DaisySP

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adenv

Author: shensley

```
Trigger-able envelope with adjustable min/max, and independent per-segment time control.
```

TODO: - Add Cycling - Implement Curve (its only linear for now). - Maybe make this an AD__sr that has AD/AR/A__sr modes.

Envelope Segments

Distinct stages that the phase of the envelope can be located in. - IDLE = located at phase location 0, and not currently running - ATTACK = First segment of envelope where phase moves from MIN value to MAX value - DECAY = Second segment of envelope where phase moves from MAX to MIN value - LAST = The final segment of the envelope (currently decay)

```
enum {
        ADENV_SEG_IDLE,
        ADENV_SEG_ATTACK,
        ADENV_SEG_DECAY,
        ADENV_SEG_LAST,
};
```

init

Initializes the ad envelope

float sample__rate - sample rate of the audio engine being run.

Defaults - current segment = idle - curve = linear - phase = 0 - min = 0 - max = 1

```
    void init(float sample_rate);
```

process

processes the current sample of the envelope. Returns the current envelope value. This should be called once per sample period.

```
    float process();
```

trigger

Starts or retriggers the envelope.

```
    inline void trigger() {_trigger = 1; }
```

Setters

set_segment_time

Sets the length of time(secondsVERIFYTHIS) for a specific segment.

```
inline void set_time(uint8_t seg, float time)
```

set_curve_scalar

Sets the amount of curve applied. Input range: -1 to 1. - At -1, curve = full logarithmic - At 1, curve = full exponential - At 0, curve = linear

```
inline void set_curve_scalar(float scalar) { _curve_scalar = scalar; }
```

set_min

Sets the minimum value of the envelope output Input range: -FLT_MAX, to FLT_MAX

```
inline void set_min(float min) {_min = min; }
```

set_max

Sets the maximum value of the envelope output Input range: -FLT_MAX, to FLT_MAX

```
inline void set_max(float max) {_max = max; }
```

Getters

current_segment

Returns the segment of the envelope that the phase is currently located in.

```
inline void current_segment();
```

Crossfade Performs a crossfade between two signals

Original author: Paul Batchelor

Ported from Soundpipe by Andrew Ikenberry added curve option for constant power, etc. initialization processing set position between two signals. range: 0-1 set curve of crossfade.

dcblock

Removes DC component of a signal

init

Initializes dcblock module

```
        void init(float sample_rate);
```

process

performs dcblock process

```
        float process(float in);
```

decimator

Performs downsampling and bitcrush effects

init

Initializes downsample module

```
void init();
```

process

Applies downsample and bitcrush effects to input signal. Returns one sample.
This should be called once per sample period.

```
float process(float input);
```

Setters

set_downsample_factor

Sets amount of downsample Input range:

```
inline void set_downsample_factor (float downsample_factor)
```

set_bitcrush_factor

Sets amount of bitcrushing Input range:

```
inline void set_bitcrush_factor (float bitcrush_factor)
```

Getters

get_downsample_factor

Returns current setting of downsample

```
inline float get_downsample_factor () { return _downsample_factor; }
```

get_bitcrush_factor

Returns current setting of bitcrush

```
inline float get_bitcrush_factor () { return _bitcrush_factor; }
```

Simple Delay Line November 2019 By: shensley

User manages their own buffer in order to provide flexibility without dynamic memory.

TODO: add some sort of type flexibility – for now its just floats. dsy_pstream.h

Ported from Csound - October 2019

Definitions from Csound: pvsanal pvsfread pvsynth pvsadsyn pvscross pvsmaska = (overloaded, – should probably just pvsset(&src, &dest))

More or less for starting purposes we really only need pvsanal, and pvsynth to get started

line

creates a line segment signal

init

Initializes line module.

```
void init(float sample_rate);
```

process

Processes line segment. Returns one sample. Expected input:

```
float process(uint8_t *finished);
```

start

Begin creation of line. Arguments: - start - beginning value - end - ending value - dur - duration in seconds of line segment

```
void start(float start, float end, float dur);
```

metro

Creates a clock signal at a specific frequency.

init

Initializes metro module.

Arguments: - freq: frequency at which new clock signals will be generated Input Range: - sample_rate: sample rate of audio engine Input range:

```
void init(float freq, float sample_rate);
```

process

checks current state of metro object and updates state if necesary.

```
uint8_t process();
```

Setters

set_freq

Sets frequency at which metro module will run at.

```
void set_freq(float freq);
```

Getters

get_freq

Returns current value for frequency.

```
inline float get_freq() { return freq_; }
```

nlfilt

port by: stephen hensley, December 2019

Non-linear filter.

The four 5-coefficients: a, b, d, C, and L are used to configure different filter types.

Structure for Dobson/Fitch nonlinear filter

Revised Formula from Risto Holopainen 12 Mar 2004

`Y{n} =tanh(a Y{n-1} + b Y{n-2} + d Y^2{n-L} + X{n} - C)`

Though traditional filter types can be made, the effect will always respond differently to different input.

This Source is a heavily modified version of the original source from Csound.

