Gamma: C++ Generic Synthesis Library
Tutorial
Version 0.9.5

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Contents

1 Gamma Overview

Gamma is a cross-platform, C++ library for doing generic synthesis and filtering of numerical data. It contains numerous mathematical functions related to synthesis together with an assortment of sequence generators and filters for signal processing. It is oriented towards real-time sound and graphics rendering, but is equally useful for non-real-time tasks.

1.1 Generic Types

Gamma objects are templated on their value type allowing them to be used with any arithmetic data types, such as scalars, vectors, and complex numbers. Processing algorithms are designed as much as possible to be data type and domain independent. This allows a common set of functions and objects to be used for a variety of modalities, such as sound or graphics, and in arbitrary domains, such as time, space, or frequency. Algorithms are type generic which means they can process any type of data that has the appropriate mathematical operators defined (typically +, -, *, and /). The following illustrates how a variety of sequence generators can be constructed from a generic recursive multiplication class, `RMul`:

```
RMul<float      > rMul(0.99);                   // decaying envelope
RMul<Vec3<>     > rMul(Vec3<>(0.9,0.8,0.7));     // 3 decaying envelopes
RMul<Complex<> > rMul(Polar<>(1.00, M_2PI/8));  // complex sinusoid
RMul<Complex<> > rMul(Polar<>(0.99, M_2PI/8));  // complex exponential
```

1.2 Generators and Filters

Processing objects are typically either a generator or a filter and can be either domain observers or not. Objects adopt a standard processing operator interface similar to functional syntax. Generators return their next value with the `()` operator and filters return their next value with the `(T)` operator, where `T` is a value type.

```
gen::RAdd rAdd(1, 0);                         // recursively add 1 starting at 0
for(int i=0; i<4; ++i) cout << rAdd();        // prints "1 2 3 4"

fil::Delay1 delay1(0);                        // 1 element delay starting with 0
for(int i=1; i<5; ++i) cout << delay1(i);     // prints "0 1 2 3"
```

1.3 Domain Synchronization

A sampling domain is a discretely-spaced independent variable along which a function is evaluated. In Gamma, domains are abstracted through an observer design pattern into a `Sync` subject and a `Synced` observer. The `Sync` holds a sampling rate/interval amount and notifies its attached `Synced`s whenever it changes value. The following example shows how we can create an audio sample rate subject with an observer:

```
struct MySynced : public Synced {

    // This is called whenever the sampling domain changes
    // 'dr' is the new SPU divided by the old SPU
    void onResync(double dr);
};

Sync sync;
sync.spu(44100);    // set samples/unit (in our case 44,100 samples/sec)

MySynced synced;
synced.spu();       // returns 1 (the default)
sync << synced;     // make 'synced' an observer of 'sync'
synced.spu();       // returns 44,100
sync.spu(8000);     // change the sample rate, calls onResync of 'synced'
synced.spu();       // returns 8000
```

2 Obtaining and Building

The Gamma source code can be downloaded from http://mat.ucsb.edu/gamma/. This is where to get the most recent *stable* releases. To obtain a more current, but possibly (and most likely) unstable version, you can check out the source anonymously through SVN via

```
svn co https://svn.mat.ucsb.edu/svn/gamma/trunk gamma
```

The simplest way to build the library is to use GNU Make. Instructions for setting up each platform with Make follow. Further instructions can be found in the `README` file in the root directory.

2.1 Linux Compilation

The simplest way to build the library is to use GNU Make (see section 2.4 below). Use either Synaptic or `apt-get` to install the most recent developer versions of libsndfile and PortAudio (v19). If using `apt-get`, you will do something like

```
sudo apt-get update
sudo apt-get install portaudio19-dev libsndfile1-dev
```

Gamma has also been tested to work with Puredyne, a Linux live CD distribution that conveniently bundles with it the important media-related libraries. Unfortunately, it does not come with `g++` installed, so you have to install it manually on boot-up:

```
sudo apt-get update
sudo apt-get install g++-4.4
```

2.2 Mac OSX Compilation

You can build the library using either GNU Make or Xcode. If you are using Make, see section 2.4 below. To use Xcode, you will need to install the developer tools from Apple. You can get them for free (after registration) from http://developer.apple.com/. The Xcode project is located at `project/xcode/gamma.xcodeproj`.

2.3 Windows Compilation

There is currently no provided Visual Studio project. Visual Studio Express can be downloaded for free from Microsoft. It will be necessary to link to libsndfile and PortAudio v19. Alternatively, you can use Cygwin or MinGW which provide a Linux-like environment natively from within Windows. They can be used to build the Gamma sources using their provided version of GNU Make. These methods have not been thoroughly tested, so you can contribute to Gamma if you inform the author of your working solution.

2.4 Building With Make

If you had to install Make, test that it is working by opening a terminal and typing 'make –version'. Before running Make, ensure that the correct build options are set in the file `Makefile.config`. These can be set directly in `Makefile.config` or passed in as options to Make as `OPTION=value`. If you are using Linux/Puredyne, you will also need to uncomment the line `USING_PUREDYNE = 1`. To build the Gamma library, run `make` and hope for the best.

