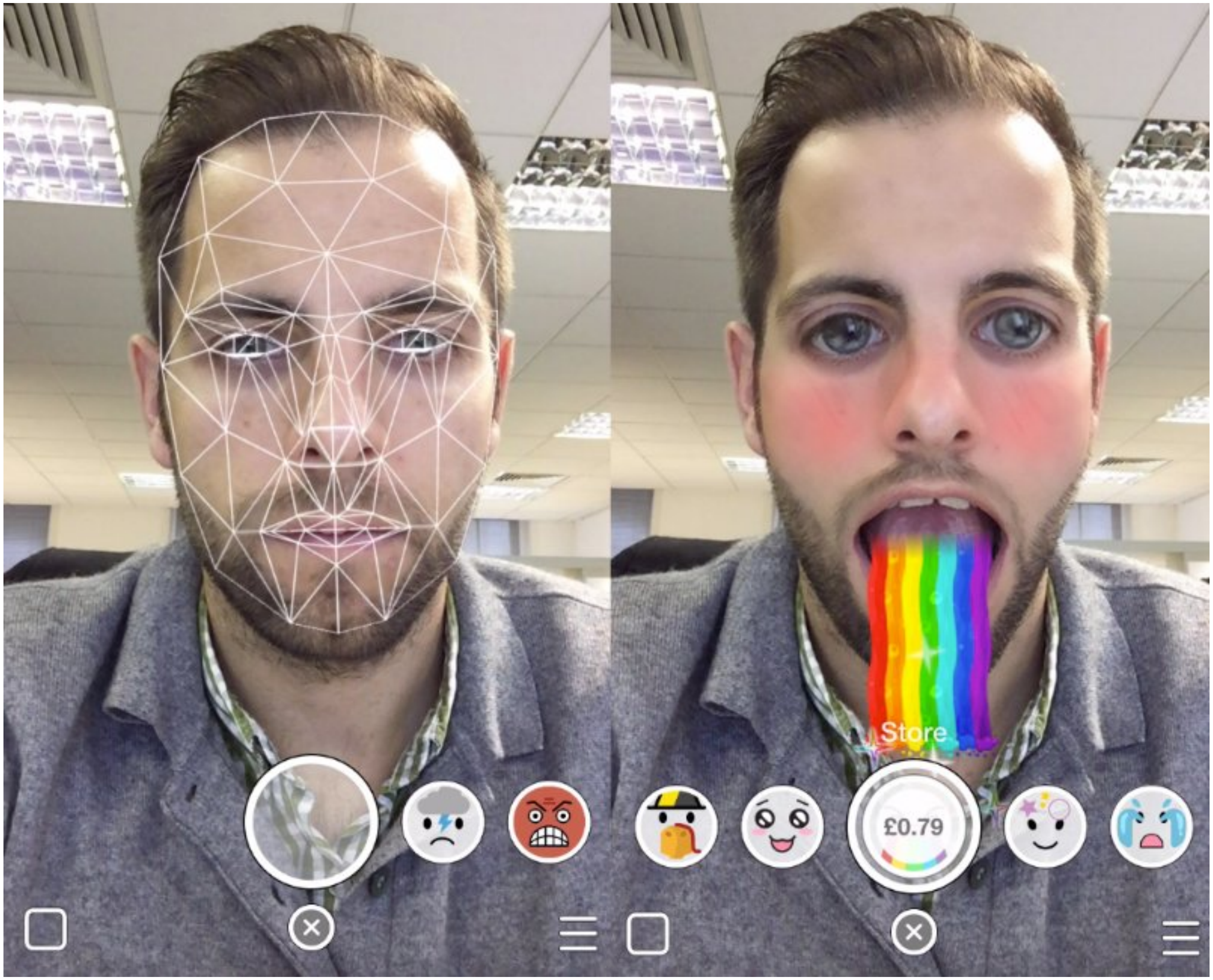


Image Processing - Saves Lives

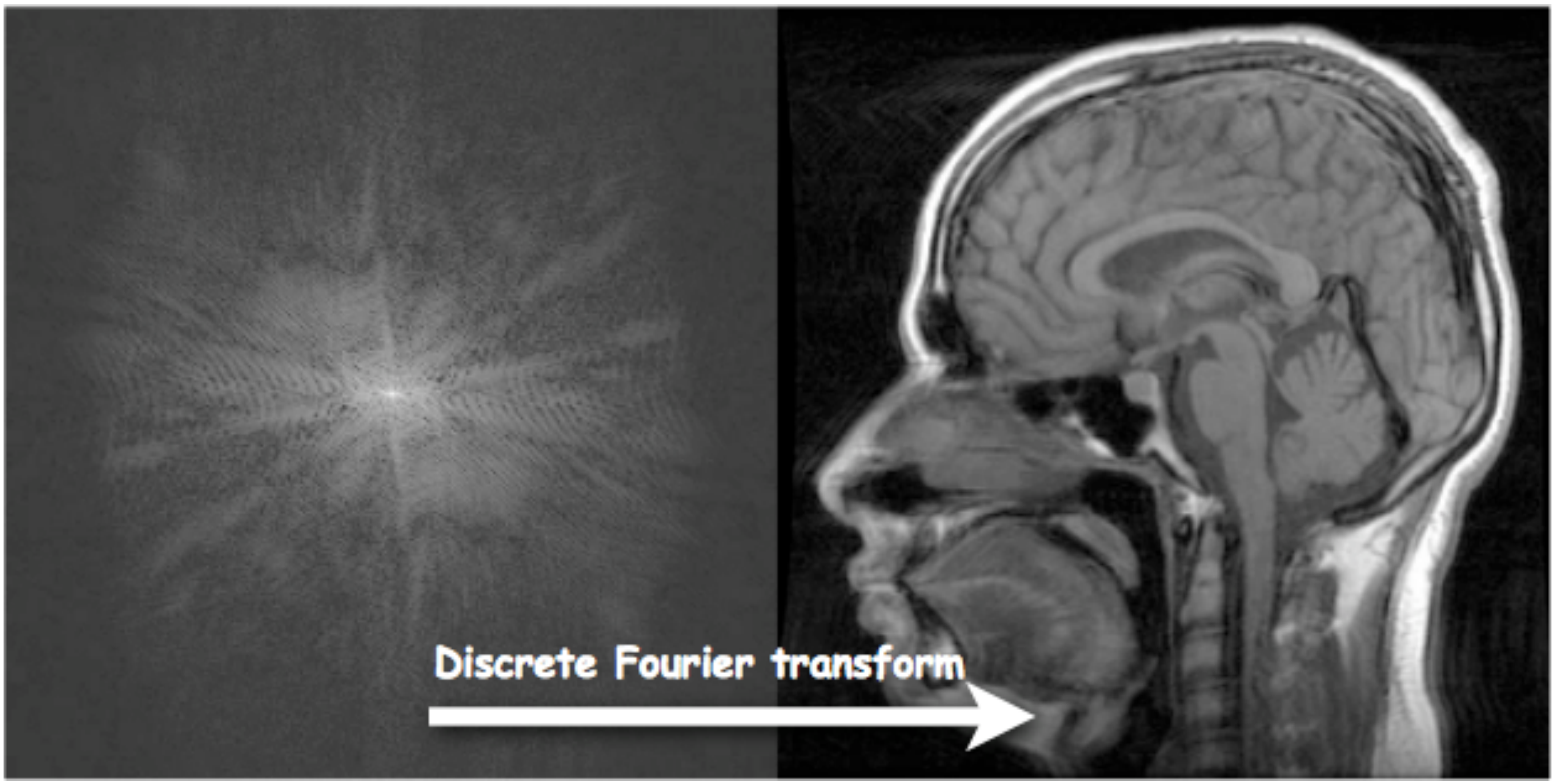
Canadian 'swirl face' jailed in Thailand

August 15, 2008

Images released by Interpol in 2007 show the 'unswirling' of the internet pictures that led to the capture of Christopher Paul Neil.

