musicdsp.org source code archive, no-frills version

Please notice that this version does NOT contain the comments you can find in the fancy version...
You can find the fancy version on http://www.musicdsp.org.

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(Allmost) Ready-to-use oscillators

Type : waveform generation
References : Ross Bencina, Olli Niemitalo, ...

Notes :
Ross Bencina: original source code poster
Olli Niemitalo: UpdateWithCubicInterpolation

Code :

```
//this code is meant as an EXAMPLE

//uncomment if you need an FM oscillator
//define FM_OSCILLATOR

/*
members are:

float phase;
int TableSize;
float sampleRate;

float *table, dtable0, dtable1, dtable2, dtable3;

->these should be filled as folows... (remember to wrap around!!!)
table[i] = the wave-shape
dtable0[i] = table[i+1] - table[i];
dtable1[i] = (3.f*(table[i]-table[i+1])-table[i-1]+table[i+2])/2.f
dtable2[i] = 2.f*table[i+1]+table[i-1]-(5.f*table[i]+table[i+2])/2.f
dtable3[i] = (table[i+1]-table[i-1])/2.f
*/

float Oscillator::UpdateWithoutInterpolation(float frequency)
{
        int i = (int) phase;

        phase += (sampleRate/(float TableSize)/frequency;

        if(phase >= (float)TableSize)
                phase -= (float)TableSize;

#ifdef FM_OSCILLATOR
        if(phase < 0.f)
                phase += (float)TableSize;
#endif

        return table[i] ;
}
```

```
float Oscillator::UpdateWithLinearInterpolation(float frequency)
{
        int i = (int) phase;
        float alpha = phase - (float) i;

        phase += (sampleRate/(float)TableSize)/frequency;

        if(phase >= (float)TableSize)
                phase -= (float)TableSize;

#ifdef FM_OSCILLATOR
        if(phase < 0.f)
                phase += (float)TableSize;
#endif

        /*
        dtable0[i] = table[i+1] - table[i]; //remember to wrap around!!!
        */

        return table[i] + dtable0[i]*alpha;
}

float Oscillator::UpdateWithCubicInterpolation( float frequency )
{
        int i = (int) phase;
        float alpha = phase - (float) i;

        phase += (sampleRate/(float)TableSize)/frequency;

        if(phase >= (float)TableSize)
                phase -= (float)TableSize;

#ifdef FM_OSCILLATOR
        if(phase < 0.f)
                phase += (float)TableSize;
#endif

        /* //remember to wrap around!!!
        dtable1[i] = (3.f*(table[i]-table[i+1])-table[i-1]+table[i+2])/2.f
        dtable2[i] = 2.f*table[i+1]+table[i-1]-
(5.f*table[i]+table[i+2])/2.f
        dtable3[i] = (table[i+1]-table[i-1])/2.f
        */

        return ((dtable1[i]*alpha + dtable2[i])*alpha +
dtable3[i])*alpha+table[i];
}
```

16-to-8-bit first-order dither

Type : First order error feedforward dithering code
References : Posted by Jon Watte

Notes :
This is about as simple a dithering algorithm as you can implement, but it's likely to sound better than just truncating to N bits.

Note that you might not want to carry forward the full difference for infinity. It's probably likely that the worst performance hit comes from the saturation conditionals, which can be avoided with appropriate instructions on many DSPs and integer SIMD type instructions, or CMOV.

Last, if sound quality is paramount (such as when going from > 16 bits to 16 bits) you probably want to use a higher-order dither function found elsewhere on this site.

Code :

```
// This code will down-convert and dither a 16-bit signed short
// mono signal into an 8-bit unsigned char signal, using a first
// order forward-feeding error term dither.

#define uchar unsigned char

void dither_one_channel_16_to_8( short * input, uchar * output, int count,
int * memory )
{
  int m = *memory;
  while( count-- > 0 ) {
    int i = *input++;
    i += m;
    int j = i + 32768 - 128;
    uchar o;
    if( j < 0 ) {
      o = 0;
    }
    else if( j > 65535 ) {
      o = 255;
    }
    else {
      o = (uchar)((j>>8)&0xff);
    }
    m = ((j-32768+128)-i);
    *output++ = o;
  }
  *memory = m;
}
```

18dB/oct resonant 3 pole LPF with tanh() dist

References : Posted by Josep M Comajuncosas
Linked file : lpf18.zip
Linked file : lpf18.sme

Notes :
Implementation in CSound and Sync Modular...

1st and 2nd order pink noise filters

Type : Pink noise
References : Posted by umminger@umminger.com

Notes :
Here are some new lower-order pink noise filter coefficients.

These have approximately equiripple error in decibels from 20hz to 20khz at a 44.1khz sampling rate.

1st order, ~ +/- 3 dB error (not recommended!)
num = [0.05338071119116 -0.03752455712906]
den = [1.00000000000000 -0.97712493947102]

2nd order, ~ +/- 0.9 dB error
num = [0.04957526213389 -0.06305581334498 0.01483220320740]
den = [1.00000000000000 -1.80116083982126 0.80257737639225]

2 Wave shaping things

References : Posted by Frederic Petrot

Notes :
Makes nice saturations effects that can be easilly computed using cordic
First using a atan function:
y1 using k=16
max is the max value you can reach (32767 would be a good guess)
Harmonics scale down linealy and not that fast

Second using the hyperbolic tangent function:
y2 using k=2
Harmonics scale down linealy very fast

Code :
```
y1 = (max>>1) * atan(k * x/max)

y2 = max * th(x/max)
```

303 type filter with saturation

Type : Runge-Kutta Filters
References : Posted by Hans Mikelson
Linked file : filters001.txt (this linked file is included below)

Notes :
I posted a filter to the Csound mailing list a couple of weeks ago that has a 303 flavor to it. It basically does wacky distortions to the sound. I used Runge-Kutta for the diff eq. simulation though which makes it somewhat sluggish.

This is a CSound score!!

Linked files

```
; ORCHESTRA
;-----------------------------------------------------------
; Runge-Kutta Filters
; Coded by Hans Mikelson June, 2000
;-----------------------------------------------------------
sr      =       44100                   ; Sample rate
kr      =       44100                   ; Kontrol rate
ksmps   =       1                       ; Samples/Kontrol period
nchnls  =       2                       ; Normal stereo
        zakinit  50, 50

;-------------------------------------------------------------
; Envelope (Knob twisting simulation)
;-------------------------------------------------------------
       instr  1

idur   =      p3         ; Duration
iamp   =      p4         ; Amplitude
ilps   =      p5         ; Loops
iofst  =      p6         ; Offset
itabl  =      p7         ; Table
ioutch =      p8         ; Output channel
iphase =      p9         ; Phase

kout   oscili iamp, ilps/idur, itabl, iphase  ; Create the envelope
       zkw    kout+iofst, ioutch      ; Send out to the zak channel

      endin

;-------------------------------------------------------------
; Runge-Kutta Freaky Filter
;-------------------------------------------------------------
       instr  7
```

```
idur   =      p3           ; Duration
iamp   =      p4           ; Amplitude
kfco2  zkr    p5           ; Filter cutoff
kq1    zkr    p6           ; Q
ih     =      .001         ; Diff eq step size
ipanl  =      sqrt(p8)     ; Pan left
ipanr  =      sqrt(1-p8)   ; Pan right
ifqc   =      cpspch(p9)   ; Pitch to frequency
kpb    zkr    p10          ; Pentic bounce frequency
kpa    zkr    p11          ; Pentic bounce amount
kasym  zkr    p12          ; Q assymmetry amount
kasep  zkr    p13          ; Q asymmetry separation


kdclck linseg 0, .02, 1, idur-.04, 1, .02, 0 ; Declick envelope
kfco1  expseg 1, idur, .1
kfco   =      kfco1*kfco2
kq     =      kq1*kfco^1.2*.1
kfc    =      kfco/8/sr*44100

ay     init   0
ay1    init   0
ay2    init   0
ay3    init   0
axs    init   0
avxs   init   0
ax     vco   1, ifqc, 2, 1, 1, 1        ; Square wave

  ; R-K Section 1
  afdbk =      kq*ay/(1+exp(-ay*3*kasep)*kasym)   ; Only oscillate in one
direction
  ak11  =      ih*((ax-ay1)*kfc-afdbk)
  ak21  =      ih*((ax-(ay1+.5*ak11))*kfc-afdbk)
  ak31  =      ih*((ax-(ay1+.5*ak21))*kfc-afdbk)
  ak41  =      ih*((ax-(ay1+ak31))*kfc-afdbk)
  ay1   =      ay1+(ak11+2*ak21+2*ak31+ak41)/6

  ; R-K Section 2
  ak12  =      ih*((ay1-ay2)*kfc)
  ak22  =      ih*((ay1-(ay2+.5*ak12))*kfc)
  ak32  =      ih*((ay1-(ay2+.5*ak22))*kfc)
  ak42  =      ih*((ay1-(ay2+ak32))*kfc)
  ay2   =      ay2+(ak12+2*ak22+2*ak32+ak42)/6

  ; Pentic bounce equation
  ax3    =      -.1*ay*kpb
  aaxs   =      (ax3*ax3*ax3*ax3*ax3+ay2)*1000*kpa     ; Update acceleration

  ; R-K Section 3
```

```
  ak13  =      ih*((ay2-ay3)*kfc+aaxs)
  ak23  =      ih*((ay2-(ay3+.5*ak13))*kfc+aaxs)
  ak33  =      ih*((ay2-(ay3+.5*ak23))*kfc+aaxs)
  ak43  =      ih*((ay2-(ay3+ak33))*kfc+aaxs)
  ay3   =      ay3+(ak13+2*ak23+2*ak33+ak43)/6

  ; R-K Section 4
  ak14  =      ih*((ay3-ay)*kfc)
  ak24  =      ih*((ay3-(ay+.5*ak14))*kfc)
  ak34  =      ih*((ay3-(ay+.5*ak24))*kfc)
  ak44  =      ih*((ay3-(ay+ak34))*kfc)
  ay    =      ay+(ak14+2*ak24+2*ak34+ak44)/6

aout   =      ay*iamp*kdclck*.07 ; Apply amp envelope and declick

       outs   aout*ipanl, aout*ipanr  ; Output the sound

       endin

-- Hans Mikelson <hljmm@charter.net>
```

3rd order Spline interpollation

References : Posted by Dave from Muon Software, originally from Josh Scholar

Notes :
(from Joshua Scholar about Spline interpollation in general...)
According to sampling theory, a perfect interpolation could be found by replacing each sample with a sinc function centered on that sample, ringing at your target nyquest frequency, and at each target point you just sum all of contributions from the sinc functions of every single point in source.
The sinc function has ringing that dies away very slowly, so each target sample will have to have contributions from a large neighborhood of source samples. Luckily, by definition the sinc function is bandwidth limited, so once we have a source that is prefilitered for our target nyquest frequency and reasonably oversampled relative to our nyquest frequency, ordinary interpolation techniques are quite fruitful even though they would be pretty useless if we hadn't oversampled.

We want an interpolation routine that at very least has the following characteristics:

1. Obviously it's continuous. But since finite differencing a signal (I don't really know about true differentiation) is equivalent to a low frequency attenuator that drops only about 6 dB per octave, continuity at the higher derivatives is important too.

2. It has to be stiff enough to find peaks when our oversampling missed them. This is where what I said about the combination the sinc function's limited bandwidth and oversampling making interpolation possible comes into play.

I've read some papers on splines, but most stuff on splines relates to graphics and uses a control point descriptions that is completely irrelevant to our sort of interpolation. In reading this stuff I quickly came to the conclusion that splines:

1. Are just piecewise functions made of polynomials designed to have some higher order continuity at the transition points.

2. Splines are highly arbitrary, because you can choose arbitrary derivatives (to any order) at each transition. Of course the more you specify the higher order the polynomials will be.

3. I already know enough about polynomials to construct any sort of spline. A polynomial through 'n' points with a derivative specified at 'm[1]' points and second derivatives specified at 'm[2]' points etc. will be a polynomial of the order n-1+m[1]+m[2]...

A way to construct third order splines (that admittedly doesn't help you construct higher order splines), is to linear interpolate between two parabolas. At each point (they are called knots)

you have a parabola going through that point, the previous and the next point. Between each point you linearly interpolate between the polynomials for each point. This may help you imagine splines.

As a starting point I used a polynomial through 5 points for each knot and used MuPad (a free Mathematica like program) to derive a polynomial going through two points (knots) where at each point it has the same first two derivatives as a 4th order polynomial through the surrounding 5 points. My intuition was that basing it on polynomials through 3 points wouldn't be enough of a neighborhood to get good continuity. When I tested it, I found that not only did basing it on 5 point polynomials do much better than basing it on 3 point ones, but that 7 point ones did nearly as badly as 3 point ones. 5 points seems to be a sweet spot.

However, I could have set the derivatives to a nearly arbitrary values - basing the values on those of polynomials through the surrounding points was just a guess.

I've read that the math of sampling theory has different interpretation to the sinc function one where you could upsample by making a polynomial through every point at the same order as the number of points and this would give you the same answer as sinc function interpolation (but this only converges perfectly when there are an infinite number of points). Your head is probably spinning right now - the only point of mentioning that is to point out that perfect interpolation is exactly as stiff as a polynomial through the target points of the same order as the number of target points.

Code :

```
//interpolates between L0 and H0 taking the previous (L1) and next (H1)
points into account
inline float ThirdInterp(const float x,const float L1,const float L0,const
float H0,const float H1)
{
    return
    L0 +
    .5f*
    x*(H0-L1 +
       x*(H0 + L0*(-2) + L1 +
          x*( (H0 - L0)*9 + (L1 - H1)*3 +
             x*((L0 - H0)*15 + (H1 -  L1)*5 +
                x*((H0 - L0)*6 + (L1 - H1)*2 )))));
}
```

5-point spline interpollation

Type : interpollation
References : Joshua Scholar, posted by David Waugh
Code :

```
//nMask = sizeofwavetable-1 where sizeofwavetable is a power of two.
double interpolate(double* wavetable, int nMask, double location)
{
    /* 5-point spline*/

    int nearest_sample = (int) location;
    double x = location - (double) nearest_sample;

    double p0=wavetable[(nearest_sample-2)&nMask];
    double p1=wavetable[(nearest_sample-1)&nMask];
    double p2=wavetable[nearest_sample];
    double p3=wavetable[(nearest_sample+1)&nMask];
    double p4=wavetable[(nearest_sample+2)&nMask];
    double p5=wavetable[(nearest_sample+3)&nMask];

    return p2 + 0.04166666666*x*((p3-p1)*16.0+(p0-p4)*2.0
    + x *((p3+p1)*16.0-p0-p2*30.0- p4
    + x *(p3*66.0-p2*70.0-p4*33.0+p1*39.0+ p5*7.0- p0*9.0
    + x *( p2*126.0-p3*124.0+p4*61.0-p1*64.0- p5*12.0+p0*13.0
    + x *((p3-p2)*50.0+(p1-p4)*25.0+(p5-p0)*5.0)))));
};
```

Alias-free waveform generation with analog filtering

Type : waveform generation
References : Posted by Magnus Jonsson
Linked file : synthesis001.txt (this linked file is included below)

Notes :
(see linkfile)

Linked files

```
alias-free waveform generation with analog filtering
----------------------------------------------------

Ok, here is how I did it. I'm not 100% sure that everything is correct.
I'll demonstrate it on a square wave instead, although i havent tried it.

The impulse response of an analog pole is:

r(t) = exp(p*x)   if t >= 0,
       0          if t < 0

notice that if we know r(t) we can get r(t+1) = r(t)*exp(p).

You all know what the waveform looks like. It's a constant -1
followed by constant 1 followed by .....

We need to convolve the impulse response with the waveform and sample it at discrete
intervals.
What if we assume that we already know the result of the last sample?
Then we can "move" the previous result backwards by multiplying it with exp(p) since
r(t+1) = r(t)*exp(p)

Now the problem is reduced to integrate only between the last sample and the current
sample, and add that to the result.

some pseudo-code:

while forever
{
        result *= exp(pole);
        phase += freq;
        result += integrate(waveform(phase-freq*t), exp(t*pole), t=0..1);
}

integrate(square(phase-freq*t), exp(t*pole), t=0..1)

The square is constant except for when it changes sign.
Let's find out what you get if you integrate a constant multiplied with
exp(t*pole) =)

integrate(k*exp(t*pole)) = k*exp(t*pole)/pole
k = square(phase-freq*t)

and with t from 0 to 1, that becomes
```

```
k*exp(pole)/pole-k/pole = (exp(pole)-1)*k/pole

the only problem left to solve now is the jumps from +1 to -1 and vice versa.

you first calculate (exp(pole)-1)*k/pole like you'd normally do

then you detect if phase goes beyond 0.5 or 1. If so, find out exactly
where between the samples this change takes place.

subtract integrate(-2*exp(t*pole), t=0..place) from the result to
undo the error

Since I am terribly bad at explaining things, this is probably just a mess to you :)

Here's the (unoptimised) code to do this with a saw:
note that sum and pole are complex.

float getsample()
{
        sum *= exp(pole);

        phase += freq;

        sum += (exp(pole)*((phase-freq)*pole+freq)-(phase*pole+freq))/pole;

        if (phase >= 0.5)
        {
                float x = (phase-0.5)/freq;

                sum -= exp(pole*x)-1.0f;

                phase -= 1;
        }

        return sum.real();
}

There's big speedup potential in this i think.

Since the filtering is done "before" sampling,
aliasing is reduced, as a free bonus. with high cutoff
and high frequencies it's still audible though, but
much less than without the filter.

If aliasing is handled in some other way,
a digital filter will sound just as well, that's what i think.

-- Magnus Jonsson <zeal@mail.kuriren.nu>
```

Alien Wah

References : Nasca Octavian Paul (paulnasca[AT]email.ro)
Linked file : alienwah.c (this linked file is included below)

Notes :
"I found this algoritm by "playing around" with complex numbers. Please email me your opinions about it.

Paul."

Linked files

```
/*
 Alien-Wah by Nasca Octavian Paul from Tg. Mures, Romania
 e-mail:  <paulnasca@email.ro> or <paulnasca@yahoo.com>.
*/

/*
 The algorithm was found by me by mistake(I was looking for something else);
 I called this effect "Alien Wah" because sounds a bit like wahwah, but more strange.
 The ideea of this effect is very simple: It is a feedback delay who uses complex
numbers.
 If x[] represents the input and y[] is the output, so a simple feedback delay looks
like this:
 y[n]=y[n-delay]*fb+x[n]*(1-fb)

 'fb' is a real number between 0 and 1.
 If you change the fb with a complex number who has the MODULUS smaller than 1, it
will look like this.

 fb=R*(cos(alpha)+i*sin(alpha));  i^2=-1; R<1;
 y[n]=y[n-delay]*R*(cos(alpha)+i*sin(alpha))+x[n]*(1-R);

 alpha is the phase of the number and is controlled by the LFO(Low Frequency
Oscillator).
 If the 'delay' parameter is low, the effect sounds more like wah-wah,
 but if it is big, the effect will sound very interesting.
 The input x[n] has the real part of the samples from the wavefile and the imaginary
part is zero.
 The output of this effect is the real part of y[n].

 Here it is a simple and unoptimised implementation of the effect. All parameters
should be changed at compile time.
 It was tested only with Borland C++ 3.1.

 Please send me your opinions about this effect.
 Hope you like it (especially if you are play to guitar).
 Paul.
*/

/*
Alien Wah Parameters

 freq       - "Alien Wah" LFO frequency
```

```
 startphase - "Alien Wah" LFO startphase (radians), needed for stereo
 fb         - "Alien Wah" FeedBack (0.0 - low feedback, 1.0 = 100% high feedback)
 delay      -  delay in samples at 44100 KHz (recomanded from 5 to 50...)
*/

#include <complex.h>
#include <fcntl.h>
#include <sys\stat.h>
#include <io.h>
#include <stdio.h>
#include <math.h>

/*
 .raw files are raw files (without header), signed 16 bit,mono
*/
#define infile "a.raw" //input file
#define outfile "b.raw" //input file
#define samplerate 44100

#define bufsize 1024
int buf1[bufsize];//input buffer
int buf2[bufsize];//output buffer


#define lfoskipsamples 25 // How many samples are processed before compute the lfo
value again

struct params
{
   float freq,startphase,fb;
   int delay;
} awparams;
//alien wah internal parameters

struct alienwahinternals
{
 complex *delaybuf;
 float lfoskip;
 long int t;
 complex c;
 int k;
} awint;

//effect initialisation
void init(float freq,float startphase,float fb,int delay){
  awparams.freq=freq;
  awparams.startphase=startphase;
  awparams.fb=fb/4+0.74;
  awparams.delay=(int)(delay/44100.0*samplerate);
  if (delay<1) delay=1;
  awint.delaybuf=new complex[awparams.delay];
  int i;
  for (i=0;i<delay;i++) awint.delaybuf[i]=complex(0,0);
  awint.lfoskip=freq*2*3.141592653589/samplerate;
  awint.t=0;
}
```

```
//process buffer
void process()
{
 int i;
 float lfo,out;
 complex outc;
 for(i=0;i<bufsize;i++)
 {
   if (awint.t++%lfoskipsamples==0)
   {
      lfo=(1+cos(awint.t*awint.lfoskip+awparams.startphase));
      awint.c=complex(cos(lfo)*awparams.fb,sin(lfo)*awparams.fb);
   };
   outc=awint.c*awint.delaybuf[awint.k]+(1-awparams.fb)*buf1[i];
   awint.delaybuf[awint.k]=outc;
   if ((++awint.k)>=awparams.delay)
      awint.k=0;
   out=real(outc)*3;  //take real part of outc
   if (out<-32768) out=-32768;
   else if (out>32767) out=32767; //Prevents clipping
   buf2[i]=out;
 };
}

int main()
{
  char f1,f2;
  int readed;
  long int filereaded=0;
  printf("\n");
  f1=open(infile,O_RDONLY|O_BINARY);
  remove(outfile);
  f2=open(outfile,O_BINARY|O_CREAT,S_IWRITE);
  long int i;

  init(0.6,0,0.5,20);  //effects parameters

  do
  {
    readed=read(f1,buf1,bufsize*2);

    process();

    write(f2,buf2,readed);
    printf("%ld  bytes \r",filereaded);
    filereaded+=readed;
  }while (readed==bufsize*2);

  delete(awint.delaybuf);
  close(f1);
  close(f2);
  printf("\n\n");

  return(0);
}
```

All-Pass Filters, a good explanation

Type : information
References : Posted by Olli Niemitalo
Linked file : filters002.txt (this linked file is included below)

Linked files

```
All-Pass Filters
----------------

A true allpass is not just:

|
|
|     |
|     |     .
|     |     |     |     .
|_____|_____|_____|_____|_____|_____|_____.

Instead, the first peak is negative and has a carefully chosen height.

      |
      |     |     .
 _____|_____|_____|_____|_____|_____._____.
|
|
|
|
|

This can be achieved by adding scaled in(T) to the scaled output of:

out(T) = in(T) + out(T-delay)*gain

Choosing the scalings right... Looking at the critical frequencies,
those that go A) a number of full cycle per peak, and those that go B) a
number of full cycles and a half-cycle per peak, we can set the
constraints to eliminate cancellation:

A) The sum of all peaks must be 1.

B) The sum of odd peaks minus the sum of even peaks (including the
negative peak which we give number zero) must be 1.

Let's get into business now that we know what we want. We call the
amplitude of the negative peak "a" and the amplitude of the first postive
peak "b". The ratio between adjacent positive peaks, the feedback gain, is
denoted by "g".
```

A) The sum of positive peaks is a geometric series and simplifies to b/(1-g). Our first constraint becomes:

a + b/(1-g) = 1.

B) Using similar math... The sum of odd peaks is b/(1-g^2). The sum of even peaks is a + b*g/(1-g^2). So the second constraint is formed:

b/(1-g^2) - (a + b*g/(1-g^2)) = 1.

Solving the two equations we get:

a = -g
b = 1-g^2

Here i had a GREAT phewwww feeling of relief looking at http://harmony-central.com/Effects/Articles/Reverb/allpass.html

Choosing g is up to you. You can even make it negative, just remember to keep |g| < 1. Gosh, perhaps complex g could be used to create a pulsating response (forget i said that)!

We can write the allpass routine in a programmer-wise pleasant way, still preserving ease of mathematical analysis:

```
mid(T) = in(T) + g*mid(T-delay);
out(T) = (1/g-g)*mid(T) - (1/g)*in(T);
```

That can be thought of as two filters put in serial and a third one in parallel with them. The first of the two serial ones is the delay with feedback. The other "filters" are just different gains.

For the first code line, the frequency response is the usual set peaks:

```
         1
--------------------
1 - g e^(-i w delay)
```

Adding the mixing on the second code line gives the flat-magnitude frequency response of the whole allpass system:

```
|       1/g - g               |
| -------------------- - 1/g | | = 1
| 1 - g e^(-i w delay)        |
```

The phase response is a wavy one: (formula not double-checked!)

```
        (g^2-1) sin(delay w)
atan ------------------------
     (g^2+1) cos(delay w) - 2g
```

I hope this cleared things out - and that there aren't fatal errors! :)

-- Olli Niemitalo <oniemita@mail.student.oulu.fi>

Reprise des calculs (T = delay) :

$$H = \frac{1/g - g}{1 - g\,e^{-iwT}} - \frac{1}{g}$$

$$H = \frac{1 - g^2 - (1 - g\,e^{-iwT})}{(1 - g\,e^{-iwT})\,g}$$

$$H = \frac{-g + e^{-iwT}}{1 - g\,e^{-iwT}}$$

$$H = e^{-iwT} \cdot \frac{1 - g\,e^{iwT}}{1 - g\,e^{-iwT}}$$

Le numerateur et le denominateur sont conjugues. d'ou :

$|H| = 1$

et

$$\arg(H) = -wT - 2 \cdot \arctan\frac{g \cdot \sin(wT)}{1 - g \cdot \cos(wT)}$$

Ce dephasage est le meme que celui trouve par Olli mais a l'avantage de separer le retard global et le dephasage.

-- Laurent de Soras <ldesoras@club-internet.fr>

More generally (in fact, maximally generally) you will get an all-pass response with any transfer function of the form :

$$H(z) = b \cdot \frac{z^{-n} \cdot \sum_i a(i) \cdot z^i}{\sum_i \mathrm{conj}(a(i)) \cdot z^{-i}}$$

where $|b| = 1$ and of course n should be large enough that your filter is causal.

-- Frederick Umminger <fumminger@my-Deja.com>

Allocating aligned memory

Type : memory allocation
References : Posted by Benno Senoner

Notes :
we waste up to align_size + sizeof(int) bytes when we alloc a memory area.
We store the aligned_ptr - unaligned_ptr delta in an int located before the aligned area.
This is needed for the free() routine since we need to free all the memory not only the aligned area.
You have to use aligned_free() to free the memory allocated with aligned_malloc() !

Code :

```
/* align_size has to be a power of two !! */
void *aligned_malloc(size_t size, size_t align_size) {

  char *ptr,*ptr2,*aligned_ptr;
  int align_mask = align_size - 1;

  ptr=(char *)malloc(size + align_size + sizeof(int));
  if(ptr==NULL) return(NULL);

  ptr2 = ptr + sizeof(int);
  aligned_ptr = ptr2 + (align_size - ((size_t)ptr2 & align_mask));


  ptr2 = aligned_ptr - sizeof(int);
  *((int *)ptr2)=(int)(aligned_ptr - ptr);

  return(aligned_ptr);
}

void aligned_free(void *ptr) {

  int *ptr2=(int *)ptr - 1;
  ptr -= *ptr2;
  free(ptr);
}
```

Bandlimited sawtooth synthesis

Type : DSF BLIT
References : Posted by emanuel.landeholm [AT] telia.com
Linked file : synthesis002.txt (this linked file is included below)

Notes :
This is working code for synthesizing a bandlimited sawtooth waveform. The algorithm is DSF BLIT + leaky integrator. Includes driver code.

There are two parameters you may tweak:

1) Desired attenuation at nyquist. A low value yields a duller sawtooth but gets rid of those annoying CLICKS when sweeping the frequency up real high. Must be strictly less than 1.0!

2) Integrator leakiness/cut off. Affects the shape of the waveform to some extent, esp. at the low end. Ideally you would want to set this low, but too low a setting will give you problems with DC.