3 Gamma Objects

3.1 Generators

Generators produce a continuous stream of samples such as a periodic function or a continuous curve. They are *function objects* in that they store some internal state and act just like a mathematical function, both syntactically and semantically. Given some generator, g, we can command it to generate its next sample by calling $g()$. Note that this call looks and acts just like a nullary mathematical function, that is, g is the function and $g()$ evaluates the function returning its output. It differs from a function, however, in that it may return a different value each time it is called. Since generators are also *objects*, many also have parameters to control their output. The following code example shows a typical use case of a generator, `Gen`, setting its parameters and generating samples.

```
Gen g;           // define a new generator
g.freq(100);     // set its frequency parameter
g.phase(0.25);   // set its phase parameter

float v;
v = g();         // generate first sample
v = g();         // generate second sample
```

Since generators simply return a sample when evaluated, one can form algebraic expressions with them using standard C++ operators. The next example shows how two oscillators can be summed together and then multiplied by an envelope.

```
Osc osc1, osc2;                 // define two oscillators
Env env;                        // define an envelope

float v;
v = (osc1() + osc2()) * env();  // generate first sample
v = (osc1() + osc2()) * env();  // generate second sample
```

Oscillators

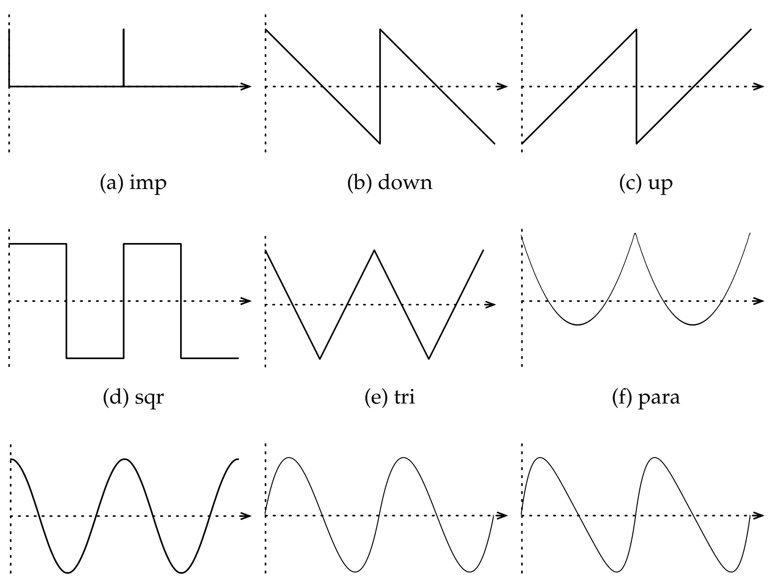
[generator/(oscAccum, oscSweep, oscLFO1, oscLFO2, oscOsc, oscImpulse)]

Oscillators produce a periodic waveform with adjustable phase and frequency. They typically have `freq()` and `phase()` methods for setting their frequency and phase.

`Accum` accumulates an internal fixed-point phase at a specified frequency. Requiring only a single fixed-point addition per iteration, it is the fastest generator. It can be used as a timer or as the input to a wave-shaping table or function. **`Sweep`** is an `Accum` that cycles linearly through the interval [0, 1) at a specified frequency.

`Sine` is the "go to" sine wave oscillator. It produces a high-quality sine wave using a polynomial approximation.

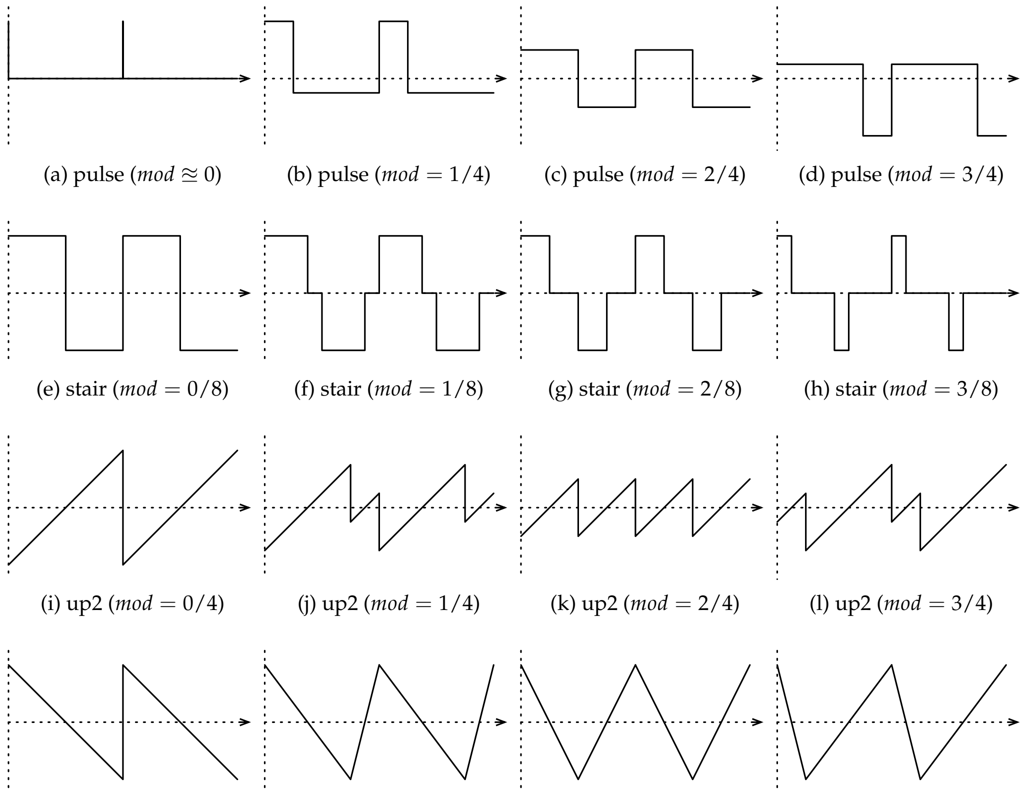
`LFO` (low-frequency oscillator) produces various piece-wise polynomial waveforms derived from the phase of an `Accum`. Since the waveforms are not band-limited, they should only be used as low-frequency modulation sources. Figure 1 shows the static waveforms that can be produced.