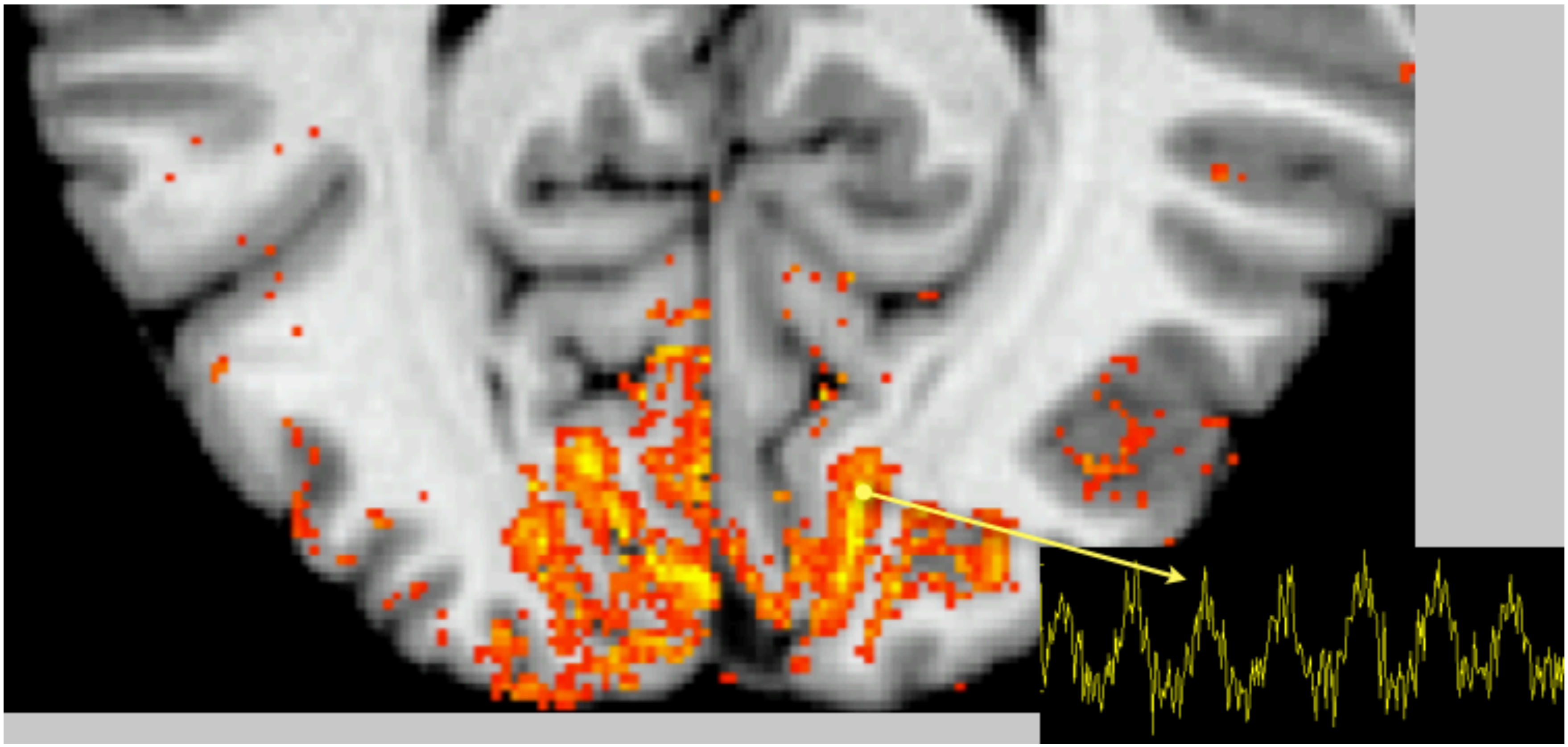
Example III: MRI

k-space (raw data) Image



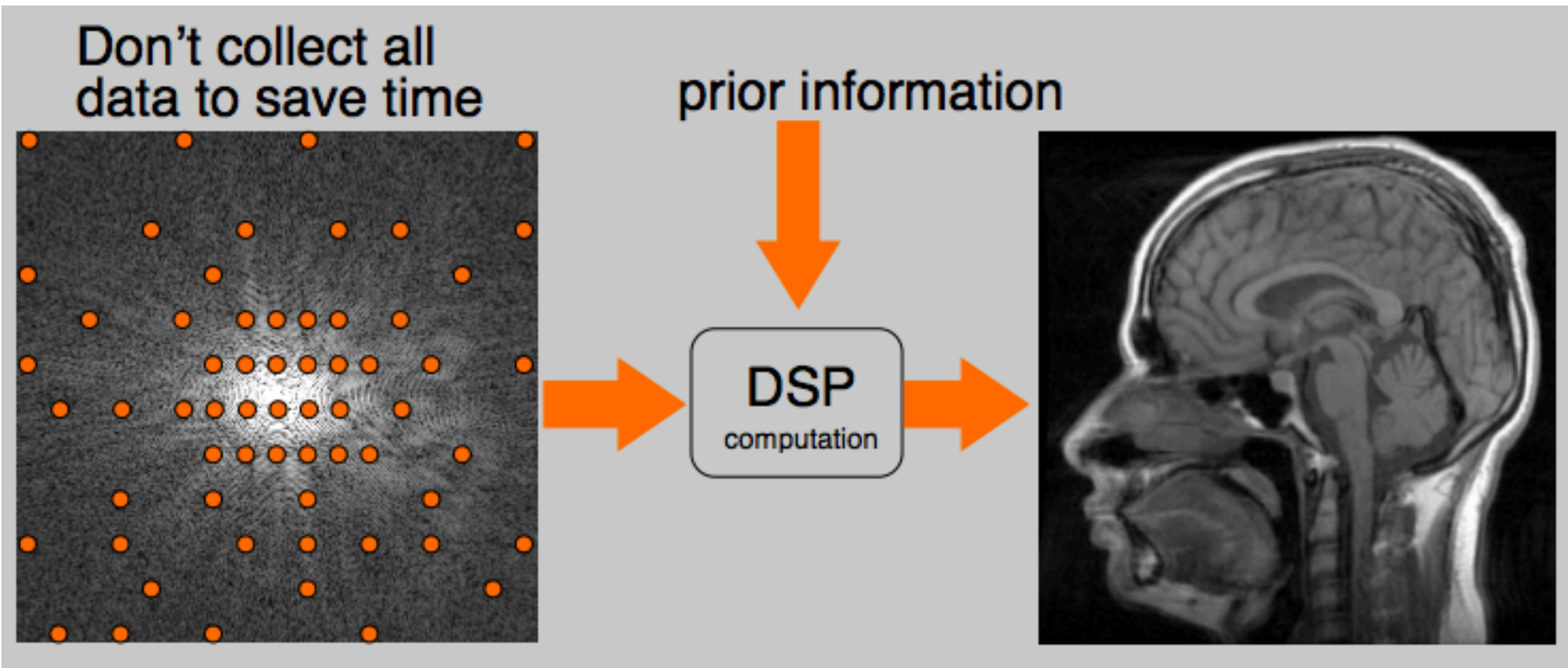
fMRI example

- Sensitivity to blood oxygenation
  - response to brain activity Convert from one signal to another