Have fun!
/Emanuel Landeholm

(see linked file)

Linked files

```
#include <stdio.h>
#include <stdlib.h>
#include <math.h>

/* Bandlimited synthesis of sawtooth by
 * leaky integration of a DSF BLIT
 *
 * Emanuel Landeholm, March 2002
 * emanuel.landeholm@telia.com
 *
 * Provided \"as is\".
 * Free as in Ef Are Ee Ee.
 */

double pi    = 3.14159265358979323846264338327950 29L;
double twopi = 6.28318530717958647692528676655900 58L;

/* Leaky integrator/first order lowpass which
 * shapes the impulse train into a nice
 * -6dB/octave spectrum
 *
 * The cutoff frequency needs to be pretty lowish
 * or the sawtooth will suffer phase distortion
 * at the low end.
 */
```

```
typedef struct
{
  double x1, y1;
  double a, b;
} lowpass_t;

/* initializes a lowpass, sets cutoff/leakiness */

void init_lowpass(lowpass_t *lp, double cutoff)
{
  double Omega;

  lp->x1 = lp->y1 = 0.0;

  Omega = atan(pi * cutoff);
  lp->a = -(1.0 - Omega) / (1.0 + Omega);
  lp->b = (1.0 - lp->b) / 2.0;
}

double update_lowpass(lowpass_t *lp, double x)
{
  double y;

  y = lp->b * (x + lp->x1) - lp->a * lp->y1;

  lp->x1 = x;
  lp->y1 = y;

  return y;
}

/* dsf blit datatype
 *
 */

typedef struct
{
  double phase;    /* phase accumulator */
  double aNQ;      /* attenuation at nyquist */
  double curcps;   /* current frequency, updated once per cycle */
  double curper;   /* current period, updated once per cycle */
  lowpass_t leaky; /* leaky integrator */
  double N;        /* # partials */
  double a;        /* dsf parameter which controls roll-off */
  double aN;       /* former to the N */
} blit_t;

/* initializes a blit structure
 *
 * The aNQ parameter is the desired attenuation
```

```
 * at nyquist. A low value yields a duller
 * sawtooth but gets rid of those annoying CLICKS
 * when sweeping the frequency up real high. |aNQ|
 * must be strictly less than 1.0! Find a setting
 * which works for you.
 *
 * The cutoff parameter controls the leakiness of
 * the integrator.
 */

void init_blit(blit_t *b, double aNQ, double cutoff)
{
  b->phase  = 0.0;
  b->aNQ = aNQ;
  b->curcps = 0.0;
  b->curper = 0.0;
  init_lowpass(&b->leaky, cutoff);
}

/* Returns a sawtooth computed from a leaky integration
 * of a DSF bandlimited impulse train.
 *
 * cps (cycles per sample) is the fundamental
 * frequency: 0 -> 0.5 == 0 -> nyquist
 */

double update_blit(blit_t *b, double cps)
{
  double P2, beta, Nbeta, cosbeta, n, d, blit, saw;

  if(b->phase >= 1.0 || b->curcps == 0.0)
    {
      /* New cycle, update frequency and everything
       * that depends on it
       */

      if(b->phase >= 1.0)
b->phase -= 1.0;

      b->curcps = cps;        /* this cycle\'s frequency */
      b->curper = 1.0 / cps;  /* this cycle\'s period */

      P2 = b->curper / 2.0;
      b->N = 1.0 + floor(P2); /* # of partials incl. dc */

      /* find the roll-off parameter which gives
       * the desired attenuation at nyquist
       */

      b->a   = pow(b->aNQ, 1.0 / P2);
      b->aN  = pow(b->a,   b->N);
```

```
    }

  beta = twopi * b->phase;

  Nbeta = b->N * beta;
  cosbeta = cos(beta);

  /* The dsf blit is scaled by 1 / period to give approximately the same
   * peak-to-peak over a wide range of frequencies.
   */

  n = 1.0 -
    b->aN * cos(Nbeta) -
    b->a * (cosbeta - b->aN * cos(Nbeta - beta));
  d = b->curper * (1.0 + b->a * (-2.0 * cosbeta + b->a));

  b->phase += b->curcps; /* update phase */

  blit = n / d - b->curcps; /* This division can only fail if |a| == 1.0
     * Subtracting the fundamental frq rids of DC
     */

  saw = update_lowpass(&b->leaky, blit); /* shape blit spectrum into a saw */

  return saw;
}

/* driver code - writes headerless 44.1 16 bit PCM to stdout */

static int clipped = 0;

static void ADC_out(double x)
{
  short s;

  if(x > 1.0)
    {
      ++clipped;
      x = 1.0;
    }
  else if(x < -1.0)
    {
      ++clipped;
      x = -1.0;
    }

  s = 32767.0 * x;

  fwrite(&s, sizeof(s), 1, stdout);
}

int main(int argc, char **argv)
```

```
{
  int i, L;
  double x, cps, cm;
  blit_t b;

  L = 1000000;

  init_blit(&b, 0.5, 0.0001);

  /* sweep from 40 to 20000 Hz */

  cps = 40.0 / 44100.0;
  cm = pow(500.0, 1.0 / (double)L);

  for(i = 0; i < L; ++i)
    {
      x = 2.0 * update_blit(&b, cps);
      ADC_out(x);

      cps *= cm;
    }

  fprintf(stderr, \"%d values were clipped\\n\", clipped);

  return 0;
}
```

Bandlimited waveform generation

Type : waveform generation
References : Posted by Joe Wright
Linked file : bandlimited.cpp (this linked file is included below)
Linked file : bandlimited.pdf

Notes :
(see linkfile)

Linked files

```
// An example of generating the sawtooth and parabola wavetables
// for storage to disk.
//
// SPEED=sampling rate, e.g. 44100.0f
// TUNING=pitch of concert A, e.g. 440.0f

//////////////////////////
// Wavetable reverse lookup
// Given a playback rate of the wavetable, what is wavetables index?
//
// rate = f.wavesize/fs e.g. 4096f/44100
// max partials = nyquist/f = wavesize/2rate  e.g. 2048/rate
//
// using max partials we could then do a lookup to find the wavetables index
// in a pre-calculated table
//
// however, we could skip max partials, and lookup a table based on a
// function of f (or rate)
//
// the first few midi notes (0 - 9) differ by < 1 so there are duplicates
// values of (int) f.
// therefore, to get an index to our table (that indexes the wavetables)
// we need 2f
//
// to get 2f from rate we multiply by the constant
// 2f = 2.fs/wavesize  e.g. 88200/4096
//
// our lookup table will have a length>25087 to cover the midi range
// we'll make it 32768 in length for easy processing


        int a,b,n;
        float* data;
        float* sinetable=new float[4096];
        float* datap;
        for(b=0;b<4096;b++)
                sinetable[b]=sin(TWOPI*(float)b/4096.0f);
        int partials;
        int partial;
        int partialindex,reverseindex,lastnumpartials;
        float max,m;
        int* reverse;
```

```
        // sawtooth

                data=new float[128*4096];
                reverse=new int[32768];

                reverseindex=0;
                partialindex=0;
                lastnumpartials=-1;

                for(n=0;n<128;n++)
                {
                        partials=(int)((SPEED*0.5f)/float(TUNING*(float)pow(2,(float)
(n-69)/12.0f))); //(int) NYQUIST/f
                        if(partials!=lastnumpartials)
                        {
                                datap=&data[partialindex*4096];
                                for(b=0;b<4096;b++)
                                        datap[b]=0.0f;  //blank wavetable
                                for(a=0;a<partials;a++)
                                {
                                        partial=a+1;
                                        m=cos((float)a*HALFPI/(float)partials);
//gibbs
                                        m*=m;  //gibbs
                                        m/=(float)partial;
                                        for(b=0;b<4096;b++)

datap[b]+=m*sinetable[(b*partial)%4096];
                                }
                                lastnumpartials=partials;
                                a=int(2.0f*TUNING*(float)pow(2,(float) (n-
69)/12.0f)); //2f
                                for(b=reverseindex;b<=a;b++)
                                        reverse[b]=partialindex;
                                reverseindex=a+1;
                                partialindex++;
                        }
                }

                for(b=reverseindex;b<32768;b++)
                        reverse[b]=partialindex-1;

                ar << (int) partialindex; //number of waveforms
                ar << (int) 4096; //waveform size (in samples)

                max=0.0;
                for(b=0;b<4096;b++)
                {
                        if(fabs(*(data+b))>max)  //normalise to richest waveform (0)
                                max=(float)fabs(*(data+b));
                }
                for(b=0;b<4096*partialindex;b++)
                {
                        *(data+b)/=max;
                }
```

```
                //ar.Write(data,4096*partialindex*sizeof(float));
                //ar.Write(reverse,32768*sizeof(int));

                delete [] data;
                delete [] reverse;
        }
        // end sawtooth

        // parabola

                data=new float[128*4096];
                reverse=new int[32768];

                reverseindex=0;
                partialindex=0;
                lastnumpartials=-1;

                float sign;

                for(n=0;n<128;n++)
                {
                        partials=(int)((SPEED*0.5f)/float(TUNING*(float)pow(2,(float)
(n-69)/12.0f)));
                        if(partials!=lastnumpartials)
                        {
                                datap=&data[partialindex*4096];
                                for(b=0;b<4096;b++)
                                        datap[b]=PI*PI/3.0f;
                                sign=-1.0f;
                                for(a=0;a<partials;a++)
                                {
                                        partial=a+1;
                                        m=cos((float)a*HALFPI/(float)partials);
//gibbs
                                        m*=m;  //gibbs
                                        m/=(float)(partial*partial);
                                        m*=4.0f*sign;
                                        for(b=0;b<4096;b++)

datap[b]+=m*sinetable[((b*partial)+1024)%4096]; //note, parabola uses cos
                                        sign=-sign;
                                }
                                lastnumpartials=partials;
                                a=int(2.0f*TUNING*(float)pow(2,(float) (n-
69)/12.0f)); //2f
                                for(b=reverseindex;b<=a;b++)
                                        reverse[b]=partialindex;
                                reverseindex=a+1;
                                partialindex++;
                        }
                }

                for(b=reverseindex;b<32768;b++)
                        reverse[b]=partialindex-1;
```

```
                ar << (int) partialindex; //number of waveforms
                ar << (int) 4096; //waveform size (in samples)

                max=0.0;
                for(b=0;b<4096;b++)
                {
                        if(fabs(*(data+b))>max)  //normalise to richest waveform (0)
                                max=(float)fabs(*(data+b));
                }
                max*=0.5;
                for(b=0;b<4096*partialindex;b++)
                {
                        *(data+b)/=max;
                        *(data+b)-=1.0f;
                }

                //ar.Write(data,4096*partialindex*sizeof(float));
                //ar.Write(reverse,32768*sizeof(int));

                delete [] data;
                delete [] reverse;
        }
        // end parabola


////////////////////////////////////////////////////////////////////////
// An example of playback of a sawtooth wave
// This is not optimised for easy reading
// When optimising you'll need to get this in assembly (especially those
// float to int conversions)
////////////////////////////////////////////////////////

#define WAVETABLE_SIZE          (1 << 12)
#define WAVETABLE_SIZEF         WAVETABLE_SIZE.0f
#define WAVETABLE_MASK          (WAVETABLE_SIZE - 1)

float index;
float rate;
int wavetableindex;
float ratetofloatfactor;
float* wavetable;

void setupnote(int midinote /*0 - 127*/)
{
        float f=TUNING*(float)pow(2,(float) (midinote-69)/12.0f));
        rate=f*WAVETABLE_SIZEF/SPEED;
        ratetofloatfactor=2.0f*SPEED/WAVETABLE_SIZEF;
        index=0.0f;
        wavetableindex=reverse[(int)(2.0f*f)];
        wavetable=&sawtoothdata[wavetableindex*WAVETABLE_SIZE];
}

void generatesample(float* buffer,int length)
{
        int currentsample,
        int nextsample;
```

```
        float m;
        float temprate;
        while(length--)
        {
                currentsample=(int) index;
                nextsample=(currentsample+1) & WAVETABLE_MASK;
                m=index-(float) currentsample; //fractional part
                *buffer++=(1.0f-m)*wavetable[currentsample]+m*wavetable[nextsample];
//linear interpolation
                rate*=slide; //slide coeffecient if required
                temprate=rate*fm; //frequency modulation if required
                index+=temprate;
                if(index>WAVETABLE_SIZEF)
                {
                        //new cycle, respecify wavetable for sliding
                        wavetableindex=reverse[(int)(ratetofloatfactor*temprate)];
                        wavetable=&sawtoothdata[wavetableindex*WAVETABLE_SIZE];
                        index-=WAVETABLE_SIZEF;
                }
        }
}
```

Bandlimited waveform generation with hard sync

References : Posted by Emanuel Landeholm
Linked file : http://www.algonet.se/~e-san/hardsync.tar.gz

Bandlimited waveforms synopsis.

References : Joe Wright
Linked file : waveforms.txt (this linked file is included below)

Notes :
(see linkfile)

Linked files

```
From: "Joe Wright" <joe@nyrsound.com>
To: <music-dsp@shoko.calarts.edu>
Subject: Re: waveform discussions
Date: Tue, 19 Oct 1999 14:45:56 +0100

After help from this group and reading various literature I have finished my
waveform engine.  As requested, I am now going to share some of the things
are have learnt from a practical viewpoint.

Problem:

The waveforms of interest are sawtooth, square and triangle.

The waveforms must be bandlimited (i.e. fitting under Nyquist).  This
precludes simple generation of the waveforms.  For example, the
analogue/continuous formular (which has infinite harmonics):

s(t) = (t*f0) mod 1  (t=time, f0=frequency of note)

procudues aliasing that cannot be filtered out when converted to the
digital/discrete form:

s(n) = (f0*n*Ts) mod 1 (n=sample, Ts equals 1/sampling rate)


The other condition of this problem is that the waveforms are generatable in
real-time.  Additionally, bonuses are given for solutions which allow
pulse-width modulation and triangle asymmetry.


The generation of these waves is non-triaval and below is discussed three
techniques to solve the problem - wavetables, approximation through
processing of |sinewave| and BLIT integration.  BLIT integration is
discussed in depth.


Wavetables:

You can generate a wavetable for the waveform by summing sine waves up to
the Nyquist frequency.  For example, the sawtooth waveform can be generated
```

by:

s(n) = Sum[k=1,n] (1/k *sin(2PI*f0*k*Ts)) where f0*n < Ts/2

The wavetable can then be played back, pitched up or down subject that pitched f*n < Ts/2. Anything lower will not alias but it may lack some higher harmonics if pitched too low.

To cover these situations, use multiple wavetables describing different frequency ranges within which it is fine to pitch up or down.

You may need to compromise between number of wavetables and accuracy because of memory considerations (especially if over-sampling). This means some wavetables will have to cover larger ranges than they should. As long as the range is too far in the lower direction rather than higher, you will not alias (you will just miss some higher harmonics).

With wavetables you can add a sawtooth to an inverted sawtooth offset in time, to produce a pulse/square wave. Vary the offset to vary the pulse width. Asymmetry of triangle waves is not possible (as far as I know) though.

Approximation through processing of |sinewave|

This method is discussed in detail in 'Modeling Analog Synthesis with DSPs' - Computer Music Journal, 21:4 pp.23 - 41, Winter 1997, Lane, Hoory, Marinez and Wang.

The basic idea starts with the generation of a sawtooth by feeding abs(sin(n)) into a lowpass followed by a highpass filter. This approximates (quite well supposedly) a bandlimited sawtooth. Unfortunately, details of what the cutoff for the lowpass should be were not precise in the paper (although the highpass cutoff was given).

Square wave and triangle wave approximation was implemented by subtracting abs(sin(n)) and abs(sin(n/2)). With different gains, pulse width (and I presume asymmetry) were possible.

For more information, consult the paper.

BLIT intergration

This topic refers to the paper 'Alias-Free Digital Synthesis of Classic Analog Waveforms' by Tim Stilson and Julius Smith of CCRMA. The paper can be found at http://www-ccrma.stanford.edu/~stilti/papers

BLIT stands for bandlimited impluse train. I'm not going to go into the theory, you'll have to read the paper for that. However, simply put, a pulse train of the form 10000100001000 etc... is not bandlimited.

The formular for a BLIT is as follows:

BLIT(x) = (m/p) * (sin(PI*x*m/p)/(m*sin(PI*x/p))

x=sample number ranging from 1 to period
p=period in samples (fs/f0). Although this should theorectically not be an interger, I found pratically speaking it needs to be.
m=2*((int)p/2)+1 (i.e. when p=odd, m=p otherwise m=p+1)

[note] in the paper they describe m as the largest odd interger not exceeding the period when in fact their formular for m (which is the one that works in practive) makes it (int)p or (int)p +1

As an extension, we also have a bipolar version:

BP-BLIT k (x)=BLIT(x) - BLIT(x+k) (where k is in the range [0,period])

Now for the clever bit. Lets start with the square/rectangle wave. Through intergration we get:

Rect(n) = Sum(i=0,n) (BP-BLIT k0 (i) - C4)

C4 is a DC offset which for the BP-BLIT is zero. This gives a nice iteration for rect(n):

Rect(n) = Rect(n-1) + BP-BLIT k0 (n) where k0 is the pulse width between [0,period] or in practive [1,period-1]

A triangle wave is given by:

Tri(n) = Sum(i=0,n) (Rect(k) - C6)

Tri(n) = Tri(n-1) + Rect(n) - C6

C6 = k0/period

The triangle must also be scaled:

Tri(b) = Tri(n-1) + g(f,d)*(Rect(n) - C6)

where g(f,d) = 0.99 / (period* d * (d-1)) d=k0/period

Theorietcally it could be 1.00 / ... but I found numerical error sometimes pushed it over the edge. The paper actually states 2.00/... but for some reason I find this to be incorrect.

Lets look at some rough and ready code. I find the best thing to do is to generate one period at a time and then reset everything. The numerical errors over one period are negligable (based on 32bit float) but if you keep

on going without resetting, the errors start to creep in.

```
    float period = samplingrate/frequency;
    float m=2*(int)(period/2)+1.0f;
    float k=(int)(k0*period);
    float g=0.99f/(periodg*(k/period)*(1-k/period));
    float t;
    float bpblit;
    float square=0.0f;
    float triangle=0.0f;
    for(t=1;t<=period;t++)
    {
        bpblit=sin(PI*(t+1)*m/period)/(m*sin(PI*(t+1)/period));
        bpblit-=sin(PI*(t+k)*m/period)/(m*sin(PI*(t+k)/period));
        square+=bpblit;
        triangle+=g*(square+k/period);
    }
    square=0;
    triangle=0;
```

Highly un-optimised code but you get the point. At each sample(t) the output values are square and triangle respectively.

Sawtooth:

This is given by:

```
Saw(n) = Sum(k=0,n) (BLIT(k) - C2)   [as opposed to BP-BLIT]
Saw(n) = Saw(n-1) + BLIT(n) - C2
```

Now, C2 is a bit tricky. Its the average of BLIT for that period (i.e. 1/n * Sum (k=0,n) (BLIT(k)). [Note] Blit(0) = 0.

I found the best way to deal with this is to have a lookup table which you have generated and saved to disk as a file which contains a value of C2 for every period you are interested in. This is because I know of no easy way to generate C2 in real-time.

Last thing. My implementation of BLIT gives negative values. Therefore my sawtooth is +C2 rather than -C2.

I hope this helps, any questions don't hestitate to contact me.

Joe Wright - Nyr Sound Ltd
http://www.nyrsound.com
info@nyrsound.com

Bandlimited waveforms...

References : Posted by Paul Kellet

Notes :
(Quoted from Paul's mail)
Below is another waveform generation method based on a train of sinc functions (actually an alternating loop along a sinc between t=0 and t=period/2).

The code integrates the pulse train with a dc offset to get a sawtooth, but other shapes can be made in the usual ways... Note that 'dc' and 'leak' may need to be adjusted for very high or low frequencies.

I don't know how original it is (I ought to read more) but it is of usable quality, particularly at low frequencies. There's some scope for optimisation by using a table for sinc, or maybe a a truncated/windowed sinc?

I think it should be possible to minimise the aliasing by fine tuning 'dp' to slightly less than 1 so the sincs join together neatly, but I haven't found the best way to do it. Any comments gratefully received.

Code :
```
float p=0.0f;      //current position
float dp=1.0f;     //change in postion per sample
float pmax;        //maximum position
float x;           //position in sinc function
float leak=0.995f; //leaky integrator
float dc;          //dc offset
float saw;         //output


//set frequency...

  pmax = 0.5f * getSampleRate() / freqHz;
  dc = -0.498f/pmax;


//for each sample...

  p += dp;
  if(p < 0.0f)
  {
    p = -p;
    dp = -dp;
  }
  else if(p > pmax)
  {
```

```
  p = pmax + pmax - p;
  dp = -dp;
}

x= pi * p;
if(x < 0.00001f)
   x=0.00001f; //don't divide by 0

saw = leak*saw + dc + (float)sin(x)/(x);
```

Base-2 exp

References : Posted by Laurent de Soras

Notes :
Linear approx. between 2 integer values of val. Uses 32-bit integers. Not very efficient but fastest than exp()
This code was designed for x86 (little endian), but could be adapted for big endian processors. Laurent thinks you just have to change the (*(1 + (int *) &ret)) expressions and replace it by (*(int *) &ret). However, He didn't test it.

Code :

```
inline double fast_exp2 (const double val)
{
   int    e;
   double ret;

   if (val >= 0)
   {
      e = int (val);
      ret = val - (e - 1);
      ((*(1 + (int *) &ret)) &= ~(2047 << 20)) += (e + 1023) << 20;
   }
   else
   {
      e = int (val + 1023);
      ret = val - (e - 1024);
      ((*(1 + (int *) &ret)) &= ~(2047 << 20)) += e << 20;
   }
   return (ret);
}
```

Biquad C code

References : Posted by Tom St Denis
Linked file : biquad.c (this linked file is included below)

Notes :
Implementation of the RBJ cookbook, in C.

Linked files

```
/* Simple implementation of Biquad filters -- Tom St Denis
 *
 * Based on the work

Cookbook formulae for audio EQ biquad filter coefficients
---------------------------------------------------------
by Robert Bristow-Johnson, pbjrbj@viconet.com  a.k.a. robert@audioheads.com

 * Available on the web at

http://www.smartelectronix.com/musicdsp/text/filters005.txt

 * Enjoy.
 *
 * This work is hereby placed in the public domain for all purposes, whether
 * commercial, free [as in speech] or educational, etc.  Use the code and please
 * give me credit if you wish.
 *
 * Tom St Denis -- http://tomstdenis.home.dhs.org
*/

/* this would be biquad.h */
#include &lt;math.h&gt;
#include &lt;stdlib.h&gt;

#ifndef M_LN2
#define M_LN2      0.69314718055994530942
#endif

#ifndef M_PI
#define M_PI           3.14159265358979323846
#endif

/* whatever sample type you want */
typedef double smp_type;

/* this holds the data required to update samples thru a filter */
typedef struct {
    smp_type a0, a1, a2, a3, a4;
    smp_type x1, x2, y1, y2;
}
biquad;
```

```
extern smp_type BiQuad(smp_type sample, biquad * b);
extern biquad *BiQuad_new(int type, smp_type dbGain, /* gain of filter */
                          smp_type freq,             /* center frequency */
                          smp_type srate,            /* sampling rate */
                          smp_type bandwidth);       /* bandwidth in octaves */

/* filter types */
enum {
    LPF, /* low pass filter */
    HPF, /* High pass filter */
    BPF, /* band pass filter */
    NOTCH, /* Notch Filter */
    PEQ, /* Peaking band EQ filter */
    LSH, /* Low shelf filter */
    HSH /* High shelf filter */
};

/* Below this would be biquad.c */
/* Computes a BiQuad filter on a sample */
smp_type BiQuad(smp_type sample, biquad * b)
{
    smp_type result;

    /* compute result */
    result = b-&gt;a0 * sample + b-&gt;a1 * b-&gt;x1 + b-&gt;a2 * b-&gt;x2 -
        b-&gt;a3 * b-&gt;y1 - b-&gt;a4 * b-&gt;y2;

    /* shift x1 to x2, sample to x1 */
    b-&gt;x2 = b-&gt;x1;
    b-&gt;x1 = sample;

    /* shift y1 to y2, result to y1 */
    b-&gt;y2 = b-&gt;y1;
    b-&gt;y1 = result;

    return result;
}

/* sets up a BiQuad Filter */
biquad *BiQuad_new(int type, smp_type dbGain, smp_type freq,
smp_type srate, smp_type bandwidth)
{
    biquad *b;
    smp_type A, omega, sn, cs, alpha, beta;
    smp_type a0, a1, a2, b0, b1, b2;

    b = malloc(sizeof(biquad));
    if (b == NULL)
        return NULL;

    /* setup variables */
    A = pow(10, dbGain /40);
```

```
omega = 2 * M_PI * freq /srate;
sn = sin(omega);
cs = cos(omega);
alpha = sn * sinh(M_LN2 /2 * bandwidth * omega /sn);
beta = sqrt(A + A);

switch (type) {
case LPF:
    b0 = (1 - cs) /2;
    b1 = 1 - cs;
    b2 = (1 - cs) /2;
    a0 = 1 + alpha;
    a1 = -2 * cs;
    a2 = 1 - alpha;
    break;
case HPF:
    b0 = (1 + cs) /2;
    b1 = -(1 + cs);
    b2 = (1 + cs) /2;
    a0 = 1 + alpha;
    a1 = -2 * cs;
    a2 = 1 - alpha;
    break;
case BPF:
    b0 = alpha;
    b1 = 0;
    b2 = -alpha;
    a0 = 1 + alpha;
    a1 = -2 * cs;
    a2 = 1 - alpha;
    break;
case NOTCH:
    b0 = 1;
    b1 = -2 * cs;
    b2 = 1;
    a0 = 1 + alpha;
    a1 = -2 * cs;
    a2 = 1 - alpha;
    break;
case PEQ:
    b0 = 1 + (alpha * A);
    b1 = -2 * cs;
    b2 = 1 - (alpha * A);
    a0 = 1 + (alpha /A);
    a1 = -2 * cs;
    a2 = 1 - (alpha /A);
    break;
case LSH:
    b0 = A * ((A + 1) - (A - 1) * cs + beta * sn);
    b1 = 2 * A * ((A - 1) - (A + 1) * cs);
    b2 = A * ((A + 1) - (A - 1) * cs - beta * sn);
    a0 = (A + 1) + (A - 1) * cs + beta * sn;
```

```
        a1 = -2 * ((A - 1) + (A + 1) * cs);
        a2 = (A + 1) + (A - 1) * cs - beta * sn;
        break;
    case HSH:
        b0 = A * ((A + 1) + (A - 1) * cs + beta * sn);
        b1 = -2 * A * ((A - 1) + (A + 1) * cs);
        b2 = A * ((A + 1) + (A - 1) * cs - beta * sn);
        a0 = (A + 1) - (A - 1) * cs + beta * sn;
        a1 = 2 * ((A - 1) - (A + 1) * cs);
        a2 = (A + 1) - (A - 1) * cs - beta * sn;
        break;
    default:
        free(b);
        return NULL;
    }

    /* precompute the coefficients */
    b-&gt;a0 = b0 /a0;
    b-&gt;a1 = b1 /a0;
    b-&gt;a2 = b2 /a0;
    b-&gt;a3 = a1 /a0;
    b-&gt;a4 = a2 /a0;

    /* zero initial samples */
    b-&gt;x1 = b-&gt;x2 = 0;
    b-&gt;y1 = b-&gt;y2 = 0;

    return b;
}
/* crc==3062280887, version==4, Sat Jul  7 00:03:23 2001 */
```

Bit quantization/reduction effect

Type : Bit-level noise-generating effect
References : Posted by Jon Watte

Notes :
This function, run on each sample, will emulate half the effect of running your signal through a Speak-N-Spell or similar low-bit-depth circuitry.

The other half would come from downsampling with no aliasing control, i e replicating every N-th sample N times in the output signal.

Code :
```
short keep_bits_from_16( short input, int keepBits ) {
  return (input & (-1 << (16-keepBits)));
}
```

Calculate notes (java)

Type : Java class for calculating notes with different in params
References : Posted by larsby[AT]elak[DOT]org
Linked file : Frequency.java (this linked file is included below)

Notes :
Converts between string notes and frequencies and back. I vaguely remember writing bits of it, and I got it off the net somwhere so dont ask me

- Larsby

Linked files

```
public class Frequency extends Number
{

    private static final double PITCH_OF_A4 = 57D;
    private static final double FACTOR = 12D / Math.log(2D);
    private static final String NOTE_SYMBOL[] = {
        "C", "C#", "D", "D#", "E", "F", "F#", "G", "G#", "A",
        "A#", "B"
    };
    public static float frequencyOfA4 = 440F;
    private float frequency;

    public static final double getPitch(float f)
    {
        return getPitch(f);
    }

    public static final double getPitch(double d)
    {
        return 57D + FACTOR * Math.log(d / (double)frequencyOfA4);
    }

    public static final float getFrequency(double d)
    {
        return (float)(Math.exp((d - 57D) / FACTOR) * (double)frequencyOfA4);
    }

    public static final String makeNoteSymbol(double d)
    {
        int i = (int)(d + 120.5D);
        StringBuffer stringbuffer = new StringBuffer(NOTE_SYMBOL[i % 12]);
        stringbuffer.append(Integer.toString(i / 12 - 10));
        return new String(stringbuffer);
    }

    public static float valueOf(String s)
        throws IllegalArgumentException
    {
        try
        {
```

```
            return (new Float(s)).floatValue();
        }
        catch(NumberFormatException _ex) { }
        try
        {
            return getFrequency(parseNoteSymbol(s));
        }
        catch(IllegalArgumentException _ex)
        {
            throw new IllegalArgumentException("Neither a floating point number nor a
valid note symbol.");
        }
    }

    public static final int parseNoteSymbol(String s)
        throws IllegalArgumentException
    {
        s = s.trim().toUpperCase();
        for(int i = NOTE_SYMBOL.length - 1; i >= 0; i--)
        {
            if(!s.startsWith(NOTE_SYMBOL[i]))
                continue;
            try
            {
                return i + 12 *
Integer.parseInt(s.substring(NOTE_SYMBOL[i].length()).trim());
            }
            catch(NumberFormatException _ex) { }
            break;
        }

        throw new IllegalArgumentException("not valid note symbol.");
    }

    public static void transformPitch(TextComponent textcomponent, boolean flag)
    {
        boolean flag1 = false;
        String s = textcomponent.getText();
        if(flag)
        {
            try
            {

textcomponent.setText(Integer.toString((int)(getFrequency(parseNoteSymbol(s)) +
0.5F)));
                return;
            }
            catch(IllegalArgumentException _ex)
            {
                flag1 = true;
            }
            return;
        }
        try
        {
```

```
            textcomponent.setText(makeNoteSymbol(getPitch((new
Float(s)).floatValue())));
            return;
        }
        catch(NumberFormatException _ex)
        {
            flag1 = true;
        }
    }

    public Frequency(float f)
    {
        frequency = 1.0F;
        frequency = f;
    }

    public Frequency(String s)
        throws IllegalArgumentException
    {
        frequency = 1.0F;
        frequency = valueOf(s);
    }

    public byte byteValue()
    {
        return (byte)(int)(frequency + 0.5F);
    }

    public short shortValue()
    {
        return (short)(int)(frequency + 0.5F);
    }

    public long longValue()
    {
        return (long)(frequency + 0.5F);
    }

    public int intValue()
    {
        return (int)(frequency + 0.5F);
    }

    public float floatValue()
    {
        return frequency;
    }

    public double doubleValue()
    {
        return (double)frequency;
    }

    public String toString()
    {
        return Integer.toString(intValue());
```

```
    }

    public String toNoteSymbol()
    {
        return makeNoteSymbol(getPitch(frequency));
    }

    public static void main(String[] args)
    {
        System.out.println(Frequency.parseNoteSymbol("C2"));
        System.out.println(Frequency.getFrequency(24));
    }
}
```

Class for waveguide/delay effects

Type : IIR filter
References : Posted by arguru[AT]smartelectronix.com

Notes :
Flexible-time, non-sample quantized delay , can be used for stuff like waveguide synthesis or time-based (chorus/flanger) fx.

MAX_WG_DELAY is a constant determining MAX buffer size (in samples)

Code :
```
class cwaveguide
{
public:
cwaveguide(){clear();}
virtual ~cwaveguide(){};

void clear()
{
counter=0;
for(int s=0;s<MAX_WG_DELAY;s++)
buffer[s]=0;
}

inline float feed(float const in,float const feedback,double const delay)
{
// calculate delay offset
double back=(double)counter-delay;

// clip lookback buffer-bound
if(back<0.0)
back=MAX_WG_DELAY+back;

// compute interpolation left-floor
int const index0=floor_int(back);

// compute interpolation right-floor
int index_1=index0-1;
int index1=index0+1;
int index2=index0+2;

// clip interp. buffer-bound
if(index_1<0)index_1=MAX_WG_DELAY-1;
if(index1>=MAX_WG_DELAY)index1=0;
if(index2>=MAX_WG_DELAY)index2=0;

// get neighbourgh samples
```

```
float const y_1= buffer [index_1];
float const y0 = buffer [index0];
float const y1 = buffer [index1];
float const y2 = buffer [index2];

// compute interpolation x
float const x=(float)back-(float)index0;

// calculate
float const c0 = y0;
float const c1 = 0.5f*(y1-y_1);
float const c2 = y_1 - 2.5f*y0 + 2.0f*y1 - 0.5f*y2;
float const c3 = 0.5f*(y2-y_1) + 1.5f*(y0-y1);

float const output=((c3*x+c2)*x+c1)*x+c0;

// add to delay buffer
buffer[counter]=in+output*feedback;

// increment delay counter
counter++;

// clip delay counter
if(counter>=MAX_WG_DELAY)
counter=0;

// return output
return output;
}

float buffer[MAX_WG_DELAY];
int counter;
};
```

Clipping without branching

Type : Min, max and clip
References : Posted by Laurent de Soras

Notes :
It may reduce accuracy for small numbers. I.e. if you clip to [-1; 1], fractional part of the result will be quantized to 23 bits (or more, depending on the bit depth of the temporary results). Thus, 1e-20 will be rounded to 0. The other (positive) side effect is the denormal number elimination.