(a) imp (b) down (c) up (d) sqr (e) tri (f) para (g) cos (h) even3 (i) even5

Figure 1: `LFO` static waveforms (two cycles shown)

Figure 2 shows the dynamic waveforms whose shape can be changed smoothly with the `mod` parameter.



(a) pulse ($mod \approxeq 0$) (b) pulse ($mod = 1/4$) (c) pulse ($mod = 2/4$) (d) pulse ($mod = 3/4$)

(e) stair ($mod = 0/8$) (f) stair ($mod = 1/8$) (g) stair ($mod = 2/8$) (h) stair ($mod = 3/8$)

(i) up2 ($mod = 0/4$) (j) up2 ($mod = 1/4$) (k) up2 ($mod = 2/4$) (l) up2 ($mod = 3/4$)

(m) line2 ($mod = 0/4$) (n) line2 ($mod = 1/4$) (o) line2 ($mod = 2/4$) (p) line2 ($mod = 3/4$)

Figure 2: `LFO` dynamic waveforms (two cycles shown)

`Osc` is a periodic function generator that reads values from a stored function table. `Osc` is the most general type of oscillator as any waveform can be stored in its table. Through its template parameters, one can specify the table element type, interpolation mode, and read tap behavior. An `Osc` can either use its own internal table or reference an existing table. The following illustrates how to create a linearly-interpolation `Osc` with its own sine wave table.

```
Osc<float, ipl::Linear> src(
    100,    // frequency
    0,      // phase
    512     // table size (must be power of 2)
);

// fill table with a sine wave
for(int i=0; i<src.size(); ++i)
    src[i] = sin(i*M_2PI / src.size());
```

Since often times several oscillators need to share the same table, `Osc` can also reference an externally allocated table. External tables can either be an `ArrayPow2` or another `Osc`.

```
Osc<> oscRef;                // to reference external table
```

```
Osc<> oscOwn(100,0,512);     // has own table
ArrayPow2 table(512);        // declare a wavetable

oscRef.source(table);        // use an ArrayPow2
oscRef.source(oscOwn);       // use another Osc table
```

Impulse generates a band-limited impulse train having either even or odd harmonics. This is accomplished by using a Chebyshev polynomial of the second kind, U_n, where

$$U_n(\cos\theta) = \frac{\sin((n+1)\theta)}{\sin(\theta)} = \sum_{k=0}^{\lfloor n/2 \rfloor} \cos([2k + n \bmod 1]\theta).$$

Saw generates a band-limited saw wave by integrating a band-limited impulse wave.

Square generates a band-limited square wave by integrating a band-limited odd-harmonic impulse wave.

DSF (discrete summation formula) produces a band-limited harmonic series with variable amplitude and frequency ratios. The amplitude ratio determines the amplitude scaling factor of harmonics as frequency increases. The frequency ratio determines the spacing between harmonics with respect to the fundamental frequency.

CSine produces a complex sinusoid using recursive complex multiplication. This generator is useful for creating frequency shifting effects.

Sample Playback

[generator/player]

Player is used for playback and looping of sound samples. In the simplest scenario, its constructor takes a path to a sound file. Template parameters can be used to specify its interpolation type and looping mode. If the playback rate is to be fixed at 1, it is best to use truncating (no) interpolation. The following illustrates some typical uses of `Player`:

```
// Looping sample player using cubic interpolation
Player<float, ipl::Cubic, tap::Wrap> looper;
looper.load("pathto/MyDrumLoop.wav");           // load a sound file

// One-shot sample player using truncating interpolation
Player<float, ipl::Trunc, tap::Clip> oneshot("pathto/MySound.aif");

float s1 = looper();    // read sample; loops when end is reached
float s2 = oneshot();   // read sample; stays at end when reached
```

`Player` can also handle multi-channel samples. To do so, one must first read the channel samples, then advance the read tap.

```
Array<float> stereoBuffer;           // 2-channel deinterleaved buffer
Player<> p;
p.buffer(stereoBuffer, 44100, 2);   // have player read from the external buffer
```

```
float s1 = p.read(0);                // read left channel sample
float s2 = p.read(1);                // read right channel sample
p.advance();                         // advance read tap
```

Noise

[generator/noise]

The noise generators produce three different spectra classified by their falloff amount. **NoiseWhite** falls off at 0 dB/octave (uniformly distributed), **NoisePink** falls off at 3 dB/octave, and **NoiseBrown** falls off at 6 dB/octave.