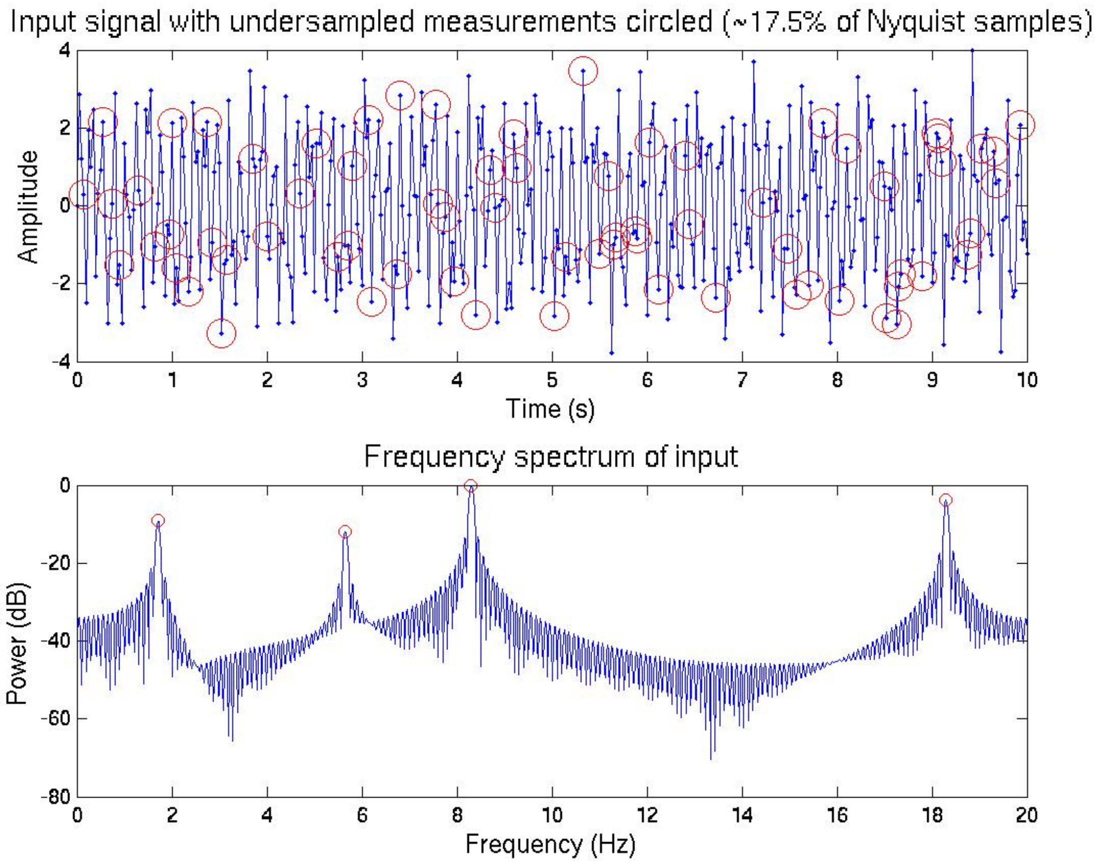


Compressive Sampling

- Compression meets sampling



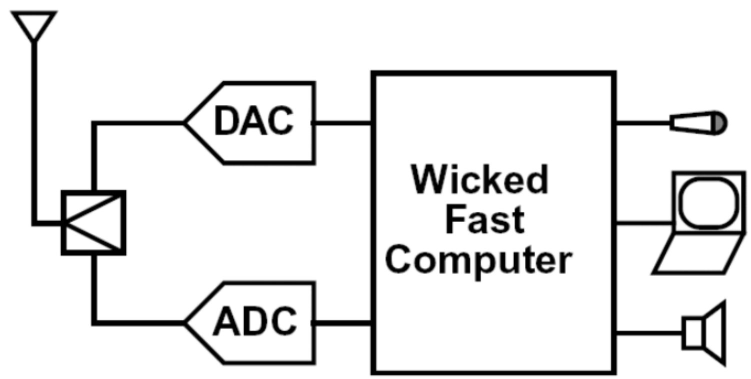
Example: Sum of Sinusoids



- Sense signal randomly M times
  - M > C·μ2(Φ,Ψ)·S·log N
- Recover with linear program