Code :

```
float max (float x, float a)
{
   x -= a;
   x += fabs (x);
   x *= 0.5;
   x += a;
   return (x);
}

float min (float x, float b)
{
   x = b - x;
   x += fabs (x)
   x *= 0.5;
   x = b - x;
   return (x);
}

float clip (float x, float a, float b)
{
   x1 = fabs (x-a);
   x2 = fabs (x-b);
   x = x1 + (a+b);
   x -= x2;
   x *= 0.5;
   return (x);
}
```

Coefficients for Daubechies wavelets 1-38

Type : wavelet transform
References : Computed by Kazuo Hatano, Compiled and verified by Olli Niemitalo
Linked file : daub.h (this linked file is included below)

Linked files

```
/* Coefficients for Daubechies wavelets 1-38
 * ----------------------------------------
 * 2000.08.10 - Some more info added.
 *
 * Computed by Kazuo Hatano, Aichi Institute of Technology.
 * ftp://phase.etl.go.jp/pub/phase/wavelet/index.html
 *
 * Compiled and verified by Olli Niemitalo.
 *
 * Discrete Wavelet Transformation (DWT) breaks a signal down into
 * subbands distributed logarithimically in frequency, each sampled
 * at a rate that has a natural proportion to the frequencies in that
 * band. The traditional fourier transformation has no time domain
 * resolution at all, or when done using many short windows on a
 * longer data, equal resolution at all frequencies. The distribution
 * of samples in the time and frequency domain by DWT is of form:
 *
 * log f
 *  |XXXXXXXXXXXXXXXX  X = a sample
 *  |X X X X X X X X   f = frequency
 *  |X   X   X   X     t = time
 *  |X       X
 *  |X
 *   ----------------t
 *
 * Single
 * subband decomposition         and      reconstruction:
 *
 *     -> high -> decimate -------------> dilute -> high
 *    |   pass    by 2      high subband  by 2      pass \
 * in |                                                  + out
 *    |                                                /  =in
 *     -> low  -> decimate -------------> dilute -> low
 *        pass    by 2      low subband   by 2      pass
 *
 * This creates two subbands from the input signal, both sampled at half
 * the original frequency. The filters approximate halfband FIR filters
 * and are determined by the choice of wavelet. Using Daubechies wavelets
 * (and most others), the data can be reconstructed to the exact original
 * even when the halfband filters are not perfect. Note that the amount
```

```
* of information (samples) stays the same throughout the operation.
*
* Decimation by 2: ABCDEFGHIJKLMNOPQR -> ACEGIKMOQ
*   Dilution by 2: ACEGIKMOQ -> A0C0E0G0I0K0M0O0Q0
*
* To get the logarithmic resolution in frequency, the low subband is
* re-transformed, and again, the low subband from this transformation
* gets the same treatment etc.
*
* Decomposition:
*
*     -> high -> decimate ---------------------------------> subband0
*    |   pass    by 2
* in |                 -> high -> decimate ---------------> subband1
*    |                |   pass    by 2
*     -> low -> decim |                 -> high -> decim -> subband2
*        pass   by 2  |                |   pass    by 2
*                     -> low -> decim |
*                        pass   by 2  |   .   down to what suffices
*                                     -> .    or if periodic data,
*                                        .     until short of data
* Reconstruction:
*
* subband0 ----------------------------------> dilute -> high
*                                               by 2      pass \
* subband1 ------------------> dilute -> high                   + out
*                               by 2      pass \               /  =in
* subband2 -> dilute -> high                   + dilute -> low
*             by 2      pass \               /  by 2      pass
*                             + dilute -> low
* Start   .                  /  by 2      pass
* here!   . -> dilute -> low
*         .    by 2      pass
*
* In a real-time application, the filters introduce delays, so you need
* to compensate them by adding additional delays to less-delayed higher
* bands, to get the summation work as intended.
*
* For periodic signals or windowed operation, this problem doesn't exist -
* a single subband transformation is a matrix multiplication, with wrapping
* implemented in the matrix:
*
* Decomposition:
*
* |L0|   |C0  C1  C2  C3               | |I0|     L = lowpass output
* |H0|   |C3 -C2  C1 -C0               | |I1|     H = highpass output
* |L1|   |        C0  C1  C2  C3       | |I2|     I = input
* |H1| = |        C3 -C2  C1 -C0       | |I3|     C = coefficients
* |L2|   |                C0  C1  C2  C3| |I4|
```

```
 * |H2|   |               C3 -C2  C1 -C0| |I5|
 * |L3|   |C2  C3                 C0  C1| |I6|
 * |H3|   |C1 -C0                 C3 -C2| |I7|     Daubechies 4-coef:
 *
 *      1+sqrt(3)        3+sqrt(3)        3-sqrt(3)        1-sqrt(3)
 * C0 = ---------   C1 = ---------   C2 = ---------   C3 = ---------
 *      4 sqrt(2)        4 sqrt(2)        4 sqrt(2)        4 sqrt(2)
 *
 * Reconstruction:
 *
 * |I0|   |C0  C3                 C2  C1| |L0|
 * |I1|   |C1 -C2                 C3 -C0| |H0|
 * |I2|   |C2  C1  C0  C3               | |L1|
 * |I3| = |C3 -C0  C1 -C2               | |H1|
 * |I4|   |        C2  C1  C0  C3       | |L2|
 * |I5|   |        C3 -C0  C1 -C2       | |H2|
 * |I6|   |                C2  C1  C0  C3| |L3|
 * |I7|   |                C3 -C0  C1 -C2| |H3|
 *
 * This file contains the lowpass FIR filter coefficients. Highpass
 * coefficients you get by reversing tap order and multiplying by
 * sequence 1,-1, 1,-1, ... Because these are orthogonal wavelets, the
 * analysis and reconstruction coefficients are the same.
 *
 * A coefficient set convolved by its reverse is an ideal halfband lowpass
 * filter multiplied by a symmetric windowing function. This creates the
 * kind of symmetry in the frequency domain that enables aliasing-free
 * reconstruction. Daubechies wavelets are the minimum-phase, minimum
 * number of taps solutions for a number of vanishing moments (seven in
 * Daub7 etc), which determines their frequency selectivity.
 */

const double Daub1[2] = {
 7.07106781186547524400844362104849039284835937688474036588339e-01,
 7.07106781186547524400844362104849039284835937688474036588339e-01};

const double Daub2[4] = {
 4.82962913144534143374871599864448683816952419504202275201171 5e-01,
 8.36516303737807905575293780916873203459370388348439293495341 4e-01,
 2.24143868042013381025972762240400355467883518184271761387168 3e-01,
-1.29409522551260381174449418812024164174534450659965256907001 6e-01};

const double Daub3[6] = {
 3.32670552950082615998511589139005630012923399245068359708470 5e-01,
 8.06891509311092576494493604088713490519297394994823618165092 0e-01,
 4.59877502118491570095151942147616720808110177431492306643386 7e-01,
-1.35011020010254588696389906699374480562219845223781191975686 2e-01,
-8.54412738820266616928191691817733115361976389880866297635174 8e-02,
 3.52262918857095366027406647155100293277583879174316103989340 6e-02};
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const double Daub4[8] = {
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 7.14846570552915647089921955273992603707608401099308175845011 0e-01,
 6.30880767929858907881716338300615220203222922677195117405747 3e-01,
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 3.08413818355607636272193625349590501703148217200340334182121 9e-02,
 3.28830116668851997354075135492443886645419411375497125972727 8e-02,
-1.05974017850690321048832085240272291810999649063764198348497 4e-02};

const double Daub5[10] = {
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-3.22448695846383746484797550621349283135649841637984722543426 8e-02,
 7.75714938400457135231304893886018198062309945201252798321014 6e-02,
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-1.25807519990819994685097399317757929492045916260978502016923 2e-02,
 3.33572528547377127799818341581735574763652474230531509970642 8e-03};

const double Daub6[12] = {
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-1.29766867567261935562289605876585460845233749223581470159931 0e-01,
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const double Daub7[14] = {
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-1.43906003928564975405068362213046001795273570549908483440175 3e-01,
-2.24036184993874982638140420233250964475783089677324655266509 5e-01,
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-1.80164070404749091526826291273955096258565146964109062532 3864e-03,
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const double Daub8[16] = {
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const double Daub9[18] = {
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const double Daub10[20] = {
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const double Daub11[22] = {
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const double Daub12[24] = {
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const double Daub13[26] = {
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const double Daub14[28] = {
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const double Daub15[30] = {
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const double Daub16[32] = {
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const double Daub17[34] = {
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const double Daub18[36] = {
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const double Daub19[38] = {
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const double Daub20[40] = {
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 1.600719935641106973482800861166599685169395465055048951307626e-01,
```

```
 3.30775781411014651149363753440461175480076867704157703075730 6e-01,
 4.96591175311718097659917114771870893935241483895172608756441 9e-01,
 4.93356078517100797572848534699731706496951362359435909111580 4e-01,
 2.13050571355578513828674335345856245125562466595116044512230 7e-01,
-1.82867667708335890797554850794623913521822318504141063292481 5e-01,
-3.21675637808997862848347172540691636192984194052818905900254 8e-01,
-6.22665060478243222664336016047876584756586210104559718031049 0e-02,
 2.32125963835353108502870810428599499867161556366285807926299 6e-01,
 1.49985119618717019958640345378892730729822602826260302863575 8e-01,
-1.41795685973059621671005314452233027639259105537583065451908 0e-01,
-1.59912565158244361828853321452353493780420884438610263917769 3e-01,
 8.56381215561510574161221781436916531348712964553600185027698 7e-02,
 1.41414734073382680088468311937917059409260617491575528349615 3e-01,
-5.65864586307273814568178765784332064681550941063511423494790 2e-02,
-1.14731170710744375239414401945894277971566548923016995020102 2e-01,
 4.30958954330476428813787122361603062424656868359540879207860 2e-02,
 8.72043982620397501191071416415445676207378612423308847185586 8e-02,
-3.66051034028742956737207103950677237256793871094343283890824 7e-02,
-6.17662087084131599360473670561324624189749778237337911398117e-02,
 3.19898775315378063081838113636685902613703545057663113417687 5e-02,
 4.00549811051159482095208708624111430903857737936673295964854 8e-02,
-2.68914938808945143855085176771596731341789039328723670007207 1e-02,
-2.31141340205493168085691355358562124892530386554020335718076 8e-02,
 2.09046452556552434021598236535134209467026149152683167268224 4e-02,
 1.12904972786859648427008148776154423285111589144984396715165 7e-02,
-1.47018820653986821370898640281660504564848122466243511408824 5e-02,
-4.13130665603108927412323110332674572318813454852093815799570 2e-03,
 9.21478503219718051203153487018173400352286164590389450430228 6e-03,
 5.62571574840353200574156559488114875706670343721452210174094 1e-04,
-5.07131450921834809393506141750566300200682132395875264964032 9e-03,
 7.16982182106401925778416536489491562188854149677337043588958 5e-04,
 2.40069778189097318389230691408259214398414055021013013953519 3e-03,
-8.44862666553777500906893785146585697325136301092400331464361 2e-04,
-9.42461407722737796401594227178009828391023063990801877858891 0e-04,
 5.81075975053286366202032106367819663340955570698147672398831 2e-04,
 2.81763925038067074601804896753560819012352318061296106260367 2e-04,
-3.03102046072661199360062902032978468249647710647042778774785 5e-04,
-4.55568269668420274688683005987764360677217149927938344795290e-05,
 1.26204335016617070538234653713181770136152238790491733595870 5e-04,
-1.15540910383371719262847904798346095338195934264237417582286 3e-05,
-4.17514164854039779729632506577571130919741192628941246828080 1e-05,
 1.33417614992135038254750345728606092221807003133013760142732 4e-05,
 1.03735918404559979563225833501006510352495984496609487021768 7e-05,
-6.45673042846961916037991043961757542098697239413712195380623 6e-06,
-1.55084435011860257585338014852591299940129247318553439574037 1e-06,
 2.14996026993966520778954819979077059689025240507639488560603 8e-06,
-8.48708758607259307186980526608942662960647987698222184083309 8e-08,
-5.18773373887414442600847468337854236806631000060282309600918 7e-07,
 1.39637754550835548122796158105996118451987250249346201026463 3e-07,
```

```
 8.4003510468959655269335871767812795079530806692593187229105 23e-08,
-4.8847579374592867620821854116087639640410103921019148549181 57e-08,
-5.4242748002872985111266841748544149284475217106644764109739 81e-09,
 1.0347045392748584809240464909528039373282395372229081594510 39e-08,
-1.4363294877951357068545398569792759111836284765216362516608 49e-09,
-1.3491977539834488218503817708897863012467413043079349559971 11e-09,
 5.2611325573575984945357666387726245721003322091989796590770 82e-10,
 6.7323364901893086857406269641826231597597675367248440301645 51e-11,
-8.2782565225381347273306929381589911153353846117958747675217 31e-11,
 1.1016929345994545511508326221602242312801953629194985409136 58e-11,
 6.2915373170395085815809136208591408358528863089895841981661 74e-12,
-2.4847892375636428570433612145027607236114685918332626758522 42e-12,
 2.6264965040652520704882828764705253798514295383894815764546 18e-14,
 1.8086612362745305822670848463439593770859220190678081456352 63e-13,
-4.2498178195714630069666163715542065728631225627449167965564 74e-14,
-4.5633971621273731091016916430479237477965634491940756218544 91e-16,
 2.0450996767889889078022725644023100953986410928193671672529 52e-15,
-4.4053070424834613424490271398383016110068352854550501558428 65e-16,
 4.3045968395587900162518674771227915088496976880581690531344 63e-17,
-1.7161524510887441887324042817379642777130260872242482355410 71e-18};
```

Constant-time exponent of 2 detector

References : Posted by Brent Lehman (mailbjl[AT]yahoo.com)

Notes :
In your common FFT program, you want to make sure that the frame you're working with has a size that is a power of 2. This tells you in just a few operations. Granted, you won't be using this algorithm inside a loop, so the savings aren't that great, but every little hack helps ;)

Code :
```
// Quit if size isn't a power of 2
if ((-size ^ size) & size) return;

// If size is an unsigned int, the above might not compile.
// You'd want to use this instead:
if (((~size + 1) ^ size) & size) return;
```

Conversions on a PowerPC

Type : motorola ASM conversions
References : Posted by James McCartney
Code :

```
double ftod(float x) { return (double)x;
00000000: 4E800020  blr
    // blr == return from subroutine, i.e. this function is a noop

float dtof(double x) { return (float)x;
00000000: FC200818  frsp        fp1,fp1
00000004: 4E800020  blr

int ftoi(float x) { return (int)x;
00000000: FC00081E  fctiwz      fp0,fp1
00000004: D801FFF0  stfd        fp0,-16(SP)
00000008: 8061FFF4  lwz         r3,-12(SP)
0000000C: 4E800020  blr

int dtoi(double x) { return (int)x;
00000000: FC00081E  fctiwz      fp0,fp1
00000004: D801FFF0  stfd        fp0,-16(SP)
00000008: 8061FFF4  lwz         r3,-12(SP)
0000000C: 4E800020  blr

double itod(int x) { return (double)x;
00000000: C8220000  lfd         fp1,@1558(RTOC)
00000004: 6C608000  xoris       r0,r3,$8000
00000008: 9001FFF4  stw         r0,-12(SP)
0000000C: 3C004330  lis         r0,17200
00000010: 9001FFF0  stw         r0,-16(SP)
00000014: C801FFF0  lfd         fp0,-16(SP)
00000018: FC200828  fsub        fp1,fp0,fp1
0000001C: 4E800020  blr

float itof(int x) { return (float)x;
00000000: C8220000  lfd         fp1,@1558(RTOC)
00000004: 6C608000  xoris       r0,r3,$8000
00000008: 9001FFF4  stw         r0,-12(SP)
0000000C: 3C004330  lis         r0,17200
00000010: 9001FFF0  stw         r0,-16(SP)
00000014: C801FFF0  lfd         fp0,-16(SP)
00000018: EC200828  fsubs       fp1,fp0,fp1
0000001C: 4E800020  blr
```

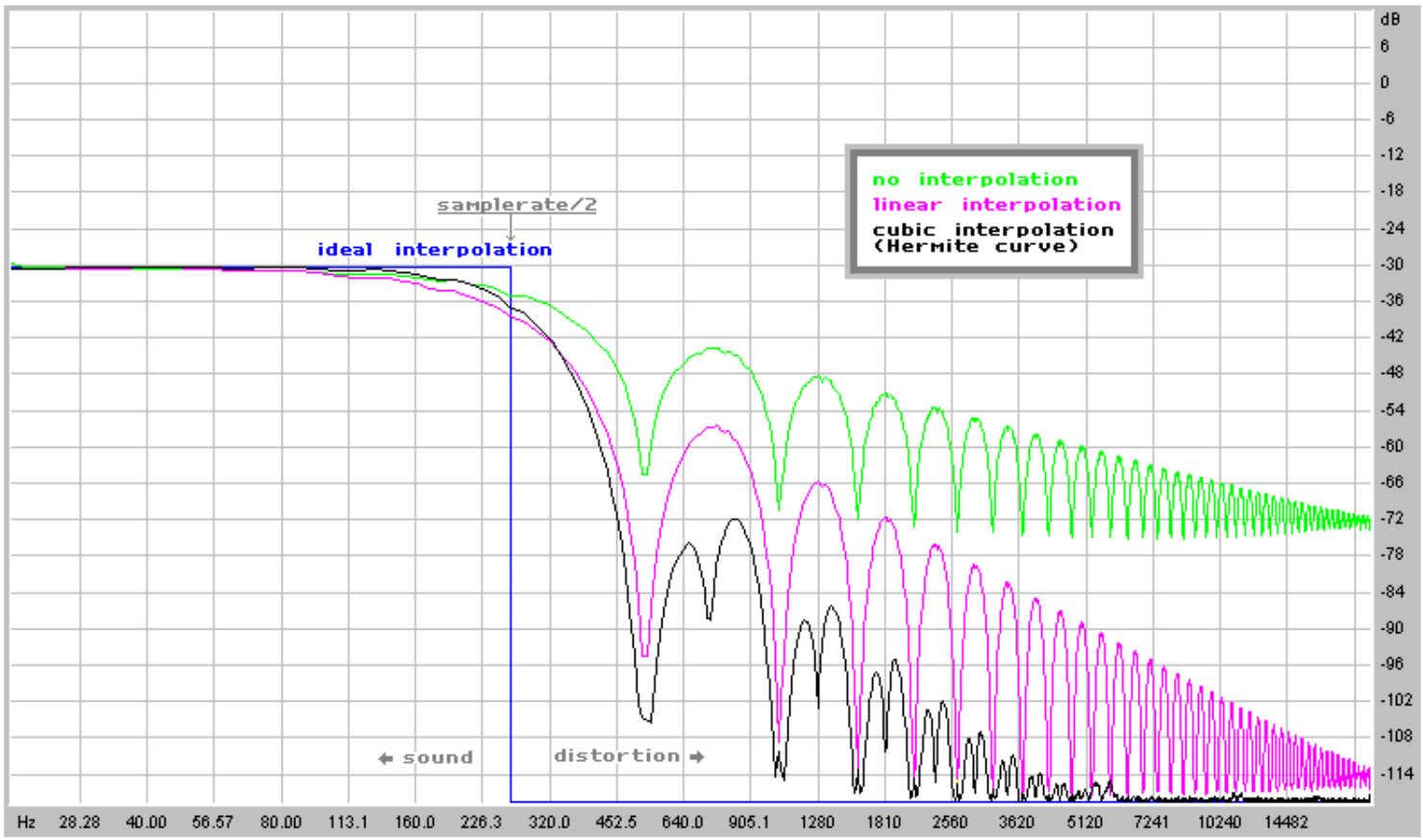
Cubic interpollation

Type : interpollation
References : Posted by Olli Niemitalo
Linked file : other001.gif (this linked file is included below)

Notes :
(see linkfile)
finpos is the fractional, inpos the integer part.

Code :
```
xm1 = x [inpos - 1];
x0  = x [inpos + 0];
x1  = x [inpos + 1];
x2  = x [inpos + 2];
a = (3 * (x0-x1) - xm1 + x2) / 2;
b = 2*x1 + xm1 - (5*x0 + x2) / 2;
c = (x1 - xm1) / 2;
y [outpos] = (((a * finpos) + b) * finpos + c) * finpos + x0;
```

Linked files



Cubic polynomial envelopes

Type : envellope generation
References : Posted by Andy Mucho

Notes :
This function runs from:
startlevel at Time=0
midlevel at Time/2
endlevel at Time
At moments of extreme change over small time, the function can generate out of range (of the 3 input level) numbers, but isn't really a problem in actual use with real numbers, and sensible/real times..

Code :

```
time = 32
startlevel = 0
midlevel = 100
endlevel = 120
k = startlevel + endlevel - (midlevel * 2)
r = startlevel
s = (endlevel - startlevel - (2 * k)) / time
t = (2 * k) / (time * time)
bigr = r
bigs = s + t
bigt = 2 * t

for(int i=0;i<time;i++)
{
bigr = bigr + bigs
bigs = bigs + bigt
}
```

Delay time calculation for reverberation

References : Posted by Andy Mucho

Notes :
This is from some notes I had scribbled down from a while back on automatically calculating diffuse delays. Given an intial delay line gain and time, calculate the times and feedback gain for numlines delay lines..

Code :

```
int   numlines = 8;
float t1 = 50.0;        // d0 time
float g1 = 0.75;        // d0 gain
float rev = -3*t1 / log10 (g1);

for (int n = 0; n < numlines; ++n)
{
  float dt = t1 / pow (2, (float (n) / numlines));
  float g = pow (10, -((3*dt) / rev));
  printf ("d%d t=%.3f g=%.3f\n", n, dt, g);
}

The above with t1=50.0 and g1=0.75 yields:

d0 t=50.000 g=0.750
d1 t=45.850 g=0.768
d2 t=42.045 g=0.785
d3 t=38.555 g=0.801
d4 t=35.355 g=0.816
d5 t=32.421 g=0.830
d6 t=29.730 g=0.843
d7 t=27.263 g=0.855

To go more diffuse, chuck in dual feedback paths with a one cycle delay
effectively creating a phase-shifter in the feedback path, then things get
more exciting.. Though what the optimum phase shifts would be I couldn't
tell you right now..
```

Denormal DOUBLE variables, macro

References : Posted by Jon Watte

Notes :
Use this macro if you want to find denormal numbers and you're using doubles...

Code :
```
#if PLATFORM_IS_BIG_ENDIAN
#define INDEX 0
#else
#define INDEX 1
#endif
inline bool is_denormal( double const & d ) {
  assert( sizeof( d ) == 2*sizeof( int ) );
  int l = ((int *)&d)[INDEX];
  return (l&0x7fe00000) != 0;
}
```

Denormal numbers

References : Compiled by Merlijn Blaauw
Linked file : other001.txt (this linked file is included below)

Notes :
this text describes some ways to avoid denormalisation. Denormalisation happens when FPU's go mad processing very small numbers

---

Linked files

```
Denormal numbers
----------------

Here's a small recap on all proposed solutions to prevent the FPU from
denormalizing:
When you feed the FPU really small values (what's the exact 'threshold'
value?) the CPU will go into denormal mode to maintain precision; as such
precision isn't required for audio applications and denormal operations are
MUCH slower than normal operations, we want to avoid them.

All methods have been proposed by people other than me, and things I say
here may be inaccurate or completely wrong :). 'Algorithms' have not been
tested and can be implemented more efficiently no doubt.
If I made some horrible mistakes, or left some stuff out, please let me/the
list know.


** Checking denormal processing solution:

To detect if a denormal number has occured, just trace your code and
look up on the STAT FPU register ... if 0x0002 flag is set then a denormal
operation occured (this flag stays fixed until next FINIT)

** Checking denormal macro solution:

NOTES:

This is the least computationally efficient method of the lot, but has the
advantage of being inaudible.
Please note that in every feedback loop, you should also check for
denormals (rendering it useless on algorithms with loads of filters,
feedback loops, etc).

CODE:

#define IS_DENORMAL(f) (((*(unsigned int *)&f)&0x7f800000)==0)

// inner-loop:

is1 = *(++in1); is1 = IS_DENORMAL(is1) ? 0.f : is1;
is2 = *(++in2); is2 = IS_DENORMAL(is2) ? 0.f : is2;

** Adding noise solution:
```

NOTES:

Less overhead than the first solution, but still 2 mem accesses. Because a number of the values of denormalBuf will be denormals themselves, there will always be *some* denormal overhead. However, a small percentage denormals probably isn't a problem.
Use this eq. to calculate the appropriate value of id (presuming rand() generates evenly distrubuted values):
id = 1/percentageOfDenormalsAllowed * denormalThreshold
(I do not know the exact value of the denormalThreshold, the value at which the FPU starts to denormal).

Possible additions to this algorithm include, noiseshaping the noise buffer, which would allow a smaller value of percentageOfDenormalsAllowed without becomming audible - however, in some algorithms, with filters and such, I think this might cause the noise to be removed, thus rendering it useless. Checking for denormals on noise generation might have a similar effect I suspect.

CODE:

```
// on construction:
float **denormalBuf = new float[getBlockSize()];

float id = 1.0E-20;

for (int i = 0; i < getBlockSize(); i++)
{
   denormalBuf[i] = (float)rand()/32768.f * id;
}

// inner-loop:

float noise = *(++noiseBuf);
is1 = *(++in1) + noise;
is2 = *(++in2) + noise;
..
```

** Flipping number solution:

NOTES:

In my opinion the way-to-go method for most applications; very little overhead (no compare/if/etc or memory access needed), there isn't a percentage of the values that will denormal and it will be inaudible in most cases.
The exact value of id will probably differ from algorithm to algorithm, but the proposed value of 1.0E-30 seemed too small to me.

CODE:

```
// on construction:
float id = 1.0E-25;
```

```
// inner-loop:
is1 = *(++in1) + id;
is2 = *(++in2) + id;
id = -id;
..
```

** Adding offset solution:

NOTES:

This is the most efficient method of the lot and is also inaudible.
However, some filters will probably remove the added DC offset, thus rendering it useless.

CODE:

```
// inner-loop:

is1 = *(++in1) + 1.0E-25;
is2 = *(++in2) + 1.0E-25;
```

** Fix-up solution

You can also walk through your filter and clamp any numbers which are close enough to being denormal at the end of each block, as long as your blocks are not too large. For instance, if you implement EQ using a bi-quad, you can check the delay slots at the end of each process() call, and if any slot has a magnitude of 10^-15 or smaller, you just clamp it to 0. This will ensure that your filter doesn't run for long times with denormal numbers; ideally (depending on the coefficients) it won't reach 10^-35 from the 10^-15 initial state within the time of one block of samples.
That solution uses the least cycles, and also has the nice property of generating absolute-0 output values for long stretches of absolute-0 input values; the others don't.

Denormal numbers, the meta-text

References : Laurent de Soras
Linked file : denormal.pdf

Notes :
This very interesting paper, written by Laurent de Soras (www.ohmforce.com) has everything you ever wanted to know about denormal numbers! And it obviously descibes how you can get rid of them too!

(see linked file)

DFT

Type : fourier transform
References : Posted by Andy Mucho
Code :

```
AnalyseWaveform(float *waveform, int framesize)
{
   float aa[MaxPartials];
   float bb[MaxPartials];
   for(int i=0;i<partials;i++)
   {
     aa[i]=0;
     bb[i]=0;
   }

   int hfs=framesize/2;
   float pd=pi/hfs;
   for (i=0;i<framesize;i++)
   {
     float w=waveform[i];
     int im = i-hfs;
     for(int h=0;h<partials;h++)
     {
        float th=(pd*(h+1))*im;
        aa[h]+=w*cos(th);
        bb[h]+=w*sin(th);
     }
   }
   for (int h=0;h<partials;h++)
       amp[h]= sqrt(aa[h]*aa[h]+bb[h]*bb[h])/hfs;
}
```

Digital RIAA equalization filter coefficients

Type : RIAA
References : Posted by Frederick Umminger

Notes :
Use at your own risk. Confirm correctness before using. Don't assume I didn't goof something up.

-Frederick Umminger

Code :

```
The "turntable-input software" thread inspired me to generate some
coefficients for a digital RIAA equalization filter. These coefficients
were found by matching the magnitude response of the s-domain transfer
function using some proprietary Matlab scripts. The phase response may or
may not be totally whacked.

The s-domain transfer function is

R3(1+R1*C1*s)(1+R2*C2*s)/(R1(1+R2*C2*s) + R2(1+R1*C1*s) +
R3(1+R1*C1*s)(1+R2*C2*s))

where

R1 = 883.3k
R2 = 75k
R3 = 604
C1 = 3.6n
C2 = 1n

This is based on the reference circuit found in
http://www.hagtech.com/pdf/riaa.pdf

The coefficients of the digital transfer function b(z^-1)/a(z^-1) in
descending powers of z, are:

44.1kHz
b = [ 0.02675918611906  -0.04592084787595   0.01921229297239]
a = [ 1.00000000000000  -0.73845850035973  -0.17951755477430]
error +/- 0.25dB

48kHz
b = [  0.02675918611906  -0.04592084787595   0.01921229297239]
a = [  1.00000000000000  -0.73845850035973  -0.17951755477430]
error +/- 0.15dB

88.2kHz
```

```
b = [  0.04872204977233  -0.09076930609195   0.04202280710877]
a = [ 1.00000000000000  -0.85197860443215  -0.10921171201431]
error +/- 0.01dB


96kHz
b = [ 0.05265477122714  -0.09864197097385   0.04596474352090  ]
a = [  1.00000000000000  -0.85835597216218  -0.10600020417219 ]
error +/- 0.006dB
```

Discrete Summation Formula (DSF)

References : Stylson, Smith and others... (posted by Alexander Kritov)

Notes :
Buzz uses this type of synth.
For cool sounds try to use variable,
for example a=exp(-x/12000)*0.8 // x- num.samples

Code :

```
double DSF (double x,  // input
            double a,  // a<1.0
            double N,  // N<SmplFQ/2,
            double fi) // phase
{
  double s1 = pow(a,N-1.0)*sin((N-1.0)*x+fi);
  double s2 = pow(a,N)*sin(N*x+fi);
  double s3 = a*sin(x+fi);
  double s4 =1.0 - (2*a*cos(x)) +(a*a);
  if (s4==0)
     return 0;
  else
     return (sin(fi) - s3 - s2 +s1)/s4;
}
```

Dither code

Type : Dither with noise-shaping
References : Posted by Paul Kellett

Notes :
This is a simple implementation of highpass triangular-PDF dither (a good general-purpose dither) with optional 2nd-order noise shaping (which lowers the noise floor by 11dB below 0.1 Fs).
The code assumes input data is in the range +1 to -1 and doesn't check for overloads!

To save time when generating dither for multiple channels you can re-use lower bits of a previous random number instead of calling rand() again. e.g. r3=(r1 & 0x7F)<<8;

Code :

```
  int   r1, r2;                 //rectangular-PDF random numbers
  float s1, s2;                 //error feedback buffers
  float s = 0.5f;               //set to 0.0f for no noise shaping
  float w = pow(2.0,bits-1);    //word length (usually bits=16)
  float wi= 1.0f/w;
  float d = wi / RAND_MAX;      //dither amplitude (2 lsb)
  float o = wi * 0.5f;          //remove dc offset
  float in, tmp;
  int   out;

//for each sample...

  r2=r1;                                //can make HP-TRI dither by
  r1=rand();                            //subtracting previous rand()

  in += s * (s1 + s1 - s2);             //error feedback
  tmp = in + o + d * (float)(r1 - r2); //dc offset and dither

  out = (int)(w * tmp);                 //truncate downwards
  if(tmp<0.0f) out--;                   //this is faster than floor()

  s2 = s1;
  s1 = in - wi * (float)out;            //error
```

Dithering

References : Paul Kellett
Linked file : nsdither.txt (this linked file is included below)

Notes :
(see linked file)

Linked files

```
Noise shaped dither  (March 2000)
-------------------

This is a simple implementation of highpass triangular-PDF dither with
2nd-order noise shaping, for use when truncating floating point audio
data to fixed point.

The noise shaping lowers the noise floor by 11dB below 5kHz (@ 44100Hz
sample rate) compared to triangular-PDF dither. The code below assumes
input data is in the range +1 to -1 and doesn't check for overloads!

To save time when generating dither for multiple channels you can do
things like this:  r3=(r1 & 0x7F)<<8; instead of calling rand() again.


  int   r1, r2;               //rectangular-PDF random numbers
  float s1, s2;               //error feedback buffers
  float s = 0.5f;             //set to 0.0f for no noise shaping
  float w = pow(2.0,bits-1);  //word length (usually bits=16)
  float wi= 1.0f/w;
  float d = wi / RAND_MAX;    //dither amplitude (2 lsb)
  float o = wi * 0.5f;        //remove dc offset
  float in, tmp;
  int   out;


//for each sample...

  r2=r1;                                //can make HP-TRI dither by
  r1=rand();                            //subtracting previous rand()

  in += s * (s1 + s1 - s2);             //error feedback
  tmp = in + o + d * (float)(r1 - r2); //dc offset and dither

  out = (int)(w * tmp);                 //truncate downwards
```

```
  if(tmp<0.0f) out--;                   //this is faster than floor()

  s2 = s1;
  s1 = in - wi * (float)out;            //error


--
paul.kellett@maxim.abel.co.uk
http://www.maxim.abel.co.uk
```

Double to Int

Type : pointer cast (round to zero, or 'trunctate')
References : Posted by many people, implementation by Andy M00cho

Notes :
-Platform independant, literally. You have IEEE FP numbers, this will work, as long as your not expecting a signed integer back larger than 16bits :)
-Will only work correctly for FP numbers within the range of [-32768.0,32767.0]
-The FPU must be in Double-Precision mode

Code :

```
typedef double lreal;
typedef float  real;
typedef unsigned long uint32;
typedef long int32;

   //2^36 * 1.5, (52-_shiftamt=36) uses limited precision to floor
   //16.16 fixed point representation

const lreal _double2fixmagic = 68719476736.0*1.5;
const int32 _shiftamt        = 16;

#if BigEndian_
        #define iexp_                           0
        #define iman_                           1
#else
        #define iexp_                           1
        #define iman_                           0
#endif //BigEndian_

// Real2Int
inline int32 Real2Int(lreal val)
{
   val= val + _double2fixmagic;
   return ((int32*)&val)[iman_] >> _shiftamt;
}

// Real2Int
inline int32 Real2Int(real val)
{
   return Real2Int ((lreal)val);
}

For the x86 assembler freaks here's the assembler equivalent:

__double2fixmagic    dd 000000000h,042380000h
```

```
fld    AFloatingPoint Number
fadd   QWORD PTR __double2fixmagic
fstp   TEMP
movsx  eax,TEMP+2
```

Early echo's with image-mirror technique

References : Donald Schulz
Linked file : early_echo.c (this linked file is included below)
Linked file : early_echo_eng.c (this linked file is included below)

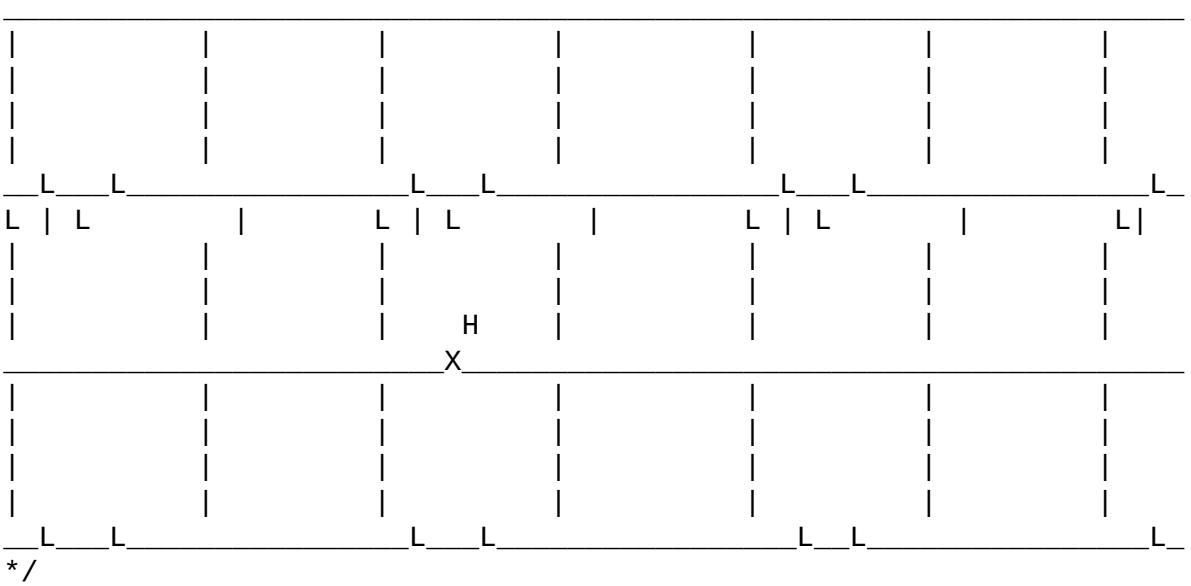
Notes :
(see linked files)
Donald Schulz's code for computing early echoes using the image-mirror method. There's an english and a german version.

