Computer Music - March 1st

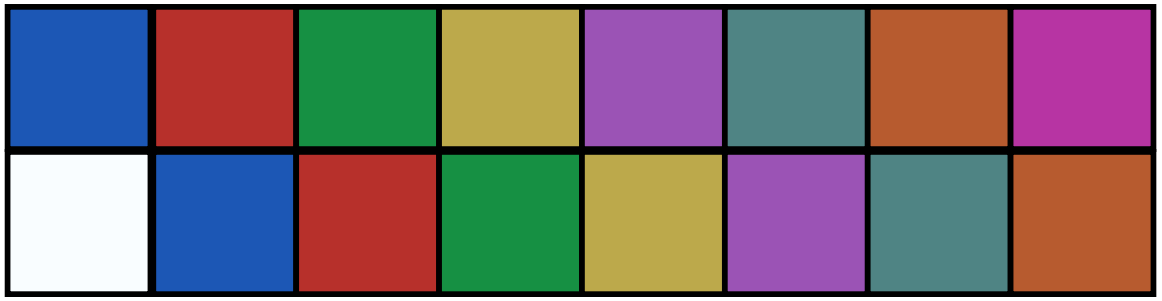
Power conservation

Artificial Reverberation

Fractional and Variable Delays

Doppler Effect/Pitch Shifting

Introduction to Filters



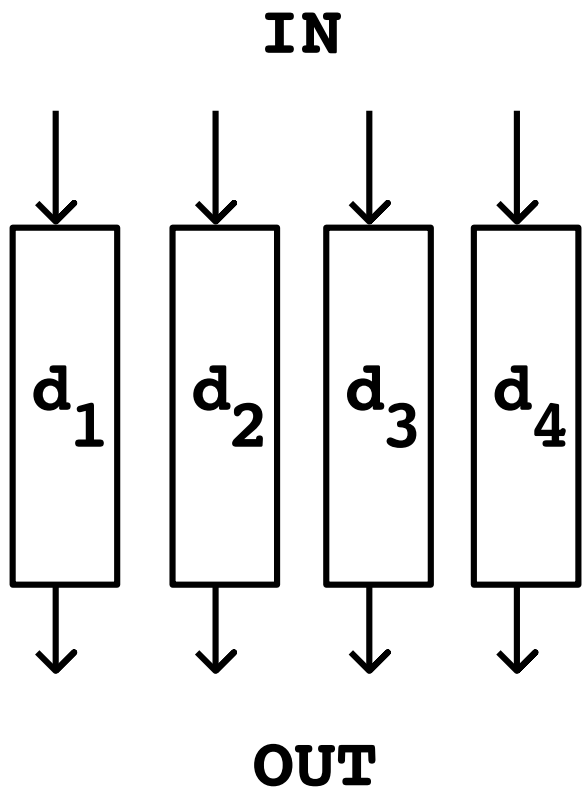
Johanna Devaney

Review

- Delays
  - Canon
  - Echoes
  - Filtering
  - Altered room quality
- Recirculating Delay Networks

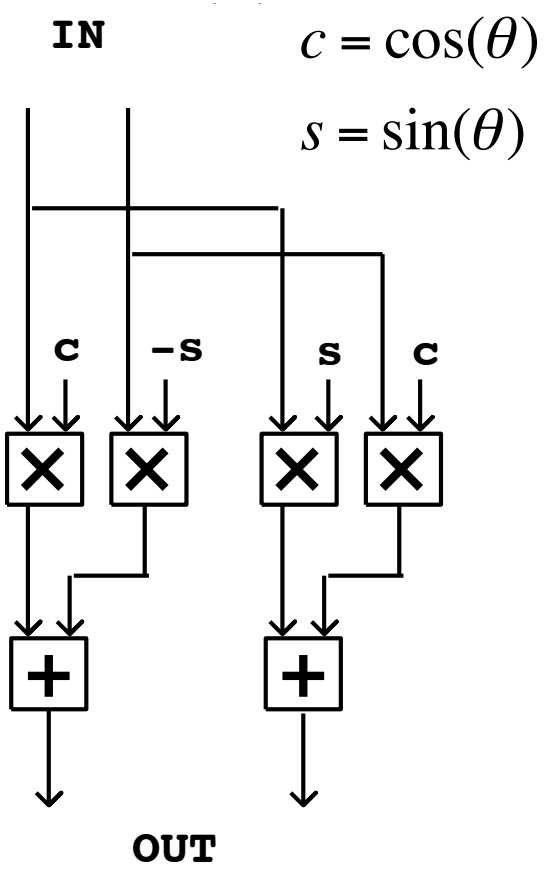
Power conservation

- If power of the input (IN) equals power of output (OUT) the system is unitary
- If a system is unitary then it has a flat frequency response
- This can be preserved even with rotations and reflections of the signal

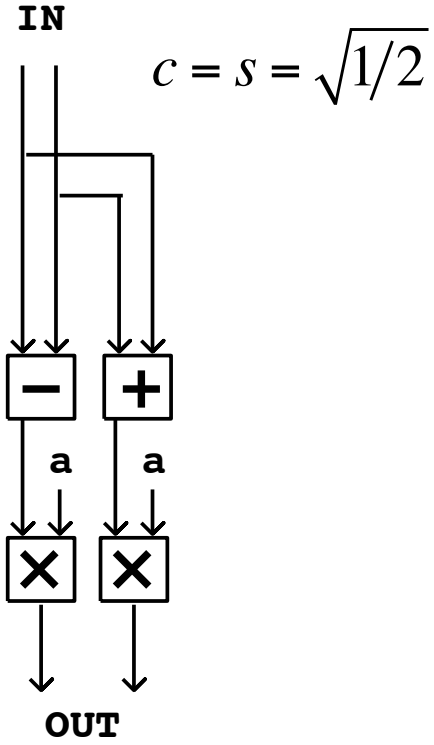


Power conservation

Rotation of two signals

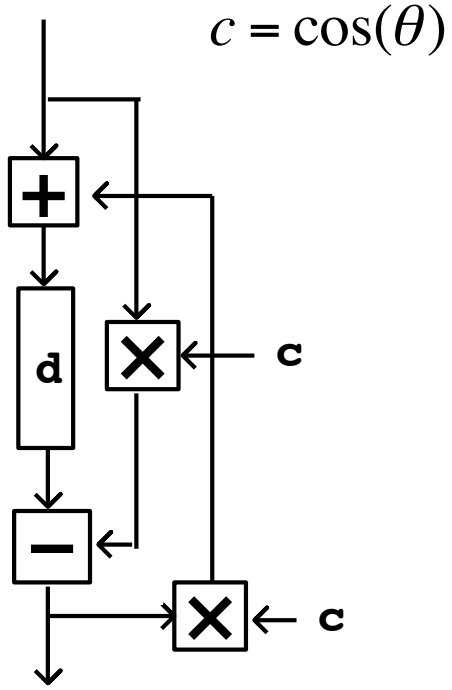


Rotation by $\theta = \pi/4$



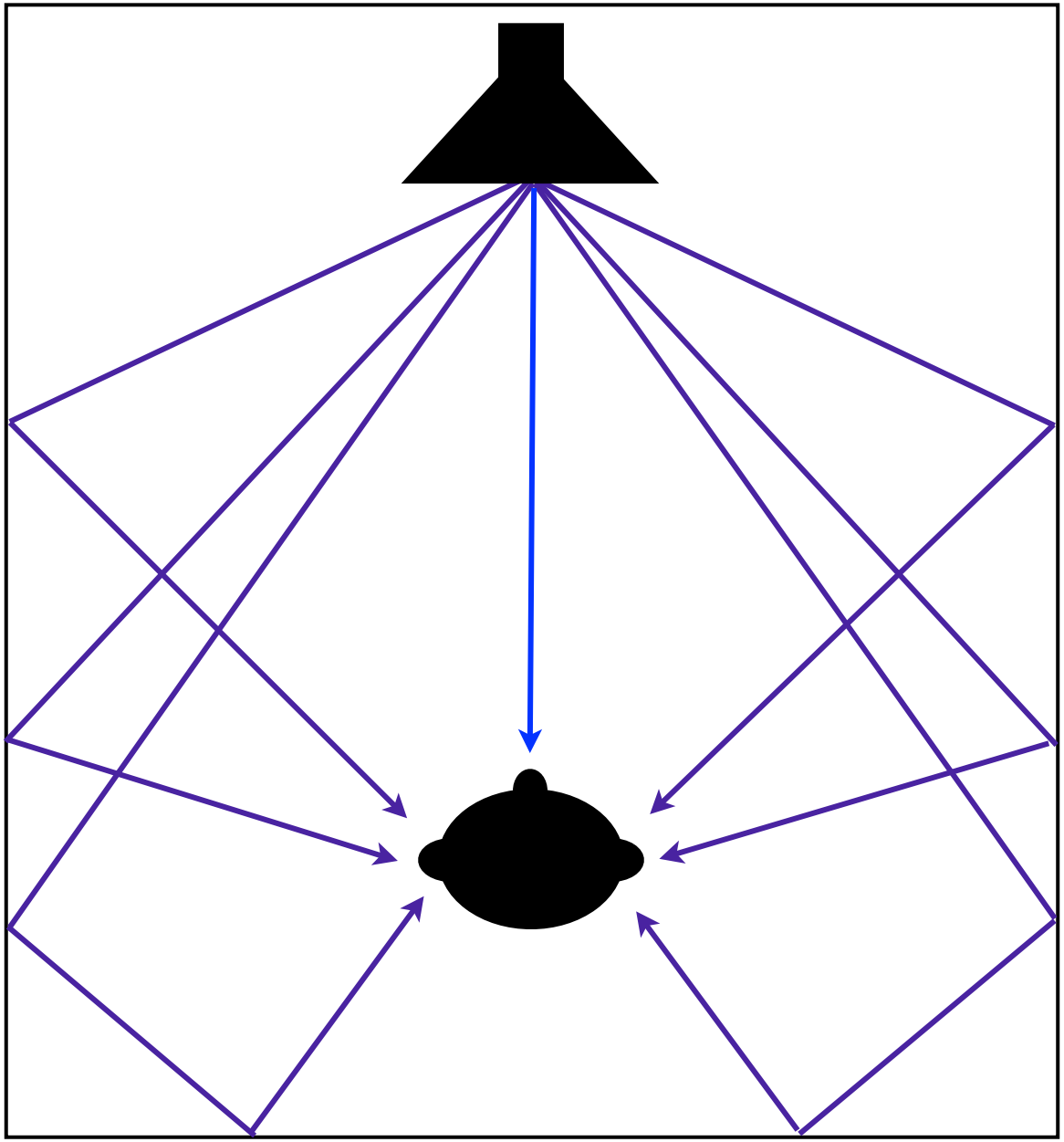
Useful for designing reverb

Recirculating network



All-pass filter: the phase of frequencies around the cut-off frequency are modified