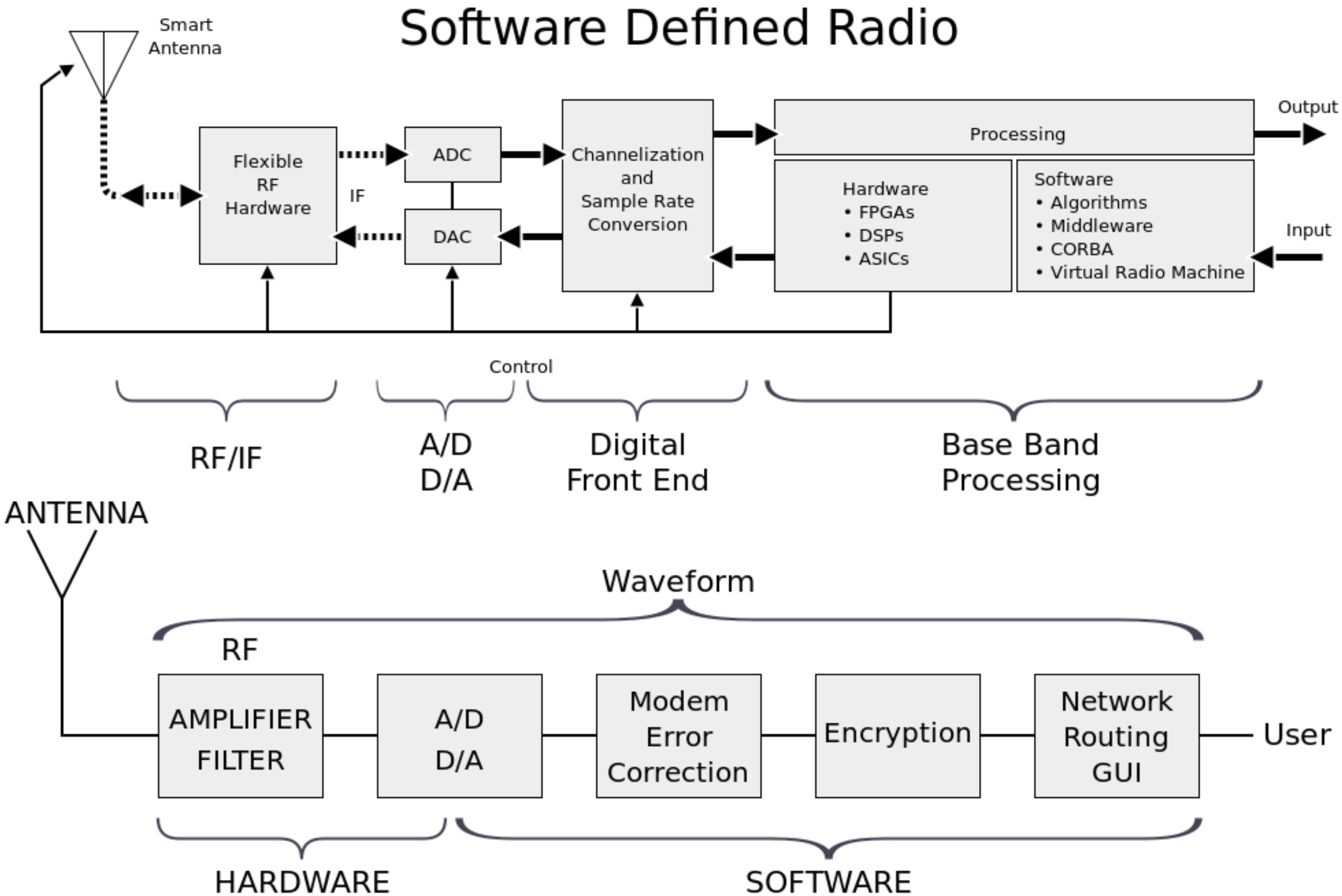
Example IV: Software Defined Radio

- Traditional radio:
  - Hardware receiver/mixers/demodulators/filtering
  - Outputs analog signals or digital bits
- Software Defined Radio:
  - Uses RF front end for baseband signal
  - High speed ADC digitizes samples
  - All processing chain done in software



[Schreier, "ADCs and DACs: Marching Towards the Antenna," GIRAFE workshop, ISSCC 2003]

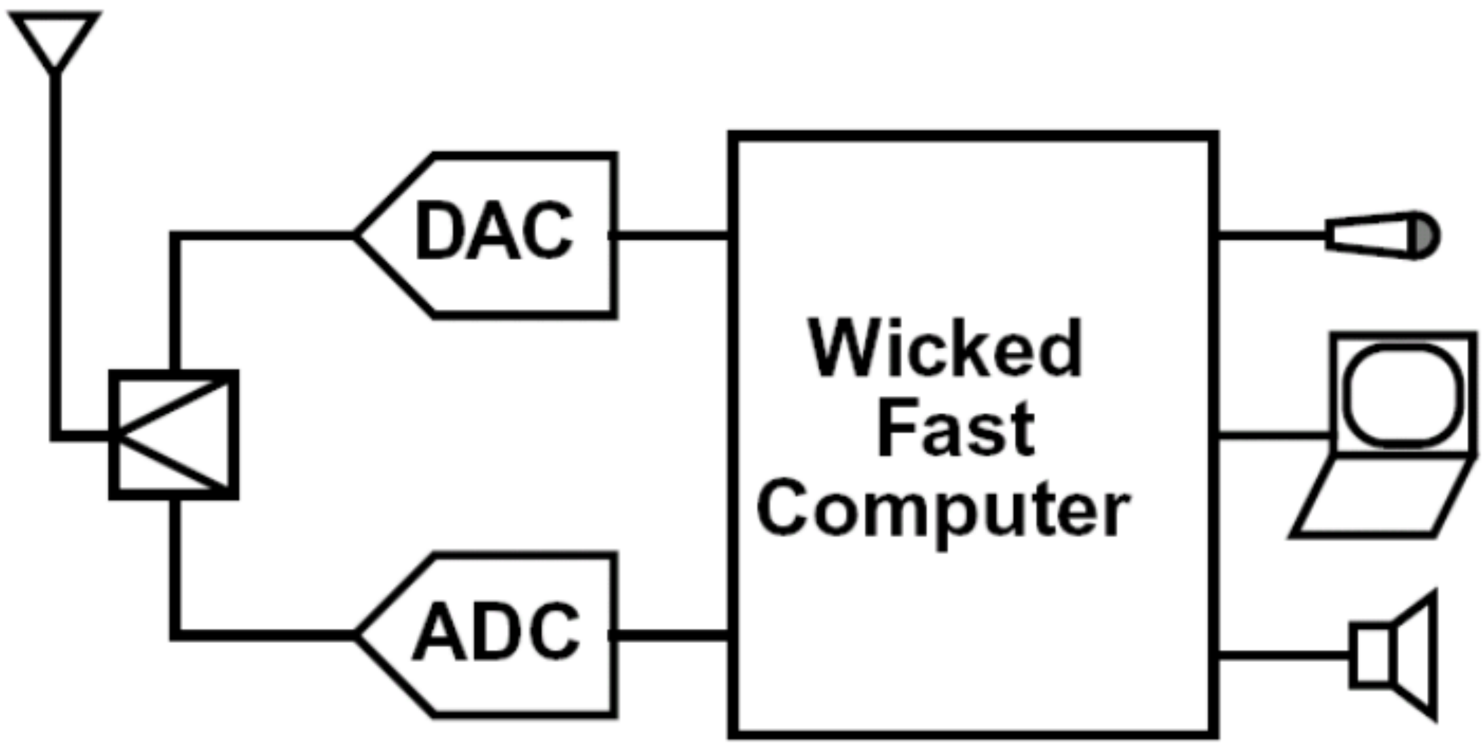
Software Defined Radio



Software Defined Radio

- Advantages:
  - Flexibility
  - Upgradable
  - Sophisticated processing
  - Ideal processing chain
  - Not approximate like in analog hardware
- Already used in consumer electronics
  - Cellphone baseband processors
  - Wifi, GPS, etc....