Linked files

```
/*
From: "Donald Schulz" <d.schulz@gmx.de>
To: <music-dsp@shoko.calarts.edu>
Subject: [music-dsp] Image-mirror method source code
Date: Sun, 11 Jun 2000 15:01:51 +0200

A while ago I wrote a program to calculate early echo responses.
As there seems to be some interest in this, I now post it into the
public domain.

Have phun,

Donald.
*/

/*****************************************************************************
 *
 * Early Echo Computation using image-mirror method
 *
 * Position of listener, 2 sound-sources, room-size may be set.
 * Four early echo responses are calculated (from left sound source and
 * right sound source to left and right ear). Angle with which the sound
 * meets the ears is taken into account.
 * The early echo response from left sound source to left ear is printed
 * to screen for demonstration, the first table contains the delay times
 * and the second one the weights.
 *
 * Program is released into the public domain.
 *
 * Sorry for german comments :-(
 * Some frequently used german words:
 * hoerpos : listening position
 * breite : width
 * laenge : length
 * hoehe : height
 * daempfung : damping
 * links : left
 * rechts : right
 * Ohr    : ear
 * Winkel : angle
 * Wichtung : weight
```

```
 * Normierung : normalization
 *
 *
 * If someone does some improvements on this, I (Donald, d.schulz@gmx.de)
 * would be happy to get the improved code.
 *
 ************************************************************************/



#include <math.h>
#include <stdio.h>

#define early_length 0x4000     /* Laenge der Puffer fuer early-echo */
#define early_length_1 0x3fff

#define early_tap_num 20        /* Anzahl an early-echo taps */

#define breite 15.0              /* 15 m breiter Raum (x)*/
#define laenge 20.0             /* 20 m lang (y) */
#define hoehe 10.0               /* 10 m hoch (z)*/
#define daempfung 0.999         /* Daempfungsfaktor bei Reflexion */

#define hoerposx 7.91            /* hier sitzt der Hoerer (linkes Ohr) */
#define hoerposy 5.0
#define hoerposz 2.0

#define leftposx 5.1            /* hier steht die linke Schallquelle */
#define leftposy 16.3
#define leftposz 2.5

#define rightposx 5.9            /* hier steht die rechte Schallquelle */
#define rightposy 6.3
#define rightposz 1.5

#define i_length 32             /* Laenge des Eingangs-Zwischenpuffers */
#define i_length_1 31
#define o_length 32             /* Laenge des Ausgangs-Zwischenpuffers */
#define o_length_1 31


float *early_l2l;  /* linker Kanal nach linkem Ohr */
float *early_l2r;  /* linker Kanal nach rechtem Ohr */
float *early_r2l;  /* rechter Kanal nach linkem Ohr */
float *early_r2r;  /* rechter Kanal nach rechtem Ohr */

int early_pos=0;

int e_delays_l2l[early_tap_num];       /* Delays der early-echos */
float e_values_l2l[early_tap_num];     /* Gewichtungen der delays */
int e_delays_l2r[early_tap_num];
float e_values_l2r[early_tap_num];
int e_delays_r2l[early_tap_num];
float e_values_r2l[early_tap_num];
int e_delays_r2r[early_tap_num];
float e_values_r2r[early_tap_num];
```

```
/* Early-echo Berechnung mittels Spiegelquellenverfahren

Raummodell:
H - Hoererposition
L - Spiegelschallquellen des linken Kanales
U - Koordinatenursprung
Raum sei 11 meter breit und 5 meter lang (1 Zeichen = 1 meter)
Linker Kanal stehe bei x=2  y=4  z=?
Hoerer stehe bei x=5 y=1 z=?
```




```
*/

main()
{
  int i,j,select;
  float dist_max;
  float x,y,z,xref,yref,zref;
  float x_pos,y_pos,z_pos;
  float distance,winkel;
  float wichtung;
  float normierungr,normierungl;


  early_l2l=(float *)malloc(early_length*sizeof(float));
  early_l2r=(float *)malloc(early_length*sizeof(float));;
  early_r2l=(float *)malloc(early_length*sizeof(float));;
  early_r2r=(float *)malloc(early_length*sizeof(float));;


  /* Erst mal Echos loeschen: */
  for (i=0;i<early_length;i++)
  early_l2l[i]=early_l2r[i]=early_r2l[i]=early_r2r[i]=0.0;

  dist_max=300.0*early_length/44100.0;  /* 300 m/s Schallgeschwindigkeit */

  /* Echo vom LINKEN Kanal auf linkes/rechtes Ohr berechnen */

  for (x=-ceil(dist_max/(2*laenge));x<=ceil(dist_max/(2*laenge));x++)
  for (y=-ceil(dist_max/(2*breite));y<=ceil(dist_max/(2*breite));y++)
```

```
for (z=-ceil(dist_max/(2*hoehe));z<=ceil(dist_max/(2*hoehe));z++)
{
  xref=2*x*breite;
  yref=2*y*laenge;
  zref=2*z*hoehe;
  for (select=0;select<8;select++)   /* vollstaendige Permutation */
  {
    if (select&1) x_pos=xref+leftposx;
    else x_pos=xref-leftposx;
    if (select&2) y_pos=yref+leftposy;
    else y_pos=yref-leftposy;
    if (select&4) z_pos=zref+leftposz;
    else z_pos=zref-leftposz;
    /* Jetzt steht die absolute Position der Quelle in ?_pos */

    /* Relative Position zum linken Ohr des Hoerers bestimmen: */
    x_pos-=hoerposx;
    y_pos-=hoerposy;
    z_pos-=hoerposz;


    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147)<early_length)
    {
      /* Einfallswinkel im Bereich -Pi/2 bis Pi/2 ermitteln: */
      winkel=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=links=>Verstaerkung=1 Pi=rechts=>Verstaerkung=0.22 (?) */
        winkel+=3.1415926/2;     wichtung = 1 - winkel/4;
      }
      else            /* Klang kommt von hinten: */
      {
        winkel-=3.1415926/2;  wichtung= 1 + winkel/4;
      }
      /* Early-echo gemaess Winkel und Entfernung gewichten: */
      early_l2l[(int) (distance*147.)]+=wichtung/(pow(distance,3.1));
    }

    /* Relative Position zum rechten Ohr des Hoerers bestimmen: */
    x_pos-=0.18;  /* Kopf ist 18 cm breit */

    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147)<early_length)
    {
      /* Einfallswinkel im Bereich -Pi/2 bis Pi/2 ermitteln: */
      winkel=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=links=>Verstaerkung=1 Pi=rechts=>Verstaerkung=0.22 (?) */
        winkel-=3.1415926/2;     wichtung = 1 + winkel/4;
      }
      else            /* Klang kommt von hinten: */
      {
        winkel+=3.1415926/2;  wichtung= 1 - winkel/4;
```

```
      }
      /* Early-echo gemaess Winkel und Entfernung gewichten: */
      early_l2r[(int) (distance*147.)]+=wichtung/(pow(distance,3.1));
    }
  }
}

/* Echo vom RECHTEN Kanal auf linkes/rechtes Ohr berechnen */

for (x=-ceil(dist_max/(2*laenge));x<=ceil(dist_max/(2*laenge));x++)
for (y=-ceil(dist_max/(2*breite));y<=ceil(dist_max/(2*breite));y++)
for (z=-ceil(dist_max/(2*hoehe));z<=ceil(dist_max/(2*hoehe));z++)
{
  xref=2*x*breite;
  yref=2*y*laenge;
  zref=2*z*hoehe;
  for (select=0;select<8;select++)   /* vollstaendige Permutation */
  {
    if (select&1) x_pos=xref+rightposx;
    else x_pos=xref-rightposx;
    if (select&2) y_pos=yref+rightposy;
    else y_pos=yref-rightposy;
    if (select&4) z_pos=zref+rightposz;
    else z_pos=zref-rightposz;
    /* Jetzt steht die absolute Position der Quelle in ?_pos */

    /* Relative Position zum linken Ohr des Hoerers bestimmen: */
    x_pos-=hoerposx;
    y_pos-=hoerposy;
    z_pos-=hoerposz;


    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147.)<early_length)
    {
      /* Einfallswinkel im Bereich -Pi/2 bis Pi/2 ermitteln: */
      winkel=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=links=>Verstaerkung=1 Pi=rechts=>Verstaerkung=0.22 (?) */
        winkel+=3.1415926/2;     wichtung = 1 - winkel/4;
      }
      else            /* Klang kommt von hinten: */
      {
        winkel-=3.1415926/2;  wichtung= 1 + winkel/4;
      }
      /* Early-echo gemaess Winkel und Entfernung gewichten: */
      early_r2l[(int) (distance*147.)]+=wichtung/(pow(distance,3.1));
    }


    /* Und jetzt Early-Echo zweiter Kanal auf LINKES Ohr berechnen */
    x_pos-=0.18; /* Kopfbreite addieren */

    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
```

```
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147)<early_length)
    {
      /* Einfallswinkel im Bereich -Pi/2 bis Pi/2 ermitteln: */
      winkel=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=links=>Verstaerkung=1 Pi=rechts=>Verstaerkung=0.22 (?) */
        winkel-=3.1415926/2;     wichtung = 1 + winkel/4;
      }
      else            /* Klang kommt von hinten: */
      {
        winkel+=3.1415926/2;  wichtung= 1 - winkel/4;
      }
      /* Early-echo gemaess Winkel und Entfernung gewichten: */
      early_r2r[(int) (distance*147.)]+=wichtung/(pow(distance,3.1));
    }
  }
}


/* Und jetzt aus berechnetem Echo die ersten early_tap_num Werte holen */
/* Erst mal e's zuruecksetzen: */
for (i=0;i<early_tap_num;i++)
{
  e_values_l2l[i]=e_values_l2r[i]=0.0;
  e_delays_l2l[i]=e_delays_l2r[i]=0;  /* Unangenehme Speicherzugriffe vermeiden */
  e_values_r2l[i]=e_values_r2r[i]=0.0;
  e_delays_r2l[i]=e_delays_r2r[i]=0;
}

/* und jetzt e_delays und e_values extrahieren: */
j=0;
normierungl=normierungr=0.0;
for(i=0;i<early_length;i++)
{
  if (early_l2l[i]!=0)
  {
    e_delays_l2l[j]=i;
    e_values_l2l[j]=early_l2l[i];
    normierungl+=early_l2l[i];
    j++;
  }
  if (j==early_tap_num) break;
}
j=0;
for(i=0;i<early_length;i++)
{
  if (early_l2r[i]!=0)
  {
    e_delays_l2r[j]=i;
    e_values_l2r[j]=early_l2r[i];
    normierungr+=early_l2r[i];
    j++;
  }
```

```
    if (j==early_tap_num) break;
  }
  j=0;
  for(i=0;i<early_length;i++)
  {
    if (early_r2l[i]!=0)
    {
      e_delays_r2l[j]=i;
      e_values_r2l[j]=early_r2l[i];
      normierungl+=early_r2l[i];
      j++;
    }
    if (j==early_tap_num) break;
  }
  j=0;
  for(i=0;i<early_length;i++)
  {
    if (early_r2r[i]!=0)
    {
      e_delays_r2r[j]=i;
      e_values_r2r[j]=early_r2r[i];
      normierungr+=early_r2r[i];
      j++;
    }
    if (j==early_tap_num) break;
  }

  /* groessere von beiden Normierungen verwenden: */
  if (normierungr>normierungl) normierungr=normierungl;

  for (j=0;j<early_tap_num;j++)
  {
    e_values_l2l[j]/=normierungr;
    e_values_l2r[j]/=normierungr;
    e_values_r2l[j]/=normierungr;
    e_values_r2r[j]/=normierungr;
  }
  /* Ausgeben nur der l2l-Werte fuer schnelles Reverb */
  printf("int e_delays[%d]={",early_tap_num);
  for (j=0;j<(early_tap_num-1);j++)
    printf("%d, ",e_delays_l2l[j]);
  printf("%d};\n\n",e_delays_l2l[j]);

  printf("float e_values[%d]={",early_tap_num);
  for (j=0;j<(early_tap_num-1);j++)
    printf("%0.4f, ",e_values_l2l[j]);
  printf("%0.4f};\n\n",e_values_l2l[j]);

}
```

```
/*
From: "Donald Schulz" <d.schulz@gmx.de>
To: <music-dsp@shoko.calarts.edu>
Subject: [music-dsp] Image-mirror method source code
Date: Sun, 11 Jun 2000 15:01:51 +0200

A while ago I wrote a program to calculate early echo responses.
As there seems to be some interest in this, I now post it into the
public domain.

Have phun,

Donald.
*/

/*

i have taken the liberty of renaming some of donald's variables to make
his code easier to use for english speaking, non-german speakers. i did
a simple search and replace on all the german words mentioned in the
note below. all of the comments are still in german, but the english
variable names make the code easier to follow, at least for me. i don't
think that i messed anything up by doing this, but if something's not
working you might want to try the original file, just in case.

douglas

*/

/*****************************************************************************
 *
 * Early Echo Computation using image-mirror method
 *
 * Position of listener, 2 sound-sources, room-size may be set.
 * Four early echo responses are calculated (from left sound source and
 * right sound source to left and right ear). Angle with which the sound
 * meets the ears is taken into account.
 * The early echo response from left sound source to left ear is printed
 * to screen for demonstration, the first table contains the delay times
 * and the second one the weights.
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 * Program is released into the public domain.
 *
 * Sorry for german comments :-(
 * Some frequently used german words:
 * hoerpos : listening position
 * breite : width
 * laenge : length
 * hoehe : height
 * daempfung : damping
 * links : left
 * rechts : right
 * Ohr    : ear
 * Winkel : angle
 * Wichtung : weight
```
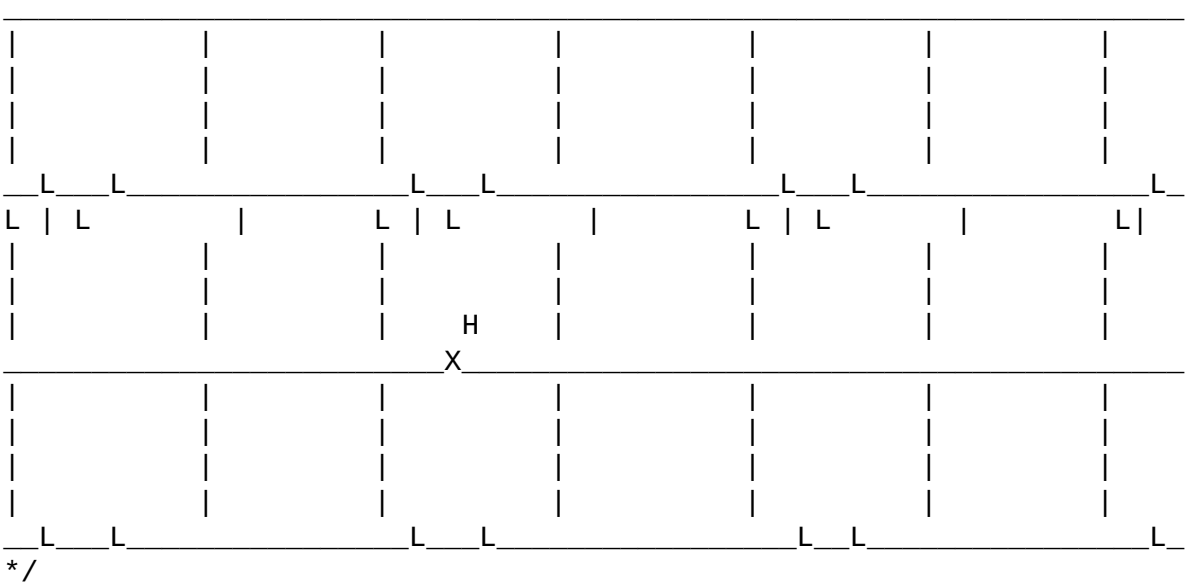

```
 * Normierung : normalization
 *
 *
 * If someone does some improvements on this, I (Donald, d.schulz@gmx.de)
 * would be happy to get the improved code.
 *
 ************************************************************************/



#include <math.h>
#include <stdio.h>

#define early_length 0x4000     /* Length der Puffer fuer early-echo */
#define early_length_1 0x3fff

#define early_tap_num 20        /* Anzahl an early-echo taps */

#define width 15.0              /* 15 m widthr Raum (x)*/
#define length 20.0             /* 20 m lang (y) */
#define height 10.0               /* 10 m hoch (z)*/
#define damping 0.999         /* Dampingsfaktor bei Reflexion */

#define listening_positionx 7.91            /* hier sitzt der Hoerer (linkes ear) */
#define listening_positiony 5.0
#define listening_positionz 2.0

#define leftposx 5.1            /* hier steht die linke Schallquelle */
#define leftposy 16.3
#define leftposz 2.5

#define rightposx 5.9            /* hier steht die rechte Schallquelle */
#define rightposy 6.3
#define rightposz 1.5

#define i_length 32             /* Length des Eingangs-Zwischenpuffers */
#define i_length_1 31
#define o_length 32             /* Length des Ausgangs-Zwischenpuffers */
#define o_length_1 31


float *early_l2l;  /* linker Kanal nach linkem ear */
float *early_l2r;  /* linker Kanal nach rechtem ear */
float *early_r2l;  /* rechter Kanal nach linkem ear */
float *early_r2r;  /* rechter Kanal nach rechtem ear */

int early_pos=0;

int e_delays_l2l[early_tap_num];       /* Delays der early-echos */
float e_values_l2l[early_tap_num];     /* Geweighten der delays */
int e_delays_l2r[early_tap_num];
float e_values_l2r[early_tap_num];
int e_delays_r2l[early_tap_num];
float e_values_r2l[early_tap_num];
int e_delays_r2r[early_tap_num];
float e_values_r2r[early_tap_num];
```

```
/* Early-echo Berechnung mittels Spiegelquellenverfahren

Raummodell:
H - Hoererposition
L - Spiegelschallquellen des linken Kanales
U - Koordinatenursprung
Raum sei 11 meter breit und 5 meter lang (1 Zeichen = 1 meter)
Linker Kanal stehe bei x=2  y=4  z=?
Hoerer stehe bei x=5 y=1 z=?
```



```
*/

main()
{
  int i,j,select;
  float dist_max;
  float x,y,z,xref,yref,zref;
  float x_pos,y_pos,z_pos;
  float distance,angle;
  float weight;
  float normalizationr,normalizationl;


  early_l2l=(float *)malloc(early_length*sizeof(float));
  early_l2r=(float *)malloc(early_length*sizeof(float));;
  early_r2l=(float *)malloc(early_length*sizeof(float));;
  early_r2r=(float *)malloc(early_length*sizeof(float));;


  /* Erst mal Echos loeschen: */
  for (i=0;i<early_length;i++)
  early_l2l[i]=early_l2r[i]=early_r2l[i]=early_r2r[i]=0.0;

  dist_max=300.0*early_length/44100.0;  /* 300 m/s Schallgeschwindigkeit */

  /* Echo vom LINKEN Kanal auf linkes/rechtes ear berechnen */

  for (x=-ceil(dist_max/(2*length));x<=ceil(dist_max/(2*length));x++)
  for (y=-ceil(dist_max/(2*width));y<=ceil(dist_max/(2*width));y++)
```

```
for (z=-ceil(dist_max/(2*height));z<=ceil(dist_max/(2*height));z++)
{
  xref=2*x*width;
  yref=2*y*length;
  zref=2*z*height;
  for (select=0;select<8;select++)   /* vollstaendige Permutation */
  {
    if (select&1) x_pos=xref+leftposx;
    else x_pos=xref-leftposx;
    if (select&2) y_pos=yref+leftposy;
    else y_pos=yref-leftposy;
    if (select&4) z_pos=zref+leftposz;
    else z_pos=zref-leftposz;
    /* Jetzt steht die absolute Position der Quelle in ?_pos */

    /* Relative Position zum linken ear des Hoerers bestimmen: */
    x_pos-=listening_positionx;
    y_pos-=listening_positiony;
    z_pos-=listening_positionz;


    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147)<early_length)
    {
      /* Einfallsangle im Bereich -Pi/2 bis Pi/2 ermitteln: */
      angle=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=left=>Verstaerkung=1 Pi=right=>Verstaerkung=0.22 (?) */
        angle+=3.1415926/2;     weight = 1 - angle/4;
      }
      else            /* Klang kommt von hinten: */
      {
        angle-=3.1415926/2;  weight= 1 + angle/4;
      }
      /* Early-echo gemaess angle und Entfernung gewichten: */
      early_l2l[(int) (distance*147.)]+=weight/(pow(distance,3.1));
    }

    /* Relative Position zum rechten ear des Hoerers bestimmen: */
    x_pos-=0.18;  /* Kopf ist 18 cm breit */

    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147)<early_length)
    {
      /* Einfallsangle im Bereich -Pi/2 bis Pi/2 ermitteln: */
      angle=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=left=>Verstaerkung=1 Pi=right=>Verstaerkung=0.22 (?) */
        angle-=3.1415926/2;     weight = 1 + angle/4;
      }
      else            /* Klang kommt von hinten: */
      {
        angle+=3.1415926/2;  weight= 1 - angle/4;
```

```
      }
      /* Early-echo gemaess angle und Entfernung gewichten: */
      early_l2r[(int) (distance*147.)]+=weight/(pow(distance,3.1));
    }
  }
}

/* Echo vom RECHTEN Kanal auf linkes/rechtes ear berechnen */

for (x=-ceil(dist_max/(2*length));x<=ceil(dist_max/(2*length));x++)
for (y=-ceil(dist_max/(2*width));y<=ceil(dist_max/(2*width));y++)
for (z=-ceil(dist_max/(2*height));z<=ceil(dist_max/(2*height));z++)
{
  xref=2*x*width;
  yref=2*y*length;
  zref=2*z*height;
  for (select=0;select<8;select++)   /* vollstaendige Permutation */
  {
    if (select&1) x_pos=xref+rightposx;
    else x_pos=xref-rightposx;
    if (select&2) y_pos=yref+rightposy;
    else y_pos=yref-rightposy;
    if (select&4) z_pos=zref+rightposz;
    else z_pos=zref-rightposz;
    /* Jetzt steht die absolute Position der Quelle in ?_pos */

    /* Relative Position zum linken ear des Hoerers bestimmen: */
    x_pos-=listening_positionx;
    y_pos-=listening_positiony;
    z_pos-=listening_positionz;


    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
    /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
    if ((distance*147.)<early_length)
    {
      /* Einfallsangle im Bereich -Pi/2 bis Pi/2 ermitteln: */
      angle=atan(y_pos/x_pos);
      if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
      {  /* 0=left=>Verstaerkung=1 Pi=right=>Verstaerkung=0.22 (?) */
        angle+=3.1415926/2;     weight = 1 - angle/4;
      }
      else            /* Klang kommt von hinten: */
      {
        angle-=3.1415926/2;  weight= 1 + angle/4;
      }
      /* Early-echo gemaess angle und Entfernung gewichten: */
      early_r2l[(int) (distance*147.)]+=weight/(pow(distance,3.1));
    }


    /* Und jetzt Early-Echo zweiter Kanal auf LINKES ear berechnen */
    x_pos-=0.18; /* Kopfwidth addieren */

    distance=sqrt(pow(x_pos,2)+pow(y_pos,2)+pow(z_pos,2));
```

```
      /* distance*147 = distance/300[m/s]*44100[ATW/s]; bei 32kHZ-> *106 */
      if ((distance*147)<early_length)
      {
        /* Einfallsangle im Bereich -Pi/2 bis Pi/2 ermitteln: */
        angle=atan(y_pos/x_pos);
        if (y_pos>0)   /* Klang kommt aus vorderem Bereich: */
        {  /* 0=left=>Verstaerkung=1 Pi=right=>Verstaerkung=0.22 (?) */
          angle-=3.1415926/2;     weight = 1 + angle/4;
        }
        else            /* Klang kommt von hinten: */
        {
          angle+=3.1415926/2;  weight= 1 - angle/4;
        }
        /* Early-echo gemaess angle und Entfernung gewichten: */
        early_r2r[(int) (distance*147.)]+=weight/(pow(distance,3.1));
      }
    }
  }


  /* Und jetzt aus berechnetem Echo die ersten early_tap_num Werte holen */
  /* Erst mal e's zuruecksetzen: */
  for (i=0;i<early_tap_num;i++)
  {
    e_values_l2l[i]=e_values_l2r[i]=0.0;
    e_delays_l2l[i]=e_delays_l2r[i]=0;  /* Unangenehme Speicherzugriffe vermeiden */
    e_values_r2l[i]=e_values_r2r[i]=0.0;
    e_delays_r2l[i]=e_delays_r2r[i]=0;
  }

  /* und jetzt e_delays und e_values extrahieren: */
  j=0;
  normalizationl=normalizationr=0.0;
  for(i=0;i<early_length;i++)
  {
    if (early_l2l[i]!=0)
    {
      e_delays_l2l[j]=i;
      e_values_l2l[j]=early_l2l[i];
      normalizationl+=early_l2l[i];
      j++;
    }
    if (j==early_tap_num) break;
  }
  j=0;
  for(i=0;i<early_length;i++)
  {
    if (early_l2r[i]!=0)
    {
      e_delays_l2r[j]=i;
      e_values_l2r[j]=early_l2r[i];
      normalizationr+=early_l2r[i];
      j++;
    }
```

```
    if (j==early_tap_num) break;
  }
  j=0;
  for(i=0;i<early_length;i++)
  {
    if (early_r2l[i]!=0)
    {
      e_delays_r2l[j]=i;
      e_values_r2l[j]=early_r2l[i];
      normalizationl+=early_r2l[i];
      j++;
    }
    if (j==early_tap_num) break;
  }
  j=0;
  for(i=0;i<early_length;i++)
  {
    if (early_r2r[i]!=0)
    {
      e_delays_r2r[j]=i;
      e_values_r2r[j]=early_r2r[i];
      normalizationr+=early_r2r[i];
      j++;
    }
    if (j==early_tap_num) break;
  }

  /* groessere von beiden normalizationen verwenden: */
  if (normalizationr>normalizationl) normalizationr=normalizationl;

  for (j=0;j<early_tap_num;j++)
  {
    e_values_l2l[j]/=normalizationr;
    e_values_l2r[j]/=normalizationr;
    e_values_r2l[j]/=normalizationr;
    e_values_r2r[j]/=normalizationr;
  }
  /* Ausgeben nur der l2l-Werte fuer schnelles Reverb */
  printf("int e_delays[%d]={",early_tap_num);
  for (j=0;j<(early_tap_num-1);j++)
    printf("%d, ",e_delays_l2l[j]);
  printf("%d};\n\n",e_delays_l2l[j]);

  printf("float e_values[%d]={",early_tap_num);
  for (j=0;j<(early_tap_num-1);j++)
    printf("%0.4f, ",e_values_l2l[j]);
  printf("%0.4f};\n\n",e_values_l2l[j]);

}
```

ECE320 project: Reverberation w/ parameter control from PC

References : Posted by Brahim Hamadicharef (project by Hua Zheng and Shobhit Jain)
Linked file : rev.txt (this linked file is included below)

Notes :
rev.asm
ECE320 project: Reverberation w/ parameter control from PC
Hua Zheng and Shobhit Jain
12/02/98 ~ 12/11/98
(se linked file)