Artificial Reverberation



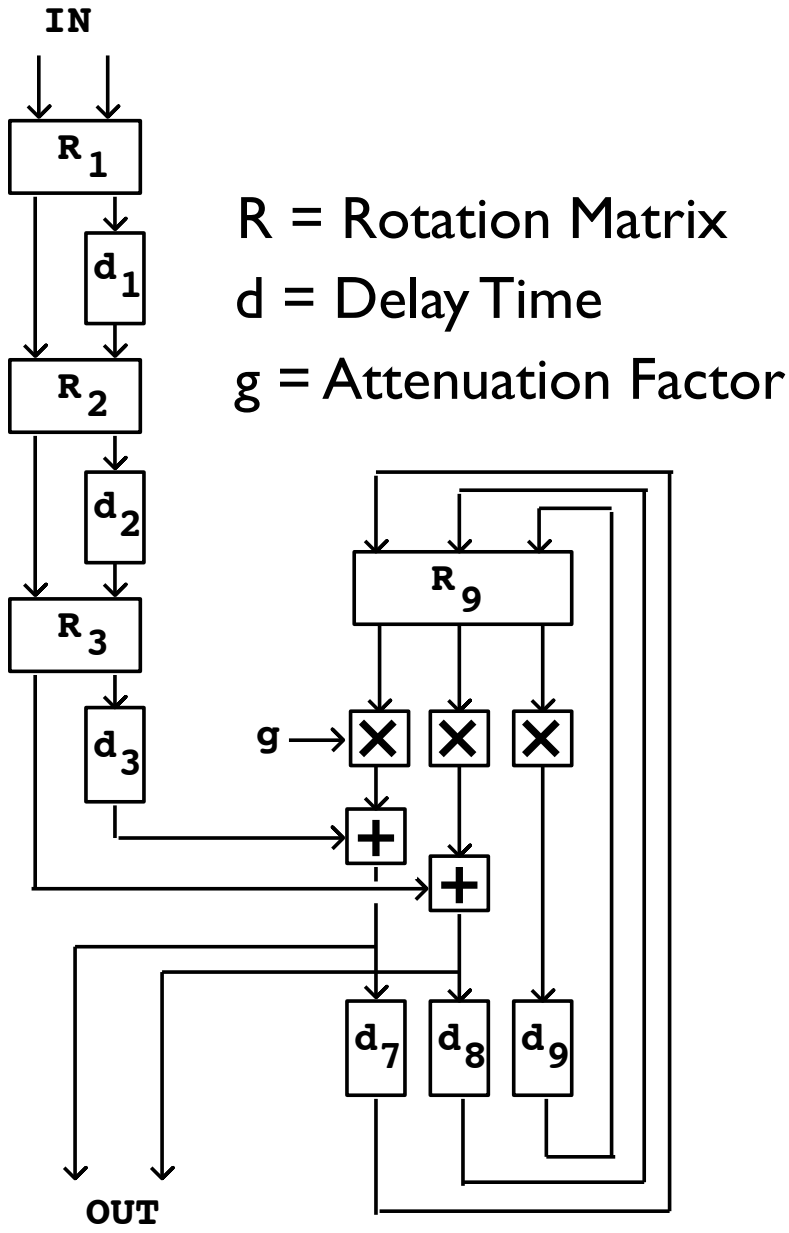
Direct

Reflections

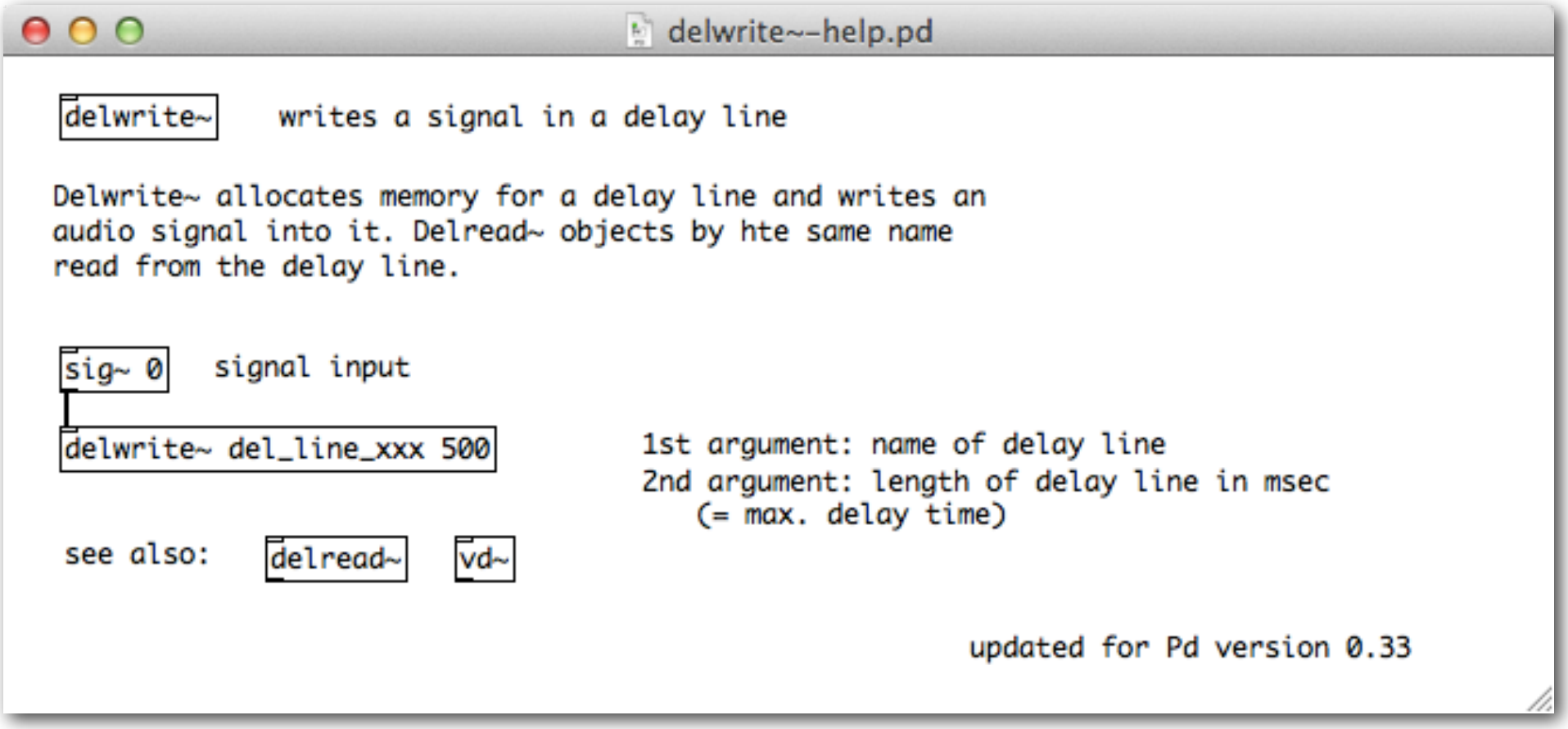
Adapted from: Roads, C. 1996. Computer Music Tutorial. MIT Press. 475

Artificial Reverberation

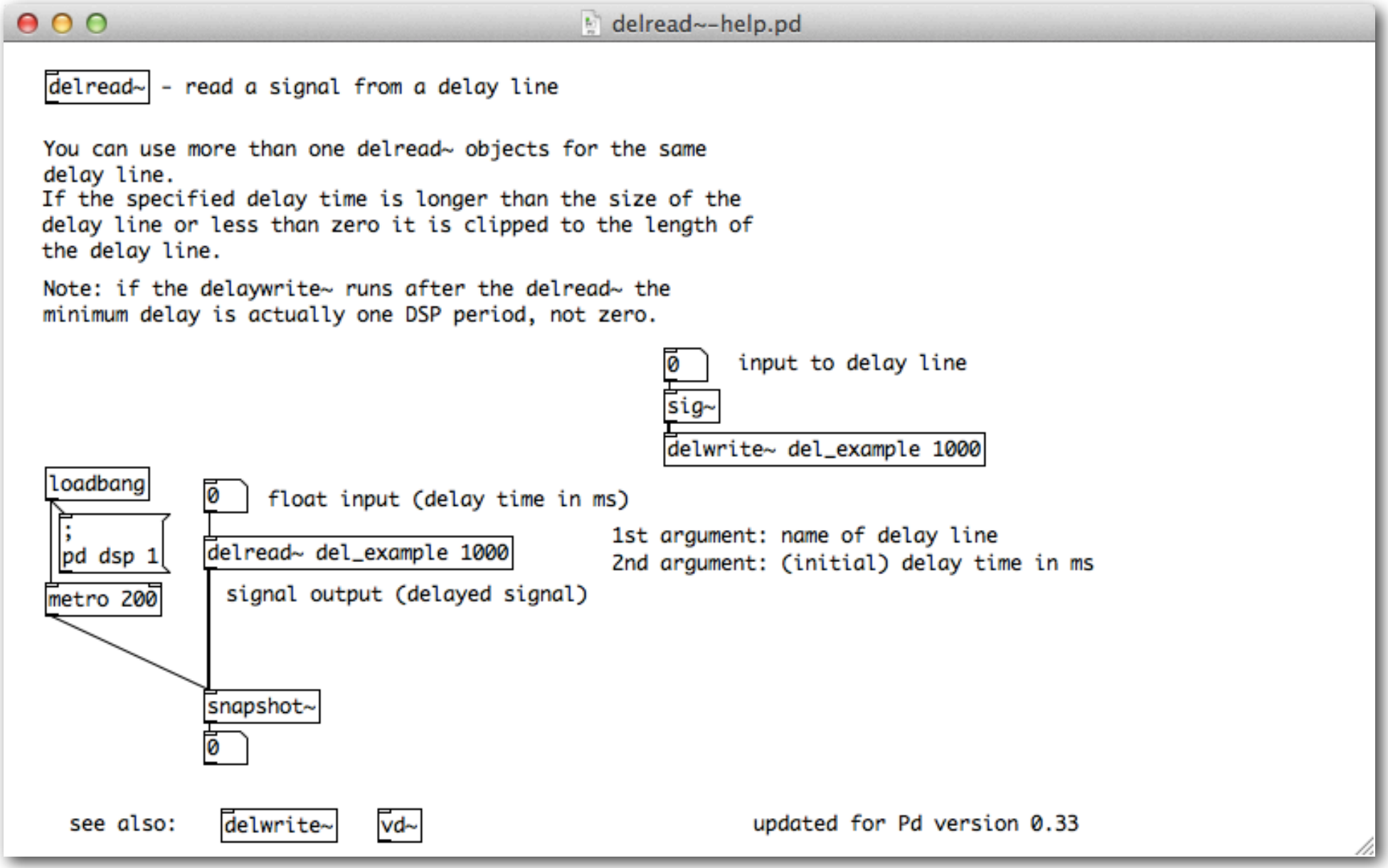
- Things to watch out for
  - Delay lines length
    - coloration can occur if the lines are too short
  - Echo density
    - should be at least 1000/second