Envelopes

[generator/(envDecay, envAD, envSeg, envSegNoise, envSegExp)]

Envelopes (or envelope generators) are used to control low-frequency modulation of a source signal's amplitude. Envelopes typically start at zero amplitude, vary continuously within the interval [0, 1], and end at zero. Envelope classes share two common methods—`done()` which indicates whether the envelope is finished and `reset()` which starts the envelope over at the beginning.

Decay produces an exponentially decaying envelope given some starting value. Many naturally occurring sounds have an exponentially decaying amplitude. This envelope is very fast to compute only requiring a single multiply per sample. Since it does not actually reach zero, `done()` returns true when the amplitude falls below 0.001 (-60 dB).

Env is a continuous piece-wise exponential curve envelope with N number of segments. It consists of three attribute arrays—`lengths()`, `curves()`, and `levels()`. The size of lengths and curves is N and the size of levels is $N + 1$. Lengths stores the length or duration of each segment. Curves stores the exponential curvature, c, of each segment where $c < 0$ approaches rapidly, $c = 0$ is linear, and $c > 0$ approaches slowly. Natural sounding envelopes will have $c < 0$. Levels stores the amplitude amounts at the endpoints of each segment. Setting levels 1 and 2, for instance, sets the endpoint levels of segment 1. `Env` has a special level point called `sustainPoint()` that instructs the envelope to hold at this level until `release()` is called. The sustain can be disabled by setting `sustainPoint()` to N or by calling `sustainDisable()`.

AD is an `Env` that generates a two segment attack-decay envelope. It is useful for synthesizing simple percussive-like sounds. The lengths are [attack, decay] and the levels are [0, 1, 0].

ADSR is an `Env` that generates a three segment attack-decay-sustain-release envelope. This envelope is useful for simulating the transient, steady-state, and release portions of an instrument. The lengths are [attack, decay, release] and the levels are [0, 1, sustain, 0]. Its prototypical shape is shown in Fig. 3.

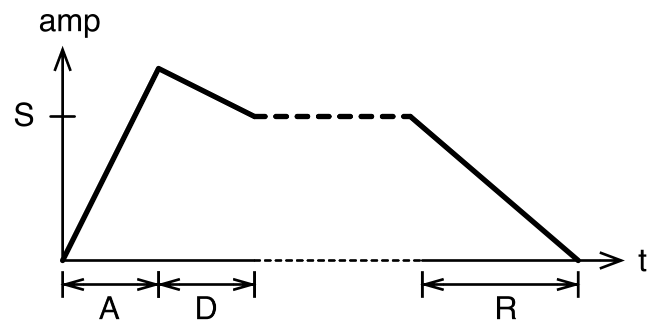


Figure 3: An ADSR prototypical envelope shape.

Seg generates an envelope by interpolating between a sequence of breakpoints. A `Seg` can be used to downsample another generator, such as a noise generator, to produce a more slowly varying curve. The following example shows how to create a noise signal made of line segments (AKA band-limited noise):

```
Seg<float, iplSeq::Linear> seg(1.); // linear segment with length of 1
NoiseWhite<> noise;                 // white (uniform) noise generator

float s = seg(noise);               // generate next linear-piecewise white noise
```

SegExp is like `Seg` but generates an exponential curve between breakpoints.

3.2 Filters

[filter/freqResponses]

A filter applies a particular transformation to a stream of input samples to produce a new stream of output samples. Filters behave identically to the generators discussed above (3.1) with one important difference—they act like unary functions rather than nullary functions. The following example shows how the output of a generator is filtered.

```
Gen g;       // define a generator
Filter f;    // define a filter

float v;
v = g();     // generate first sample
v = f(v);    // filter first sample
v = f(g()); // generate and filter second sample
```

FIR

An FIR (finite impulse response) filter produces new values from a linear combination of its previous inputs. It is relatively easy to construct FIR filters with a linear-phase response by using symmetrical coefficients. Linear-phase responses have the desirable characteristic of not distorting the phase of the input signal. The disadvantage of FIR filters is that very high orders are required to obtain steep frequency responses.

Because of this, it often becomes necessary to implement FIRs using an FFT.

`MovingAvg` is a moving average filter that outputs the average of its previous N input values. It acts as a low-pass filter and does very well at preserving the shape of the input signal (due to its linear-phase response). It is quite unique amongst FIRs in that it can be implemented with computational complexity $O(1)$, independent of its order. Plots of its frequency response magnitude are shown in Fig. 4.

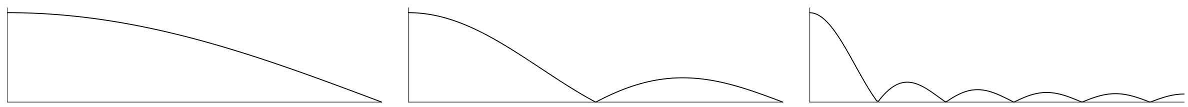


Figure 4: `MovingAvg` frequency response magnitudes for orders $N = 1, 3, 10$ (left to right).

`Notch` is a two-zero notch filter with linear-phase. The notch center frequency, f_c, can be controlled, but its bandwidth cannot. Plots of its frequency response magnitude are shown in Fig. 5. As can be seen, when $f_c < f_s/4$, the filter has a high-pass characteristic and when $f_c > f_s/4$, it has a low-pass characteristic.