Linked files

```
; rev.asm
;
; ECE320 project: Reverberation w/ parameter control from PC
;
; Hua Zheng and Shobhit Jain
; 12/02/98 ~ 12/11/98
;
; bugs fixed
;       wrong co coefficients
;       using current out point to calculate new in point
;       r6 changed in set_dl (now changed to r4)
;       initialize er delaylines to be 0 causes no output -- program memory
;       periodic pops: getting garbage because external memory is configured to 16k

;================================================
; Initialization
;================================================

SAMPLERATE equ 48
           nolist
           include 'core302.asm'
           list

;---------------------------
; Variables and storage setup
;---------------------------

RESET   equ     255             ; serial data reset character
n_para  equ     29              ; number of parameters expected from serial port
delayline_number        equ     12
delay_pointers  equ     24
er_number       equ     3
co_number       equ     6
ap_number       equ     3

        org     x:$0
ser_data        ds      n_para  ; location of serial data chunk
delays                  ; default rev parameters: delay length
        dc      967                     ; er1 $3c7
```

```
        dc      1867                    ; er2 $74b
        dc      3359                    ; er3 $d1f
        dc      2399                    ; co1 $95f
        dc      2687                    ; co2 $a7f
        dc      2927                    ; co3 $b6f
        dc      3271                    ; co4 $cc7
        dc      3457                    ; co5 $d81
        dc      3761                    ; co6 $eb1
        dc      293                     ; ap1 $125
        dc      83                      ; ap2 $53
        dc      29                      ; ap3 $1d
coeffs                  ; default rev parameters: coefficients
        dc      0.78                    ; er1
        dc      0.635                   ; er2
        dc      0.267                   ; er3
;       dc      0
;       dc      0
;       dc      0
        dc      0.652149                ; co1 $7774de
        dc      0.301209
        dc      0.615737                ; co2 $53799e
        dc      0.334735
        dc      0.586396                ; co3 $4ed078
        dc      0.362044
        dc      0.546884                ; co4 $4b0f06
        dc      0.399249
        dc      0.525135                ; co5 $4337a0
        dc      0.419943
        dc      0.493653                ; co6 $3f3006
        dc      0.450179
        dc      0.2                     ; brightness
        dc      0.4                     ; mix
comb    ds      6                       ; one sample delay in comb filters
in      ds      1                       ; input sample
lpf     ds      1                       ; one sample delay in LPF
dl_p    ds      delay_pointers  ; delayline in/out pointers

dl_er_l equ     $1000           ; max er delayline length 4096/48=85.3ms
dl_co_l equ     $1000           ; max co delayline length 85.3ms
dl_ap_l equ     $200            ; max ap delayline length 512/48=10.67ms

        org     p:$1000
dl_er1  dsm     dl_er_l ; er1 delayline
dl_er2  dsm     dl_er_l ; er2 delayline
dl_er3  dsm     dl_er_l ; er3 delayline
        org     y:$8000
dl_co1  dsm     dl_co_l ; co1 delayline
dl_co2  dsm     dl_co_l ; co2 delayline
dl_co3  dsm     dl_co_l ; co3 delayline
dl_co4  dsm     dl_co_l ; co4 delayline
dl_co5  dsm     dl_co_l ; co5 delayline
dl_co6  dsm     dl_co_l ; co6 delayline
        org     y:$F000
dl_ap1  dsm     dl_ap_l ; ap1 delayline
```

```
dl_ap2  dsm     dl_ap_l ; ap2 delayline
dl_ap3  dsm     dl_ap_l ; ap3 delayline

;---------------------------
; Memory usage
;---------------------------
; P:$0000 -- $0200      core file
; P:$0200 -- $0300      progam
;
; X:$0000 -- $1BFF      data 7168=149.3ms               serial data, parameters,
pointers
; Y:$0000 -- $1BFF      data 7168=149.3ms               not used
; P:$1000 -- $4FFF      data 16384=341.3ms      er(85*3=255ms)
; Y:$8000 -- $FFFF      data 32768=682.67ms     co(80*6=480ms) and ap(10*3=30ms)
;
; X,Y:$1C00 -- $1BFF    reserved for system
;

;================================================
; Main program
;================================================

        org     p:$200
main

;---------------------------
; Initialization
;---------------------------

        move    #0,x0
;       move    #dl_er1,r0
;       move    #dl_er_l,y0
;       do      #er_number,clear_dl_er_loop
;       rep     y0
;               movem   x0,p:(r0)+
;       nop
;clear_dl_er_loop

        move    #dl_co1,r0
        move    #dl_co_l,y0
        do      #co_number,clear_dl_co_loop
        rep     y0
                move    x0,y:(r0)+
        nop
clear_dl_co_loop

        move    #dl_ap1,r0
        move    #dl_ap_l,y0
        do      #ap_number,clear_dl_ap_loop
        rep     y0
                move    x0,y:(r0)+
        nop
clear_dl_ap_loop

        move    #comb,r0
```

```
        rep     #co_number
                move    x0,x:(r0)+      ; init comb filter states
        move    #lpf,r0
        move    x0,x:(r0)               ; init LPF state

        move    #ser_data,r6    ; incoming data buffer pointer
        move    #(n_para-1),m6

        jsr     set_dl          ; set all delayline pointers

        ; initialize SCI
        movep   #$0302,x:M_SCR  ; R/T enable, INT disable
        movep   #$006a,x:M_SCCR   ; SCP=0, CD=106 (9636 bps)
        movep   #7,x:M_PCRE

;---------------------------
; Main loop
;       Register usage
;               r0: delayline pointer pointer
;               r1: coefficients pointer
;               r2: comb filter internal state pointer
;               r4,r5: used in delayline subroutine
;               r6: incoming buffer pointer
;               a: output of each segment
;---------------------------

main_loop:

        move    #dl_p,r0
        move    #coeffs,r1

        waitdata        r3,buflen,1
        move    x:(r3)+,a
        move    a,x:<in         ; save input sample for mix

;---------------------------
; Early reflection
;---------------------------
;       temp = in;
;       for (int i=0; i<earlyRefNum; i++)
;       {
;               in = delayLineEarlyRef[c][i]->tick(in);
;               temp += earlyRefCoeff[i] * in;
;       }
;       return temp;

        move    a,b                             ; b=temp=in
        move    #(dl_er_l-1),n6
        do      #er_number,er_loop
                jsr     use_dl_er
                move    a,y0                    ; y0=a=in (delayline out)
                move    x:(r1)+,x0              ; x0=coeff
                mac     x0,y0,b         ; b=temp
er_loop
        asr     #2,b,a
```

```
        move    b,a

;---------------------------
; Comb filter
;---------------------------
;       float temp1 = 0., temp2;
;       for (int i=0; i<combNum; i++)
;       {
;               temp2 = delayLineComb[c][i]->tick
;                       (in + combCoeff[i] * combLastOut[c][i]);
;               combLastOut[c][i] = temp2+combDamp[i]*combLastOut[c][i];
;               temp1 += temp2;
;       }
;       return temp1 / float(combNum);

        move    #comb,r2
        move    a,y1
        clr     b
        move    #(dl_co_l-1),n6
        do      #co_number,co_loop
                move    y1,a                    ; a=in
                move    x:(r1)+,x0              ; x0=coeff
                move    x:(r2),y0               ; y0=lastout
                mac     x0,y0,a x:(r1)+,x0      ; x0=damp
                jsr     use_dl
                move    a,x1                    ; a=x1=temp2
                mac     x0,y0,a         ; a=lastout
                move    a,x:(r2)+
                add     x1,b                    ; b=temp1
co_loop
;       asr     #2,b,a
        move    b,a

;---------------------------
; All-pass filter
;---------------------------
;       float temp1, temp2;
;       for (int i=0; i<allPassNum; i++)
;       {
;               temp1 = delayLineAllPass[c][i]->lastOut();
;               temp2 = allPassCoeff[i] * (in - temp1);
;               delayLineAllPass[c][i]->tick(in + temp2);
;               in = temp1 + temp2;
;       }
;       return in;

        move    #1,n0
        move    #0.7,x1
        move    #(dl_ap_l-1),n6
        do      #ap_number,ap_loop
                move    y:(r0+n0),x0    ; x0=temp1
                sub     x0,a                    ; a=in-temp1
                move    a,y0
                mpy     x1,y0,b         ; b=temp2
                add     b,a                     ; a=in+temp2
```

```
                jsr     use_dl
                add     x0,b                    ; b=temp1+temp2
                move    b,a                     ; a=in
ap_loop

;---------------------------
; Brightness
;---------------------------
;       lastout = lastout + BW * (in - lastout);
        move    x:<lpf,b
        sub     b,a             x:(r1)+,x0      ; a=in-lastout, x0=bri
        move    a,y0
        mpy     x0,y0,a
        add     b,a
        move    a,x:<lpf

;---------------------------
; Mix
;---------------------------
;       out = (1-mix)*in + mix*out = in + mix * (out - in);
        move    x:<in,y0                        ; y0=in
        sub     y0,a            x:(r1)+,x0      ; a=out-in, x0=mix
        move    a,y1                            ; y1=out-in
        mpy     x0,y1,b y0,a            ; b=mix*(out-in), a=in
        add     b,a

;---------------------------
; Spit out
;---------------------------
        move    a,y:(r3)+
        move    a,y:(r3)+
        move    (r3)+

;---------------------------
; Get parameters and reformat them
;---------------------------
        jclr    #2,x:M_SSR,main_loop            ; do nothing if no new data arrived

        movep   x:M_SRXL,a                              ; get next 8-bit word from
SCI

        cmp     #RESET,a
        jeq     reformat_data                   ; if it's RESET, then reformat data

        move    a,x:(r6)+                               ; save one incoming data for
later reformatting

        jmp     main_loop

reformat_data:
; order of parameters:
;       er1 delay, er1 coeff, er2 ..., er3 ...
;       co1 delay, coeff_c, coeff_d, co2 ... , ... , co6
;       ap1 delay, ap2, ap3
;       brightness
```

```
;       mix

        move    #ser_data,r0
        move    #delays,r1
        move    #coeffs,r2

        do      #3,format_er_loop
                move    x:(r0)+,a               ; er delay
                asr     #20,a,a
                ; max delay 4096=2^12, max value 256=2^8, scale 256/4096=2^-4
                move    a0,x:(r1)+
                move    x:(r0)+,a               ; er coeff
                asr     #9,a,a
                move    a0,x:(r2)+
format_er_loop

        move    #>$000001,x0
        do      #6,format_co_loop
                move    x:(r0)+,a               ; co delay
                asr     #20,a,a         ; max delay 4096=2^12
                move    a0,a1
;try this:              asl     #4,a,a
                or      x0,a
                move    a1,x:(r1)+
                move    x:(r0)+,a               ; co coeff
                asr     #9,a,a
                move    a0,x:(r2)+
                move    x:(r0)+,a               ; co damping
                asr     #9,a,a
                move    a0,x:(r2)+
format_co_loop

        do      #3,format_ap_loop
                move    x:(r0)+,a               ; ap delay
                asr     #23,a,a         ; max delay 528=2^9
                move    a0,a1
                or      x0,a
                move    a1,x:(r1)+
format_ap_loop

        jsr     set_dl

        move    x:(r0)+,a                       ; brightness
        asr     #9,a,a
        move    a0,x:(r2)+

        move    x:(r0)+,a                       ; mix
        asr     #9,a,a
        move    a0,x:(r2)+

        jmp     main_loop

;================================================
; Set all delayline length subroutine
;       IN:     nothing
```

```
;       OUT:    out pointer UNCHANGED
;               in pointer = out + length e.g. (#(dl_p+3))=(#(dl_p+4))+x:(r4)
;               r4=r4+1: next delay length
;================================================
set_dl:
        move    #(dl_p+1),r5    ; first out pointer
        move    #dl_er1,r4
        move    r4,x:(r5)+              ; initial out point=delayline starting
address
        move    (r5)+
        move    #dl_er2,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_er3,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_co1,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_co2,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_co3,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_co4,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_co5,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_co6,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_ap1,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_ap2,r4
        move    r4,x:(r5)+
        move    (r5)+
        move    #dl_ap3,r4
        move    r4,x:(r5)+
        move    (r5)+

        move    #delays,r4                      ; delayline length
        move    #(dl_p+1),r5            ; first out pointer
        move    #2,n5
        do      #delayline_number,set_dl_loop
                move    x:(r4)+,x0              ; x0=length
                move    x:(r5)-,a               ; a=out pointer
                add     x0,a
                move    a,x:(r5)+               ; in=out+length
                move    (r5)+n5         ; next out pointer
set_dl_loop
```

```
        rts

;================================================
; Access delayline subroutine
;       IN:     in and out pointers in r4,r5
;               modulo (delayline length-1) in n6
;               input sample in a
;       OUT:    in and out pointers modulo incremented
;               output sample in a
;================================================
;       inputs[inPoint++] = sample;
;       inPoint &= lengthm1;
;       lastOutput = inputs[outPoint++];
;       outPoint &= lengthm1;
;       return lastOutput;
use_dl:
        move    n6,m4
        move    n6,m5
        move    x:(r0)+,r4      ; in point
        move    x:(r0)-,r5      ; out point
        move    a,y:(r4)+       ; queue in
        move    y:(r5)+,a       ; queue out
        move    r4,x:(r0)+      ; modulo incremented in point
        move    r5,x:(r0)+      ; modulo incremented out point
        rts

use_dl_er:                      ; using P memory
        move    n6,m4
        move    n6,m5
        move    x:(r0)+,r4      ; in point
        move    x:(r0)-,r5      ; out point
        movem   a,p:(r4)+       ; queue in
        movem   p:(r5)+,a       ; queue out
        move    r4,x:(r0)+      ; modulusly incremented in point
        move    r5,x:(r0)+      ; modulusly incremented out point
        rts
```

Envelope detector

References : Posted by Bram

Notes :
Basicaly a one-pole LP filter with different coefficients for attack and release fed by the abs() of the signal. If you don't need different attack and decay settings, just use in->abs()->LP

Code :

```
//attack and release in milliseconds
float ga = (float) exp(-1/(SampleRate*attack));
float gr = (float) exp(-1/(SampleRate*release));

float envelope=0;

for(...)
{
  //get your data into 'input'
  EnvIn = abs(input);

  if(envelope < EnvIn)
  {
     envelope *= ga;
     envelope += (1-ga)*EnvIn;
  }
  else
  {
     envelope *= gr;
     envelope += (1-gr)*EnvIn;
  }
  //envelope now contains.........the envelope ;)
}
```

Exponential parameter mapping

References : Posted by Russell Borogove

Notes :
Use this if you want to do an exponential map of a parameter (mParam) to a range (mMin - mMax).
Output is in mData...

Code :
```
float logmax = log10f( mMax );
float logmin = log10f( mMin );
float logdata = (mParam * (logmax-logmin)) + logmin;

mData = powf( 10.0f, logdata );
if (mData < mMin)
{
  mData = mMin;
}
if (mData > mMax)
{
  mData = mMax;
}
```

Fast binary log approximations

Type : C code
References : Posted by musicdsp.org[AT]mindcontrol.org

Notes :
This code uses IEEE 32-bit floating point representation knowledge to quickly compute approximations to the log2 of a value. Both functions return under-estimates of the actual value, although the second flavour is less of an under-estimate than the first (and might be sufficient for using in, say, a dBV/FS level meter).

Running the test program, here's the output:

0.1: -4 -3.400000
1: 0 0.000000
2: 1 1.000000
5: 2 2.250000
100: 6 6.562500

Code :

```
// Fast logarithm (2-based) approximation
// by Jon Watte

#include <assert.h>

int floorOfLn2( float f ) {
  assert( f > 0. );
  assert( sizeof(f) == sizeof(int) );
  assert( sizeof(f) == 4 );
  return (((*(int *)&f)&0x7f800000)>>23)-0x7f;
}

float approxLn2( float f ) {
  assert( f > 0. );
  assert( sizeof(f) == sizeof(int) );
  assert( sizeof(f) == 4 );
  int i = (*(int *)&f);
  return (((i&0x7f800000)>>23)-0x7f)+(i&0x007fffff)/(float)0x800000;
}

// Here's a test program:

#include <stdio.h>

// insert code from above here

int
```

```
main()
{
  printf( "0.1: %d  %f\n", floorOfLn2( 0.1 ), approxLn2( 0.1 ) );
  printf( "1:   %d  %f\n", floorOfLn2( 1. ), approxLn2( 1. ) );
  printf( "2:   %d  %f\n", floorOfLn2( 2. ), approxLn2( 2. ) );
  printf( "5:   %d  %f\n", floorOfLn2( 5. ), approxLn2( 5. ) );
  printf( "100: %d  %f\n", floorOfLn2( 100. ), approxLn2( 100. ) );
  return 0;
}
```

Fast exp2 approximation

References : Posted by Laurent de Soras

Notes :
Partial approximation of exp2 in fixed-point arithmetic. It is exactly :
[0 ; 1[-> [0.5 ; 1[
f : x |-> 2^(x-1)
To get the full exp2 function, you have to separate the integer and fractionnal part of the number. The integer part may be processed by bitshifting. Process the fractionnal part with the function, and multiply the two results.
Maximum error is only 0.3 % which is pretty good for two mul ! You get also the continuity of the first derivate.

-- Laurent

Code :

```
// val is a 16-bit fixed-point value in 0x0 - 0xFFFF ([0 ; 1[)
// Returns a 32-bit fixed-point value in 0x80000000 - 0xFFFFFFFF ([0.5 ;
1[)
unsigned int fast_partial_exp2 (int val)
{
   unsigned int   result;

   __asm
   {
      mov eax, val
      shl eax, 15           ; eax = input [31/31 bits]
      or  eax, 080000000h   ; eax = input + 1  [32/31 bits]
      mul eax
      mov eax, edx          ; eax = (input + 1) ^ 2 [32/30 bits]
      mov edx, 2863311531   ; 2/3 [32/32 bits], rounded to +oo
      mul edx               ; eax = 2/3 (input + 1) ^ 2 [32/30 bits]
      add edx, 1431655766   ; + 4/3 [32/30 bits] + 1
      mov result, edx
   }

   return (result);
}
```

Fast in-place Walsh-Hadamard Transform

Type : wavelet transform
References : Posted by Timo H Tossavainen

Notes :
IIRC, They're also called walsh-hadamard transforms.
Basically like Fourier, but the basis functions are squarewaves with different sequencies.
I did this for a transform data compression study a while back.
Here's some code to do a walsh hadamard transform on long ints in-place (you need to divide by n to get transform) the order is bit-reversed at output, IIRC.
The inverse transform is the same as the forward transform (expects bit-reversed input). i.e. x = 1/n * FWHT(FWHT(x)) (x is a vector)

Code :
```
void inline wht_bfly (long& a, long& b)
{
        long tmp = a;
        a += b;
        b = tmp - b;
}

// just a integer log2
int inline l2 (long x)
{
        int l2;
        for (l2 = 0; x > 0; x >>=1)
        {
                ++ l2;
        }

        return (l2);
}

////////////////////////////////////////
// Fast in-place Walsh-Hadamard Transform //
////////////////////////////////////////

void FWHT (std::vector& data)
{
  const int log2 = l2 (data.size()) - 1;
  for (int i = 0; i < log2; ++i)
  {
    for (int j = 0; j < (1 << log2); j += 1 << (i+1))
    {
       for (int k = 0; k < (1<<i); ++k)
       {
```

```
                wht_bfly (data [j + k], data [j + k + (1<<i)]);
            }
        }
    }
}
```

Fast log2

References : Posted by Laurent de Soras

Code :

```
inline float fast_log2 (float val)
{
   assert (val > 0);

   int * const  exp_ptr = reinterpret_cast <int *> (&val);
   int          x = *exp_ptr;
   const int    log_2 = ((x >> 23) & 255) - 128;
   x &= ~(255 << 23);
   x += 127 << 23;
   *exp_ptr = x;

   return (val + log_2);
}
```

Fast sine and cosine calculation

Type : waveform generation
References : Lot's or references... Check Julius O. SMith mainly
Code :

```
init:
float a = 2.f*(float)sin(Pi*frequency/samplerate);

float s[2];

s[0] = 0.5f;
s[1] = 0.f;

loop:
s[0] = s[0] - a*s[1];
s[1] = s[1] + a*s[0];
output_sine = s[0];
output_cosine = s[1]
```

Fast sine wave calculation

Type : waveform generation
References : James McCartney in Computer Music Journal, also the Julius O. Smith paper

Notes :
(posted by Niels Gorisse)
If you change the frequency, the amplitude rises (pitch lower) or lowers (pitch rise) a LOT I fixed the first problem by thinking about what actually goes wrong. The answer was to recalculate the phase for that frequency and the last value, and then continue normally.

Code :

```
Variables:
ip = phase of the first output sample in radians
w = freq*pi / samplerate
b1 = 2.0 * cos(w)

Init:
y1=sin(ip-w)
y2=sin(ip-2*w)

Loop:
y0 = b1*y1 - y2
y2 = y1
y1 = y0

output is in y0 (y0 = sin(ip + n*freq*pi / samplerate), n= 0, 1, 2, ... I
*think*)

Later note by James McCartney:
if you unroll such a loop by 3 you can even eliminate the assigns!!

y0 = b1*y1 - y2
y2 = b1*y0 - y1
y1 = b1*y2 - y0
```

Fast square wave generator

Type : NON-bandlimited osc...
References : Posted by Wolfgang (wschneider[AT]nexoft.de)

Notes :
Produces a square wave -1.0f .. +1.0f.
The resulting waveform is NOT band-limited, so it's propably of not much use for syntheis. It's rather useful for LFOs and the like, though.

Code :
```
Idea: use integer overflow to avoid conditional jumps.

// init:
typedef unsigned long ui32;

float sampleRate = 44100.0f; // whatever
float freq = 440.0f; // 440 Hz
float one = 1.0f;
ui32 intOver = 0L;
ui32 intIncr = (ui32)(4294967296.0 / hostSampleRate / freq));

// loop:
(*((ui32 *)&one)) &= 0x7FFFFFFF; // mask out sign bit
(*((ui32 *)&one)) |= (intOver & 0x80000000);
intOver += intIncr;
```

FFT

References : Toth Laszlo
Linked file : rvfft.ps
Linked file : rvfft.cpp (this linked file is included below)

Notes :
A paper (postscript) and some C++ source for 4 different fft algorithms, compiled by Toth Laszlo from the Hungarian Academy of Sciences Research Group on Artificial Intelligence.
Toth says: "I've found that Sorensen's split-radix algorithm was the fastest, so I use this since then (this means that you may as well delete the other routines in my source - if you believe my results)."

Linked files

```
//
//                      FFT library
//
//  (one-dimensional complex and real FFTs for array
//  lengths of 2^n)
//
//      Author: Toth Laszlo (tothl@inf.u-szeged.hu)
//
//      Research Group on Artificial Intelligence
//  H-6720 Szeged, Aradi vertanuk tere 1, Hungary
//
//      Last modified: 97.05.29
/////////////////////////////////////////////////////

#include <math.h>
#include <stdlib.h>
#include "pi.h"

/////////////////////////////////////////////////////
//Gives back "i" bit-reversed where "size" is the array
//length
//currently none of the routines call it

long bitreverse(long i, long size){

        long result,mask;

        result=0;
        for(;size>1;size>>=1){
                mask=i&1;
                i>>=1;
                result<<=1;
                result|=mask;
        }
```

```
/*     __asm{       the same in assebly
     mov eax,i
         mov ecx,sizze
     mov ebx,0
   l:shr eax,1
     rcl ebx,1
         shr ecx,1
         cmp ecx,1
     jnz l
     mov result,ebx
     }*/
         return result;
}

////////////////////////////////////////////////////
//Bit-reverser for the Bruun FFT
//Parameters as of "bitreverse()"

long bruun_reverse(long i, long sizze){

        long result, add;

        result=0;
        add=sizze;

        while(1){
        if(i!=0) {
                while((i&1)==0) { i>>=1; add>>=1;}
                i>>=1; add>>=1;
                result+=add;
        }
        else {result<<=1;result+=add; return result;}
        if(i!=0) {
                while((i&1)==0) { i>>=1; add>>=1;}
                i>>=1; add>>=1;
                result-=add;
        }
        else {result<<=1;result-=add; return result;}
        }
}

/*assembly version
long bruun_reverse(long i, long sizze){

        long result;

        result=0;

        __asm{
                mov edx,sizze
                mov eax,i
```

```
                mov ebx,0
          l: bsf cx,eax
                jz kesz1
                inc cx
                shr edx,cl
             add ebx,edx
                shr eax,cl
                bsf cx,eax
                jz kesz2
                inc cx
                shr edx,cl
                sub ebx,edx
                shr eax,cl
                jmp l
kesz1:
                shl ebx,1

                add ebx,edx
                jmp vege
kesz2:
                shl ebx,1
                sub ebx,edx

  vege: mov result,ebx
        }
        return result;
}*/

//////////////////////////////////////////////////////
// Decimation-in-freq radix-2 in-place butterfly
// data: array of doubles:
// re(0),im(0),re(1),im(1),...,re(size-1),im(size-1)
// it means that size=array_length/2 !!
//
// suggested use:
// intput in normal order
// output in bit-reversed order
//
// Source: Rabiner-Gold: Theory and Application of DSP,
// Prentice Hall,1978

void dif_butterfly(double *data,long size){

        long angle,astep,dl;
        double xr,yr,xi,yi,wr,wi,dr,di,ang;
        double *l1, *l2, *end, *ol2;

     astep=1;
         end=data+size+size;
     for(dl=size;dl>1;dl>>=1,astep+=astep){
                                          l1=data;
```

```
                                        l2=data+dl;
                        for(;l2<end;l1=l2,l2=l2+dl){
                                                        ol2=l2;
                              for(angle=0;l1<ol2;l1+=2,l2+=2){
                                 ang=2*pi*angle/size;
                                 wr=cos(ang);
                                 wi=-sin(ang);
                                 xr=*l1+*l2;
                                 xi=*(l1+1)+*(l2+1);
                                 dr=*l1-*l2;
                                 di=*(l1+1)-*(l2+1);
                                 yr=dr*wr-di*wi;
                                 yi=dr*wi+di*wr;
                                 *(l1)=xr;*(l1+1)=xi;
                                 *(l2)=yr;*(l2+1)=yi;
                                 angle+=astep;
                                                          }
                                        }
          }
    }

    ////////////////////////////////////////////////////
    // Decimation-in-time radix-2 in-place inverse butterfly
    // data: array of doubles:
    // re(0),im(0),re(1),im(1),...,re(size-1),im(size-1)
    // it means that size=array_length/2 !!
    //
    // suggested use:
    // intput in bit-reversed order
    // output in normal order
    //
    // Source: Rabiner-Gold: Theory and Application of DSP,
    // Prentice Hall,1978

    void inverse_dit_butterfly(double *data,long size){

          long angle,astep,dl;
          double xr,yr,xi,yi,wr,wi,dr,di,ang;
          double *l1, *l2, *end, *ol2;

       astep=size>>1;
           end=data+size+size;
       for(dl=2;astep>0;dl+=dl,astep>>=1){
                                        l1=data;
                                        l2=data+dl;
                        for(;l2<end;l1=l2,l2=l2+dl){
                                                        ol2=l2;
                              for(angle=0;l1<ol2;l1+=2,l2+=2){
                                 ang=2*pi*angle/size;
                                 wr=cos(ang);
```

```
                                wi=sin(ang);
                                xr=*l1;
                                xi=*(l1+1);
                                yr=*l2;
                                yi=*(l2+1);
                                dr=yr*wr-yi*wi;
                                di=yr*wi+yi*wr;
                                *(l1)=xr+dr;*(l1+1)=xi+di;
                                *(l2)=xr-dr;*(l2+1)=xi-di;
                                angle+=astep;
                                                        }
                                          }
         }
}

//////////////////////////////////////////////////////
// data shuffling into bit-reversed order
// data: array of doubles:
// re(0),im(0),re(1),im(1),...,re(size-1),im(size-1)
// it means that size=array_length/2 !!
//
// Source: Rabiner-Gold: Theory and Application of DSP,
// Prentice Hall,1978

void unshuffle(double *data, long size){

        long i,j,k,l,m;
        double re,im;

        //old version - turned out to be a bit slower
    /*for(i=0;i<size-1;i++){
             j=bitreverse(i,size);
             if (j>i){   //swap
                    re=data[i+i];im=data[i+i+1];
                    data[i+i]=data[j+j];data[i+i+1]=data[j+j+1];
                    data[j+j]=re;data[j+j+1]=im;
                         }
        }*/

l=size-1;
m=size>>1;
for (i=0,j=0; i<l ; i++){
          if (i<j){
                                re=data[j+j]; im=data[j+j+1];
                                data[j+j]=data[i+i]; data[j+j+1]=data[i+i+1];
                                data[i+i]=re; data[i+i+1]=im;
                                }
          k=m;
          while (k<=j){
                                j-=k;
```

```
                                k>>=1;
                                }
            j+=k;
        }
    }

    ////////////////////////////////////////////////////
    // used by realfft
    // parameters as above
    //
    // Source: Brigham: The Fast Fourier Transform
    // Prentice Hall, ?

    void realize(double *data, long size){

            double xr,yr,xi,yi,wr,wi,dr,di,ang,astep;
            double *l1, *l2;

            l1=data;
            l2=data+size+size-2;
        xr=*l1;
        xi=*(l1+1);
        *l1=xr+xi;
        *(l1+1)=xr-xi;
            l1+=2;
            astep=pi/size;
        for(ang=astep;l1<=l2;l1+=2,l2-=2,ang+=astep){
                                xr=(*l1+*l2)/2;
                                yi=(-(*l1)+(*l2))/2;
                                yr=(*(l1+1)+*(l2+1))/2;
                                xi=(*(l1+1)-*(l2+1))/2;
                                wr=cos(ang);
                                wi=-sin(ang);
                                dr=yr*wr-yi*wi;
                                di=yr*wi+yi*wr;
                                *l1=xr+dr;
                                *(l1+1)=xi+di;
                                *l2=xr-dr;
                                *(l2+1)=-xi+di;
            }
    }

    ////////////////////////////////////////////////////
    // used by inverse realfft
    // parameters as above
    //
    // Source: Brigham: The Fast Fourier Transform
    // Prentice Hall, ?

    void unrealize(double *data, long size){
```

```
        double xr,yr,xi,yi,wr,wi,dr,di,ang,astep;
        double *l1, *l2;

        l1=data;
        l2=data+size+size-2;
    xr=(*l1)/2;
    xi=(*(l1+1))/2;
    *l1=xr+xi;
    *(l1+1)=xr-xi;
        l1+=2;
        astep=pi/size;
    for(ang=astep;l1<=l2;l1+=2,l2-=2,ang+=astep){
                                xr=(*l1+*l2)/2;
                                yi=-(-(*l1)+(*l2))/2;
                                yr=(*(l1+1)+*(l2+1))/2;
                                xi=(*(l1+1)-*(l2+1))/2;
                                wr=cos(ang);
                                wi=-sin(ang);
                                dr=yr*wr-yi*wi;
                                di=yr*wi+yi*wr;
                                *l2=xr+dr;
                                *(l1+1)=xi+di;
                                *l1=xr-dr;
                                *(l2+1)=-xi+di;
        }
}

////////////////////////////////////////////////////
// in-place Radix-2 FFT for complex values
// data: array of doubles:
// re(0),im(0),re(1),im(1),...,re(size-1),im(size-1)
// it means that size=array_length/2 !!
//
// output is in similar order, normalized by array length
//
// Source: see the routines it calls ...

void fft(double *data, long size){

        double *l, *end;

        dif_butterfly(data,size);
        unshuffle(data,size);

        end=data+size+size;
        for(l=data;l<end;l++){*l=*l/size;};
}


////////////////////////////////////////////////////
// in-place Radix-2 inverse FFT for complex values
```

```
// data: array of doubles:
// re(0),im(0),re(1),im(1),...,re(size-1),im(size-1)
// it means that size=array_length/2 !!
//
// output is in similar order, NOT normalized by
// array length
//
// Source: see the routines it calls ...

void ifft(double* data, long size){

        unshuffle(data,size);
        inverse_dit_butterfly(data,size);
}

////////////////////////////////////////////////////
// in-place Radix-2 FFT for real values
// (by the so-called "packing method")
// data: array of doubles:
// re(0),re(1),re(2),...,re(size-1)
//
// output:
// re(0),re(size/2),re(1),im(1),re(2),im(2),...,re(size/2-1),im(size/2-1)
// normalized by array length
//
// Source: see the routines it calls ...

void realfft_packed(double *data, long size){

        double *l, *end;
        double div;

        size>>=1;
        dif_butterfly(data,size);
        unshuffle(data,size);
        realize(data,size);

        end=data+size+size;
        div=size+size;
        for(l=data;l<end;l++){*l=*l/div;};
}


////////////////////////////////////////////////////
// in-place Radix-2 inverse FFT for real values
// (by the so-called "packing method")
// data: array of doubles:
// re(0),re(size/2),re(1),im(1),re(2),im(2),...,re(size/2-1),im(size/2-1)
//
// output:
// re(0),re(1),re(2),...,re(size-1)
```

```
// NOT normalized by array length
//
//Source: see the routines it calls ...

void irealfft_packed(double *data, long size){

        double *l, *end;

        size>>=1;
        unrealize(data,size);
        unshuffle(data,size);
        inverse_dit_butterfly(data,size);

        end=data+size+size;
        for(l=data;l<end;l++){*l=(*l)*2;};
}


//////////////////////////////////////////////////////
// Bruun FFT for real values
// data: array of doubles:
// re(0),re(1),re(2),...,re(size-1)
//
// output:
// re(0),re(size/2),...,re(i),im(i)... pairs in
// "bruun-reversed" order
// normalized by array length
//
// Source:
// Bruun: z-Transform DFT Filters and FFT's
// IEEE Trans. ASSP, ASSP-26, No. 1, February 1978
//
// Comments:
// (this version is implemented in a manner that every
// cosine is calculated only once;
// faster than the other version (see next)

void realfft_bruun(double *data, long size){

        double *end, *l0, *l1, *l2, *l3;
        long dl, dl2, dl_o, dl2_o, i, j, k, kk;
        double d0,d1,d2,d3,c,c2,p4;

        end=data+size;
        //first filterings, when there're only two taps
        size>>=1;
        dl=size;
        dl2=dl/2;
        for(;dl>1;dl>>=1,dl2>>=1){
                l0=data;
                l3=data+dl;
```

```
        for(i=0;i<dl;i++){
                d0=*l0;
                d2=*l3;
                *l0=d0+d2;
                *l3=d0-d2;
                l0++;
                l3++;
        }
}
l0=data;l1=data+1;
d0=*l0;d1=*l1;
*l0=d0+d1;*l1=d0-d1;
l1+=2;
*l1=-(*l1);

//the remaining filterings

p4=pi/(2*size);
j=1;
kk=1;
dl_o=size/2;
dl2_o=size/4;
while(dl_o>1){
for(k=0;k<kk;k++){
c2=p4*bruun_reverse(j,size);
c=2*cos(c2);
c2=2*sin(c2);
dl=dl_o;
dl2=dl2_o;
for(;dl>1;dl>>=1,dl2>>=1){
                l0=data+((dl*j)<<1);
                l1=l0+dl2;l2=l0+dl;l3=l1+dl;
                for(i=0;i<dl2;i++){
                        d1=(*l1)*c;
                        d2=(*l2)*c;
                        d3=*l3+(*l1);
                        d0=*l0+(*l2);
                        *l0=d0+d1;
                        *l1=d3+d2;
                        *l2=d0-d1;
                        *l3=d3-d2;
                        l0++;
                        l1++;
                        l2++;
                        l3++;
                }
        }
        //real conversion
        l3-=4;
        *l3=*l3-c*(*l0)/2;
```

```
                *l0=-c2*(*l0)/2;
                *l1=*l1+c*(*l2)/2;
                *l2=-c2*(*l2)/2;
        j++;
        }
        dl_o>>=1;
        dl2_o>>=1;
        kk<<=1;
        }

        //division with array length
        size<<=1;
        for(i=0;i<size;i++) data[i]=data[i]/size;
}

////////////////////////////////////////////////////
// Bruun FFT for real values
// data: array of doubles:
// re(0),re(1),re(2),...,re(size-1)
//
// output:
// re(0),re(size/2),re(1),im(1),re(2),im(2),...,re(size/2-1),im(size/2-1)
// normalized by array length
//
// Source: see the routines it calls ...
void realfft_bruun_unshuffled(double *data, long size){

        double *data2;
        long i,j,k;

        realfft_bruun(data,size);
        //unshuffling - cannot be done in-place (?)
        data2=(double *)malloc(size*sizeof(double));
        for(i=1,k=size>>1;i<k;i++){
                j=bruun_reverse(i,k);
                data2[j+j]=data[i+i];
                data2[j+j+1]=data[i+i+1];
        }
        for(i=2;i<size;i++) data[i]=data2[i];
        free(data2);
}


////////////////////////////////////////////////////
// Bruun FFT for real values
// data: array of doubles:
// re(0),re(1),re(2),...,re(size-1)
//
// output:
// re(0),re(size/2),...,re(i),im(i)... pairs in
// "bruun-reversed" order
```

```
// normalized by array length
//
// Source:
// Bruun: z-Transform DFT Filters and FFT's
// IEEE Trans. ASSP, ASSP-26, No. 1, February 1978
//
// Comments:
// (this version is implemented in a row-by-row manner;
// control structure is simpler, but there are too
// much cosine calls - with a cosine lookup table
// probably this would be slightly faster than bruun_fft

/*void realfft_bruun2(double *data, long size){

        double *end, *l0, *l1, *l2, *l3;
        long dl, dl2, i, j;
        double d0,d1,d2,d3,c,c2,p4;

        end=data+size;
        p4=pi/(size);
        size>>=1;
        dl=size;
        dl2=dl/2;
        //first filtering, when there're only two taps
        for(;dl>1;dl>>=1,dl2>>=1){
                l0=data;
                l3=data+dl;
                for(i=0;i<dl;i++){
                        d0=*l0;
                        d2=*l3;
                        *l0=d0+d2;
                        *l3=d0-d2;
                        l0++;
                        l3++;
                }
                //the remaining filterings
                j=1;
                while(l3<end){
                        l0=l3;l1=l0+dl2;l2=l0+dl;l3=l1+dl;
                        c=2*cos(p4*bruun_reverse(j,size));
                        for(i=0;i<dl2;i++){
                                d0=*l0;
                                d1=*l1;
                                d2=*l2;
                                d3=*l3;
                                *l0=d0+c*d1+d2;
                                *l2=d0-c*d1+d2;
                                *l1=d1+c*d2+d3;
                                *l3=d1-c*d2+d3;
                                l0++;
```

```
                                l1++;
                                l2++;
                                l3++;
                        }
                        j++;
                }
        }

        //the last row: transform of real data
        //the first two cells
        l0=data;l1=data+1;
        d0=*l0;d1=*l1;
        *l0=d0+d1;*l1=d0-d1;
        l1+=2;
        *l1=-(*l1);
        l0+=4;l1+=2;
        //the remaining cells
        j=1;
        while(l0<end){
                        c=p4*bruun_reverse(j,size);
                        c2=sin(c);
                        c=cos(c);
                        *l0=*l0-c*(*l1);
                        *l1=-c2*(*l1);
                        l0+=2;
                        l1+=2;
                        *l0=*l0+c*(*l1);
                        *l1=-c2*(*l1);
                        l0+=2;
                        l1+=2;
                        j++;
                        }
        //division with array length
        for(i=0;i<size;i++) data[i]=data[i]/size;
}
*/

//the same as realfft_bruun_unshuffled,
//but calls realfft_bruun2
/*void realfft_bruun_unshuffled2(double *data, long size){

        double *data2;
        long i,j,k;

        realfft_bruun2(data,size);
        //unshuffling - cannot be done in-place (?)
        data2=(double *)malloc(size*sizeof(double));
        for(i=1,k=size>>1;i<k;i++){
                j=bruun_reverse(i,k);
                data2[j+j]=data[i+i];
```

```
                data2[j+j+1]=data[i+i+1];
        }
        for(i=2;i<size;i++) data[i]=data2[i];
        free(data2);
}*/

////////////////////////////////////////////////////
// Sorensen in-place split-radix FFT for real values
// data: array of doubles:
// re(0),re(1),re(2),...,re(size-1)
//
// output:
// re(0),re(1),re(2),...,re(size/2),im(size/2-1),...,im(1)
// normalized by array length
//
// Source:
// Sorensen et al: Real-Valued Fast Fourier Transform Algorithms,
// IEEE Trans. ASSP, ASSP-35, No. 6, June 1987

void realfft_split(double *data,long n){

  long i,j,k,i5,i6,i7,i8,i0,id,i1,i2,i3,i4,n2,n4,n8;
  double t1,t2,t3,t4,t5,t6,a3,ss1,ss3,cc1,cc3,a,e,sqrt2;

  sqrt2=sqrt(2.0);
  n4=n-1;

  //data shuffling
      for (i=0,j=0,n2=n/2; i<n4 ; i++){
          if (i<j){
                              t1=data[j];
                              data[j]=data[i];
                              data[i]=t1;
                              }
          k=n2;
          while (k<=j){
                              j-=k;
                              k>>=1;
                              }
          j+=k;
      }

/*----------------------*/

        //length two butterflies
        i0=0;
        id=4;
   do{
       for (; i0<n4; i0+=id){
                        i1=i0+1;
                        t1=data[i0];
```

```
                        data[i0]=t1+data[i1];
                        data[i1]=t1-data[i1];
                    }
                id<<=1;
            i0=id-2;
            id<<=1;
        } while ( i0<n4 );

       /*----------------------*/
       //L shaped butterflies
    n2=2;
    for(k=n;k>2;k>>=1){
            n2<<=1;
            n4=n2>>2;
            n8=n2>>3;
            e = 2*pi/(n2);
            i1=0;
            id=n2<<1;
            do{
                for (; i1<n; i1+=id){
                        i2=i1+n4;
                        i3=i2+n4;
                        i4=i3+n4;
                        t1=data[i4]+data[i3];
                        data[i4]-=data[i3];
                        data[i3]=data[i1]-t1;
                        data[i1]+=t1;
                        if (n4!=1){
                                i0=i1+n8;
                                i2+=n8;
                                i3+=n8;
                                i4+=n8;
                                t1=(data[i3]+data[i4])/sqrt2;
                                t2=(data[i3]-data[i4])/sqrt2;
                                data[i4]=data[i2]-t1;
                                data[i3]=-data[i2]-t1;
                                data[i2]=data[i0]-t2;
                                data[i0]+=t2;
                        }
                 }
                     id<<=1;
                 i1=id-n2;
                 id<<=1;
              } while ( i1<n );
            a=e;
            for (j=2; j<=n8; j++){
                  a3=3*a;
                  cc1=cos(a);
                  ss1=sin(a);
                  cc3=cos(a3);
```

```
                ss3=sin(a3);
                a=j*e;
                i=0;
                id=n2<<1;
                do{
                     for (; i<n; i+=id){
                            i1=i+j-1;
                            i2=i1+n4;
                            i3=i2+n4;
                            i4=i3+n4;
                            i5=i+n4-j+1;
                            i6=i5+n4;
                            i7=i6+n4;
                            i8=i7+n4;
                            t1=data[i3]*cc1+data[i7]*ss1;
                            t2=data[i7]*cc1-data[i3]*ss1;
                            t3=data[i4]*cc3+data[i8]*ss3;
                            t4=data[i8]*cc3-data[i4]*ss3;
                            t5=t1+t3;
                            t6=t2+t4;
                            t3=t1-t3;
                            t4=t2-t4;
                            t2=data[i6]+t6;
                            data[i3]=t6-data[i6];
                            data[i8]=t2;
                            t2=data[i2]-t3;
                            data[i7]=-data[i2]-t3;
                            data[i4]=t2;
                            t1=data[i1]+t5;
                            data[i6]=data[i1]-t5;
                            data[i1]=t1;
                            t1=data[i5]+t4;
                            data[i5]-=t4;
                            data[i2]=t1;
                       }
                     id<<=1;
                     i=id-n2;
                     id<<=1;
                   } while(i<n);
            }
      }

        //division with array length
   for(i=0;i<n;i++) data[i]/=n;
}


////////////////////////////////////////////////////
// Sorensen in-place split-radix FFT for real values
// data: array of doubles:
```

```
// re(0),re(1),re(2),...,re(size-1)
//
// output:
// re(0),re(size/2),re(1),im(1),re(2),im(2),...,re(size/2-1),im(size/2-1)
// normalized by array length
//
// Source:
// Source: see the routines it calls ...

void realfft_split_unshuffled(double *data,long n){

        double *data2;
        long i,j;

        realfft_split(data,n);
        //unshuffling - not in-place
        data2=(double *)malloc(n*sizeof(double));
        j=n/2;
        data2[0]=data[0];
        data2[1]=data[j];
        for(i=1;i<j;i++) {data2[i+i]=data[i];data2[i+i+1]=data[n-i];}
        for(i=0;i<n;i++) data[i]=data2[i];
        free(data2);
}

////////////////////////////////////////////////////
// Sorensen in-place inverse split-radix FFT for real values
// data: array of doubles:
// re(0),re(1),re(2),...,re(size/2),im(size/2-1),...,im(1)
//
// output:
// re(0),re(1),re(2),...,re(size-1)
// NOT normalized by array length
//
// Source:
// Sorensen et al: Real-Valued Fast Fourier Transform Algorithms,
// IEEE Trans. ASSP, ASSP-35, No. 6, June 1987

void irealfft_split(double *data,long n){

  long i,j,k,i5,i6,i7,i8,i0,id,i1,i2,i3,i4,n2,n4,n8,n1;
  double t1,t2,t3,t4,t5,a3,ss1,ss3,cc1,cc3,a,e,sqrt2;

  sqrt2=sqrt(2.0);

n1=n-1;
n2=n<<1;
for(k=n;k>2;k>>=1){
        id=n2;
        n2>>=1;
        n4=n2>>2;
```

```
n8=n2>>3;
e = 2*pi/(n2);
i1=0;
do{
    for (; i1<n; i1+=id){
                i2=i1+n4;
                i3=i2+n4;
                i4=i3+n4;
                t1=data[i1]-data[i3];
                data[i1]+=data[i3];
                data[i2]*=2;
                data[i3]=t1-2*data[i4];
                data[i4]=t1+2*data[i4];
                if (n4!=1){
                        i0=i1+n8;
                        i2+=n8;
                        i3+=n8;
                        i4+=n8;
                        t1=(data[i2]-data[i0])/sqrt2;
                        t2=(data[i4]+data[i3])/sqrt2;
                        data[i0]+=data[i2];
                        data[i2]=data[i4]-data[i3];
                        data[i3]=2*(-t2-t1);
                        data[i4]=2*(-t2+t1);
                }
     }
         id<<=1;
     i1=id-n2;
     id<<=1;
  } while ( i1<n1 );
a=e;
for (j=2; j<=n8; j++){
      a3=3*a;
      cc1=cos(a);
      ss1=sin(a);
      cc3=cos(a3);
      ss3=sin(a3);
      a=j*e;
      i=0;
      id=n2<<1;
      do{
           for (; i<n; i+=id){
                  i1=i+j-1;
                  i2=i1+n4;
                  i3=i2+n4;
                  i4=i3+n4;
                  i5=i+n4-j+1;
                  i6=i5+n4;
                  i7=i6+n4;
                  i8=i7+n4;
```

```
                        t1=data[i1]-data[i6];
                        data[i1]+=data[i6];
                        t2=data[i5]-data[i2];
                        data[i5]+=data[i2];
                        t3=data[i8]+data[i3];
                        data[i6]=data[i8]-data[i3];
                        t4=data[i4]+data[i7];
                        data[i2]=data[i4]-data[i7];
                        t5=t1-t4;
                        t1+=t4;
                        t4=t2-t3;
                        t2+=t3;
                        data[i3]=t5*cc1+t4*ss1;
                        data[i7]=-t4*cc1+t5*ss1;
                        data[i4]=t1*cc3-t2*ss3;
                        data[i8]=t2*cc3+t1*ss3;
                        }
                  id<<=1;
                  i=id-n2;
                  id<<=1;
                } while(i<n1);
           }
        }

   /*----------------------*/
        i0=0;
        id=4;
   do{
       for (; i0<n1; i0+=id){
                        i1=i0+1;
                        t1=data[i0];
                        data[i0]=t1+data[i1];
                        data[i1]=t1-data[i1];
                }
           id<<=1;
       i0=id-2;
       id<<=1;
    } while ( i0<n1 );

/*----------------------*/

//data shuffling
      for (i=0,j=0,n2=n/2; i<n1 ; i++){
          if (i<j){
                                t1=data[j];
                                data[j]=data[i];
                                data[i]=t1;
                                }
          k=n2;
          while (k<=j){
```

```
                                j-=k;
                                k>>=1;
                                }
            j+=k;
        }
    }


    ////////////////////////////////////////////////////////
    // Sorensen in-place radix-2 FFT for real values
    // data: array of doubles:
    // re(0),re(1),re(2),...,re(size-1)
    //
    // output:
    // re(0),re(1),re(2),...,re(size/2),im(size/2-1),...,im(1)
    // normalized by array length
    //
    // Source:
    // Sorensen et al: Real-Valued Fast Fourier Transform Algorithms,
    // IEEE Trans. ASSP, ASSP-35, No. 6, June 1987

    void realfft_radix2(double *data,long n){

        double  xt,a,e, t1, t2, cc, ss;
        long  i, j, k, n1, n2, n3, n4, i1, i2, i3, i4;

            n4=n-1;
      //data shuffling
          for (i=0,j=0,n2=n/2; i<n4 ; i++){
            if (i<j){
                                xt=data[j];
                                data[j]=data[i];
                                data[i]=xt;
                                }
            k=n2;
            while (k<=j){
                                j-=k;
                                k>>=1;
                                }
            j+=k;
          }

    /* -------------------- */
        for (i=0; i<n; i += 2)
          {
             xt = data[i];
             data[i] = xt + data[i+1];
             data[i+1] = xt - data[i+1];
          }
    /* ------------------------ */
        n2 = 1;
```

```
        for (k=n;k>2;k>>=1){
                    n4 = n2;
                    n2 = n4 << 1;
                    n1 = n2 << 1;
                    e = 2*pi/(n1);
                    for (i=0; i<n; i+=n1){
                            xt = data[i];
                            data[i] = xt + data[i+n2];
                            data[i+n2] = xt-data[i+n2];
                            data[i+n4+n2] = -data[i+n4+n2];
                            a = e;
                            n3=n4-1;
                            for (j = 1; j <=n3; j++){
                                    i1 = i+j;
                                    i2 = i - j + n2;
                                    i3 = i1 + n2;
                                    i4 = i - j + n1;
                                    cc = cos(a);
                                    ss = sin(a);
                                    a += e;
                                    t1 = data[i3] * cc + data[i4] * ss;
                                    t2 = data[i3] * ss - data[i4] * cc;
                                    data[i4] = data[i2] - t2;
                                    data[i3] = -data[i2] - t2;
                                    data[i2] = data[i1] - t1;
                                    data[i1] += t1;
                            }
                    }
          }

            //division with array length
       for(i=0;i<n;i++) data[i]/=n;
    }


    ////////////////////////////////////////////////////
    // Sorensen in-place split-radix FFT for real values
    // data: array of doubles:
    // re(0),re(1),re(2),...,re(size-1)
    //
    // output:
    // re(0),re(size/2),re(1),im(1),re(2),im(2),...,re(size/2-1),im(size/2-1)
    // normalized by array length
    //
    // Source:
    // Source: see the routines it calls ...

    void realfft_radix2_unshuffled(double *data,long n){

            double *data2;
```

```
        long i,j;

        //unshuffling - not in-place
        realfft_radix2(data,n);
        data2=(double *)malloc(n*sizeof(double));
        j=n/2;
        data2[0]=data[0];
        data2[1]=data[j];
        for(i=1;i<j;i++) {data2[i+i]=data[i];data2[i+i+1]=data[n-i];}
        for(i=0;i<n;i++) data[i]=data2[i];
        free(data2);
}
```