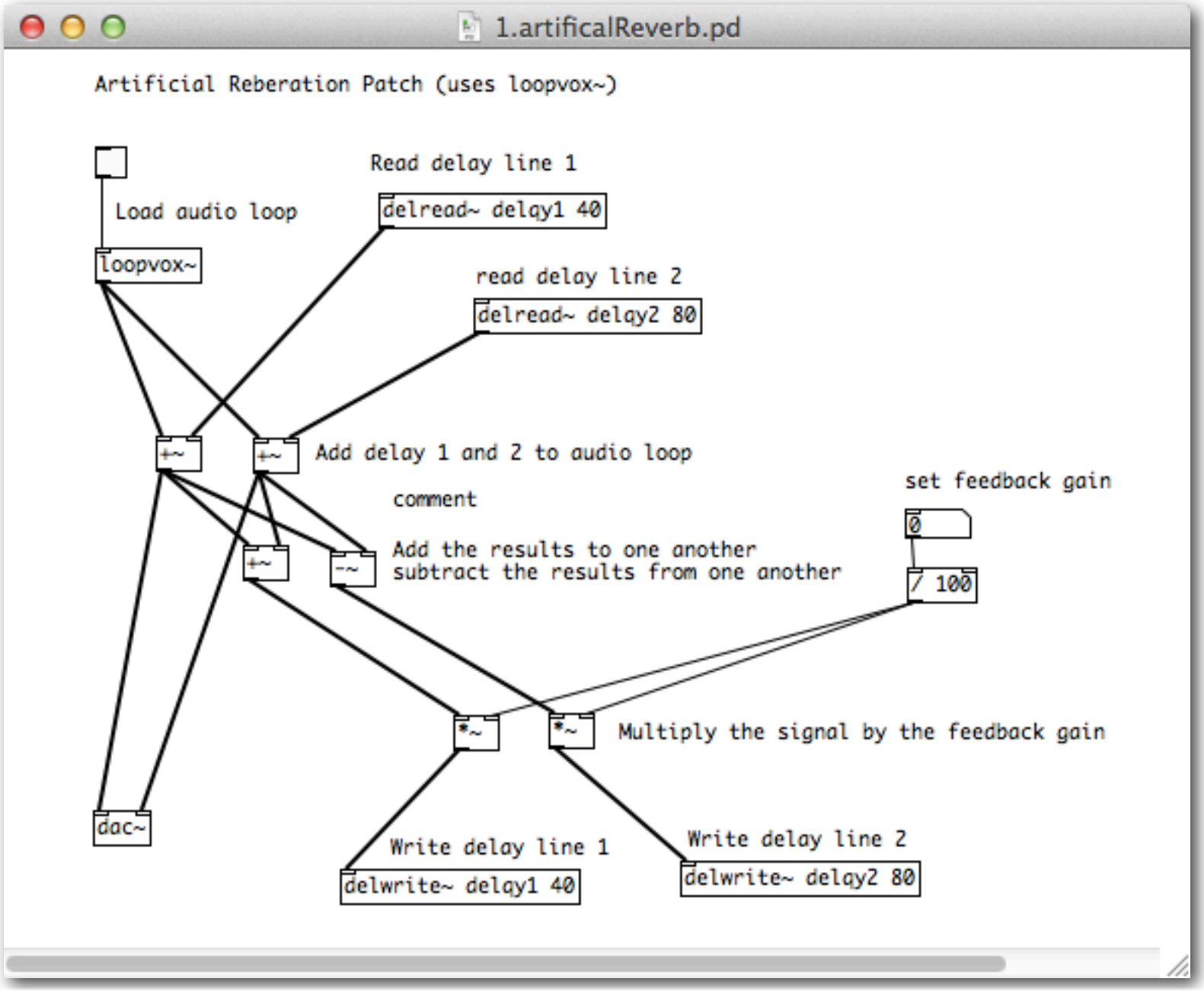
delwrite~

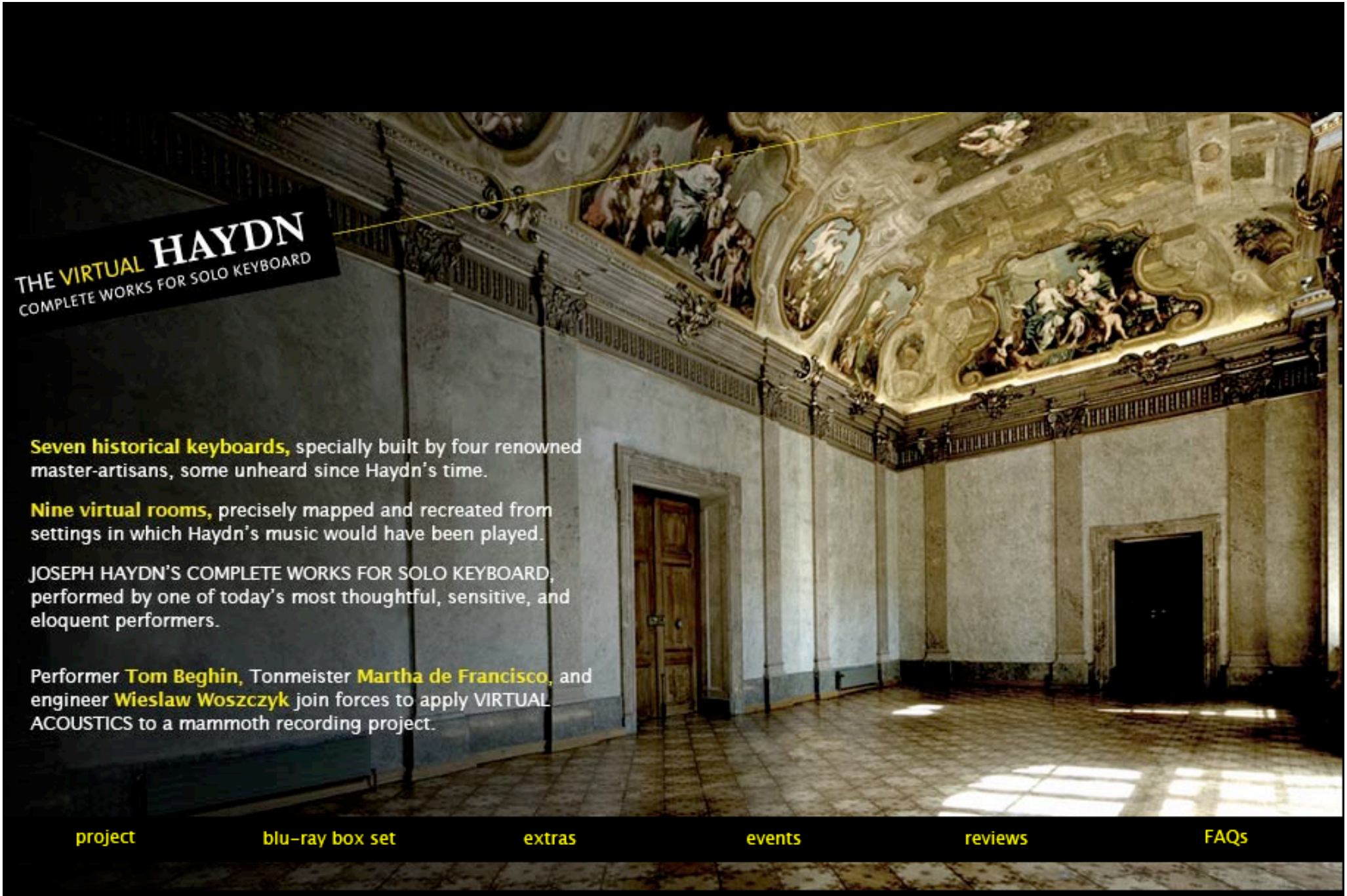

delwrite~-help.pd
delwrite~ writes a signal in a delay line
Delwrite~ allocates memory for a delay line and writes an
audio signal into it. Delread~ objects by hte same name
read from the delay line.
sig~ 0 signal input
delwrite~ del_line_xxx 500
1st argument: name of delay line
2nd argument: length of delay line in msec
(= max. delay time)
see also: delread~ vd~
updated for Pd version 0.33

delread~


delread~-help.pd
delread~ - read a signal from a delay line
You can use more than one delread~ objects for the same
delay line.
If the specified delay time is longer than the size of the
delay line or less than zero it is clipped to the length of
the delay line.
Note: if the delaywrite~ runs after the delread~ the
minimum delay is actually one DSP period, not zero.
0
input to delay line
sig~
delwrite~ del_example 1000
loadbang
0
float input (delay time in ms)
;
pd dsp 1
delread~ del_example 1000
1st argument: name of delay line
2nd argument: (initial) delay time in ms
metro 200
signal output (delayed signal)
snapshot~
0
see also:
delwrite~
vd~
updated for Pd version 0.33

Artificial Reverberation




THE VIRTUAL HAYDN
COMPLETE WORKS FOR SOLO KEYBOARD
Seven historical keyboards, specially built by four renowned master-artisans, some unheard since Haydn's time.
Nine virtual rooms, precisely mapped and recreated from settings in which Haydn's music would have been played.
JOSEPH HAYDN'S COMPLETE WORKS FOR SOLO KEYBOARD, performed by one of today's most thoughtful, sensitive, and eloquent performers.
Performer Tom Beghin, Tonmeister Martha de Francisco, and engineer Wieslaw Woszczyk join forces to apply VIRTUAL ACOUSTICS to a mammoth recording project.
project
blu-ray box set
extras
events
reviews
FAQs

Immersive Presence Lab
at Center for New
Music and Audio
Technologies (CIRMMT)
Montreal, Canada

Sonata in A Major HXVI:12 Andante
"Music Room"