Software Radio Vision



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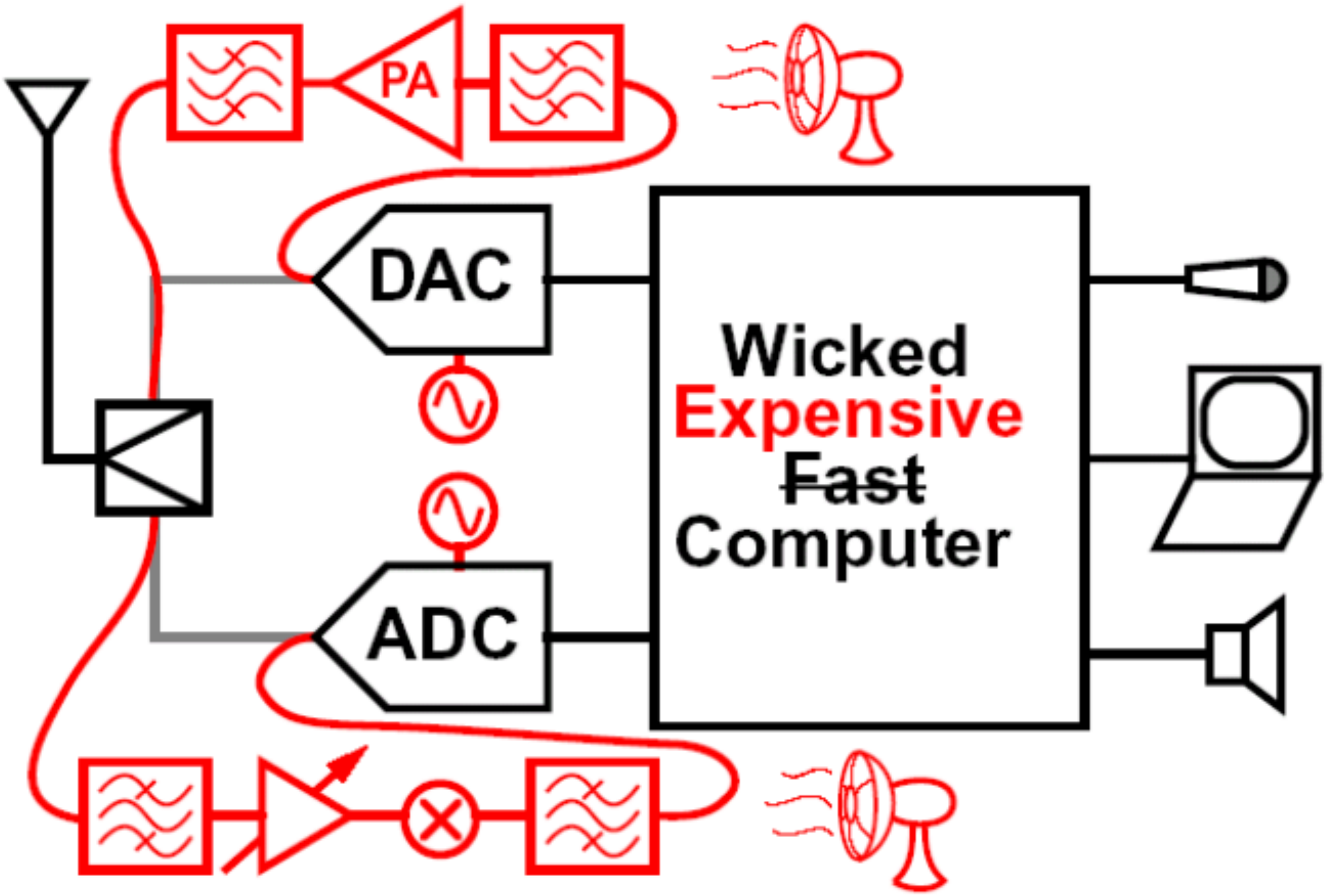
Software Radio Reality



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Future of ADC design

- Today's ADCs are extremely well optimized
- For non-incremental improvements, we must explore new ideas in signal processing that tackle ADC inefficiency at the system level
  - Manufacturing innovation
  - Compressed sensing
  - Finite innovation rate sampling
  - Other ideas?

Filter Design Example

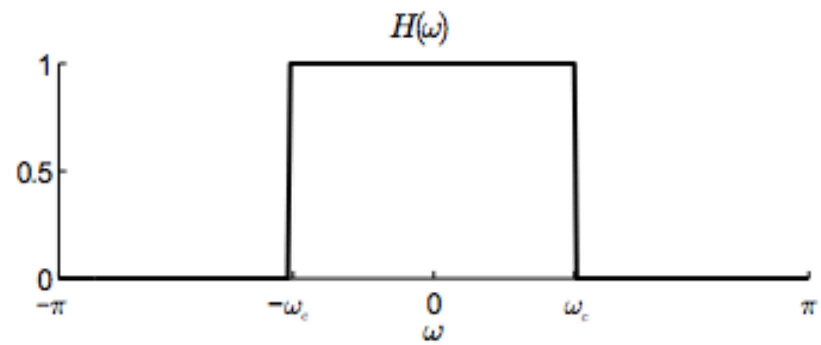
Optimal Filter Design

- Window method
  - Design Filters heuristically using windowed sinc functions
- Optimal design
  - Design a filter h[n] with $H(e^{j\omega})$
  - Approximate $H_d(e^{j\omega})$ with some optimality criteria - or satisfies specs.

Example: Ideal Low-Pass Filter

- The frequency response $H(\omega)$ of the ideal low-pass filter passes low frequencies (near $\omega = 0$) but blocks high frequencies (near $\omega = \pm\pi$)

$$H(\omega) = \begin{cases} 1 & -\omega_c \le \omega \le \omega_c \\ 0 & \text{otherwise} \end{cases}$$



- Compute the impulse response $h[n]$ given this $H(\omega)$
- Apply the inverse DTFT

$$h[n] = \int_{-\pi}^{\pi} H(\omega)\, e^{j\omega n} \frac{d\omega}{2\pi} = \int_{-\omega_c}^{\omega_c} e^{j\omega n} \frac{d\omega}{2\pi} = \left.\frac{e^{j\omega n}}{2\pi j n}\right|_{-\omega_c}^{\omega_c} = \frac{e^{j\omega_c n} - e^{-j\omega_c n}}{2\pi j n} = \frac{\omega_c}{\pi} \frac{\sin(\omega_c n)}{\omega_c n}$$

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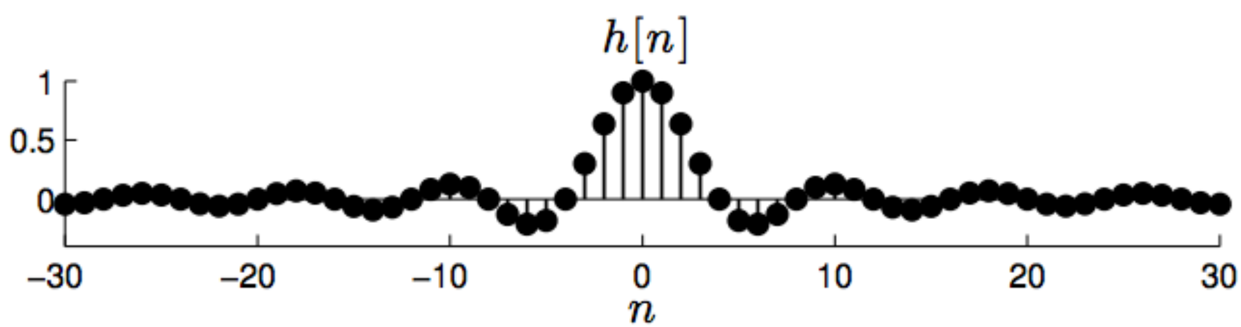