FFT classes in C++ and Object Pascal

Type : Real-to-Complex FFT and Complex-to-Real IFFT
References : Laurent de Soras (Object Pascal translation by Frederic Vanmol)
Linked file : FFTReal.zip

Notes :
(see linkfile)

Float to int

References : Posted by Ross Bencina

Notes :
intel only

Code :
```
int truncate(float flt)
{
  int i;
  static const double half = 0.5f;
  _asm
  {
     fld flt
     fsub half
     fistp i
  }
  return i
}
```

Float-to-int, coverting an array of floats

References : Posted by Stefan Stenzel

Notes :
intel only

Code :
```
void f2short(float *fptr,short *iptr,int n)
{
_asm {
    mov     ebx,n
    mov     esi,fptr
    mov     edi,iptr
    lea     ebx,[esi+ebx*4]   ; ptr after last
    mov     edx,0x80008000    ; damn endianess confuses...
    mov     ecx,0x4b004b00    ; damn endianess confuses...
    mov     eax,[ebx]         ; get last value
    push    eax
    mov     eax,0x4b014B01
    mov     [ebx],eax             ; mark end
    mov     ax,[esi+2]
    jmp     startf2slp

;   Pad with nops to make loop start at address divisible
;   by 16 + 2, e.g. 0x01408062, don't ask why, but this
;   gives best performance. Unfortumately "align 16" does
;   not seem to work with my VC.
;   below I noted the measured execution times for different
;   nop-paddings on my Pentium Pro, 100 conversions.
;   saturation:  off pos neg


   nop          ;355 546 563 <- seems to be best
;   nop          ;951 547 544
;   nop          ;444 646 643
;   nop          ;444 646 643
;   nop          ;944 951 950
;   nop          ;358 447 644
;   nop          ;358 447 643
;   nop          ;358 544 643
;   nop          ;543 447 643
;   nop          ;643 447 643
;   nop          ;1047 546 746
;   nop          ;545 954 1253
;   nop          ;545 547 661
;   nop          ;544 547 746
;   nop          ;444 947 1147
```

```
;   nop         ;444 548 545
in_range:
    mov     eax,[esi]
    xor     eax,edx
saturate:
    lea     esi,[esi+4]
    mov     [edi],ax
    mov     ax,[esi+2]
    add     edi,2
startf2slp:
    cmp     ax,cx
    je      in_range
    mov     eax,edx
    js      saturate     ; saturate neg -> 0x8000
    dec     eax          ; saturate pos -> 0x7FFF
    cmp     esi,ebx      ; end reached ?
    jb      saturate
    pop     eax
    mov     [ebx],eax    ; restore end flag
    }
}
```

Formant filter

References : Posted by Alex
Code :

```
/*
Public source code by alex@smartelectronix.com
Simple example of implementation of formant filter
Vowelnum can be 0,1,2,3,4 <=> A,E,I,O,U
Good for spectral rich input like saw or square
*/
//-----------------------------------------------------------VOWEL
COEFFICIENTS
const double coeff[5][11]= {
{ 8.11044e-06,
8.943665402, -36.83889529, 92.01697887, -154.337906, 181.6233289,
-151.8651235,   89.09614114, -35.10298511, 8.388101016, -0.923313471  ///A
},
{4.36215e-06,
8.90438318, -36.55179099, 91.05750846, -152.422234, 179.1170248,  ///E
-149.6496211,87.78352223, -34.60687431, 8.282228154, -0.914150747
},
{ 3.33819e-06,
8.893102966, -36.49532826, 90.96543286, -152.4545478, 179.4835618,
-150.315433, 88.43409371, -34.98612086, 8.407803364, -0.932568035  ///I
},
{1.13572e-06,
8.994734087, -37.2084849, 93.22900521, -156.6929844, 184.596544,   ///O
-154.3755513, 90.49663749, -35.58964535, 8.478996281, -0.929252233
},
{4.09431e-07,
8.997322763, -37.20218544, 93.11385476, -156.2530937, 183.7080141,  ///U
-153.2631681, 89.59539726, -35.12454591, 8.338655623, -0.910251753
}
};
//-----------------------------------------------------------------------
------
static double memory[10]={0,0,0,0,0,0,0,0,0,0};
//-----------------------------------------------------------------------
------
float formant_filter(float *in, int vowelnum)
{
    res= (float) ( coeff[vowelnum][0]  *in +
coeff[vowelnum][1]  *memory[0] +
coeff[vowelnum][2]  *memory[1] +
coeff[vowelnum][3]  *memory[2] +
coeff[vowelnum][4]  *memory[3] +
coeff[vowelnum][5]  *memory[4] +
coeff[vowelnum][6]  *memory[5] +
```

```
    coeff[vowelnum][7]  *memory[6] +
    coeff[vowelnum][8]  *memory[7] +
    coeff[vowelnum][9]  *memory[8] +
    coeff[vowelnum][10] *memory[9] );

    memory[9]= memory[8];
    memory[8]= memory[7];
    memory[7]= memory[6];
    memory[6]= memory[5];
    memory[5]= memory[4];
    memory[4]= memory[3];
    memory[3]= memory[2];
    memory[2]= memory[1];
    memory[1]= memory[0];
    memory[0]=(double) res;
    return res;
    }
```

Gaussian dithering

Type : Dithering
References : Posted by Aleksey Vaneev (picoder[AT]mail[DOT]ru)

Notes :
It is a more sophisticated dithering than simple RND. It gives the most low noise floor for the whole spectrum even without noise-shaping. You can use as big N as you can afford (it will not hurt), but 4 or 5 is generally enough.

Code :
```
Basically, next value is calculated this way (for RND going from -0.5 to
0.5):

dither = (RND+RND+...+RND) / N.
           \           /
            \_________/
              N times

If your RND goes from 0 to 1, then this code is applicable:

dither = (RND+RND+...+RND - 0.5*N) / N.
```

Gaussian White noise

References : Posted by Alexey Menshikov

Notes :
Code I use sometimes, but don't remember where I ripped it from.

- Alexey Menshikov

Code :

```
#define ranf() ((float) rand() / (float) RAND_MAX)

float ranfGauss (int m, float s)
{
   static int pass = 0;
   static float y2;
   float x1, x2, w, y1;

   if (pass)
   {
      y1 = y2;
   } else  {
      do {
         x1 = 2.0f * ranf () - 1.0f;
         x2 = 2.0f * ranf () - 1.0f;
         w = x1 * x1 + x2 * x2;
      } while (w >= 1.0f);

      w = (float)sqrt (-2.0 * log (w) / w);
      y1 = x1 * w;
      y2 = x2 * w;
   }
   pass = !pass;

   return ( (y1 * s + (float) m));
}
```

Gaussian White Noise

References : Posted by remage[AT]netposta.hu

Notes :
SOURCE:

Steven W. Smith:
The Scientist and Engineer's Guide to Digital Signal Processing
http://www.dspguide.com

Code :
```
#define PI 3.1415926536f

float R1 = (float) rand() / (float) RAND_MAX;
float R2 = (float) rand() / (float) RAND_MAX;

float X = (float) sqrt( -2.0f * log( R1 )) * cos( 2.0f * PI * R2 );
```

Guitar feedback

References : Posted by Sean Costello

Notes :
It is fairly simple to simulate guitar feedback with a simple Karplus-Strong algorithm (this was described in a CMJ article in the early 90's):

Code :

```
Run the output of the Karplus-Strong delay lines into a nonlinear shaping
function for distortion (i.e. 6 parallel delay lines for 6 strings, going
into 1 nonlinear shaping function that simulates an overdriven amplifier,
fuzzbox, etc.);

Run part of the output into a delay line, to simulate the distance from the
amplifier to the "strings";

The delay line feeds back into the Karplus-Strong delay lines. By
controlling the amount of the output fed into the delay line, and the
length of the delay line, you can control the intensity and pitch of the
feedback note.
```

Hermite interpollation

References : Posted by various

Notes :
These are all different ways to do the same thing : hermite interpollation. Try'm all and benchmark.

Code :

```
// original
inline float hermite1(float x, float y0, float y1, float y2, float y3)
{
    // 4-point, 3rd-order Hermite (x-form)
    float c0 = y1;
    float c1 = 0.5f * (y2 - y0);
    float c2 = y0 - 2.5f * y1 + 2.f * y2 - 0.5f * y3;
    float c3 = 1.5f * (y1 - y2) + 0.5f * (y3 - y0);

    return ((c3 * x + c2) * x + c1) * x + c0;
}

// james mccartney
inline float hermite2(float x, float y0, float y1, float y2, float y3)
{
    // 4-point, 3rd-order Hermite (x-form)
    float c0 = y1;
    float c1 = 0.5f * (y2 - y0);
    float c3 = 1.5f * (y1 - y2) + 0.5f * (y3 - y0);
    float c2 = y0 - y1 + c1 - c3;

    return ((c3 * x + c2) * x + c1) * x + c0;
}

// james mccartney
inline float hermite3(float x, float y0, float y1, float y2, float y3)
{
        // 4-point, 3rd-order Hermite (x-form)
        float c0 = y1;
        float c1 = 0.5f * (y2 - y0);
        float y0my1 = y0 - y1;
        float c3 = (y1 - y2) + 0.5f * (y3 - y0my1 - y2);
        float c2 = y0my1 + c1 - c3;

        return ((c3 * x + c2) * x + c1) * x + c0;
}

// laurent de soras
inline float hermite4(float frac_pos, float xm1, float x0, float x1, float
```

```
x2)
{
   const float    c     = (x1 - xm1) * 0.5f;
   const float    v     = x0 - x1;
   const float    w     = c + v;
   const float    a     = w + v + (x2 - x0) * 0.5f;
   const float    b_neg = w + a;

   return ((((a * frac_pos) - b_neg) * frac_pos + c) * frac_pos + x0);
}
```

Inverted parabolic envelope

Type : envellope generation
References : Posted by James McCartney
Code :

```
dur = duration in samples
midlevel = amplitude at midpoint
beglevel = beginning and ending level (typically zero)

amp = midlevel - beglevel;

rdur = 1.0 / dur;
rdur2 = rdur * rdur;

level = beglevel;
slope = 4.0 * amp * (rdur - rdur2);
curve = -8.0 * amp * rdur2;

...

for (i=0; i<dur; ++i) {
        level += slope;
        slope += curve;
}
```

Lock free fifo

References : Posted by Timo
Linked file : LockFreeFifo.h (this linked file is included below)

Notes :
Simple implementation of a lock free FIFO.

Linked files

```
#include <vector>
#include <exception>

using std::vector;
using std::exception;

template<class T> class LockFreeFifo
{
public:
        LockFreeFifo (unsigned bufsz) : readidx(0), writeidx(0), buffer(bufsz)
        {}

        T get (void)
        {
                if (readidx == writeidx)
                        throw runtime_error ("underrun");

                T result = buffer[readidx];

                if ((readidx + 1) >= buffer.size())
                        readidx = 0;
                else
                        readidx = readidx + 1;

                return result;
        }

        void put (T datum)
        {
                unsigned newidx;

                if ((writeidx + 1) >= buffer.size())
                        newidx = 0;
                else
                        newidx = writeidx + 1;

                if (newidx == readidx)
                        throw runtime_error ("overrun");

                buffer[writeidx] = datum;
```

```
                writeidx = newidx;
        }

private:

        volatile unsigned  readidx, writeidx;
        vector<T> buffer;
};
```

Look ahead limiting

References : Posted by Wilfried Welti

Notes :
use add_value with all values which enter the look-ahead area,
and remove_value with all value which leave this area. to get
the maximum value in the look-ahead area, use get_max_value.
in the very beginning initialize the table with zeroes.

If you always want to know the maximum amplitude in
your look-ahead area, the thing becomes a sorting
problem. very primitive approach using a look-up table

Code :

```
void lookup_add(unsigned section, unsigned size, unsigned value)
{
  if (section==value)
    lookup[section]++;
  else
  {
    size >>= 1;
    if (value>section)
    {
      lookup[section]++;
      lookup_add(section+size,size,value);
    }
    else
      lookup_add(section-size,size,value);
  }
}

void lookup_remove(unsigned section, unsigned size, unsigned value)
{
  if (section==value)
    lookup[section]--;
  else
  {
    size >>= 1;
    if (value>section)
    {
      lookup[section]--;
      lookup_remove(section+size,size,value);
    }
    else
      lookup_remove(section-size,size,value);
```

```
  }
}

unsigned lookup_getmax(unsigned section, unsigned size)
{
  unsigned max = lookup[section] ? section : 0;
  size >>= 1;
  if (size)
    if (max)
    {
      max = lookup_getmax((section+size),size);
      if (!max) max=section;
    }
    else
      max = lookup_getmax((section-size),size);
  return max;
}

void add_value(unsigned value)
{
  lookup_add(LOOKUP_VALUES>>1, LOOKUP_VALUES>>1, value);
}

void remove_value(unsigned value)
{
  lookup_remove(LOOKUP_VALUES>>1, LOOKUP_VALUES>>1, value);
}

unsigned get_max_value()
{
  return lookup_getmax(LOOKUP_VALUES>>1, LOOKUP_VALUES>>1);
}
```

Lowpass filter for parameter edge filtering

References : Olli Niemitalo
Linked file : filter001.gif (this linked file is included below)

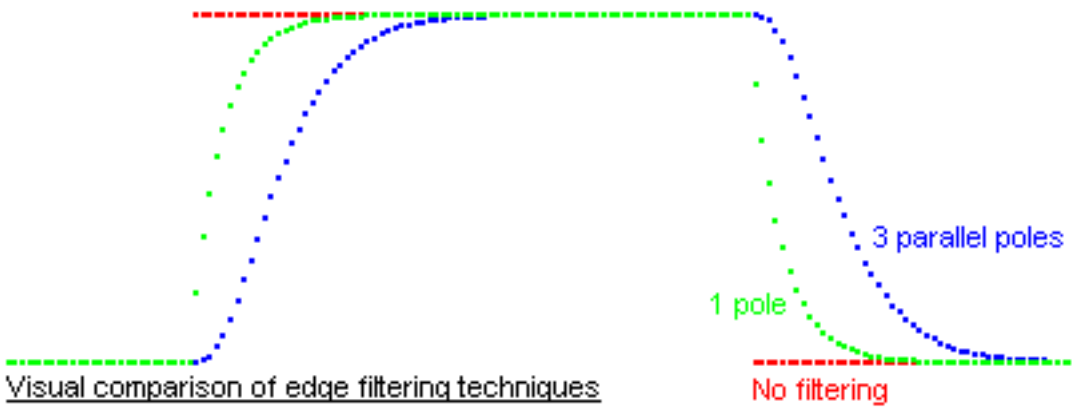
Notes :
use this filter to smooth sudden parameter changes
(see linkfile!)

Code :

```
/* - Three one-poles combined in parallel
* - Output stays within input limits
* - 18 dB/oct (approx) frequency response rolloff
* - Quite fast, 2x3 parallel multiplications/sample, no internal buffers
* - Time-scalable, allowing use with different samplerates
* - Impulse and edge responses have continuous differential
* - Requires high internal numerical precision
*/
{
        /* Parameters */
        // Number of samples from start of edge to halfway to new value
        const double        scale = 100;
        // 0 < Smoothness < 1. High is better, but may cause precision
problems
        const double        smoothness = 0.999;

        /* Precalc variables */
        double                a = 1.0-(2.4/scale); // Could also be set
directly
        double                b = smoothness;      //         -"-
        double                acoef = a;
        double                bcoef = a*b;
        double                ccoef = a*b*b;
        double                mastergain = 1.0 / (-
1.0/(log(a)+2.0*log(b))+2.0/
                        (log(a)+log(b))-1.0/log(a));
        double                again = mastergain;
        double                bgain = mastergain * (log(a*b*b)*(log(a)-
log(a*b)) /
                            ((log(a*b*b)-log(a*b))*log(a*b))
                            - log(a)/log(a*b));
        double                cgain = mastergain * (-(log(a)-log(a*b)) /
                        (log(a*b*b)-log(a*b)));

        /* Runtime variables */
        long                streamofs;
        double                areg = 0;
```

```
        double                  breg = 0;
        double                  creg = 0;

        /* Main loop */
        for (streamofs = 0; streamofs < streamsize; streamofs++)
        {
                /* Update filters */
                areg = acoef * areg + fromstream [streamofs];
                breg = bcoef * breg + fromstream [streamofs];
                creg = ccoef * creg + fromstream [streamofs];

                /* Combine filters in parallel */
                long                temp =   again * areg
                                           + bgain * breg
                                           + cgain * creg;

                /* Check clipping */
                if (temp > 32767)
                {
                        temp = 32767;
                }
                else if (temp < -32768)
                {
                        temp = -32768;
                }

                /* Store new value */
                tostream [streamofs] = temp;
        }
}
```

Linked files



Visual comparison of edge filtering techniques

LP and HP filter

Type : biquad, tweaked butterworth
References : Posted by Patrice Tarrabia
Code :

```
r  = rez amount, from sqrt(2) to ~ 0.1
f  = cutoff frequency
(from ~0 Hz to SampleRate/2 - though many
synths seem to filter only  up to SampleRate/4)

The filter algo:
out(n) = a1 * in + a2 * in(n-1) + a3 * in(n-2) - b1*out(n-1) - b2*out(n-2)

Lowpass:
      c = 1.0 / tan(pi * f / sample_rate);

      a1 = 1.0 / ( 1.0 + r * c + c * c);
      a2 = 2* a1;
      a3 = a1;
      b1 = 2.0 * ( 1.0 - c*c) * a1;
      b2 = ( 1.0 - r * c + c * c) * a1;

Hipass:
      c = tan(pi * f / sample_rate);

      a1 = 1.0 / ( 1.0 + r * c + c * c);
      a2 = -2*a1;
      a3 = a1;
      b1 = 2.0 * ( c*c - 1.0) * a1;
      b2 = ( 1.0 - r * c + c * c) * a1;
```

Magnitude and phase plot of arbitrary IIR function, up to 5th order

Type : magnitude and phase at any frequency
References : Posted by George Yohng

Notes :
Amplitude and phase calculation of IIR equation
run at sample rate "sampleRate" at frequency "F".

AMPLITUDE
---
cf_mag(F,sampleRate,
a0,a1,a2,a3,a4,a5,
b0,b1,b2,b3,b4,b5)

PHASE
---
cf_phi(F,sampleRate,
a0,a1,a2,a3,a4,a5,
b0,b1,b2,b3,b4,b5)

If you need a frequency diagram, draw a plot for
F=0...sampleRate/2

If you need amplitude in dB, use cf_lin2db(cf_mag(.....))