Sonata in G Major HXVI:6 Minuet/Trio
Haydn's House

Interpolation

- Non-integer delay times require interpolation

Linear Interpolation

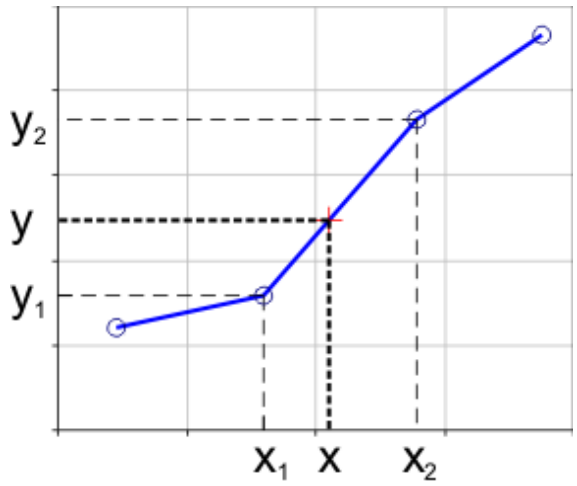


Image from Dagra Manual

Cubic Interpolation

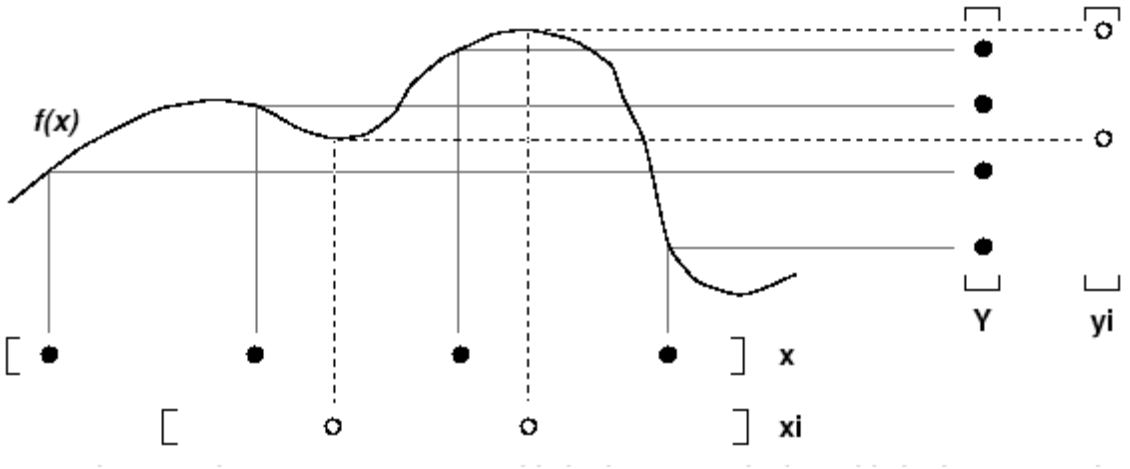
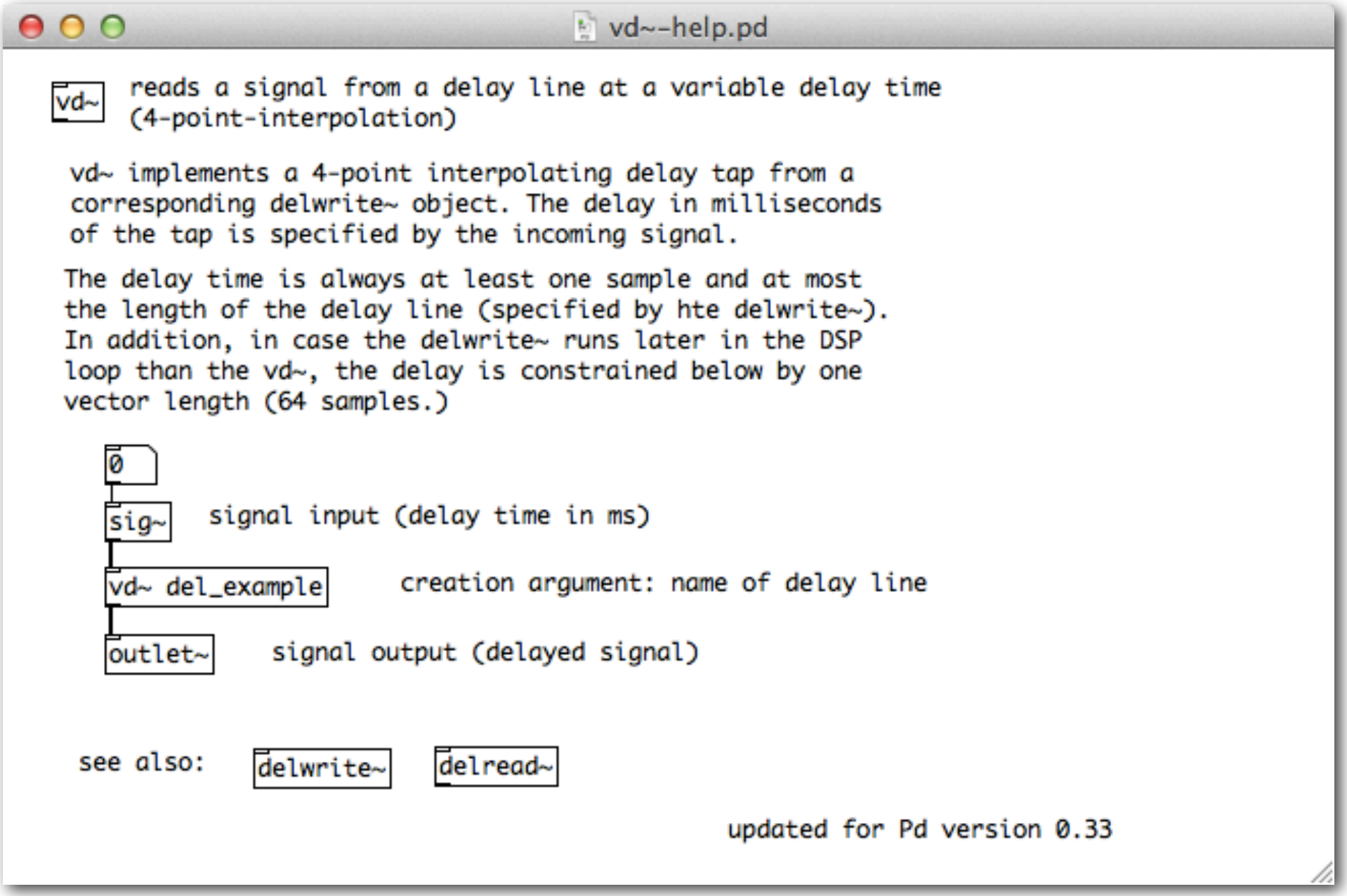


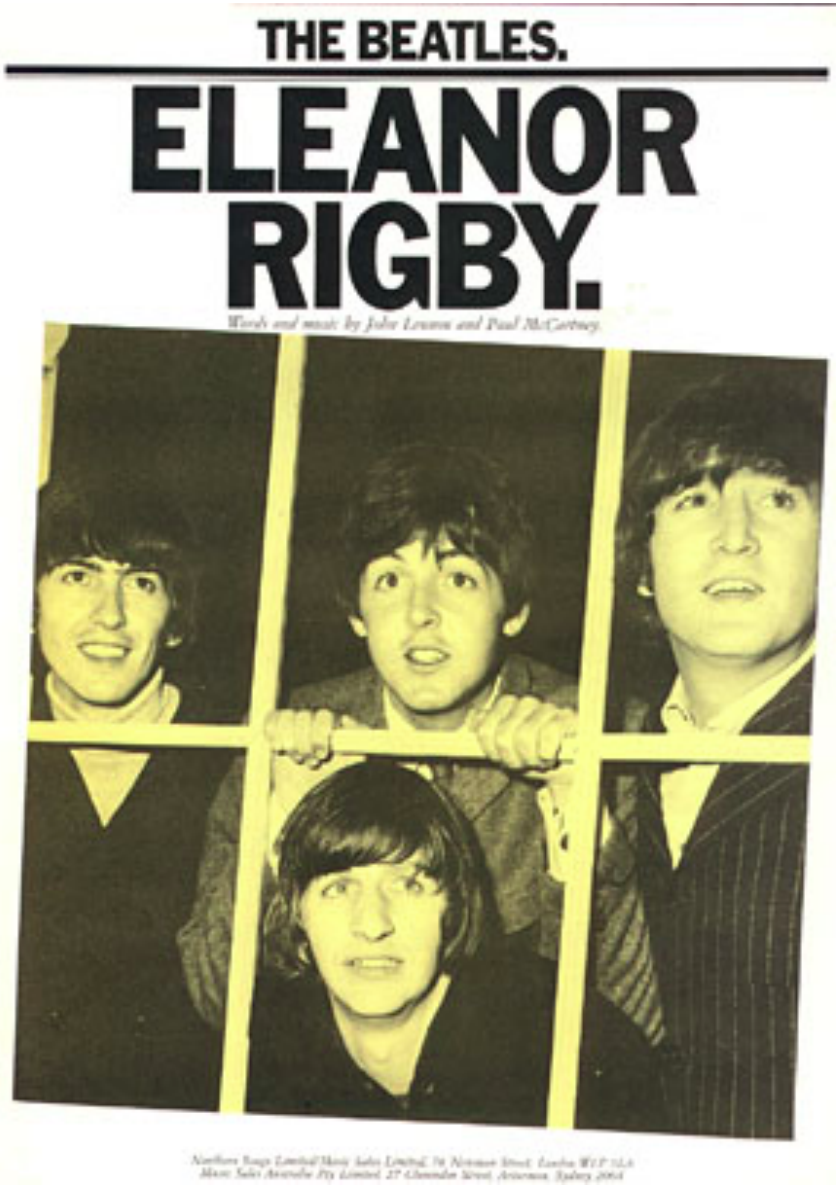
Image from Matlab Manual

- Minimum delay-time is higher with higher-order polynomials

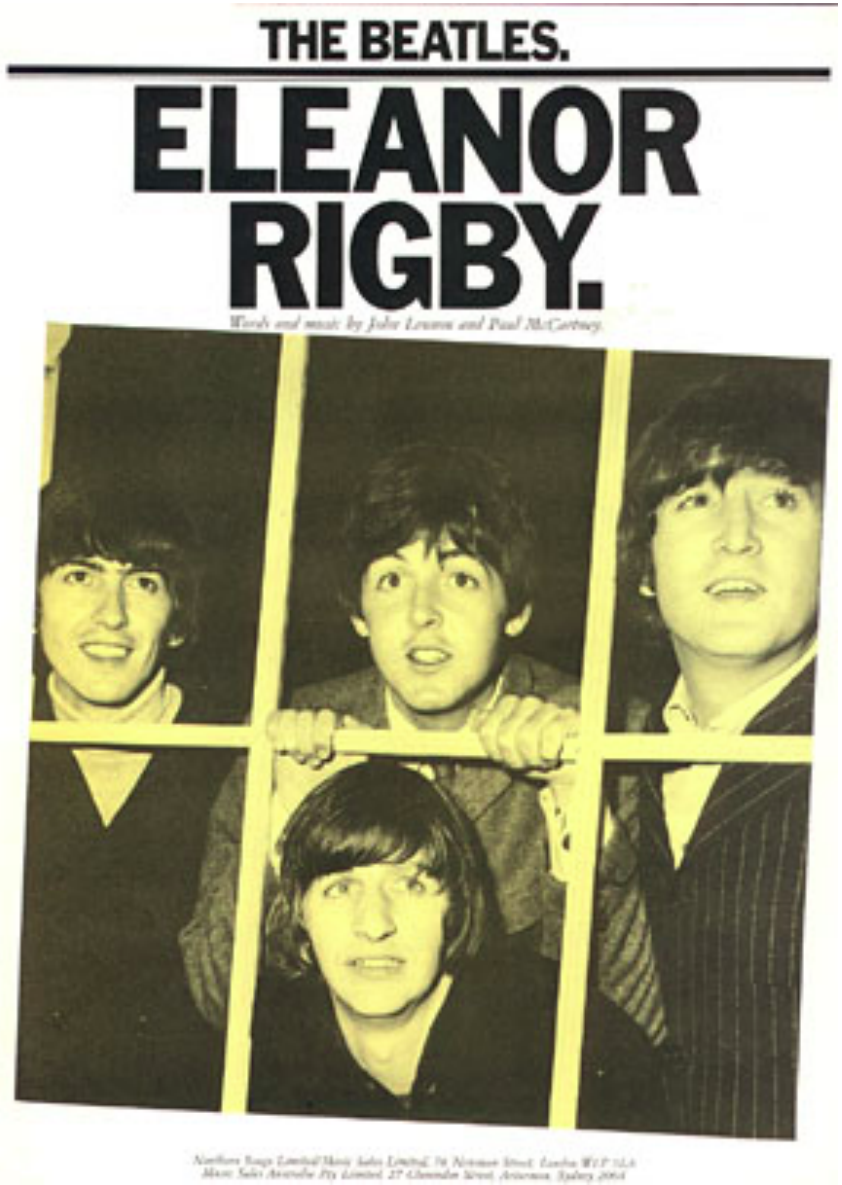
vd~


vd~-help.pd
vd~
reads a signal from a delay line at a variable delay time
(4-point-interpolation)
vd~ implements a 4-point interpolating delay tap from a
corresponding delwrite~ object. The delay in milliseconds
of the tap is specified by the incoming signal.
The delay time is always at least one sample and at most
the length of the delay line (specified by hte delwrite~).
In addition, in case the delwrite~ runs later in the DSP
loop than the vd~, the delay is constrained below by one
vector length (64 samples.)
0
sig~
signal input (delay time in ms)
vd~ del_example
creation argument: name of delay line
outlet~
signal output (delayed signal)
see also:
delwrite~
delread~
updated for Pd version 0.33

Variable delays



Variable delays



0:00-0:04 Manual Double Tracking

0:07-0:11 Manual Double Tracking

0:14 Brief Automatic Double Tracking on "Ele-a-"