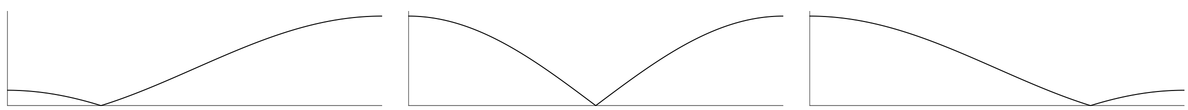


Figure 5: `Notch` frequency response magnitudes for orders $f_c = \frac{1}{8}f_s, \frac{1}{4}f_s, \frac{3}{8}f_s$ (left to right).

IIR

[filter/(onePole, biquad, blockDC)]

An IIR (infinite impulse response) filter produces new values from a linear combination of its previous inputs and outputs. IIRs can produce steep "ringing" frequency responses much more efficiently than FIRs. Their main disadvantage is that they have a non-linear phase response which can smear sharp transients in the input signal if one is not careful. Also, higher-order filters are generally not numerically stable so one typically needs to cascade several first- or second-order filters in series.

`OnePole` is a single pole filter that is useful for smoothing and integrating signals. The cut-off frequency is the frequency at which the magnitude spectrum is -3 dB. Some plots of its frequency response magnitude are shown in Fig. 6.

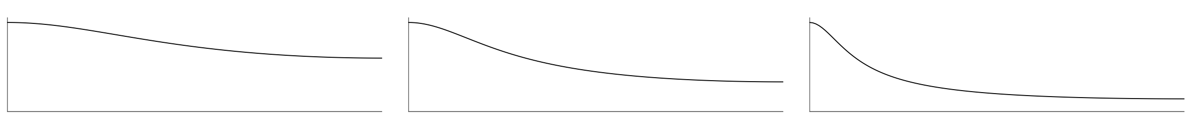


Figure 6: `OnePole` frequency response magnitudes for $f_c = 0.22f_s, 0.11f_s, 0.05f_s$ (left to right).

`Biquad` is a general-purpose second-order IIR filter. It conveniently provides resonant filter designs that can operate in low-pass, high-pass, band-pass, band-reject, or all-pass modes. The resonance amount controls the "presence" or steepness of the filter at its center frequency. For the low- and high-pass varieties, resonance is proportional to the amount of ringing at the center frequency. A resonance setting of 1 results in a maximally-flat pass band without ringing, i.e. a Butterworth filter. For the band-pass and -reject varieties, resonance is inversely proportional to the band-width of the filter. For the all-pass variety, resonance controls the locality of the phase distortion at the center frequency.

BlockDC is used for eliminating DC bias from signals. Sometimes very low frequency components will sneak into a signal, i.e., from positive feedback loops, which can create unwanted distortion due to clipping on the DAC. A `BlockDC` can be put at the end of the signal chain to eliminate these low-frequency components without coloring the rest of the signal.

Allpass1 is a first-order all-pass filter. `Allpass1` acts like a single-sample delay, however, the delay for each frequency component can be a fractional amount less than 1. The phase is shifted from 0 to 180 degrees going from DC to Nyquist. The center frequency is the frequency where the phase is shifted by 90 degrees. It is easy to construct 1-pole/1-zero low- and high-pass filters by adding or subtracting, respectively, the output of `Allpass1` from the input. The low- and high-pass filters are very similar to 1-pole filters, but have a sharper fall-off as the magnitude goes to zero at Nyquist and DC, respectively. The high-pass filter can act as a DC blocker (identical to `BlockDC`) at low cutoff. For convenience, these filters are implemented as the methods `low()` and `high()`.

Allpass2 is a second-order all-pass filter. `Allpass2` creates a frequency-dependent phase transition from 0 to 360 degrees going from DC to Nyquist. The center frequency is the frequency where the phase is shifted by 180 degrees. It is easy to construct band-pass and -reject filters by adding or subtracting, respectively, the output of `Allpass2` from the input.

Delay Lines

[filter/(delay, comb), effects/flanger]

A delay line keeps a history of its previous inputs and/or outputs. Delay lines are very similar to FIR and IIR filters, except they typically have a history several orders of magnitude higher, like 1000 samples.

Delay is an all-pass filter whose only effect is to delay its input. Delays are starting points for building many different effects including vibrato, pitch-shifting, and reverberation. The delay time can be varied dynamically as there are several interpolation modes possible— `ipl::Linear`, `ipl::Cubic`, and `ipl::AllPass`. If the delay time is not to be varied, then it is best to use either the `ipl::Trunc` or `ipl::AllPass` modes as they will not color the magnitude spectrum. `Delay` has a `write()` method to place a sample at the beginning of the delay-line and a `read()` method that returns samples in the delay-line at specified delay amounts.

```
Delay<> delay(1., 0.1); // delay with max delay of 1 sec and delay of 0.1 sec

float s = ...           // an input sample
float d;
d = delay();            // read delayed sample, d
delay(s);               // write new sample, s, to delay

d = delay(s);           // shorthand of the above

d = delay();            // feed scaled output back into input
delay(s + d*0.5);       //
```

Comb is a special kind of delay line that operates more like an IIR filter. Comb filters are typical starting points for creating flanging, chorusing, echo, and reverberation effects. Feedforward creates notches in the spectrum while feedback creates resonant peaks. If the feedforward and feedback amounts are the same magnitude, but opposite in sign, then an all-pass comb filter results. The delay time can be varied in a smooth way as with `Delay`.

