

DSP for Digital Artists

by Perry R. Cook, PhD

a 3-Day CalArts Intersession Short Course, January 14-18, 2019

Day 1: Basics, Definitions, FIR Filters, LTI Systems, Convolution, Sine Waves

Digital

Sampled/quantized signals
At regular intervals,
hold signal, convert to
digital number.

Signal

Values (pressure, voltage, etc.)
that change as function of time
Continuous: $x(t)$ $t = \text{time}$
Digital (discrete): $x(n)$ $n = \text{sample \#}$

Processing

Do various math on
sampled digital signal.
 $y(n) = F(x(n))$
Gain, Filter, Analysis

Delays (t, x, θ)

Acoustics, speed of sound
Causality (and non-causality)
(block and hop sizes)

Sinusoids (sine, cosine, exp)

Fundamental in nature
Solution of δ^2x / δ^2t
"Eigenfunctions of LTI Systems"

Parameters (Gain, θ)

Gains, Weights, phases,
(SRate), time delays, block
(window) and hop sizes

PsychoAcoustics: Definition

(a branch of PsychoPhysics)

Physical Correlate(s)

How to compute it

Loudness	Percept, Intensity	Power	Square of signal (RMS) $\sum_n x^2(n)$
Pitch	Percept, Low <-> High	Frequency	Cycles per second (spectral peak)

Log Perception and non-linear Scales

Timbre	Difference	Spectrum+	Many factors, centroid, peaks, time
Time & Timing, Tempo vs. Pitch, Resolvability of events (the 30ms boundary)			

The Just Noticeable Difference (JND) Smallest perceived (better than guessing) change

Filters:

$$y(n) = F(x(n))$$

Operations on Digital Signals

Popular Simple Filters:

Linear Gain:	$y(n) = g x(n)$	where $g = \text{some constant}$
Moving Average:	$y(n) = 0.5x(n) + 0.5x(n-1)$	(low-pass)
High-Pass Cousin:	$y(n) = x(n) - x(n-1)$	(digital differentiator)

Also called **OneZero Filter**, Averaging more samples: More Zeroes!!
Also type of **Finite Impulse Response (FIR) Filter**

Sinusoids (Sine Waves) Mass/Spring Damper, Pendulum, Rotations (Trig!), RLC Circuits

Code and Demos:

0-SamplingMore In Processing, Demonstrates Sampling, Quantization
Pendulum MassSpring RotateSine Processing Code for Physical Systems
0c-Quantize.ck 0b-PhoneBandwidth.ck Quantization and SRate Demos
ZeroX P rocessing Code to demonstrate zerocrossings
5-ZCNative.ck 6-ZCUGs.ck 7-ZCPitch.ck Zeroes2.ck ZeroXing "Pitch" Demos
1-PeakNative.ck 1b-PeaksGUI.ck Track (and display) waveform peaks
OneZero.ck FIRnthOrder.ck FIR Filters
2-SparsePower.ck 3-RMSPowerNative.ck Sum of Squares Power

Assignment: 1) Some PsychoAcoustic Metric: Measure something about you. Use JND Paradigm.

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Day 2: More Filters, Feedback Filters, More Sine Waves, Spectrum Analysis, Modal Synthesis

Some More Time-Domain “Pitch” Detection:

AMDF: pick m that minimizes: $y(n) = \sum_n |x(n) - x(n-m)|$
Autocorrelation: pick m that maximizes: $y(n) = \sum_n x(n) * x(n-m)$

Impulse Response:

$h(n) = y(n)$, in response to input $x(n) = \delta(n)$
 $\delta(n) = 1$ if $n=0$, 0 otherwise

Transfer Function:

$H(f) = Y(f)$, for every $x(t) = \sin(2\pi ft)$
Digital: $H(n) = Y(n)$, for $x(n) = \sin(2\pi fn/SR)$

Linear, Time Invariant (LTI) Systems

Linearity: Homogeneity ($\alpha x \rightarrow \alpha y$) and Superposition ($x_1 + x_2 \rightarrow y_1 + y_2$)
Time Invariance: $x(\tau + t) \rightarrow y(\tau + t)$ for all τ
OnePole: (1st Order feedback) $y(n) = gx(n) + ry(n-1)$ NOTE: $r \leq 1.0$!!!
Set r and g exactly 1.0 yields “Digital Integrator”
Set $0.0 < r < 1.0$ and $g = (1.0 - r)$ Low-pass filter
High-pass Cousin: set $-1.0 < r < 0.0$ High-pass filter

Convolution $y(n) = \sum_m x(n-m) h(m)$ implement Impulse Response as FIR filter

Filters: $y(n) = F(x(n))$ Operations on Digital Signals

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Recursive Filters: **Feedback, Auto Regressive, IIR, Pole(s)**

OnePole Filter: $y(n) = g x(n) + r y(n-1)$ NOTE: $|r| < 1.0$

NPole Filter: $y(n) = g x(n) + \sum_{m=1toN} -b_m y(n-1)$

Super Special Filter, 2-Pole “Resonator” $y(n) = g x(n) + 2r \cos(2\pi f/SR) y(n-1) - r^2 y(n-2)$

Modal Synthesis: Modes = natural resonances of system. Excite those and let them ring!

Code and Demos: See Day2.zip

Assignment: Impulse Responses & Modes: Look around, listen, find some systems. Find/Record Modal sound. Use FFTFindModes to analyze it. If you’re brave, use FFTResynth (Noise and/or Residue) to recreate it.