0:14-0:31 No Double Tracking

0:31-0:44 Automatic Double Tracking

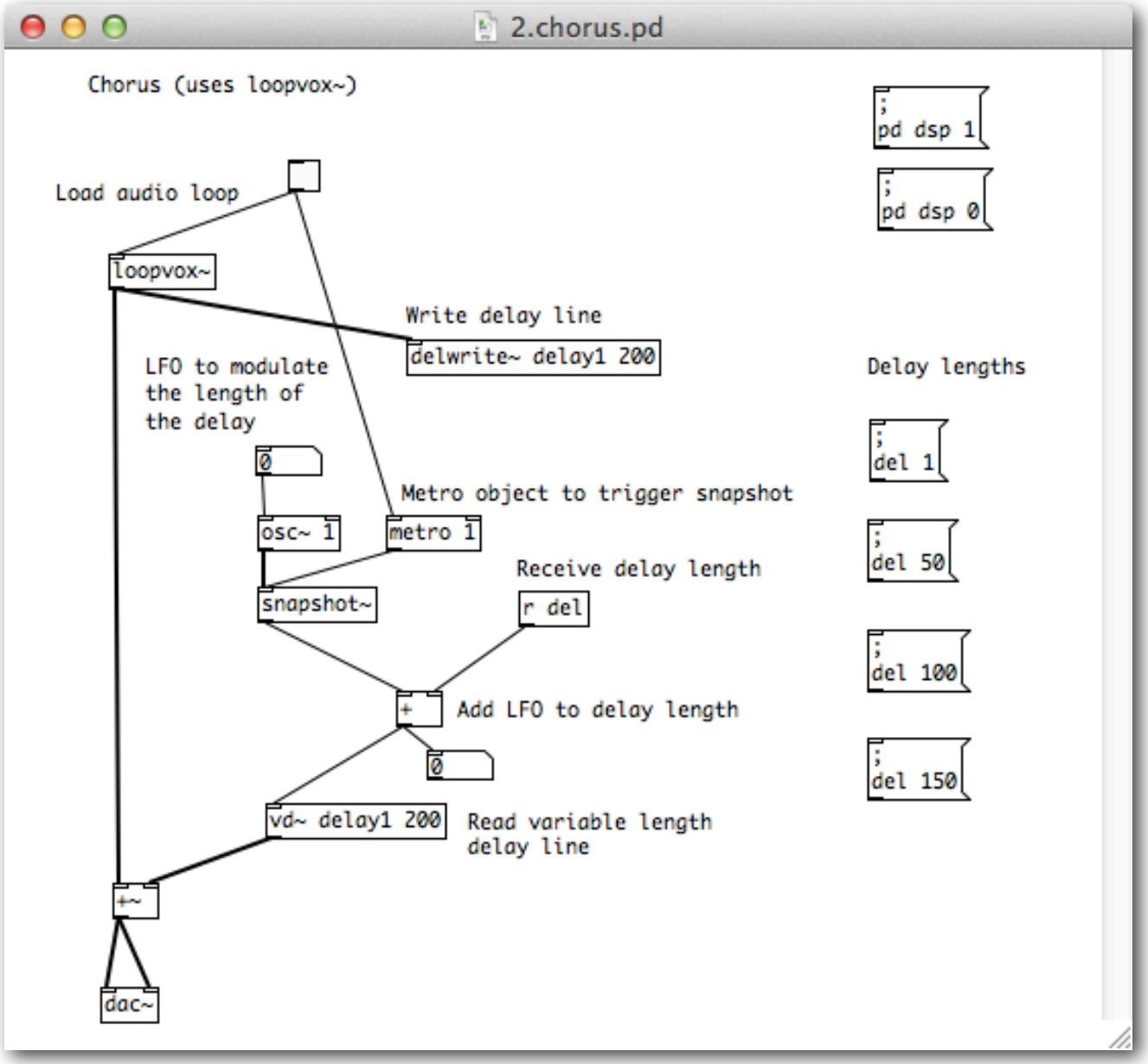
0:46-1:02 No Double Tracking

1:03-1:15 Automatic Double Tracking

1:17-1:28 Manual Double Tracking

1:31-2:02 No Double Tracking

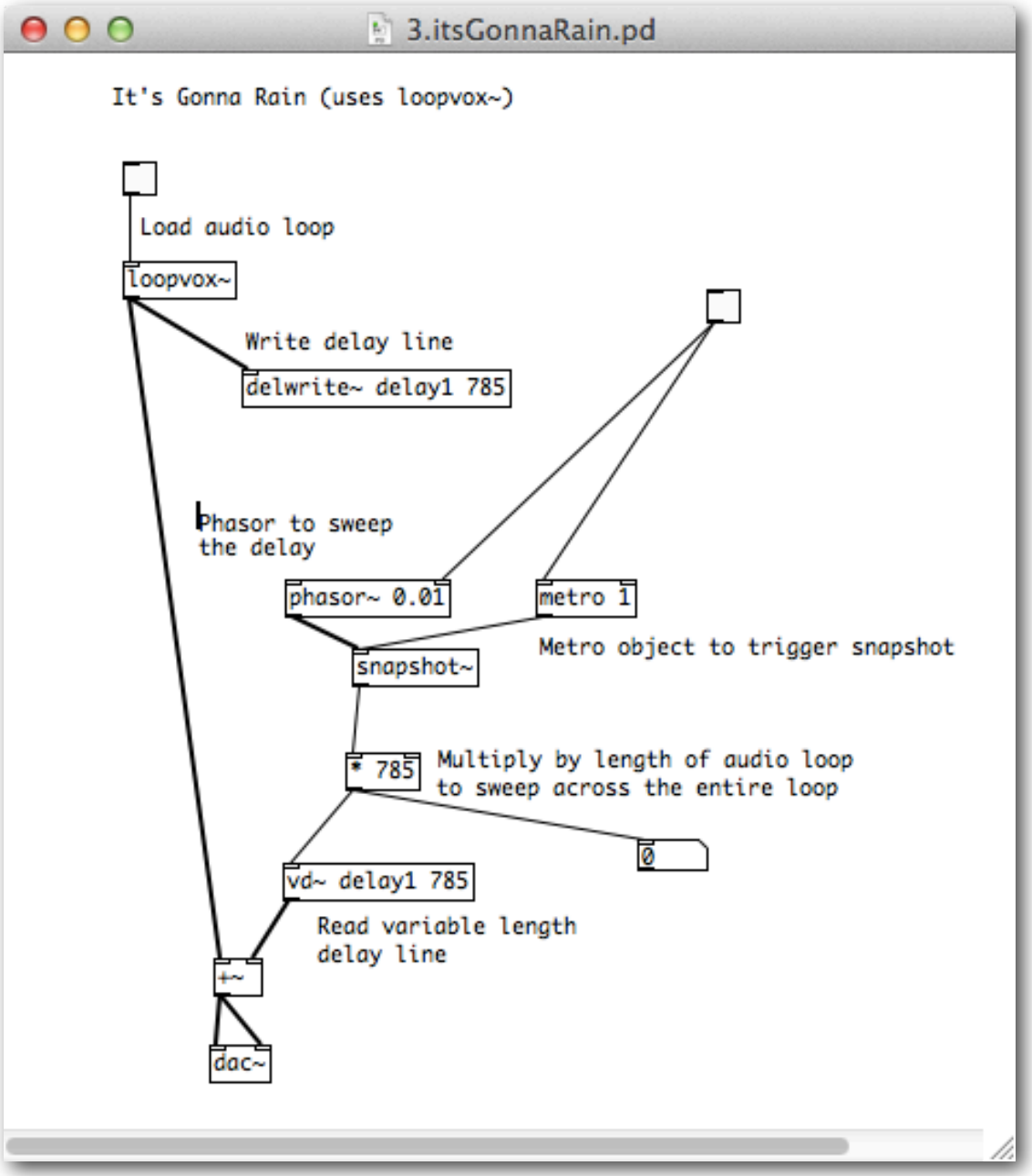
Variable delays



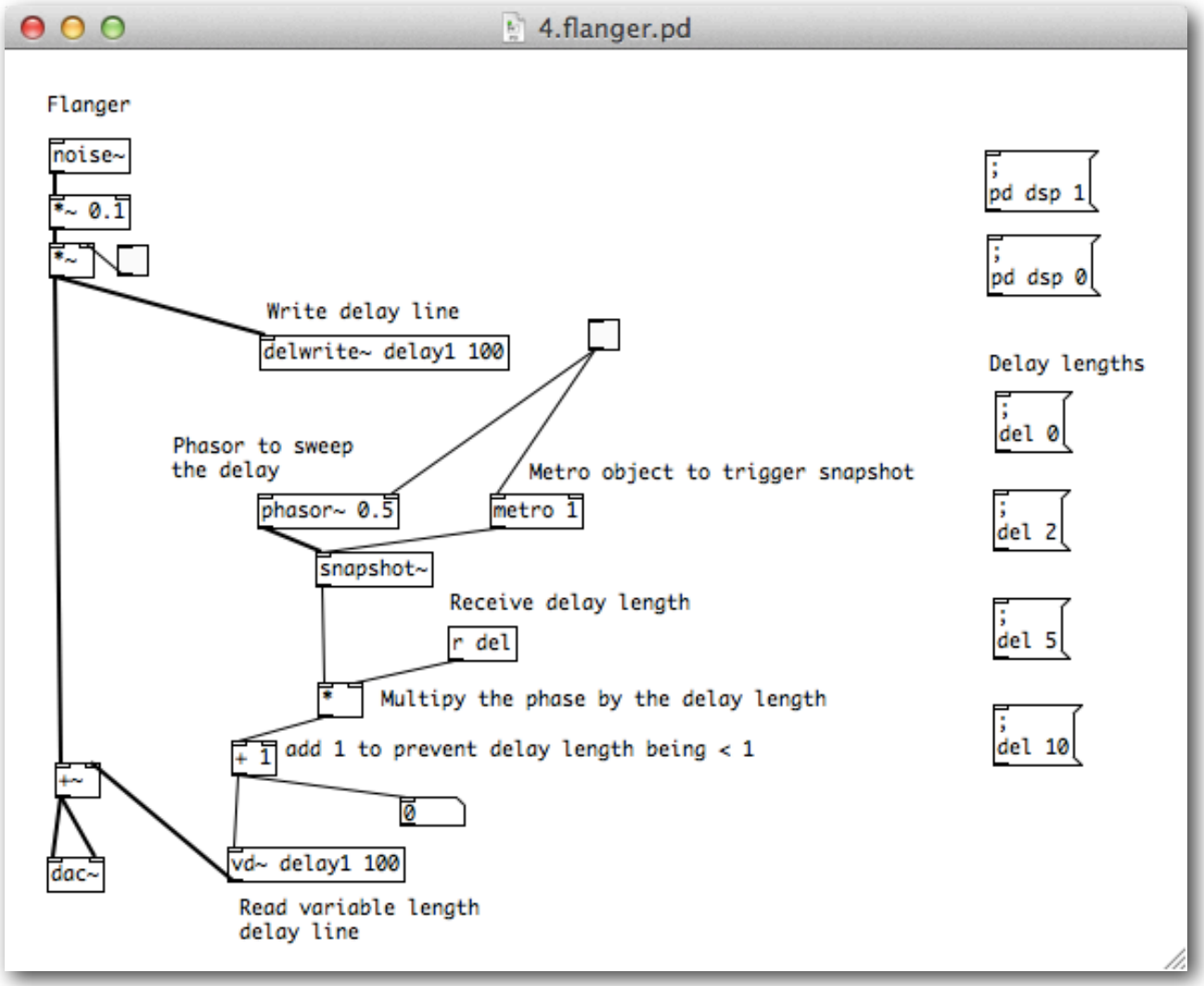
Variable delays

"It's Gonna Rain was composed in San Francisco in January 1965. The voice belongs to a young black Pentecostal preacher who called himself Brother Walter. I recorded him along with the pigeons and traffic one Sunday afternoon in Union Square in downtown San Francisco. Later at home I started playing with tape loops of his voice and, by accident, discovered the process of letting two identical loops go gradually out of phase with each other. In the first part of the piece the two loops are lined up in unison, gradually move out of phase with each other and then slowly move back into unison." - Steve Reich