Hilbert transform

[filter/HilbertFilter]

`Hilbert` produces a complex-valued (analytic) signal from a real signal. Complex signals are required to implement single-side band modulation effects. A frequency shift is accomplished by multiplying a complex-valued signal (produced from a Hilbert filter) by a complex sinusoid whose frequency determines the amount of frequency shift.

3.3 Spectral Processing

Discrete Fourier Transform

[spectral/(RFFT, CFFT)]

There are two classes for performing the discrete Fourier transform, **`RFFT`** for real-to-complex transforms and **`CFFT`** for complex-to-complex transforms. The DFT can be any size, however, it will perform best when the size is a product of small primes (e.g., powers of two). Transforms occur in-place meaning that the input and output sequences of the transform use the same memory. Single- and double-precision floating point are supported. For `RFFT`, the imaginary components at DC and Nyquist are zero and thus are not included in the output after a forward transform.

The following example demonstrates how to perform a forward and inverse transform using `RFFT`.

```
RFFT<float> dft(N);
float samples[N];           // real time/position samples

dft.forward(samples);       // perform forward transform to complex frequency domain
samples[0];                 // DC real component
samples[1];                 // 1st harmonic real component
samples[2];                 // 1st harmonic imaginary component
samples[3];                 // 2nd harmonic real component
samples[4];                 // 2nd harmonic imaginary component
...
samples[N-1];               // Nyquist real component

dft.inverse(samples);       // transform back to real time/position domain
```

The following example demonstrates how to perform a forward and inverse transform using `CFFT`.

```
CFFT<float> dft(N);
Complex<float> samples[N];  // complex time/position samples

dft.forward(samples);       // perform forward transform to complex frequency domain
samples[0];                 // complex harmonic +0
samples[1];                 // complex harmonic +1
...
samples[N-2];               // complex harmonic -2
```

```
samples[N-1];                // complex harmonic -1

dft.inverse(samples);        // transform back to complex time/position domain
```

Short-time Fourier Transform

[spectral/STFT]

The short-time Fourier transform uses overlapping windows to obtain better temporal resulotion and to avoid discontinuities between analysis windows during resynthesis. The **STFT** object can operate in a special stream mode which hides buffering and allows easy insertion into a sample loop.

```
// Setup
STFT stft(
    2048,       // window size
    2048/4,     // hop size
    0,          // pad size
    HANN,       // window type: BARTLETT, BLACKMAN, BLACKMAN_HARRIS,
                //  HAMMING, HANN, WELCH, RECTANGLE
    COMPLEX     // format of frequency samples
);

// time-domain loop
{
    float s = src();

    // stream in samples until we have enough for a DFT
    if(stft(s)){

        // frequency domain loop
        for(int k=0; k<stft.numBins(); ++k){
            stft.bin(k);    // the kth frequency sample
        }
    }

    // stream out sample from overlap add resynthesis
    s = stft();
}
```

3.4 Effects

Chorus

`Chorus` emulates multiple voices “singing” in unison (as in a choir) from a single source. Chorusing is useful for thickening up otherwise dry and monotonous sounds. `Chorus` operates by mixing its input with the input sent through two frequency modulated comb filters in parallel.

Waveshaping

Waveshaping is a process whereby a signal is mapped through a mathematical function. In many cases, a function table is used for efficiency and flexibility, but other times a direct function evaluation (such as raising to a power) is more appropriate.

The **ChebyN** class is a waveshaper that is capable of producing a weighted sum of cosine waves from a single sinusoid very efficiently. It accomplishes this by using the recursive definition of Chebyshev polynomials of the first kind, $T_n(x) = 2xT_{n-1}(x) - T_{n-2}(x)$, together with the fact that $T_n(\cos(\theta)) = \cos(n\theta)$.

```
ChebyN<16> shaper;      // produces a weighted sum of 16 cosine waves
Sine<> sine(100);       // sine wave source

// set shaper harmonic amplitudes to the reciprocal of their number
for(int k=0; k<shaper.size(); ++k)
    shaper.coef(k) = 1./(k+1);

float s = sine();       // generate the next sine sample having an amplitude of 1
s = shaper(s);          // shape the sine to produce harmonics
```

Frequency shifting

`FreqShift` shifts the spectrum of a signal up or down. Frequency shifting preserves frequency distances, but not ratios. In contrast, pitch shifting preserves frequency ratios, but not distances. For example, if the input is a harmonic signal with a fundamental frequency of 100 Hz (i.e., containing frequency components at 100, 200, 300... Hz) shifted up by 10 Hz, then the resulting spectrum will contain frequency components at 110, 210, 310.. Hz and therefore no longer be harmonic. The following shows how to shift the spectrum of a periodic waveform to create an inharmonic tone:

```
FreqShift<> fshift(10); // frequency shift by 10 Hz
LFO<> osc(100);

float s = osc.tri();    // generate triangle wave with frequencies at 100, 200, 300, ... Hz
s = fshift(s);          // shift its spectrum up 10 Hz so it has inharmonic
                        // frequencies at 110, 210, 310, ... Hz
```

Frequency shifting is straightforward to implement with complex-valued signals. Since audio processing usually occurs on real-valued signals, we need a way to translate real-valued signals to complex-valued signals. A Hilbert transform does exactly this (3.2).