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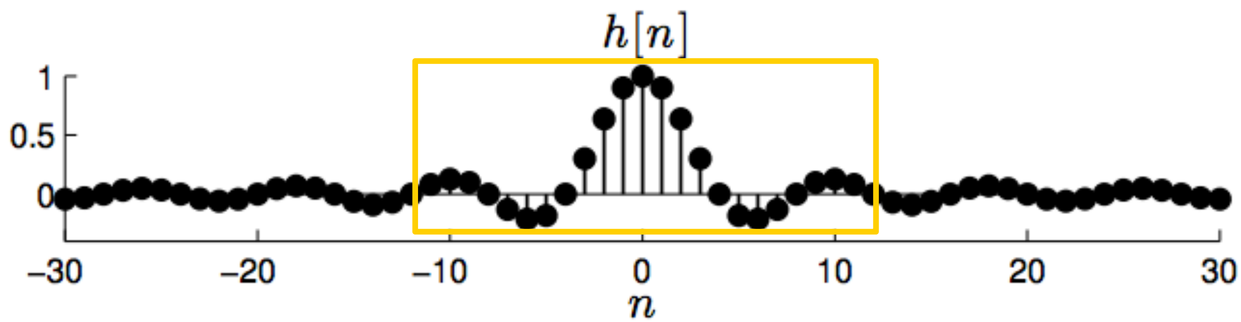
$$H(\omega) = \begin{cases} 1 & -\omega_c \le |\omega| \le \omega_c \\ 0 & \text{otherwise} \end{cases}$$

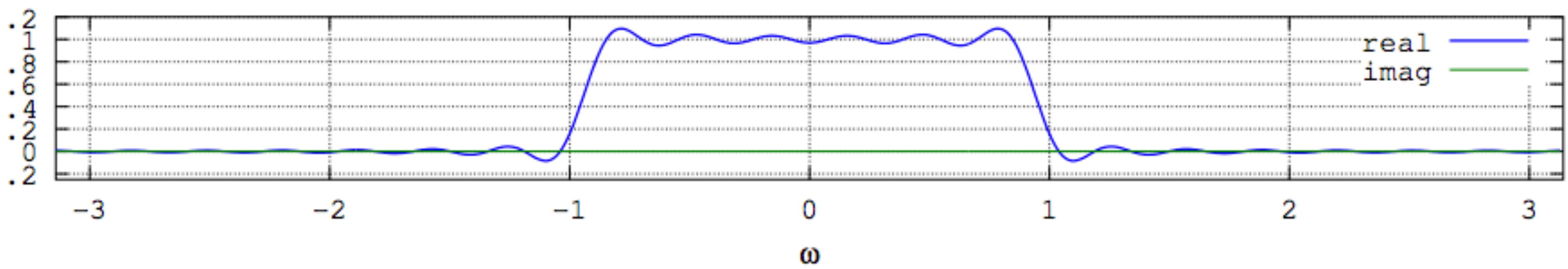
$$h[n] \;=\; 2\omega_c \frac{\sin(\omega_c n)}{\omega_c n}$$



Truncate and shift

$$h_{LP}[n] = w_N[n-N] \cdot h[n-N]$$

Example: Practical LP Filter



- Pass band smeared and rippled
  - Smearing determined by width of main lobe
  - Rippling determined by size of side lobes