Variable delays



Variable delays



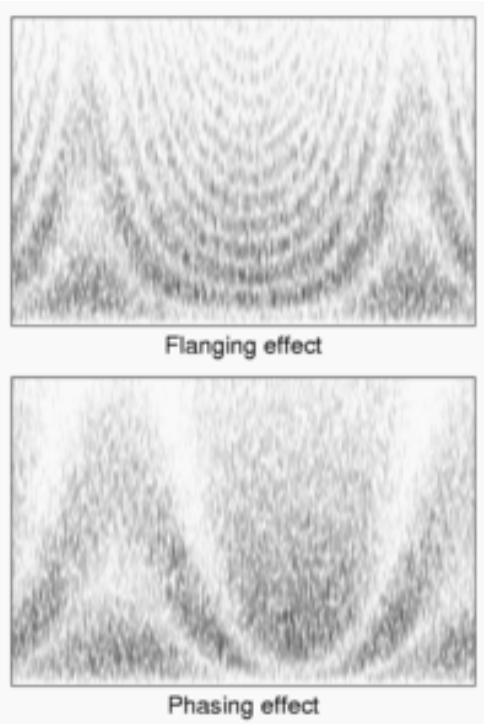
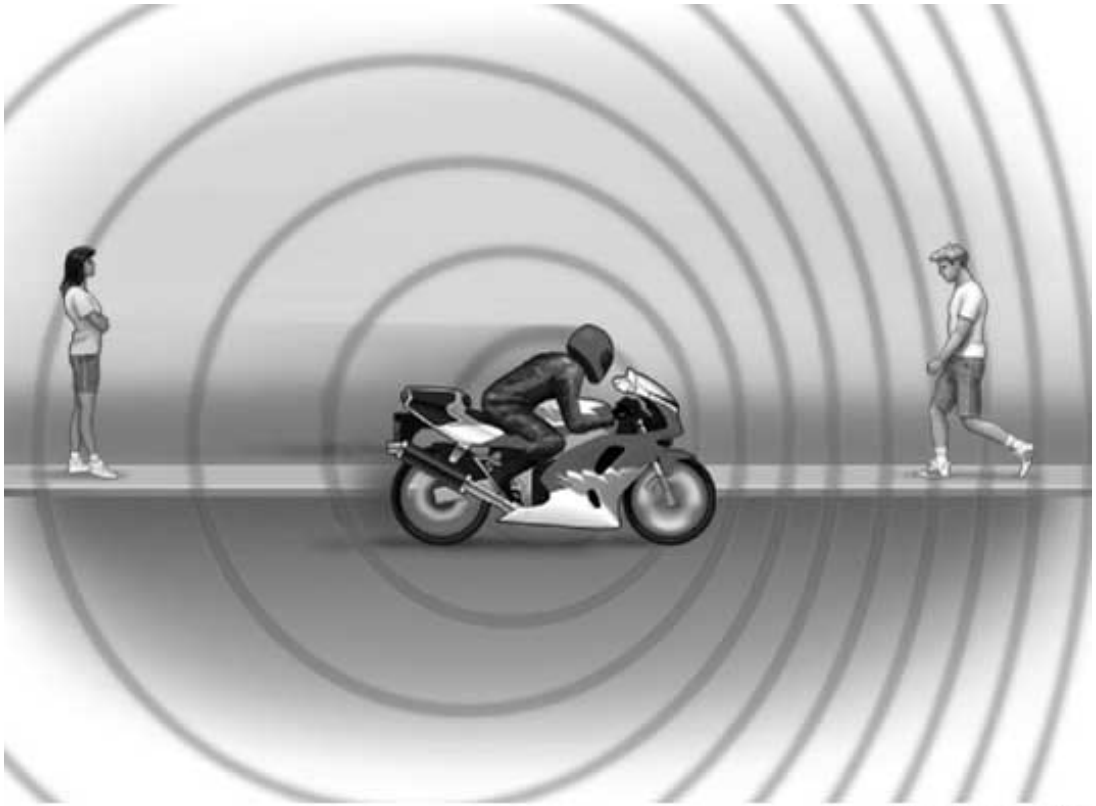


Image from Wikipedia

Doppler Effect



Precision Graphics

- Air can function as a delay line
- A sound will sound higher in pitch as an object approaches as its motion causes the sound waves to bunch together
- It will sound lower in pitch as the object passes and moves further away because the sound waves become further apart


CAMERAS - VIDEO
PAWNSHOP

Doppler Effect - Leslie Speaker

Image from Wikipedia

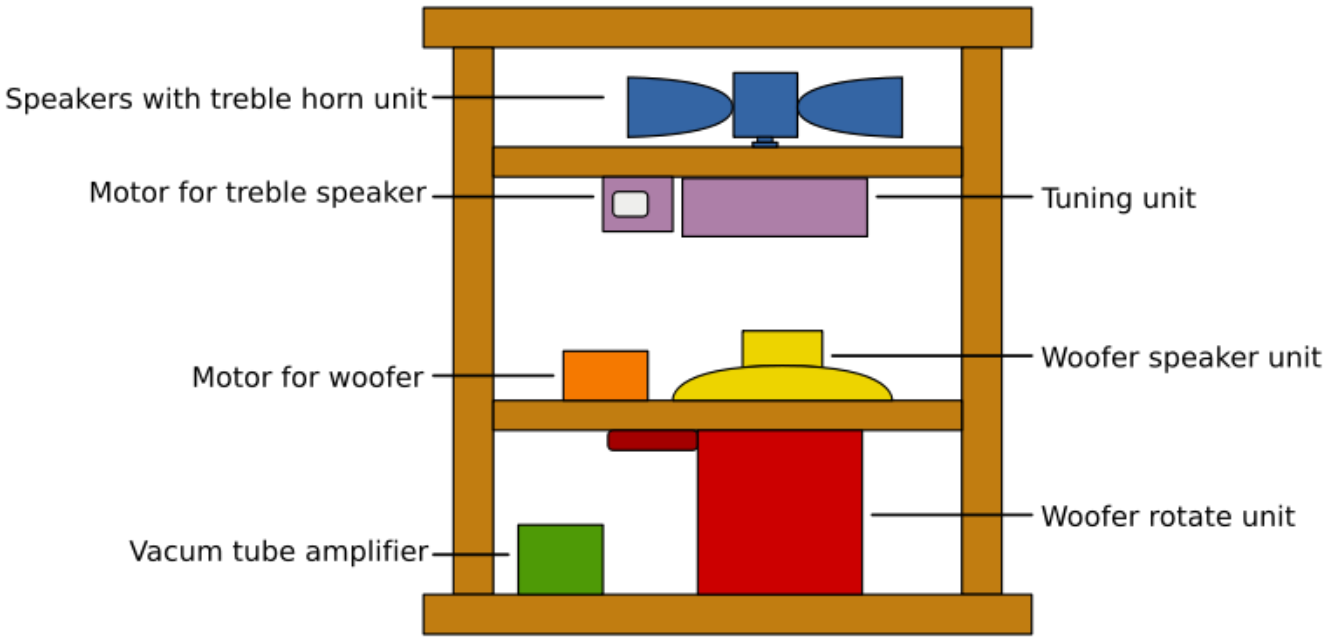
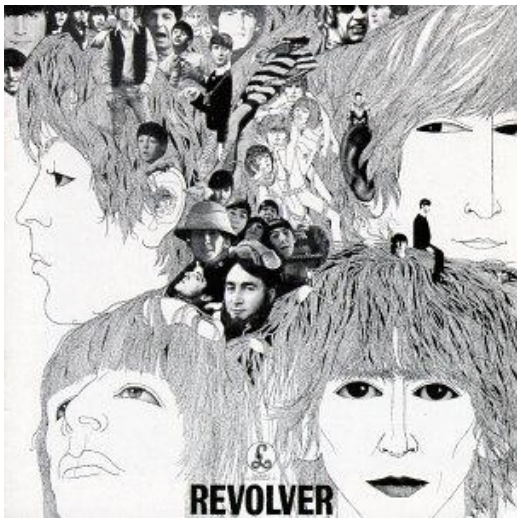


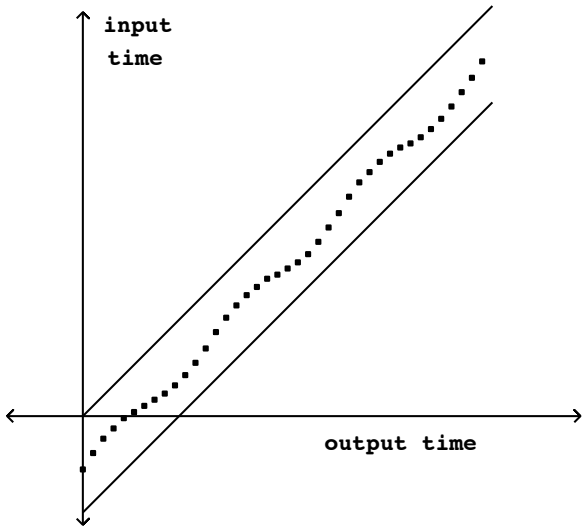
Image from Wikipedia

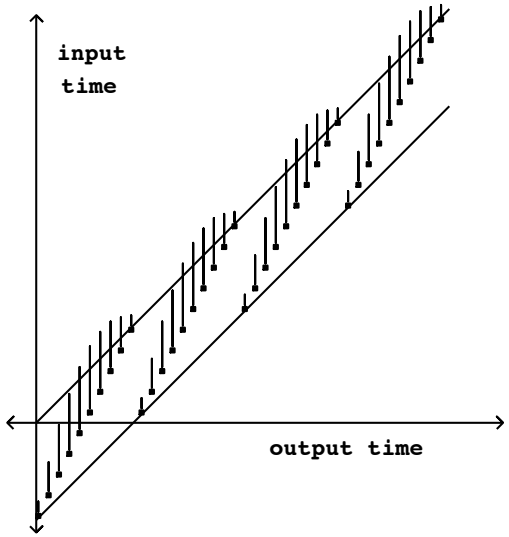
- Treble Horn and Woofer Rotate Units move
- Treble Motor/Tuning Units and Woofer Motor/ Speaker Units are stationary
- Examples
  - organ tone with variable speaker settings
  - vocal effect on "Tomorrow Never Knows"