3.5 I/O

Audio Devices

[io/audioDevice]

Communication with audio devices is done through the `AudioIO` class. An `AudioIO` allows one to open a stream to any number of channels of input and/or output on a particular audio device. When the

AudioIO is started, it will call a user supplied callback function at regular intervals. The rate at which the callback function is executed is the *sample rate* divided by the *block size* (both user specified). In the callback, it is up to the user to read samples from the input buffer(s) and/or write samples to the output buffer(s). The argument to the callback function is a reference to an AudioIOData object which holds onto the i/o buffers, user data, and other relevant information about the audio stream. Audio buffers are in a *non-interleaved* format so that samples within each channel are tightly packed.

The following code example illustrates how to open an audio stream to the default device and define a callback to process audio input.

```
#include "Gamma/AudioIO.h"

struct MyStuff{};

void audioCB(AudioIOData& io){

    MyStuff& stuff = io.user<MyStuff>();

    while(io()){
        float inSample1 = io.in(0);
        float inSample2 = io.in(1);
        io.out(0) = -inSample1;
        io.out(1) = -inSample2;
    }
}

int main(){
    MyStuff stuff;
    AudioIO audioIO(
        128,        // block size
        44100,      // sample rate (Hz)
        audioCB,    // user-defined callback
        &stuff,     // user data
        2,          // input channels to open
        2           // output
    );
    audioIO.start();
}
```

Sound Files

[io/soundFile]

The SoundFile object is used to read/write various kinds of uncompressed sound file formats, such as WAV, AIFF, AU, and FLAC. A sound file is usually comprised of a header describing the format (sample rate, amplitude resolution, compression algorithm), length (in samples), and number of channels of the data followed by a sequence of samples comprising the waveform of the sound. The data is most commonly stored in an *interleaved* format meaning that the samples iterate fastest over the channels, then over time. For example, an interleaved stereo sound would be stored as $L_1, R_1, L_2, R_2, ..., L_N, R_N$ where L_n and R_n are the left and right channel samples, respectively, at time n and N is the length of the sound. The following example shows how to open a file and store its contents into a buffer.

```
SoundFile sf("example.wav");
sf.openRead();                     // open file in read mode

double frameRate = sf.frameRate(); // get frame rate
int frames = sf.frames();          // get number of frames
int channels = sf.channels();      // get number of channels

float buf[frames * channels];      // create a buffer to store all the frames

sf.read(buf, frames);              // copy file contents to buffer
                                   // channels are interleaved
```

The next code example demonstrates how to write data to a sound file.

```
SoundFile sf("example.wav");
sf.openWrite();                    // open file in write mode

sf.frameRate(44100);               // set frame rate
sf.channels(1);                    // set number of channels
sf.format(SoundFile::WAV);         // set file format (WAV, AIFF, AU, RAW, FLAC)
sf.encoding(SoundFile::PCM_16);    // set sample encoding (PCM_S8, PCM_16, PCM_24, PCM_32,
                                   // PCM_U8, FLOAT, DOUBLE, ULAW, ALAW)

float buf[bufSize];                // create a buffer to store all the frames
sf.write(buf, bufSize);            // write buffer to file
                                   // channels are interleaved
```

Recording

[io/recording]

The `Recorder` object provides a means for buffering real-time audio so that it can be saved to a file. It is generally not practical to perform file i/o from the audio thread since it has strict timing requirements and disk operations can take an indeterminate amount of time. `Recorder` has a `write()` method for storing samples from a real-time audio thread into its buffer and a `read()` method for reading samples from a lower priority thread.

```
SoundFile sf("recording.aif");  // sound file to record to
Recorder rec(2);                // set up to record stereo audio

void setup(){
    sf.openWrite();
}

// in audio sample loop
{
    float s1, s2;               // samples of left and right channels
```

```
    ...                          // generate samples
    rec.write(s1, s2);           // write stereo frame into buffer
}

// called periodically from non-audio thread
{
    float * buf;                 // this will point to the read samples
    int n = rec.read(buf);       // copy samples from ring to read buffer
    sf.write(buf, n);            // write samples to the sound file
}
```

3.6 Event Scheduling

Gamma contains a very simple, yet flexible mechanism for scheduling and controlling processes within the audio thread from a separate lower-priority thread. The scheduling can be used to create synthesis graphs, playback note lists, and do algorithmic composition in real-time. The two main components are `Scheduler` and `Process`. **`Scheduler`** places new events into an order-of-execution tree of `Process`es that are run from the audio thread. A **`Process`** is a user-defined audio processing unit which is typically either a synth or an effect. To illustrate how these two components are used together, we will walkthrough how to construct a very common synthesis network—a number of synthesizer voices running through an effects chain. In our example, we will have a group `voices` with two voices, `voice1` and `voice2`, and a group `effects` with two effects, `chorus` and `reverb`. We will use an order-of-execution tree that calls the voices first and then runs their sum through the two effects in series. The tree is shown in Fig. 7. The position of each box is related to its child/sibling relationships in the tree and the solid arrows show the order of execution. The tree is executed in a *depth-first* fashion so that children are visited first then siblings, if any, starting at the root node. When the end of a sibling chain is reached, we go back up the child chain until another sibling is found. When we make it back up the the root, then the trace is complete.