Set b0=-1 if you have such function:

y[n] = a0*x[n] + a1*x[n-1] + a2*x[n-2] + a3*x[n-3] + a4*x[n-4] + a5*x[n-5] +
+ b1*y[n-1] + b2*y[n-2] + b3*y[n-3] + b4*y[n-4] + b5*y[n-5];

Set b0=1 if you have such function:

y[n] = a0*x[n] + a1*x[n-1] + a2*x[n-2] + a3*x[n-3] + a4*x[n-4] + a5*x[n-5] +
- b1*y[n-1] - b2*y[n-2] - b3*y[n-3] - b4*y[n-4] - b5*y[n-5];

Do not try to reverse engineer these formulae - they don't give any sense
other than they are derived from transfer function, and they work. :)

Code :

```
/*
C file can be downloaded from
http://www.yohng.com/dsp/cfsmp.c
*/


#define C_PI 3.14159265358979323846264

double cf_mag(double f,double rate,
              double a0,double a1,double a2,double a3,double a4,double a5,
              double b0,double b1,double b2,double b3,double b4,double b5)
{
    return
        sqrt((a0*a0 + a1*a1 + a2*a2 + a3*a3 + a4*a4 + a5*a5 +
              2*(a0*a1 + a1*a2 + a2*a3 + a3*a4 +
a4*a5)*cos((2*f*C_PI)/rate) +
              2*(a0*a2 + a1*a3 + a2*a4 + a3*a5)*cos((4*f*C_PI)/rate) +
              2*a0*a3*cos((6*f*C_PI)/rate) + 2*a1*a4*cos((6*f*C_PI)/rate) +
              2*a2*a5*cos((6*f*C_PI)/rate) + 2*a0*a4*cos((8*f*C_PI)/rate) +
              2*a1*a5*cos((8*f*C_PI)/rate) +
2*a0*a5*cos((10*f*C_PI)/rate))/
              (b0*b0 + b1*b1 + b2*b2 + b3*b3 + b4*b4 + b5*b5 +
              2*(b0*b1 + b1*b2 + b2*b3 + b3*b4 +
b4*b5)*cos((2*f*C_PI)/rate) +
              2*(b0*b2 + b1*b3 + b2*b4 + b3*b5)*cos((4*f*C_PI)/rate) +
              2*b0*b3*cos((6*f*C_PI)/rate) + 2*b1*b4*cos((6*f*C_PI)/rate) +
              2*b2*b5*cos((6*f*C_PI)/rate) + 2*b0*b4*cos((8*f*C_PI)/rate) +
              2*b1*b5*cos((8*f*C_PI)/rate) +
2*b0*b5*cos((10*f*C_PI)/rate)));
}


double cf_phi(double f,double rate,
              double a0,double a1,double a2,double a3,double a4,double a5,
              double b0,double b1,double b2,double b3,double b4,double b5)
{
        atan2((a0*b0 + a1*b1 + a2*b2 + a3*b3 + a4*b4 + a5*b5 +
              (a0*b1 + a1*(b0 + b2) + a2*(b1 + b3) + a5*b4 + a3*(b2 + b4) +
              a4*(b3 + b5))*cos((2*f*C_PI)/rate) +
              ((a0 + a4)*b2 + (a1 + a5)*b3 + a2*(b0 + b4) +
              a3*(b1 + b5))*cos((4*f*C_PI)/rate) +
a3*b0*cos((6*f*C_PI)/rate) +
              a4*b1*cos((6*f*C_PI)/rate) + a5*b2*cos((6*f*C_PI)/rate) +
              a0*b3*cos((6*f*C_PI)/rate) + a1*b4*cos((6*f*C_PI)/rate) +
              a2*b5*cos((6*f*C_PI)/rate) + a4*b0*cos((8*f*C_PI)/rate) +
              a5*b1*cos((8*f*C_PI)/rate) + a0*b4*cos((8*f*C_PI)/rate) +
              a1*b5*cos((8*f*C_PI)/rate) +
              (a5*b0 + a0*b5)*cos((10*f*C_PI)/rate))/
```

```
              (b0*b0 + b1*b1 + b2*b2 + b3*b3 + b4*b4 + b5*b5 +
              2*((b0*b1 + b1*b2 + b3*(b2 + b4) +
b4*b5)*cos((2*f*C_PI)/rate) +
              (b2*(b0 + b4) + b3*(b1 + b5))*cos((4*f*C_PI)/rate) +
              (b0*b3 + b1*b4 + b2*b5)*cos((6*f*C_PI)/rate) +
              (b0*b4 + b1*b5)*cos((8*f*C_PI)/rate) +
              b0*b5*cos((10*f*C_PI)/rate))),

             ((a1*b0 + a3*b0 + a5*b0 - a0*b1 + a2*b1 + a4*b1 - a1*b2 +
              a3*b2 + a5*b2 - a0*b3 - a2*b3 + a4*b3 -
              a1*b4 - a3*b4 + a5*b4 - a0*b5 - a2*b5 - a4*b5 +
              2*(a3*b1 + a5*b1 - a0*b2 + a4*(b0 + b2) - a1*b3 + a5*b3 +
              a2*(b0 - b4) - a0*b4 - a1*b5 - a3*b5)*cos((2*f*C_PI)/rate) +
              2*(a3*b0 + a4*b1 + a5*(b0 + b2) - a0*b3 - a1*b4 - a0*b5 -
a2*b5)*
              cos((4*f*C_PI)/rate) + 2*a4*b0*cos((6*f*C_PI)/rate) +
              2*a5*b1*cos((6*f*C_PI)/rate) - 2*a0*b4*cos((6*f*C_PI)/rate) -
              2*a1*b5*cos((6*f*C_PI)/rate) + 2*a5*b0*cos((8*f*C_PI)/rate) -
              2*a0*b5*cos((8*f*C_PI)/rate))*sin((2*f*C_PI)/rate))/
              (b0*b0 + b1*b1 + b2*b2 + b3*b3 + b4*b4 + b5*b5 +
              2*(b0*b1 + b1*b2 + b2*b3 + b3*b4 +
b4*b5)*cos((2*f*C_PI)/rate) +
              2*(b0*b2 + b1*b3 + b2*b4 + b3*b5)*cos((4*f*C_PI)/rate) +
              2*b0*b3*cos((6*f*C_PI)/rate) + 2*b1*b4*cos((6*f*C_PI)/rate) +
              2*b2*b5*cos((6*f*C_PI)/rate) + 2*b0*b4*cos((8*f*C_PI)/rate) +
              2*b1*b5*cos((8*f*C_PI)/rate) +
2*b0*b5*cos((10*f*C_PI)/rate)));
}

double cf_lin2db(double lin)
{
    if (lin<9e-51) return -1000; /* prevent invalid operation */
    return 20*log10(lin);
}
```

MATLAB-Tools for SNDAN

References : Posted by Markus Sapp
Linked file : other001.zip

Notes :
(see linkfile)

Measuring interpollation noise

References : Posted by Jon Watte

Notes :
You can easily estimate the error by evaluating the actual function and evaluating your interpolator at each of the mid-points between your samples. The absolute difference between these values, over the absolute value of the "correct" value, is your relative error. log10 of your relative error times 20 is an estimate of your quantization noise in dB. Example:

You have a table for every 0.5 "index units". The value at index unit 72.0 is 0.995 and the value at index unit 72.5 is 0.999. The interpolated value at index 72.25 is 0.997. Suppose the actual function value at that point was 0.998; you would have an error of 0.001 which is a relative error of 0.001002004.. log10(error) is about -2.99913, which times 20 is about -59.98. Thus, that's your quantization noise at that position in the table. Repeat for each pair of samples in the table.

Note: I said "quantization noise" not "aliasing noise". The aliasing noise will, as far as I know, only happen when you start up-sampling without band-limiting and get frequency aliasing (wrap-around), and thus is mostly independent of what specific interpolation mechanism you're using.

Millimeter to DB (faders...)

References : Posted by James McCartney

Notes :
These two functions reproduce a traditional professional mixer fader taper.

MMtoDB converts millimeters of fader travel from the bottom of the fader for a 100 millimeter fader into decibels. DBtoMM is the inverse.

The taper is as follows from the top:
The top of the fader is +10 dB
100 mm to 52 mm : -5 dB per 12 mm
52 mm to 16 mm : -10 dB per 12 mm
16 mm to 4 mm : -20 dB per 12 mm
4 mm to 0 mm : fade to zero. (in these functions I go to -200dB which is effectively zero for up to 32 bit audio.)

Code :

```
float MMtoDB(float mm)
{
        float db;

        mm = 100. - mm;

        if (mm <= 0.) {
                db = 10.;
        } else if (mm < 48.) {
                db = 10. - 5./12. * mm;
        } else if (mm < 84.) {
                db = -10. - 10./12. * (mm - 48.);
        } else if (mm < 96.) {
                db = -40. - 20./12. * (mm - 84.);
        } else if (mm < 100.) {
                db = -60. - 35. * (mm - 96.);
        } else db = -200.;
        return db;
}

float DBtoMM(float db)
{
        float mm;
        if (db >= 10.) {
                mm = 0.;
```

```
        } else if (db > -10.) {
                mm = -12./5. * (db - 10.);
        } else if (db > -40.) {
                mm = 48. - 12./10. * (db + 10.);
        } else if (db > -60.) {
                mm = 84. - 12./20. * (db + 40.);
        } else if (db > -200.) {
                mm = 96. - 1./35. * (db + 60.);
        } else mm = 100.;

        mm = 100. - mm;

        return mm;
}
```

Moog VCF

Type : 24db resonant lowpass
References : CSound source code, Stilson/Smith CCRMA paper.

Notes :
Digital approximation of Moog VCF. Fairly easy to calculate coefficients, fairly easy to process algorithm, good sound.

Code :

```
//Init
cutoff = cutoff freq in Hz
fs = sampling frequency //(e.g. 44100Hz)
res = resonance [0 - 1] //(minimum - maximum)

f = 2 * cutoff / fs; //[0 - 1]
k = 3.6*f - 1.6*f*f -1; //(Empirical tunning)
p = (k+1)*0.5;
scale = e^((1-p)*1.386249;
r = res*scale;
y4 = output;

y1=y2=y3=y4=oldx=oldy1=oldy2=oldy3=0;

//Loop
//--Inverted feed back for corner peaking
x = input - r*y4;

//Four cascaded onepole filters (bilinear transform)
y1=x*p + oldx*p - k*y1;
y2=y1*p+oldy1*p - k*y2;
y3=y2*p+oldy2*p - k*y3;
y4=y3*p+oldy3*p - k*y4;

//Clipper band limited sigmoid
y4 = y4 - (y4^3)/6;

oldx = x;
oldy1 = y1;
oldy2 = y2;
oldy3 = y3;
```

Moog VCF, variation 1

Type : 24db resonant lowpass
References : CSound source code, Stilson/Smith CCRMA paper., Paul Kellett version

Notes :
The second "q =" line previously used exp() - I'm not sure if what I've done is any faster, but this line needs playing with anyway as it controls which frequencies will self-oscillate. I think it could be tweaked to sound better than it currently does.

Highpass / Bandpass :

They are only 6dB/oct, but still seem musically useful - the 'fruity' sound of the 24dB/oct lowpass is retained.

Code :

```
// Moog 24 dB/oct resonant lowpass VCF
// References: CSound source code, Stilson/Smith CCRMA paper.
// Modified by paul.kellett@maxim.abel.co.uk July 2000

  float f, p, q;             //filter coefficients
  float b0, b1, b2, b3, b4;  //filter buffers (beware denormals!)
  float t1, t2;              //temporary buffers

// Set coefficients given frequency & resonance [0.0...1.0]

  q = 1.0f - frequency;
  p = frequency + 0.8f * frequency * q;
  f = p + p - 1.0f;
  q = resonance * (1.0f + 0.5f * q * (1.0f - q + 5.6f * q * q));

// Filter (in [-1.0...+1.0])

  in -= q * b4;                          //feedback
  t1 = b1;  b1 = (in + b0) * p - b1 * f;
  t2 = b2;  b2 = (b1 + t1) * p - b2 * f;
  t1 = b3;  b3 = (b2 + t2) * p - b3 * f;
            b4 = (b3 + t1) * p - b4 * f;
  b4 = b4 - b4 * b4 * b4 * 0.166667f;    //clipping
  b0 = in;

// Lowpass  output:  b4
// Highpass output:  in - b4;
// Bandpass output:  3.0f * (b3 - b4);
```

Moog VCF, variation 2

Type : 24db resonant lowpass
References : CSound source code, Stilson/Smith CCRMA paper., Timo Tossavainen (?) version

Notes :
in[x] and out[x] are member variables, init to 0.0 the controls:

fc = cutoff, nearly linear [0,1] -> [0, fs/2]
res = resonance [0, 4] -> [no resonance, self-oscillation]

Code :

```
Tdouble MoogVCF::run(double input, double fc, double res)
{
  double f = fc * 1.16;
  double fb = res * (1.0 - 0.15 * f * f);
  input -= out4 * fb;
  input *= 0.35013 * (f*f)*(f*f);
  out1 = input + 0.3 * in1 + (1 - f) * out1; // Pole 1
  in1  = input;
  out2 = out1 + 0.3 * in2 + (1 - f) * out2;  // Pole 2
  in2  = out1;
  out3 = out2 + 0.3 * in3 + (1 - f) * out3;  // Pole 3
  in3  = out2;
  out4 = out3 + 0.3 * in4 + (1 - f) * out4;  // Pole 4
  in4  = out3;
  return out4;
}
```

Noise Shaping Class

Type : Dithering with 9th order noise shaping
References : Posted by cshei[AT]indiana.edu
Linked file : NS9dither16.h (this linked file is included below)

Notes :
This is an implemetation of a 9th order noise shaping & dithering class, that runs quite fast (it has one function that uses Intel x86 assembly, but you can replace it with a different rounding function if you are running on a non-Intel platform). _aligned_malloc and _aligned_free require the MSVC++ Processor Pack, available from www.microsoft.com. You can replace them with "new" and "delete," but allocating aligned memory seems to make it run faster. Also, you can replace ZeroMemory with a memset that sets the memory to 0 if you aren't using Win32.
Input should be floats from -32768 to 32767 (processS will clip at these points for you, but clipping is bad when you are trying to convert floats to shorts). Note to reviewer - it would probably be better if you put the code in a file such as NSDither.h and have a link to it - it's rather long.

(see linked file)

Linked files

```
#pragma once

#include <malloc.h>
#include <rounding.h>

// F-weighted
//static const float or3Fc[3] = {1.623f, -0.982f, 0.109f};
static const float or9Fc[9] = {2.412f, -3.370f, 3.937f, -4.174f, 3.353f, -2.205f,
1.281f, -0.569f, 0.0847f};

// modified-E weighted
//static const float or2MEc[2] = {1.537f, -0.8367f};
//static const float or3MEc[3] = {1.652f, -1.049f, 0.1382f};
//static const float or9MEc[9] = {1.662f, -1.263f, 0.4827f, -0.2913f, 0.1268f, -
0.1124f, 0.03252f, -0.01265f, -0.03524f};

// improved-E weighted
//static const float or5IEc[5] = {2.033f, -2.165f, 1.959f, -1.590f, 0.6149f};
//static const float or9IEc[9] = {2.847f, -4.685f, 6.214f, -7.184f, 6.639f, -5.032f,
3.263f, -1.632f, 0.4191f};

// Simple 2nd order
//static const float or2Sc[2] = {1.0f, -0.5f};

// Much faster than C's % operator (anyway, x will never be > 2*n in this case so
this is very simple and fast)
// Tell the compiler to try to inline as much as possible, since it makes it run so
much faster since these functions get called at least once per sample
__inline my_mod(int x, int n)
{
        if(x > n) x-=n;
        return x;
}

__inline int round(float f)
```

```
{
        int r;
        __asm {
                fld f
                fistp r
        }
        return r;
}

__inline short ltos(long l)
{
        return (short)((l==(short)l) ? l : (l>>31)^0x7FFF);
}

__inline float frand()
{
        // Linear Congruential Method - got it from the book "Algorithms," by Robert
Sedgewick
        static unsigned long a = 0xDEADBEEF;

        a = a * 140359821 + 1;
        return a * (1.0f / 0xFFFFFFFF);
}

class NS9dither16
{
public:
        NS9dither16();
        ~NS9dither16();
        short processS(float samp);
        int processI(float samp);
        void reset();

private:
        int order;
        int HistPos;

        float* c;       // Coeffs
        float* EH;      // Error History
};

__inline NS9dither16::NS9dither16()
{
        order = 9;

        //c=new float[order];  // if you don't have _aligned_malloc
        c = (float*)_aligned_malloc(order*sizeof(float), 16);
        CopyMemory(c, or9Fc, order*sizeof(float));

        //EH = new float[2*order];      // if you don't have _aligned_malloc
        EH = (float*)_aligned_malloc(2*order*sizeof(float), 16);
        ZeroMemory(EH, 2*order*sizeof(float));

        // Start at top of error history - you can make it start anywhere from 0-8 if
you really want to
        HistPos=8;
```

```
}

__inline NS9dither16::~NS9dither16()
{
        //if(c) delete [] c;    // if you don't have _aligned_free
        //if(EH) delete [] EH;  // if you don't have _aligned_free
        if(c) _aligned_free(c); // don't really need "if," since it is OK to free
null pointer, but still...
        if(EH) _aligned_free(EH);
}

__inline void NS9dither16::reset()
{
        ZeroMemory(EH, 2*order*sizeof(float));
        // Start at top of error history - you can make it start anywhere from 0-8 if
you really want to
        HistPos=8;
}

// Force inline because VC++.NET doesn't inline for some reason (VC++ 6 does)
__forceinline short NS9dither16::processS(float samp)
{
        int output;

        /*for(int x=0; x<order; x++)
        {
                //samp -= c[x] * EH[(HistPos+x) % order];
                samp -= c[x] * EH[HistPos+x];
        }*/
        // Unrolled loop for faster execution
        /*samp -= c[0]*EH[HistPos] + c[1]*EH[HistPos+1] + c[2]*EH[HistPos+2] +
                        c[3]*EH[HistPos+3] + c[4]*EH[HistPos+4] + c[5]*EH[HistPos+5]
+
                        c[6]*EH[HistPos+6] + c[7]*EH[HistPos+7] +
c[8]*EH[HistPos+8];*/
        // This arrangement seems to execute 3 clock cycles faster on a P-III
        samp -= c[8]*EH[HistPos+8] + c[7]*EH[HistPos+7] + c[6]*EH[HistPos+6] +
                        c[5]*EH[HistPos+5] + c[4]*EH[HistPos+4] + c[3]*EH[HistPos+3]
+
                        c[2]*EH[HistPos+1] + c[1]*EH[HistPos+1] + c[0]*EH[HistPos];

        output = round(samp + (frand() + frand() - 1));

        //HistPos =(HistPos+8) % order; // The % operator is really slow
        HistPos = my_mod((HistPos+8), order);
        // Update buffer (both copies)
        EH[HistPos+9] = EH[HistPos] = output - samp;

        return ltos(output);
}

__forceinline int NS9dither16::processI(float samp)
{
        int output;

        /*for(int x=0; x<order; x++)
```

```
        {
                //samp -= c[x] * EH[(HistPos+x) % order];
                samp -= c[x] * EH[HistPos+x];
        }*/
        // Unrolled loop for faster execution
        /*samp -= c[0]*EH[HistPos] + c[1]*EH[HistPos+1] + c[2]*EH[HistPos+2] +
                        c[3]*EH[HistPos+3] + c[4]*EH[HistPos+4] + c[5]*EH[HistPos+5]
+
                        c[6]*EH[HistPos+6] + c[7]*EH[HistPos+7] +
c[8]*EH[HistPos+8];*/
        // This arrangement seems to execute 3 clock cycles faster on a P-III
        samp -= c[8]*EH[HistPos+8] + c[7]*EH[HistPos+7] + c[6]*EH[HistPos+6] +
                        c[5]*EH[HistPos+5] + c[4]*EH[HistPos+4] + c[3]*EH[HistPos+3]
+
                        c[2]*EH[HistPos+1] + c[1]*EH[HistPos+1] + c[0]*EH[HistPos];

        output = round(samp + (frand() + frand() - 1));

        //HistPos =(HistPos+8) % order; // The % operator is really slow
        HistPos = my_mod((HistPos+8), order);
        // Update buffer (both copies)
        EH[HistPos+9] = EH[HistPos] = output - samp;

        return output;
}
```

Notch filter

Type : 2 poles 2 zeros IIR
References : Posted by Olli Niemitalo

Notes :
Creates a muted spot in the spectrum with adjustable steepness. A complex conjugate pair of zeros on the z- plane unit circle and neutralizing poles approaching at the same angles from inside the unit circle.

Code :
```
Parameters:
0 =< freq =< samplerate/2
0 =< q < 1 (The higher, the narrower)

AlgoAlgo=double pi = 3.141592654;
double sqrt2 = sqrt(2.0);

double freq = 2050; // Change! (zero & pole angle)
double q = 0.4;     // Change! (pole magnitude)

double z1x = cos(2*pi*freq/samplerate);
double a0a2 = (1-q)*(1-q)/(2*(fabs(z1x)+1)) + q;
double a1 = -2*z1x*a0a2;
double b1 = -2*z1x*q;
double b2 = q*q;
double reg0, reg1, reg2;

unsigned int streamofs;
reg1 = 0;
reg2 = 0;

/* Main loop */
for (streamofs = 0; streamofs < streamsize; streamofs++)
{
  reg0 = a0a2 * ((double)fromstream[streamofs]
                 + fromstream[streamofs+2])
       + a1 * fromstream[streamofs+1]
       - b1 * reg1
       - b2 * reg2;

  reg2 = reg1;
  reg1 = reg0;

  int temp = reg0;

  /* Check clipping */
  if (temp > 32767) {
```

```
      temp = 32767;
    } else if (temp < -32768) temp = -32768;

    /* Store new value */
    tostream[streamofs] = temp;
  }
```

One pole LP and HP

References : Posted by Bram

Code :

```
LP:
recursion: tmp = (1-p)*in + p*tmp with output = tmp
coefficient: p = (2-cos(x)) - sqrt((2-cos(x))^2 - 1) with x =
2*pi*cutoff/samplerate
coeficient approximation: p = (1 - 2*cutoff/samplerate)^2

HP:
recursion: tmp = (p-1)*in - p*tmp with output = tmp
coefficient: p = (2+cos(x)) - sqrt((2+cos(x))^2 - 1) with x =
2*pi*cutoff/samplerate
coeficient approximation: p = (2*cutoff/samplerate)^2
```

One pole, one zero LP/HP

References : Posted by mistert[AT]inwind[DOT]it
Code :

```
void SetLPF(float fCut, float fSampling)
{
    float w = 2.0 * fSampling;
    float Norm;

    fCut *= 2.0F * PI;
    Norm = 1.0 / (fCut + w);
    b1 = (w - fCut) * Norm;
    a0 = a1 = fCut * Norm;
}

void SetHPF(float fCut, float fSampling)
{
    float w = 2.0 * fSampling;
    float Norm;

    fCut *= 2.0F * PI;
    Norm = 1.0 / (fCut + w);
    a0 = w * Norm;
    a1 = -a0;
    b1 = (w - fCut) * Norm;
}

Where
out[n] = in[n]*a0 + in[n-1]*a1 + out[n-1]*b1;
```

One zero, LP/HP

References : Posted by Bram

Notes :
LP is only 'valid' for cutoffs > samplerate/4
HP is only 'valid' for cutoffs < samplerate/4

Code :

```
theta = cutoff*2*pi / samplerate

LP:
H(z) = (1+p*z^(-1)) / (1+p)
out[i] = 1/(1+p) * in[i] + p/(1+p) * in[i-1];
p = (1-2*cos(theta)) - sqrt((1-2*cos(theta))^2 - 1)
Pi/2 < theta < Pi

HP:
H(z) = (1-p*z^(-1)) / (1+p)
out[i] = 1/(1+p) * in[i] - p/(1+p) * in[i-1];
p = (1+2*cos(theta)) - sqrt((1+2*cos(theta))^2 - 1)
0 < theta < Pi/2
```

Parallel combs delay calculation

References : Posted by Juhana Sadeharju (kouhia[AT]nic[DOT]funet[DOT]fi)

Notes :
This formula can be found from a patent related to parallel combs structure. The formula places the first echoes coming out of parallel combs to uniformly distributed sequence. If T_ ,...,T_n are the delay lines in increasing order, the formula can be derived by setting T_(k-1)/T_k = Constant and T_n/(2*T_1) = Constant, where 2*T_1 is the echo coming just after the echo T_n. I figured this out myself as it is not told in the patent. The formula is not the best which one can come up. I use a search method to find echo sequences which are uniform enough for long enough time. The formula is uniform for a short time only.
The formula doesn't work good for series allpass and FDN structures, for which a similar formula can be derived with the same idea. The search method works for these structures as well.

Phase modulation Vs. Frequency modulation

References : Posted by Bram
Linked file : SimpleOscillator.h (this linked file is included below)

Notes :
This code shows what the difference is betwee FM and PM.
The code is NOT optimised, nor should it be used like this.
It is an EXAMPLE

See linked file.

Linked files

```
////////////////////////////////////////////////
//
// this code was NEVER MEANT TO BE USED.
//
// use as EXPLANATION ONLY for the difference between
// Phase Modulation and Frequency Modulation.
// there are MANY ways to speed this code up.
//
// bram@musicdsp.org | bram@smartelectronix.com
//
// ps:
// we use the 'previous' value of the phase in all the algo's to make sure that
// the first call to the getSampleXX() function returns the wave at phase 'zero'
//
////////////////////////////////////////////////

#include "math.h";

#define Pi 3.141592f

class SimpleOscillator
{
        SimpleOscillator(const float sampleRate = 44100.f, const long tableSize =
4096)
        {
                this->tableSize = tableSize;
                this->sampleRate = sampleRate;

                phase = 0.f;

                makeTable();
        }

        ~SimpleOscillator()
        {
                delete [] table;
        }
```

```
        // normal oscillator, no modulation
        //
        float generateSample(const float frequency)
        {
                float lookupPhase = phase;

                phase += frequency * (float)tableSize / sampleRate;
                wrap(phase);

                return lookup(lookupPhase);
        }


        // frequency modulation
        // the fm input should be in HZ.
        //
        // example:
        // osc1.getSampleFM(440.f, osc2.getSample(0.5f) * 5.f )
        // would give a signal where the frequency of the signal is
        // modulated between 435hz and 445hz at a 0.5hz rate
        //
        float generateSampleFM(const float frequency, const float fm)
        {
                float lookupPhase = phase;

                phase += (frequency + fm) * (float)tableSize / sampleRate;
                wrap(phase);

                return lookup(lookupPhase);
        }


        // phase modulation
        //
        // a phase mod value of 1.f will increase the "phase" of the wave by a full
cycle
        // i.e. calling getSamplePM(440.f,1.f) will result in the "same" wave as
getSamplePM(440.f,0.f)
        //
        float generateSamplePM(const float frequency, const float pm)
        {
                float lookupPhase = phase + (float)tableSize * pm;
                wrap(lookupPhase)

                phase += frequency * (float)tableSize / sampleRate;
                wrap(phase);

                return lookup(lookupPhase);
        }


        // do the lookup in the table
        // you could use different methods here
        // like linear interpollation or higher order...
        // see musicdsp.org
```

```
        //
        float lookup(const float phase)
        {
                return table[(long)phase];
        }


        // wrap around
        //
        void wrap(float &in)
        {
                while(in < 0.f)
                        in += (float)tableSize;

                while(in >= (float)tableSize)
                        in -= (float)tableSize;

                return in;
        }


        // set the sample rate
        //
        void setSampleRate(const float sampleRate)
        {
                this->sampleRate = sampleRate;
        }


        // sets the phase of the oscillator
        // phase should probably be in 0..Pi*2
        //
        void setPhase(const float phase)
        {
                this->phase = phase / (2.f * Pi) * (float)tableSize;
                wrap(phase);
        }

private:

        float sampleRate;
        float phase;

        float *table;
        long tableSize;

        void makeTable()
        {
                table = new float[tableSize];
                for(long i=0;i<tableSize;i++)
                {
                        float x = Pi * 2.f * (float)i / (float)tableSize;
                        table[i] = (float)sin(x);
                }
        }
```

```
}
```

Phaser code

References : Posted by Ross Bencina
Linked file : phaser.cpp (this linked file is included below)

Notes :
(see linked file)

Linked files

```
/*
Date: Mon, 24 Aug 1998 07:02:40 -0700
Reply-To: music-dsp
Originator: music-dsp@shoko.calarts.edu
Sender: music-dsp
Precedence: bulk
From: "Ross Bencina" <rbencina@hotmail.com>
To: Multiple recipients of list <music-dsp>
Subject: Re: Phaser revisited [code included]
X-Comment:  Music Hackers Unite! http://shoko.calarts.edu/~glmrboy/musicdsp/music-
dsp.html
Status: RO

Hi again,

Thanks to Chris Towsend and Marc Lindahl for their helpful
contributions. I now have a working phaser and it sounds great! It seems
my main error was using a 'sub-sampled' all-pass reverberator instead of
a single sample all-pass filter [what was I thinking? :)].

I have included a working prototype (C++) below for anyone who is
interested. My only remaining doubt is whether the conversion from
frequency to delay time [ _dmin = fMin / (SR/2.f); ] makes any sense
what-so-ever.

Ross B.
*/
/*
    class: Phaser
    implemented by: Ross Bencina <rossb@kagi.com>
    date: 24/8/98

    Phaser is a six stage phase shifter, intended to reproduce the
    sound of a traditional analogue phaser effect.
    This implementation uses six first order all-pass filters in
    series, with delay time modulated by a sinusoidal.

    This implementation was created to be clear, not efficient.
    Obvious modifications include using a table lookup for the lfo,
    not updating the filter delay times every sample, and not
    tuning all of the filters to the same delay time.

    Thanks to:
```

```
    The nice folks on the music-dsp mailing list, including...
    Chris Towsend and Marc Lindahl

    ...and Scott Lehman's Phase Shifting page at harmony central:
    http://www.harmony-central.com/Effects/Articles/Phase_Shifting/

*/

#define SR (44100.f)  //sample rate
#define F_PI (3.14159f)

class Phaser{
public:
        Phaser()  //initialise to some usefull defaults...
        : _fb( .7f )
        , _lfoPhase( 0.f )
        , _depth( 1.f )
        , _zm1( 0.f )
    {
        Range( 440.f, 1600.f );
        Rate( .5f );
    }

    void Range( float fMin, float fMax ){ // Hz
        _dmin = fMin / (SR/2.f);
        _dmax = fMax / (SR/2.f);
    }

    void Rate( float rate ){ // cps
        _lfoInc = 2.f * F_PI * (rate / SR);
    }

    void Feedback( float fb ){ // 0 -> <1.
        _fb = fb;
    }

    void Depth( float depth ){  // 0 -> 1.
        _depth = depth;
    }

    float Update( float inSamp ){
        //calculate and update phaser sweep lfo...
        float d  = _dmin + (_dmax-_dmin) * ((sin( _lfoPhase ) +
1.f)/2.f);
        _lfoPhase += _lfoInc;
        if( _lfoPhase >= F_PI * 2.f )
                _lfoPhase -= F_PI * 2.f;

        //update filter coeffs
        for( int i=0; i<6; i++ )
                _alps[i].Delay( d );

        //calculate output
        float y =       _alps[0].Update(
```

```
                                    _alps[1].Update(
                        _alps[2].Update(
                         _alps[3].Update(
                          _alps[4].Update(
                           _alps[5].Update( inSamp + _zm1 * _fb ))))));
        _zm1 = y;

        return inSamp + y * _depth;
    }
private:
        class AllpassDelay{
    public:
        AllpassDelay()
                : _a1( 0.f )
            , _zm1( 0.f )
            {}

        void Delay( float delay ){ //sample delay time
                _a1 = (1.f - delay) / (1.f + delay);
        }

        float Update( float inSamp ){
                float y = inSamp * -_a1 + _zm1;
                _zm1 = y * _a1 + inSamp;

            return y;
        }
    private:
        float _a1, _zm1;
    };

    AllpassDelay _alps[6];

    float _dmin, _dmax; //range
    float _fb; //feedback
    float _lfoPhase;
    float _lfoInc;
    float _depth;

    float _zm1;
};
```

Pink noise filter

References : Posted by Paul Kellett
Linked file : pink.txt (this linked file is included below)

Notes :
(see linked file)

Linked files

```
Filter to make pink noise from white  (updated March 2000)
----------------------------------

This is an approximation to a -10dB/decade filter using a weighted sum
of first order filters. It is accurate to within +/-0.05dB above 9.2Hz
(44100Hz sampling rate). Unity gain is at Nyquist, but can be adjusted
by scaling the numbers at the end of each line.

If 'white' consists of uniform random numbers, such as those generated
by the rand() function, 'pink' will have an almost gaussian level
distribution.


  b0 = 0.99886 * b0 + white * 0.0555179;
  b1 = 0.99332 * b1 + white * 0.0750759;
  b2 = 0.96900 * b2 + white * 0.1538520;
  b3 = 0.86650 * b3 + white * 0.3104856;
  b4 = 0.55000 * b4 + white * 0.5329522;
  b5 = -0.7616 * b5 - white * 0.0168980;
  pink = b0 + b1 + b2 + b3 + b4 + b5 + b6 + white * 0.5362;
  b6 = white * 0.115926;


An 'economy' version with accuracy of +/-0.5dB is also available.

  b0 = 0.99765 * b0 + white * 0.0990460;
  b1 = 0.96300 * b1 + white * 0.2965164;
  b2 = 0.57000 * b2 + white * 1.0526913;
  pink = b0 + b1 + b2 + white * 0.1848;



---
paul.kellett@maxim.abel.co.uk
http://www.abel.co.uk/~maxim/
```