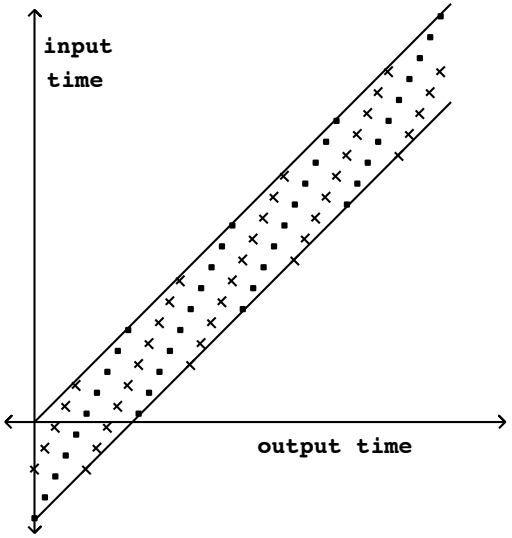


Pitch shifting

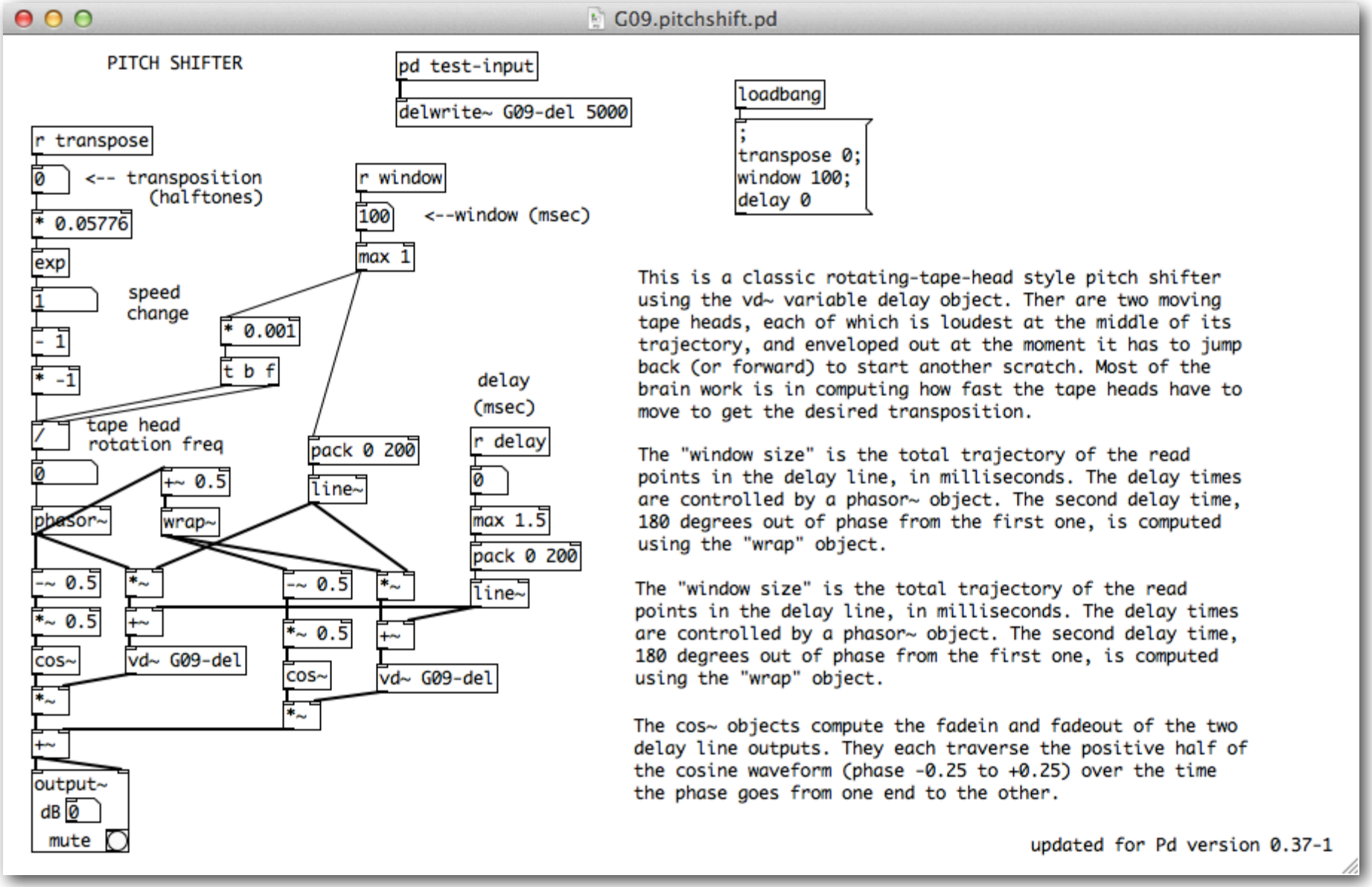
- Need to stay within minimum and maximum allowed delay
  - vibrato
  - piecewise delay with enveloping
  - overlaying two delay lines







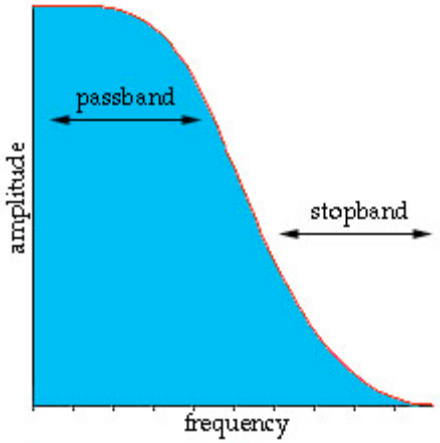
Pitch shifting



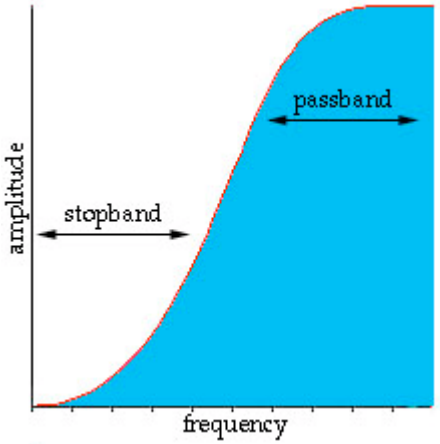
Introduction to Filters

Cut-off frequency - point above or below which frequencies are attenuated

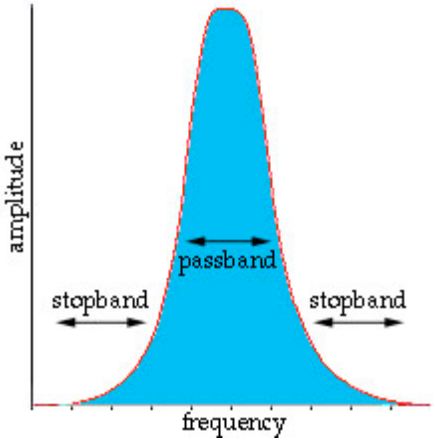
low-pass filter



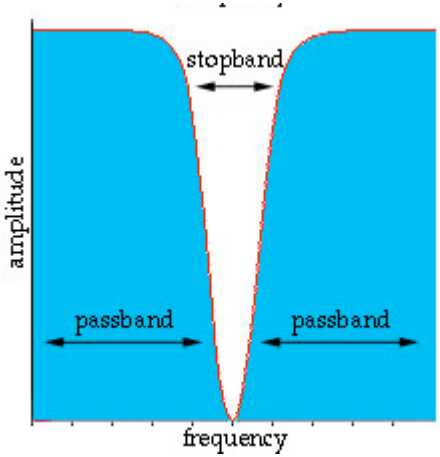
high-pass filter



band-pass filter

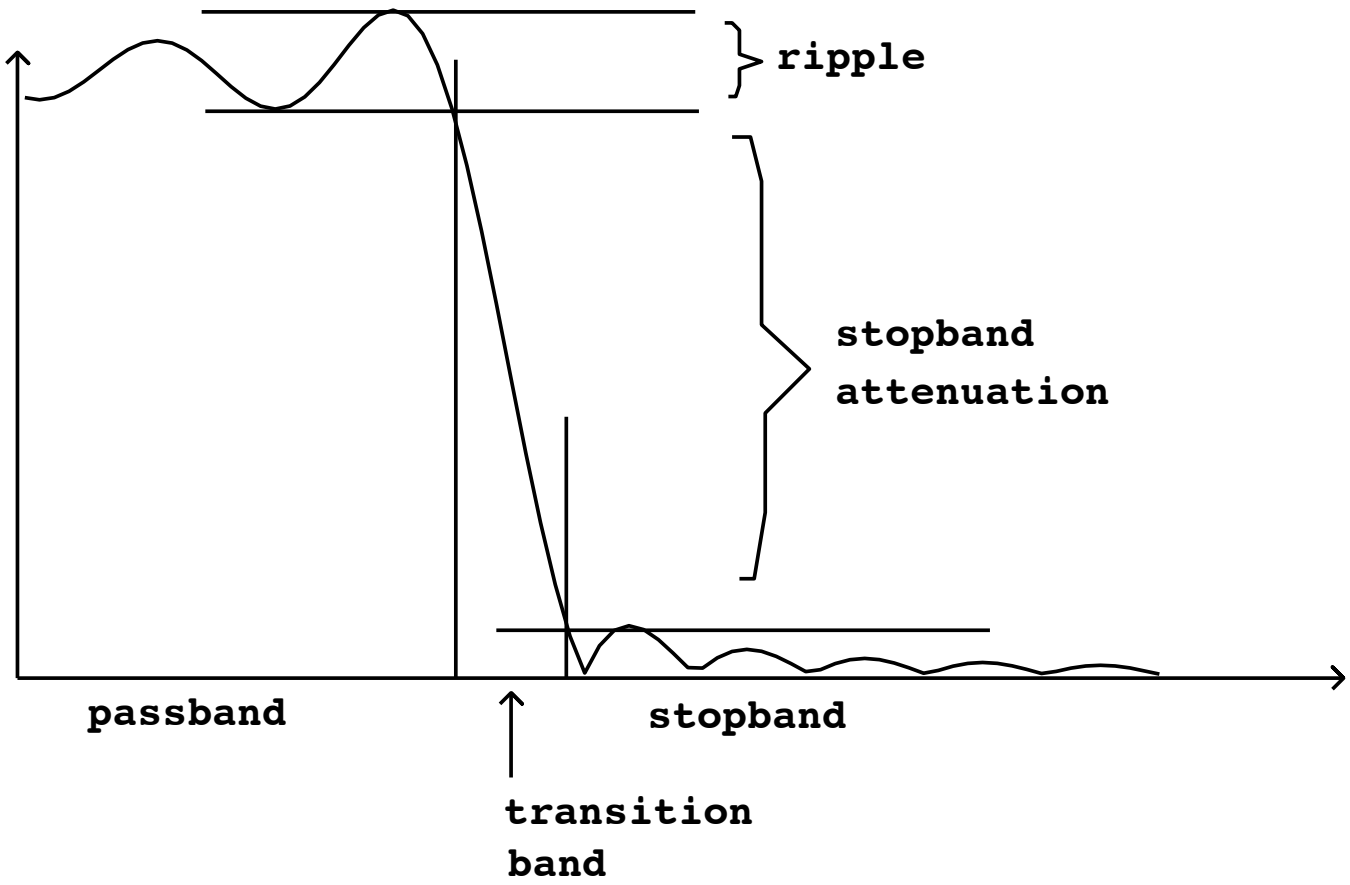


band-stop filter



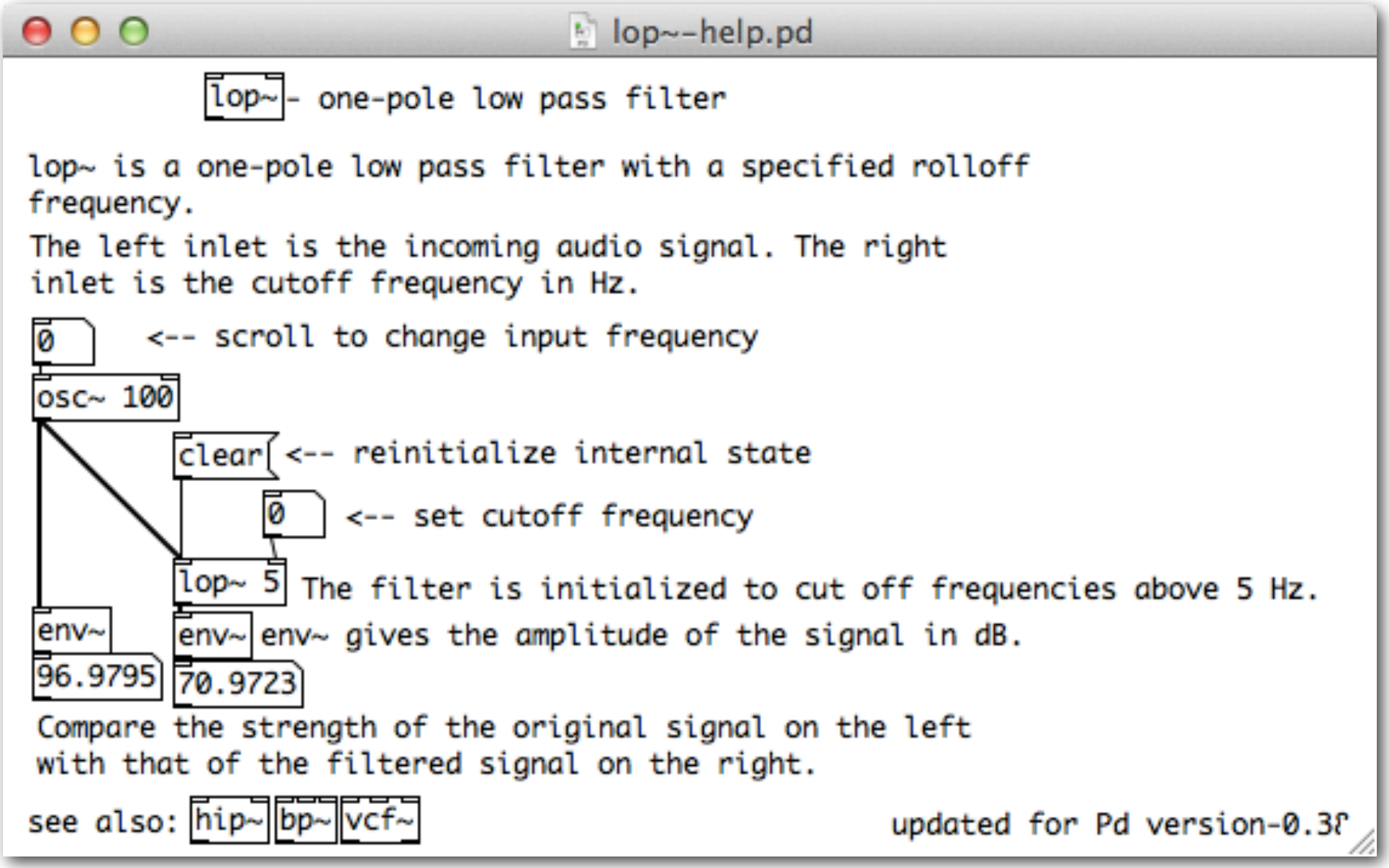
Images from: Burk, Polansky, Repetto, Roberts, and Rockmore. 2011. Music and Computers: A Theoretical and Historical Approach