- Different windows for different main/side lobe characteristics

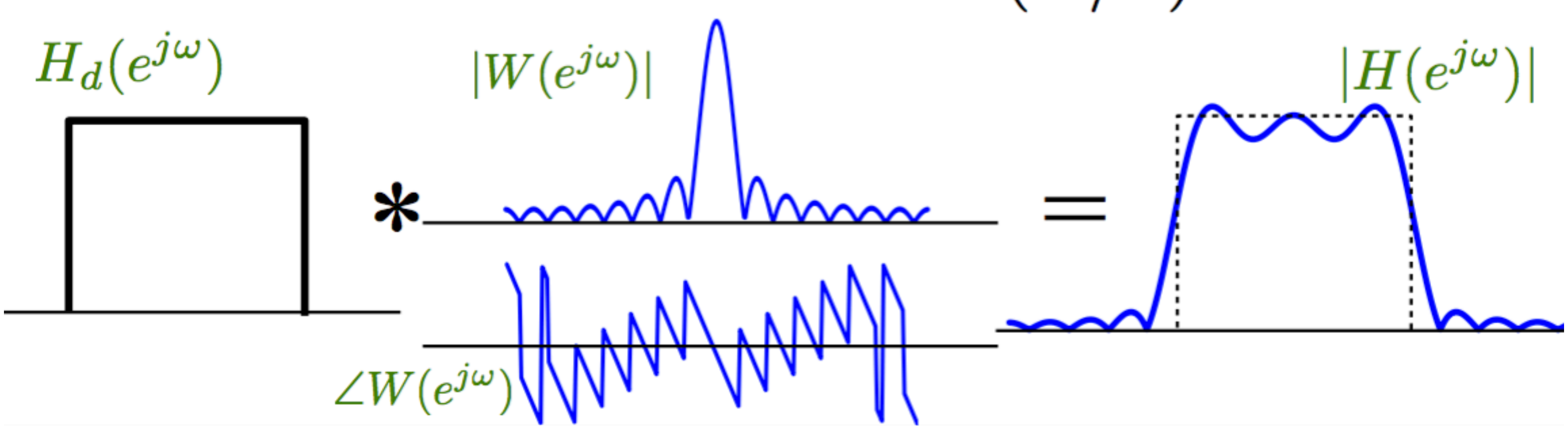
FIR Design by Windowing

- Desired filter,

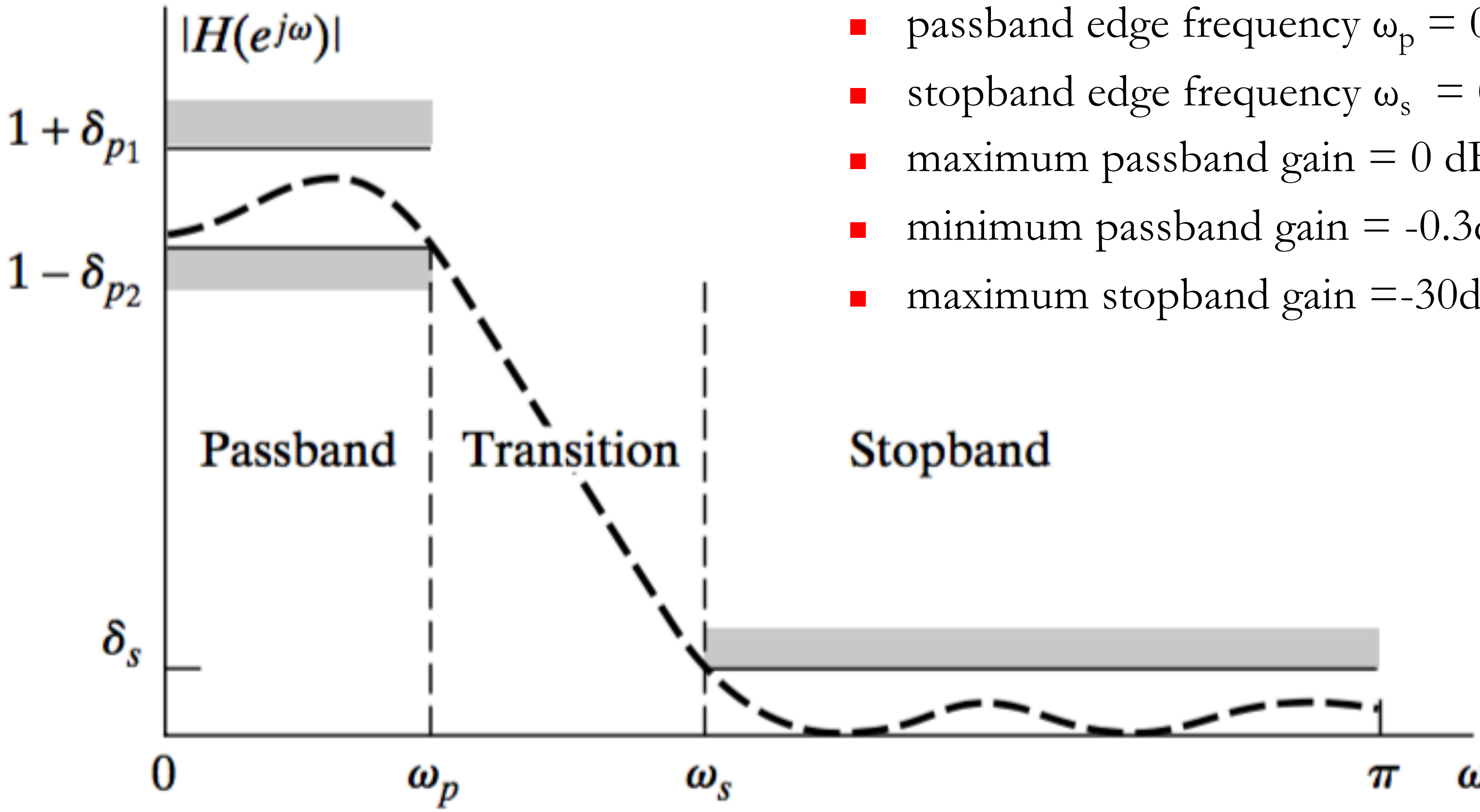
$$H(e^{j\omega}) = H_d(e^{j\omega}) * W(e^{j\omega})$$

- For Boxcar (rectangular) window

$$W(e^{j\omega}) = e^{-j\omega\frac{M}{2}} \frac{\sin(w(M+1)/2)}{\sin(w/2)}$$



Filter Specifications



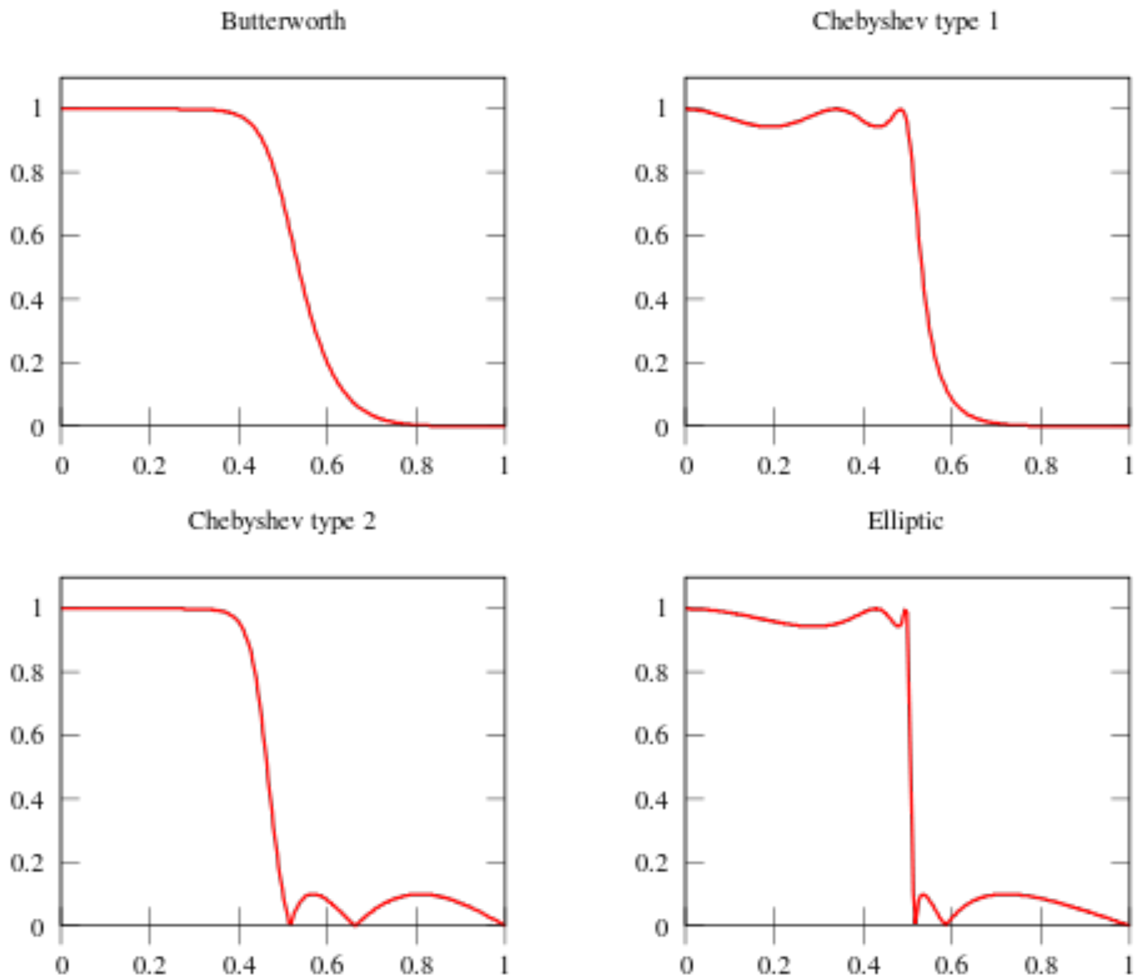
- Design specifications
  - passband edge frequency $\omega_p = 0.5\pi$
  - stopband edge frequency $\omega_s = 0.6\pi$
  - maximum passband gain = 0 dB
  - minimum passband gain = -0.3dB
  - maximum stopband gain =-30dB

CT Filters

- Butterworth
  - Monotonic in pass and stop bands
- Chebyshev, Type I
  - Equiripple in pass band and monotonic in stop band
- Chebyshev, Type II
  - Monotonic in pass band and equiripple in stop band
- Elliptic
  - Equiripple in pass and stop bands