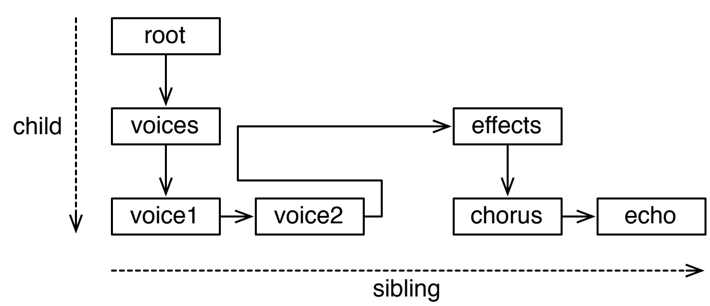


Figure 7: Order-of-execution tree for a voice/effect synthesis network.

The code to set up and run the network is:

```
Scheduler s;                              // define a new scheduler
Process& effects = s.add<Process>();      // add effects group
Process& voices  = s.add<Process>();      // add voices group

s.add<Echo>(effects);                     // add echo to effects
s.add<Chorus>(effects);                   // add chorus to effects
```

```
s.add<Voice>(voices).freq(220).dt(1);   // add first voice to voices, starts after 1 second
s.add<Voice>(voices).freq(329.6);       // add second voice to voices

s.start();                              // start scheduler

AudioIO io(256, 44100., s.audioCB, &s); // define audio i/o
Sync::master().spu(io.fps());           // set sample rate
io.start();                             // start audio i/o
```

First, the two groups `effects` and `voices` are added. The method `add` creates a new object of type given in the angle brackets and makes it the first child of the root node. Because `voices` is added last, it is the first child and is therefore executed first. Next, new effects are added as children nodes of the `effects` group and, likewise, two new voices are added to the `voices` group. The first voice is instructed to start 1 second after being added by using the `Process::dt` method. Since the voices run in parallel, the order that they execute within the `voices` group does not matter. The situation is different for the effects since they run in series. The chorus is added last and therefore is executed before the echo. After setting up the synthesis network, the last steps are to start the scheduler and then hand its callback to an `AudioIO` object.

The voices and effects are defined by making a subclass of `Process`. `Process` has a virtual method `onProcess` for defining how samples are processed. The following shows how to write a voice (the `Voice` from above) comprised of an enveloped triangle wave:

```
struct Voice : public Process{
    Voice(float freq=440, float dur=4, float tilt=0.01)
    :   osc(freq), env(dur*tilt, dur*(1-tilt))
    {}

    Voice& freq(float v){ osc.freq(v); return *this; }

    void onProcess(AudioIOData& io){
        while(io()){
            float s = osc.tri() * env() * 0.1;
            io.out(0) += s;
            io.out(1) += s;
        }
        if(env.done()) free();
    }

    LFO<> osc;
    AD<> env;
};
```

In order for `Voice` (or any other `Process`) to do useful work in the synthesis network, we must write an `onProcess` method. This method has the same signature and behavior as the `AudioIO` callback discussed above (3.5). Note that for voices we *add to* rather than *replace* the contents of the current output buffers. This is because we want our voices to sound in parallel. Also note that after the sample loop we check if the envelope is done and, if so, call the `Process` method `free`. Calling `free` removes the process from the synthesis network and subsequently marks the object for deletion by the scheduler. Writing an effect is done in a similar way as a voice. Here is code implementing an echo effect:

```
struct Echo : public Process{
    Echo(): echo(0.4, 0.323, 0, 0.8){}

    void onProcess(AudioIOData& io){
        while(io()){
            float2 s = float2(io.out(0), io.out(1));
            s = echo(s)*0.5;
            io.out(0) += s[0];
            io.out(1) += s[1];
        }
    }

    Comb<float2> echo;
};
```

Like the voice, we add to rather than replace the current output buffer. For effects, we generally will not call `free` since we want to keep them running all the time.

4 Related Work

4.1 Other C++ Synthesis Libraries

The following are other C++ sound synthesis libraries with similar goals as Gamma:

- CSL, http://fastlabinc.com/CSL/
- SndObj, http://sndobj.sourceforge.net/
- STK, https://ccrma.stanford.edu/software/stk/

	Gamma 0.9.5	CSL 5	SndObj 2.6.6	STK 4.3.1
Spectral	DFT, STFT	DFT	DFT, STFT, PV	
Scalar proc.	Yes	No	No	Yes
Vector proc.	No	Yes	Yes	Yes
Graphs	No	Yes	Yes	No
Sample type	generic	typedef	float	typedef
Extra		MIDI, OSC, Exceptions, Instrument	MIDI, Thread	MIDI, SKINI Thread, Socket
Strength	genericity	spatialization	spectral	phys. modeling
Missing	spatialization		pink noise	spectral
Dependencies	libsndfile, PortAudio	JUCE, liblo	FFTW	
License	BSD	BSD	GPL	BSD

Table 1: Sound synthesis library comparison chart

4.2 Readings

1. Mathews, M. (1969). The Technology of Computer Music. The M.I.T. Press, Boston.

2. Oppenheim, A. V. and Schafer, R. W. (1999). Discrete-time Signal Processing. Prentice Hall, New Jersey, second edition.

3. Roads, C. (1996). Computer Music Tutorial. MIT Press.

4. Smith, S. W. (2006). The Scientist and Engineer's Guide to Digital Signal Processing. California Technical Publishing.

5. Steiglitz, K. (1996). A Digital Signal Processing Primer. Addison-Wesley Publishing Company, Inc.