Introduction to Filters

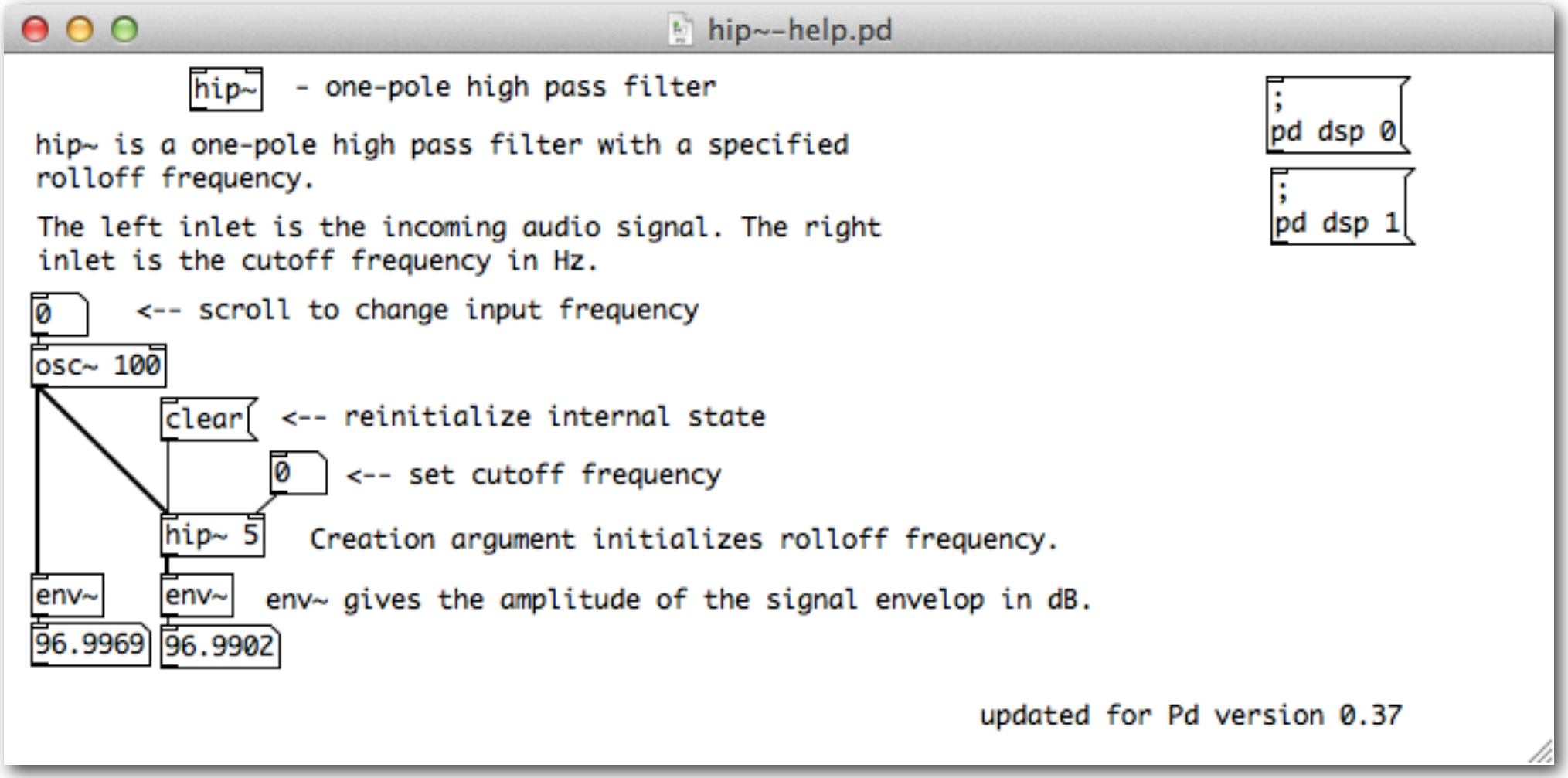


Low-pass filter frequency response in more detail

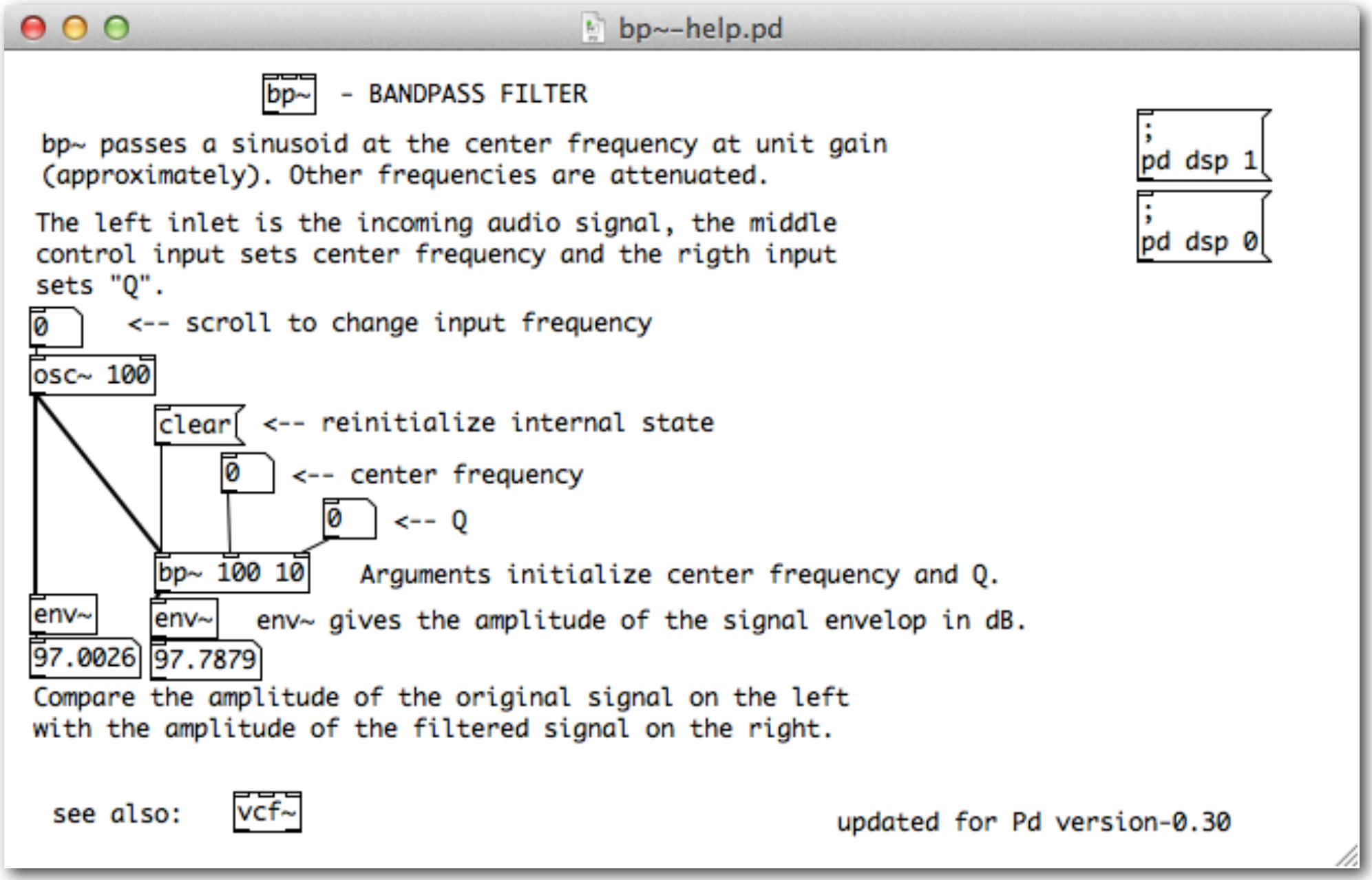
lop~


lop~-help.pd
lop~ - one-pole low pass filter
lop~ is a one-pole low pass filter with a specified rolloff frequency.
The left inlet is the incoming audio signal. The right inlet is the cutoff frequency in Hz.
0
<-- scroll to change input frequency
osc~ 100
clear
<-- reinitialize internal state
0
<-- set cutoff frequency
lop~ 5
The filter is initialized to cut off frequencies above 5 Hz.
env~
env~
env~ gives the amplitude of the signal in dB.
96.9795
70.9723
Compare the strength of the original signal on the left with that of the filtered signal on the right.
see also:
hip~
bp~
vcf~
updated for Pd version-0.3

hip~


hip~-help.pd
hip~ - one-pole high pass filter
hip~ is a one-pole high pass filter with a specified rolloff frequency.
The left inlet is the incoming audio signal. The right inlet is the cutoff frequency in Hz.
; pd dsp 0
; pd dsp 1
0
<-- scroll to change input frequency
osc~ 100
clear
<-- reinitialize internal state
0
<-- set cutoff frequency
hip~ 5
Creation argument initializes rolloff frequency.
env~
env~
env~ gives the amplitude of the signal envelop in dB.
96.9969
96.9902
updated for Pd version 0.37

bp~


bp~-help.pd
bp~ - BANDPASS FILTER
bp~ passes a sinusoid at the center frequency at unit gain (approximately). Other frequencies are attenuated.
The left inlet is the incoming audio signal, the middle control input sets center frequency and the rigth input sets "Q".
; pd dsp 1
; pd dsp 0
0
<-- scroll to change input frequency
osc~ 100
clear
<-- reinitialize internal state
0
<-- center frequency
0
<-- Q
bp~ 100 10
Arguments initialize center frequency and Q.
env~
env~
env~ gives the amplitude of the signal envelop in dB.
97.0026
97.7879
Compare the amplitude of the original signal on the left with the amplitude of the filtered signal on the right.
see also:
vcf~
updated for Pd version-0.30

Introduction to Filters