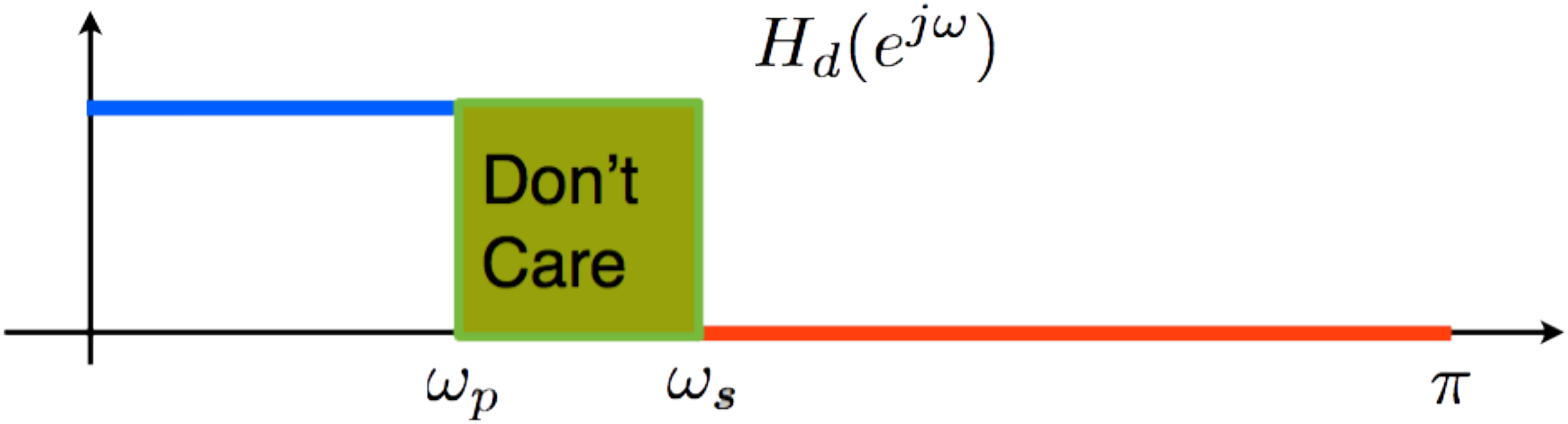
- Appendix B in textbook

FIR Design by Windowing



- Butterworth
  - Monotonic in pass and stop bands
- Chebyshev, Type I
  - Equiripple in pass band and monotonic in stop band
- Chebyshev, Type II
  - Monotonic in pass band and equiripple in stop band
- Elliptic
  - Equiripple in pass and stop bands

FIR Design by Optimality



- Least Squares:

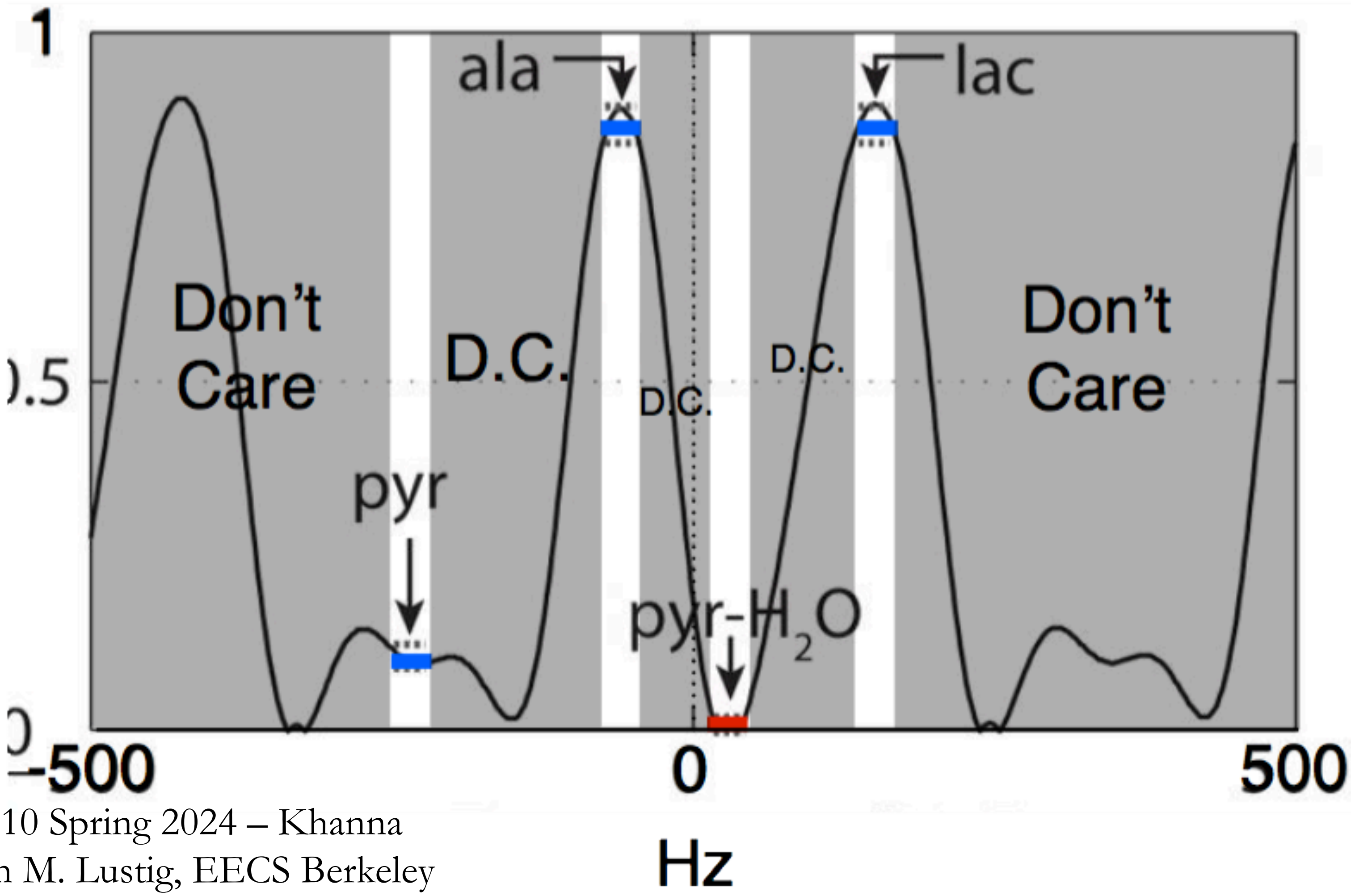
$$\text{minimize} \quad \int_{\omega \in \text{care}} |H(e^{j\omega}) - H_d(e^{j\omega})|^2 d\omega$$

- Variation: Weighted Least Squares:

$$\text{minimize} \quad \int_{-\pi}^{\pi} W(\omega)|H(e^{j\omega}) - H_d(e^{j\omega})|^2 d\omega$$

Example of Complex Filter

- Larson et. al, “Multiband Excitation Pulses for Hyperpolarized 13C Dynamic Chemical Shift Imaging” JMR 2008;194(1):121-127
- Need to design 11 taps filter with following frequency response:



Admin

- Find web, get text, start HW 0 and assigned reading…
  - http://www.seas.upenn.edu/~ese5310
  - https://edstem.org/us/courses/53754/discussion/
  - https://canvas.upenn.edu/
- Diagnostic quiz due 1/25
  - Review before or after as needed