TODO:

- make this work on a single sample instead of just on blocks at a time.

init

Initializes the nlfilt object.

```
void init();
```

process

Process the array pointed to by *in and updates the output to *out;

This works on a block of audio at once, the size of which is set with the size.

```
void process_block(float *in, float *out, size_t size);
```

setters

set_coefficients

inputs these are the five coefficients for the filter.

```
inline void set_coefficients(float a, float b, float d, float C, float L)
```

individual setters for each coefficients.

```
inline void set_a(float a) { a_ = a; }

inline void set_b(float b) { b_ = b; }

inline void set_d(float d) { d_ = d; }

inline void set_C(float C) { C_ = C; }
inline void set_L(float L) { L_ = L; }
```

oscillator

```
 Synthesis of several waveforms, including polyBLEP bandlimited waveforms.
```

example:

```
daisysp::oscillator osc;
init()
{
    osc.init(SAMPLE_RATE);
    osc.set_frequency(440);
    osc.set_amp(0.25);
    osc.set_waveform(WAVE_TRI);
}

callback(float *in, float *out, size_t size)
{
    for (size_t i = 0; i < size; i+=2)
    {
        out[i] = out[i+1] = osc.process();
    }
}
```

Waveforms

Choices for output waveforms, POLYBLEP are appropriately labeled. Others are naive forms.

```
        enum
        {
            WAVE_SIN,
            WAVE_TRI,
            WAVE_SAW,
            WAVE_RAMP,
            WAVE_SQUARE,
            WAVE_POLYBLEP_TRI,
            WAVE_POLYBLEP_SAW,
            WAVE_POLYBLEP_SQUARE,
            WAVE_LAST,
        };
```

init

Initializes the oscillator

float samplerate - sample rate of the audio engine being run, and the frequency that the process function will be called.

Defaults: - freq = 100 Hz - amp = 0.5 - waveform = sine wave.

```
void init(float samplerate)
```

set_freq

Changes the frequency of the oscillator, and recalculates phase increment.

```
inline void set_freq(const float f)
```

set_amp

Sets the amplitude of the waveform.

```
inline void set_amp(const float a) { amp = a; }
```

set_waveform

Sets the waveform to be synthesized by the process() function.

```
inline void set_waveform(const uint8_t wf) { waveform = wf < WAVE_LAST ? wf : WAVE_S
```

process

Processes the waveform to be generated, returning one sample. This should be called once per sample period.

```
float process();
```

phasor

Generates a normalized signal moving from 0-1 at the specified frequency.

TODO: I'd like to make the following things easily configurable: - Selecting which channels should be initialized/included in the sequence conversion. - Setup a similar start function for an external mux, but that seems outside the scope of this file.

init

Initializes the phasor module

sample rate, and freq are in Hz

initial phase is in radians

```
inline void init(float sample_rate, float freq, float initial_phase)
```

process

processes phasor and returns current value

```
float process();
```

Setters

set_freq

Sets frequency of the phasor in Hz

```
void set_freq(float freq);
```

Getters

get_freq

Returns current frequency value in Hz

```
inline float get_freq() { return freq_; }
```

port

Applies portamento to an input signal. At each new step value, the input is low-pass filtered to move towards that value at a rate determined by ihtim. ihtim is the “half-time” of the function (in seconds), during which the curve will traverse half the distance towards the new value, then half as much again, etc., theoretically never reaching its asymptote.

This code has been ported from Soundpipe to DaisySP by Paul Batchelor.

The Soundpipe module was extracted from the Csound opcode “portk”.

Original Author(s): Robbin Whittle, John ffitch

Year: 1995, 1998

Location: Opcodes/biquad.c

init

Initializes port module

Arguments:

- sample_rate: sample rate of audio engine
- htime: half-time of the function, in seconds.

```
void init(float sample_rate, float htime);
```

process

Applies portamento to input signal and returns processed signal.

```
float process(float in);
```

Setters

set_htime

Sets htime

```
inline void set_htime(float htime) { htime_ = htime; }
```

Getters

get_htime

returns current value of htime

```
inline float get_htime() { return htime_; }
```

reverbsc

`Stereo Reverb`

Ported from soundpipe

example:

daisysp/modules/examples/ex_reverbsc

init

Initializes the reverb module, and sets the samplerate at which the process function will be called.

```
void init(float samplerate);
```

process

process the input through the reverb, and updates values of out1, and out2 with the new processed signal.

```
void process(float in1, float in2, float *out1, float *out2);
```

set_feedabck

controls the reverb time. reverb tail becomes infinite when set to 1.0

range: 0.0 to 1.0

```
inline void set_feedback(float fb) { _feedback = fb; }
```

set_lpfreq

controls the internal dampening filter's cutoff frequency.

range: 0.0 to samplerate / 2

```
inline void set_lpfreq(float freq) { _lpfreq = freq; }
```

svf

```
Double Sampled, Stable State Variable Filter
```

Credit to Andrew Simper from musicdsp.org

This is his “State Variable Filter (Double Sampled, Stable)”

Additional thanks to Laurent de Soras for stability limit, and Stefan Diedrichsen for the correct notch output

Ported by: Stephen Hensley

example: daisysp/examples/svf/

init

Initializes the filter

float samplerate - sample rate of the audio engine being run, and the frequency that the process function will be called.

```
void init(float samplerate);
```

process

Process the input signal, updating all of the outputs.

```
void process(float in);
```

Setters

set_freq

sets the frequency of the cutoff frequency.

f must be between 0.0 and samplerate / 2

```
void set_freq(float f);
```

set_res

sets the resonance of the filter.

Must be between 0.0 and 1.0 to ensure stability.

```
void set_res(float r);
```

set_drive

sets the drive of the filter, affecting the response of the resonance of the filter..

```
inline void set_drive(float d) { _drive = d; }
```

Filter Outputs

Lowpass Filter

```
inline float low() { return _out_low; }
```

Highpass Filter

```
inline float high() { return _out_high; }
```

Bandpass Filter

```
inline float band() { return _out_band; }
```

Notch Filter

```
inline float notch() { return _out_notch; }
```

Peak Filter

```
inline float peak() { return _out_peak; }
```