Polyphase Filters

Type : polyphase filters, used for up and down-sampling
References : C++ source code by Dave from Muon Software
Linked file : BandLimit.cpp (this linked file is included below)
Linked file : BandLimit.h (this linked file is included below)

Linked files

```
CAllPassFilter::CAllPassFilter(const double coefficient)
{
	a=coefficient;

	x0=0.0;
	x1=0.0;
	x2=0.0;

	y0=0.0;
	y1=0.0;
	y2=0.0;

};


CAllPassFilter::~CAllPassFilter()
{
};


double CAllPassFilter::process(const double input)
{
	//shuffle inputs
	x2=x1;
	x1=x0;
	x0=input;

	//shuffle outputs
	y2=y1;
	y1=y0;

	//allpass filter 1
	const double output=x2+((input-y2)*a);

	y0=output;

	return output;
};


CAllPassFilterCascade::CAllPassFilterCascade(const double* coefficient, const int N)
{
	allpassfilter=new CAllPassFilter*[N];

	for (int i=0;i<N;i++)
	{
```

```
                allpassfilter[i]=new CAllPassFilter(coefficient[i]);
        }

        numfilters=N;
};


CAllPassFilterCascade::~CAllPassFilterCascade()
{
        delete[] allpassfilter;
};


double CAllPassFilterCascade::process(const double input)
{
        double output=input;

        int i=0;

        do
        {
                output=allpassfilter[i]->process(output);
                i++;

        } while (i<numfilters);

        return output;
};


CHalfBandFilter::CHalfBandFilter(const int order, const bool steep)
{
        if (steep==true)
        {
                if (order==12)  //rejection=104dB, transition band=0.01
                {
                        double a_coefficients[6]=
                        {0.036681502163648017
                        ,0.2746317593794541
                        ,0.56109896978791948
                        ,0.769741833862266
                        ,0.8922608180038789
                        ,0.962094548378084
                        };

                        double b_coefficients[6]=
                        {0.13654762463195771
                        ,0.42313861743656667
                        ,0.6775400499741616
                        ,0.839889624849638
                        ,0.9315419599631839
                        ,0.9878163707328971
                        };

                        filter_a=new CAllPassFilterCascade(a_coefficients,6);
                        filter_b=new CAllPassFilterCascade(b_coefficients,6);
```

```
}
else if (order==10)     //rejection=86dB, transition band=0.01
{
        double a_coefficients[5]=
        {0.051457617441190984
        ,0.35978656070567017
        ,0.6725475931034693
        ,0.8590884928249939
        ,0.9540209867860787
        };

        double b_coefficients[5]=
        {0.18621906251989334
        ,0.529951372847964
        ,0.7810257527489514
        ,0.9141815687605308
        ,0.985475023014907
        };

        filter_a=new CAllPassFilterCascade(a_coefficients,5);
        filter_b=new CAllPassFilterCascade(b_coefficients,5);
}
else if (order==8)      //rejection=69dB, transition band=0.01
{
        double a_coefficients[4]=
        {0.07711507983241622
        ,0.4820706250610472
        ,0.7968204713315797
        ,0.9412514277740471
        };

        double b_coefficients[4]=
        {0.2659685265210946
        ,0.6651041532634957
        ,0.8841015085506159
        ,0.9820054141886075
        };

        filter_a=new CAllPassFilterCascade(a_coefficients,4);
        filter_b=new CAllPassFilterCascade(b_coefficients,4);
}
else if (order==6)      //rejection=51dB, transition band=0.01
{
        double a_coefficients[3]=
        {0.1271414136264853
        ,0.6528245886369117
        ,0.9176942834328115
        };

        double b_coefficients[3]=
        {0.40056789819445626
        ,0.8204163891923343
        ,0.9763114515836773
        };
```

```
                filter_a=new CAllPassFilterCascade(a_coefficients,3);
                filter_b=new CAllPassFilterCascade(b_coefficients,3);
        }
        else if (order==4)      //rejection=53dB,transition band=0.05
        {
                double a_coefficients[2]=
                {0.12073211751675449
                ,0.6632020224193995
                };

                double b_coefficients[2]=
                {0.3903621872345006
                ,0.890786832653497
                };

                filter_a=new CAllPassFilterCascade(a_coefficients,2);
                filter_b=new CAllPassFilterCascade(b_coefficients,2);
        }

        else    //order=2, rejection=36dB, transition band=0.1
        {
                double a_coefficients=0.23647102099689224;
                double b_coefficients=0.7145421497126001;

                filter_a=new CAllPassFilterCascade(&a_coefficients,1);
                filter_b=new CAllPassFilterCascade(&b_coefficients,1);
        }
}
else    //softer slopes, more attenuation and less stopband ripple
{
        if (order==12)  //rejection=150dB, transition band=0.05
        {
                double a_coefficients[6]=
                {0.01677466677723562
                ,0.13902148819717805
                ,0.3325011117394731
                ,0.53766105314488
                ,0.7214184024215805
                ,0.8821858402078155
                };

                double b_coefficients[6]=
                {0.06501319274445962
                ,0.23094129990840923
                ,0.4364942348420355
                ,0.06329609551399348
                ,0.80378086794111226
                ,0.9599687404800694
                };

                filter_a=new CAllPassFilterCascade(a_coefficients,6);
                filter_b=new CAllPassFilterCascade(b_coefficients,6);
        }
        else if (order==10)     //rejection=133dB, transition band=0.05
        {
```

```
                double a_coefficients[5]=
                {0.02366831419883467
                ,0.18989476227180174
                ,0.43157318062118555
                ,0.6632020224193995
                ,0.860015542499582
                };

                double b_coefficients[5]=
                {0.09056555904993387
                ,0.3078575723749043
                ,0.5516782402507934
                ,0.7652146863779808
                ,0.95247728378667541
                };

                filter_a=new CAllPassFilterCascade(a_coefficients,5);
                filter_b=new CAllPassFilterCascade(b_coefficients,5);
        }
        else if (order==8)      //rejection=106dB, transition band=0.05
        {
                double a_coefficients[4]=
                {0.03583278843106211
                ,0.2720401433964576
                ,0.5720571972357003
                ,0.827124761997324
                };

                double b_coefficients[4]=
                {0.1340901419430669
                ,0.4243248712718685
                ,0.7062921421386394
                ,0.9415030941737551
                };

                filter_a=new CAllPassFilterCascade(a_coefficients,4);
                filter_b=new CAllPassFilterCascade(b_coefficients,4);
        }
        else if (order==6)      //rejection=80dB, transition band=0.05
        {
                double a_coefficients[3]=
                {0.06029739095712437
                ,0.4125907203610563
                ,0.7727156537429234
                };

                double b_coefficients[3]=
                {0.21597144456092948
                ,0.6043586264658363
                ,0.9238861386532906
                };

                filter_a=new CAllPassFilterCascade(a_coefficients,3);
                filter_b=new CAllPassFilterCascade(b_coefficients,3);
        }
```

```
                else if (order==4)      //rejection=70dB,transition band=0.1
                {
                        double a_coefficients[2]=
                        {0.07986642623635751
                        ,0.5453536510711322
                        };

                        double b_coefficients[2]=
                        {0.28382934487410993
                        ,0.8344118914807379
                        };

                        filter_a=new CAllPassFilterCascade(a_coefficients,2);
                        filter_b=new CAllPassFilterCascade(b_coefficients,2);
                }

                else    //order=2, rejection=36dB, transition band=0.1
                {
                        double a_coefficients=0.23647102099689224;
                        double b_coefficients=0.7145421497126001;

                        filter_a=new CAllPassFilterCascade(&a_coefficients,1);
                        filter_b=new CAllPassFilterCascade(&b_coefficients,1);
                }
        }

        oldout=0.0;
};


CHalfBandFilter::~CHalfBandFilter()
{
        delete filter_a;
        delete filter_b;
};


double CHalfBandFilter::process(const double input)
{
        const double output=(filter_a->process(input)+oldout)*0.5;
        oldout=filter_b->process(input);

        return output;
}
```

```
class CAllPassFilter
{
public:

        CAllPassFilter(const double coefficient);
        ~CAllPassFilter();
        double process(double input);

private:
```

```
        double a;

        double x0;
        double x1;
        double x2;

        double y0;
        double y1;
        double y2;
};


class CAllPassFilterCascade
{
public:
        CAllPassFilterCascade(const double* coefficients, int N);
        ~CAllPassFilterCascade();

        double process(double input);

private:
        CAllPassFilter** allpassfilter;
        int numfilters;
};


class CHalfBandFilter
{
public:
        CHalfBandFilter(const int order, const bool steep);
        ~CHalfBandFilter();

        double process(const double input);

private:
        CAllPassFilterCascade* filter_a;
        CAllPassFilterCascade* filter_b;
        double oldout;
};
```

---

pow(x,4) approximation

References : Posted by Stefan Stenzel

Notes :
Very hacked, but it gives a rough estimate of x**4 by modifying exponent and mantissa.

Code :
```
float p4fast(float in)
{
  long *lp,l;

  lp=(long *)(&in);
  l=*lp;

  l-=0x3F800000l; /* un-bias */
  l<<=2;          /* **4 */
  l+=0x3F800000l; /* bias */
  *lp=l;

  /* compiler will read this from memory since & operator had been used */
  return in;
}
```

Prewarping

Type : explanation
References : Posted by robert bristow-johnson (better known as "rbj")

Notes :
prewarping is simply recognizing the warping that the BLT introduces. to determine frequency response, we evaluate the digital H(z) at z=exp(j*w*T) and we evaluate the analog Ha(s) at s=j*W . the following will confirm the jw to unit circle mapping and will show exactly what the mapping is (this is the same stuff in the textbooks):

the BLT says: s = (2/T) * (z-1)/(z+1)

substituting: s = j*W = (2/T) * (exp(j*w*T) - 1) / (exp(j*w*T) + 1)

j*W = (2/T) * (exp(j*w*T/2) - exp(-j*w*T/2)) / (exp(j*w*T/2) + exp(-j*w*T/2))

= (2/T) * (j*2*sin(w*T/2)) / (2*cos(w*T/2))

= j * (2/T) * tan(w*T/2)

or

analog W = (2/T) * tan(w*T/2)

so when the real input frequency is w, the digital filter will behave with the same amplitude gain and phase shift as the analog filter will have at a hypothetical frequency of W. as w*T approaches pi (Nyquist) the digital filter behaves as the analog filter does as W -> inf. for each degree of freedom that you have in your design equations, you can adjust the analog design frequency to be just right so that when the deterministic BLT warping does its thing, the resultant warped frequency comes out just right. for a simple LPF, you have only one degree of freedom, the cutoff frequency. you can precompensate it so that the true cutoff comes out right but that is it, above the cutoff, you will see that the LPF dives down to -inf dB faster than an equivalent analog at the same frequencies.

Pseudo-Random generator

Type : Linear Congruential, 32bit
References : Hal Chamberlain, "Musical Applications of Microprocessors" (Posted by Phil Burk)

Notes :
This can be used to generate random numeric sequences or to synthesise a white noise audio signal.
If you only use some of the bits, use the most significant bits by shifting right.
Do not just mask off the low bits.

Code :

```
/* Calculate pseudo-random 32 bit number based on linear congruential
method. */
unsigned long GenerateRandomNumber( void )
{
   /* Change this for different random sequences. */
   static unsigned long randSeed = 22222;
   randSeed = (randSeed * 196314165) + 907633515;
   return randSeed;
}
```

Pulsewidth modulation

Type : waveform generation
References : Steffan Diedrichsen

Notes :
Take an upramping sawtooth and its inverse, a downramping sawtooth. Adding these two waves with a well defined delay between 0 and period (1/f)
results in a square wave with a duty cycle ranging from 0 to 100%.

RBJ-Audio-EQ-Cookbook

Type : Biquads for all puposes!
References : Robert Bristow-Johnson (a.k.a. RBJ)
Linked file : http://www.harmony-central.com/Computer/Programming/Audio-EQ-Cookbook.txt
Linked file : http://www.harmony-central.com/Effects/Articles/EQ_Coefficients/EQ-Coefficients.pdf

Notes :
(see linkfile)
A superb collection of filters used in a lot of (commercial) plugins and effects.
There's also a very interesting paper linked about biquad EQ filters.

Reading the compressed WA! parts in gigasampler files

References : Paul Kellett
Linked file : gigxpand.zip

Notes :
(see linkfile)
Code to read the .WA! (compressed .WAV) parts of GigaSampler .GIG files.
For related info on reading .GIG files see http://www.linuxdj.com/evo

Resonant filter

References : Posted by Paul Kellett

Notes :
This filter consists of two first order low-pass filters in series, with some of the difference between the two filter outputs fed back to give a resonant peak.

You can use more filter stages for a steeper cutoff but the stability criteria get more complicated if the extra stages are within the feedback loop.

Code :

```
//set feedback amount given f and q between 0 and 1
fb = q + q/(1.0 - f);

//for each sample...
buf0 = buf0 + f * (in - buf0 + fb * (buf0 - buf1));
buf1 = buf1 + f * (buf0 - buf1);
out = buf1;
```

Resonant IIR lowpass (12dB/oct)

Type : Resonant IIR lowpass (12dB/oct)
References : Posted by Olli Niemitalo

Notes :
Hard to calculate coefficients, easy to process algorithm

Code :
```
resofreq = pole frequency
amp = magnitude at pole frequency (approx)

double pi = 3.141592654;

/* Parameters. Change these! */
double resofreq = 5000;
double amp = 1.0;

DOUBLEWORD streamofs;
double w = 2.0*pi*resofreq/samplerate; // Pole angle
double q = 1.0-w/(2.0*(amp+0.5/(1.0+w))+w-2.0); // Pole magnitude
double r = q*q;
double c = r+1.0-2.0*cos(w)*q;
double vibrapos = 0;
double vibraspeed = 0;

/* Main loop */
for (streamofs = 0; streamofs < streamsize; streamofs++) {

  /* Accelerate vibra by signal-vibra, multiplied by lowpasscutoff */
  vibraspeed += (fromstream[streamofs] - vibrapos) * c;

  /* Add velocity to vibra's position */
  vibrapos += vibraspeed;

  /* Attenuate/amplify vibra's velocity by resonance */
  vibraspeed *= r;

  /* Check clipping */
  temp = vibrapos;
  if (temp > 32767) {
    temp = 32767;
  } else if (temp < -32768) temp = -32768;

  /* Store new value */
  tostream[streamofs] = temp;
}
```

Resonant low pass filter

Type : 24dB lowpass
References : Posted by "Zxform"
Linked file : filters004.txt (this linked file is included below)

Linked files

```
// ----------------- file filterIIR00.c begin -----------------
/*
Resonant low pass filter source code.
By baltrax@hotmail.com (Zxform)
*/

#include <stdlib.h>
#include <stdio.h>
#include <math.h>

/*****************************************************************************

FILTER.C - Source code for filter functions

    iir_filter         IIR filter floats sample by sample (real time)

*****************************************************************************/

/* FILTER INFORMATION STRUCTURE FOR FILTER ROUTINES */

typedef struct {
    unsigned int length;       /* size of filter */
    float *history;            /* pointer to history in filter */
    float *coef;               /* pointer to coefficients of filter */
} FILTER;

#define FILTER_SECTIONS   2   /* 2 filter sections for 24 db/oct filter */

typedef struct {
        double a0, a1, a2;       /* numerator coefficients */
        double b0, b1, b2;       /* denominator coefficients */
} BIQUAD;

BIQUAD ProtoCoef[FILTER_SECTIONS];      /* Filter prototype coefficients,
                                                    1 for each filter section
*/

void szxform(
    double *a0, double *a1, double *a2,     /* numerator coefficients */
    double *b0, double *b1, double *b2,   /* denominator coefficients */
    double fc,           /* Filter cutoff frequency */
    double fs,           /* sampling rate */
    double *k,           /* overall gain factor */
```

```
    float *coef);         /* pointer to 4 iir coefficients */

/*
 * ---------------------------------------------------------------------
 *
 * iir_filter - Perform IIR filtering sample by sample on floats
 *
 * Implements cascaded direct form II second order sections.
 * Requires FILTER structure for history and coefficients.
 * The length in the filter structure specifies the number of sections.
 * The size of the history array is 2*iir->length.
 * The size of the coefficient array is 4*iir->length + 1 because
 * the first coefficient is the overall scale factor for the filter.
 * Returns one output sample for each input sample.  Allocates history
 * array if not previously allocated.
 *
 * float iir_filter(float input,FILTER *iir)
 *
 *     float input        new float input sample
 *     FILTER *iir        pointer to FILTER structure
 *
 * Returns float value giving the current output.
 *
 * Allocation errors cause an error message and a call to exit.
 * ---------------------------------------------------------------------
 */
float iir_filter(input,iir)
    float input;        /* new input sample */
    FILTER *iir;        /* pointer to FILTER structure */
{
    unsigned int i;
    float *hist1_ptr,*hist2_ptr,*coef_ptr;
    float output,new_hist,history1,history2;

/* allocate history array if different size than last call */

    if(!iir->history) {
        iir->history = (float *) calloc(2*iir->length,sizeof(float));
        if(!iir->history) {
            printf("\nUnable to allocate history array in iir_filter\n");
            exit(1);
        }
    }

    coef_ptr = iir->coef;               /* coefficient pointer */

    hist1_ptr = iir->history;           /* first history */
    hist2_ptr = hist1_ptr + 1;          /* next history */

        /* 1st number of coefficients array is overall input scale factor,
         * or filter gain */
```

```
    output = input * (*coef_ptr++);

    for (i = 0 ; i < iir->length; i++)
        {
        history1 = *hist1_ptr;          /* history values */
        history2 = *hist2_ptr;

        output = output - history1 * (*coef_ptr++);
        new_hist = output - history2 * (*coef_ptr++);    /* poles */

        output = new_hist + history1 * (*coef_ptr++);
        output = output + history2 * (*coef_ptr++);      /* zeros */

        *hist2_ptr++ = *hist1_ptr;
        *hist1_ptr++ = new_hist;
        hist1_ptr++;
        hist2_ptr++;
    }

    return(output);
}


/*
 * ------------------------------------------------------------------
 *
 * main()
 *
 * Example main function to show how to update filter coefficients.
 * We create a 4th order filter (24 db/oct roloff), consisting
 * of two second order sections.
 * ------------------------------------------------------------------
 */
int main()
{
        FILTER   iir;
        float    *coef;
        double   fs, fc;     /* Sampling frequency, cutoff frequency */
        double   Q;     /* Resonance > 1.0 < 1000 */
        unsigned nInd;
        double   a0, a1, a2, b0, b1, b2;
        double   k;           /* overall gain factor */

/*
 * Setup filter s-domain coefficients
 */
                /* Section 1 */
        ProtoCoef[0].a0 = 1.0;
        ProtoCoef[0].a1 = 0;
        ProtoCoef[0].a2 = 0;
        ProtoCoef[0].b0 = 1.0;
```

```
        ProtoCoef[0].b1 = 0.765367;
        ProtoCoef[0].b2 = 1.0;

                /* Section 2 */
        ProtoCoef[1].a0 = 1.0;
        ProtoCoef[1].a1 = 0;
        ProtoCoef[1].a2 = 0;
        ProtoCoef[1].b0 = 1.0;
        ProtoCoef[1].b1 = 1.847759;
        ProtoCoef[1].b2 = 1.0;

        iir.length = FILTER_SECTIONS;         /* Number of filter sections */

/*
 * Allocate array of z-domain coefficients for each filter section
 * plus filter gain variable
 */
        iir.coef = (float *) calloc(4 * iir.length + 1, sizeof(float));
        if (!iir.coef)
        {
                printf("Unable to allocate coef array, exiting\n");
                exit(1);
        }

        k = 1.0;          /* Set overall filter gain */
        coef = iir.coef + 1;     /* Skip k, or gain */

        Q = 1;                          /* Resonance */
        fc = 5000;                  /* Filter cutoff (Hz) */
        fs = 44100;                       /* Sampling frequency (Hz) */

/*
 * Compute z-domain coefficients for each biquad section
 * for new Cutoff Frequency and Resonance
 */
        for (nInd = 0; nInd < iir.length; nInd++)
        {
                a0 = ProtoCoef[nInd].a0;
                a1 = ProtoCoef[nInd].a1;
                a2 = ProtoCoef[nInd].a2;

                b0 = ProtoCoef[nInd].b0;
                b1 = ProtoCoef[nInd].b1 / Q;      /* Divide by resonance or Q
*/
                b2 = ProtoCoef[nInd].b2;
                szxform(&a0, &a1, &a2, &b0, &b1, &b2, fc, fs, &k, coef);
                coef += 4;                        /* Point to next filter
section */
        }

        /* Update overall filter gain in coef array */
```

```
        iir.coef[0] = k;

        /* Display filter coefficients */
        for (nInd = 0; nInd < (iir.length * 4 + 1); nInd++)
                printf("C[%d] = %15.10f\n", nInd, iir.coef[nInd]);
/*
 * To process audio samples, call function iir_filter()
 * for each audio sample
 */
        return (0);
}



// ----------------- file filterIIR00.c end -----------------
>

Reposting bilinear.c just in case the other one was not the latest version.

// ----------------- file bilinear.c begin -----------------
/*
 * ------------------------------------------------------------
 *      bilinear.c
 *
 *      Perform bilinear transformation on s-domain coefficients
 *      of 2nd order biquad section.
 *      First design an analog filter and use s-domain coefficients
 *      as input to szxform() to convert them to z-domain.
 *
 * Here's the butterworth polinomials for 2nd, 4th and 6th order sections.
 *      When we construct a 24 db/oct filter, we take to 2nd order
 *      sections and compute the coefficients separately for each section.
 *
 *      n       Polinomials
 * ------------------------------------------------------------------------
 *      2       s^2 + 1.4142s +1
 *      4       (s^2 + 0.765367s + 1) (s^2 + 1.847759s + 1)
 *      6       (s^2 + 0.5176387s + 1) (s^2 + 1.414214 + 1) (s^2 + 1.931852s +
1)
 *
 *      Where n is a filter order.
 *      For n=4, or two second order sections, we have following equasions for
each
 *      2nd order stage:
 *
 *      (1 / (s^2 + (1/Q) * 0.765367s + 1)) * (1 / (s^2 + (1/Q) * 1.847759s +
1))
 *
 *      Where Q is filter quality factor in the range of
 *      1 to 1000. The overall filter Q is a product of all
```

```
 *      2nd order stages. For example, the 6th order filter
 *      (3 stages, or biquads) with individual Q of 2 will
 *      have filter Q = 2 * 2 * 2 = 8.
 *
 *      The nominator part is just 1.
 *      The denominator coefficients for stage 1 of filter are:
 *      b2 = 1; b1 = 0.765367; b0 = 1;
 *      numerator is
 *      a2 = 0; a1 = 0; a0 = 1;
 *
 *      The denominator coefficients for stage 1 of filter are:
 *      b2 = 1; b1 = 1.847759; b0 = 1;
 *      numerator is
 *      a2 = 0; a1 = 0; a0 = 1;
 *
 *      These coefficients are used directly by the szxform()
 *      and bilinear() functions. For all stages the numerator
 *      is the same and the only thing that is different between
 *      different stages is 1st order coefficient. The rest of
 *      coefficients are the same for any stage and equal to 1.
 *
 *      Any filter could be constructed using this approach.
 *
 *      References:
 *             Van Valkenburg, "Analog Filter Design"
 *             Oxford University Press 1982
 *             ISBN 0-19-510734-9
 *
 *             C Language Algorithms for Digital Signal Processing
 *             Paul Embree, Bruce Kimble
 *             Prentice Hall, 1991
 *             ISBN 0-13-133406-9
 *
 *             Digital Filter Designer's Handbook
 *             With C++ Algorithms
 *             Britton Rorabaugh
 *             McGraw Hill, 1997
 *             ISBN 0-07-053806-9
 * ----------------------------------------------------------
 */

#include <math.h>

void prewarp(double *a0, double *a1, double *a2, double fc, double fs);
void bilinear(
    double a0, double a1, double a2,    /* numerator coefficients */
    double b0, double b1, double b2,    /* denominator coefficients */
    double *k,                          /* overall gain factor */
    double fs,                          /* sampling rate */
    float *coef);                       /* pointer to 4 iir coefficients */
```

```
/*
 * ------------------------------------------------------------
 *      Pre-warp the coefficients of a numerator or denominator.
 *      Note that a0 is assumed to be 1, so there is no wrapping
 *      of it.
 * ------------------------------------------------------------
 */
void prewarp(
    double *a0, double *a1, double *a2,
    double fc, double fs)
{
    double wp, pi;

    pi = 4.0 * atan(1.0);
    wp = 2.0 * fs * tan(pi * fc / fs);

    *a2 = (*a2) / (wp * wp);
    *a1 = (*a1) / wp;
}


/*
 * ------------------------------------------------------------
 * bilinear()
 *
 * Transform the numerator and denominator coefficients
 * of s-domain biquad section into corresponding
 * z-domain coefficients.
 *
 *      Store the 4 IIR coefficients in array pointed by coef
 *      in following order:
 *             beta1, beta2    (denominator)
 *             alpha1, alpha2  (numerator)
 *
 * Arguments:
 *             a0-a2   - s-domain numerator coefficients
 *             b0-b2   - s-domain denominator coefficients
 *             k               - filter gain factor. initially set to 1
 *                               and modified by each biquad section in such
 *                               a way, as to make it the coefficient by
 *                               which to multiply the overall filter gain
 *                               in order to achieve a desired overall filter
gain,
 *                               specified in initial value of k.
 *             fs             - sampling rate (Hz)
 *             coef    - array of z-domain coefficients to be filled in.
 *
 * Return:
 *             On return, set coef z-domain coefficients
```

```
 * ------------------------------------------------------------
 */
void bilinear(
    double a0, double a1, double a2,    /* numerator coefficients */
    double b0, double b1, double b2,    /* denominator coefficients */
    double *k,           /* overall gain factor */
    double fs,           /* sampling rate */
    float *coef         /* pointer to 4 iir coefficients */
)
{
    double ad, bd;

                 /* alpha (Numerator in s-domain) */
    ad = 4. * a2 * fs * fs + 2. * a1 * fs + a0;
                 /* beta (Denominator in s-domain) */
    bd = 4. * b2 * fs * fs + 2. * b1* fs + b0;

                 /* update gain constant for this section */
    *k *= ad/bd;

                 /* Denominator */
    *coef++ = (2. * b0 - 8. * b2 * fs * fs)
                           / bd;         /* beta1 */
    *coef++ = (4. * b2 * fs * fs - 2. * b1 * fs + b0)
                           / bd; /* beta2 */

                 /* Nominator */
    *coef++ = (2. * a0 - 8. * a2 * fs * fs)
                           / ad;         /* alpha1 */
    *coef = (4. * a2 * fs * fs - 2. * a1 * fs + a0)
                           / ad;   /* alpha2 */
}


/*
 * ------------------------------------------------------------
 * Transform from s to z domain using bilinear transform
 * with prewarp.
 *
 * Arguments:
 *      For argument description look at bilinear()
 *
 *      coef - pointer to array of floating point coefficients,
 *                     corresponding to output of bilinear transofrm
 *                     (z domain).
 *
 * Note: frequencies are in Hz.
 * ------------------------------------------------------------
 */
void szxform(
    double *a0, double *a1, double *a2, /* numerator coefficients */
```

```
    double *b0, double *b1, double *b2, /* denominator coefficients */
    double fc,         /* Filter cutoff frequency */
    double fs,         /* sampling rate */
    double *k,         /* overall gain factor */
    float *coef)         /* pointer to 4 iir coefficients */
{
                /* Calculate a1 and a2 and overwrite the original values */
        prewarp(a0, a1, a2, fc, fs);
        prewarp(b0, b1, b2, fc, fs);
        bilinear(*a0, *a1, *a2, *b0, *b1, *b2, k, fs, coef);
}


// ----------------- file bilinear.c end -----------------
```

And here is how it all works.

```
// ----------------- file filter.txt begin -----------------
How to construct a kewl low pass resonant filter?

Lets assume we want to create a filter for analog synth.
The filter rolloff is 24 db/oct, which corresponds to 4th
order filter. Filter of first order is equivalent to RC circuit
and has max rolloff of 6 db/oct.

We will use classical Butterworth IIR filter design, as it
exactly corresponds to our requirements.

A common practice is to chain several 2nd order sections,
or biquads, as they commonly called, in order to achive a higher
order filter. Each 2nd order section is a 2nd order filter, which
has 12 db/oct roloff. So, we need 2 of those sections in series.

To compute those sections, we use standard Butterworth polinomials,
or so called s-domain representation and convert it into z-domain,
or digital domain. The reason we need to do this is because
the filter theory exists for analog filters for a long time
and there exist no theory of working in digital domain directly.
So the common practice is to take standard analog filter design
and use so called bilinear transform to convert the butterworth
equasion coefficients into z-domain.

Once we compute the z-domain coefficients, we can use them in
a very simple transfer function, such as iir_filter() in our
C source code, in order to perform the filtering function.
The filter itself is the simpliest thing in the world.
The most complicated thing is computing the coefficients
for z-domain.

Ok, lets look at butterworth polynomials, arranged as a series
of 2nd order sections:
```

```
 * Note: n is filter order.
 *
 *      n       Polynomials
 *      ---------------------------------------------------------------------
 *      2       s^2 + 1.4142s +1
 *      4       (s^2 + 0.765367s + 1) * (s^2 + 1.847759s + 1)
 *      6       (s^2 + 0.5176387s + 1) * (s^2 + 1.414214 + 1) * (s^2 +
1.931852s + 1)
 *
 * For n=4 we have following equasion for the filter transfer function:
 *
 *                     1                              1
 * T(s) = -------------------------- * ----------------------------
 *        s^2 + (1/Q) * 0.765367s + 1   s^2 + (1/Q) * 1.847759s + 1
 *
```

The filter consists of two 2nd order secions since highest s power is 2.
Now we can take the coefficients, or the numbers by which s is multiplied
and plug them into a standard formula to be used by bilinear transform.

Our standard form for each 2nd order secion is:

```
       a2 * s^2 + a1 * s + a0
H(s) = ----------------------
       b2 * s^2 + b1 * s + b0
```

Note that butterworth nominator is 1 for all filter sections,
which means s^2 = 0 and s^1 = 0

Lets convert standard butterworth polinomials into this form:

```
       0 + 0 + 1                  0 + 0 + 1
--€---------------------- * --------------------------
1 + ((1/Q) * 0.765367) + 1   1 + ((1/Q) * 1.847759) + 1
```

Section 1:
a2 = 0; a1 = 0; a0 = 1;
b2 = 1; b1 = 0.5176387; b0 = 1;

Section 2:
a2 = 0; a1 = 0; a0 = 1;
b2 = 1; b1 = 1.847759; b0 = 1;

That Q is filter quality factor or resonance, in the range of
1 to 1000. The overall filter Q is a product of all 2nd order stages.
For example, the 6th order filter (3 stages, or biquads)
with individual Q of 2 will have filter Q = 2 * 2 * 2 = 8.

These a and b coefficients are used directly by the szxform()
and bilinear() functions.

The transfer function for z-domain is:

```
       1 + alpha1 * z^(-1) + alpha2 * z^(-2)
H(z) = -------------------------------------
       1 + beta1 * z^(-1) + beta2 * z^(-2)
```

When you need to change the filter frequency cutoff or resonance, or Q, you call the szxform() function with proper a and b coefficients and the new filter cutoff frequency or resonance. You also need to supply the sampling rate and filter gain you want to achive. For our purposes the gain = 1.

We call szxform() function 2 times becase we have 2 filter sections. Each call provides different coefficients.

The gain argument to szxform() is a pointer to desired filter gain variable.

```
double k = 1.0;      /* overall gain factor */
```

Upon return from each call, the k argument will be set to a value, by which to multiply our actual signal in order for the gain to be one. On second call to szxform() we provide k that was changed by the previous section. During actual audio filtering function iir_filter() will use this k

Summary:

Our filter is pretty close to ideal in terms of all relevant parameters and filter stability even with extremely large values of resonance. This filter design has been verified under all variations of parameters and it all appears to work as advertized.

Good luck with it.
If you ever make a directX wrapper for it, post it to comp.dsp.

```
 *
 * ------------------------------------------------------------
 *References:
 *Van Valkenburg, "Analog Filter Design"
 *Oxford University Press 1982
 *ISBN 0-19-510734-9
 *
 *C Language Algorithms for Digital Signal Processing
 *Paul Embree, Bruce Kimble
 *Prentice Hall, 1991
 *ISBN 0-13-133406-9
 *
 *Digital Filter Designer's Handbook
 *With C++ Algorithms
 *Britton Rorabaugh
 *McGraw Hill, 1997
```

```
*ISBN 0-07-053806-9
* ------------------------------------------------------------
```

Reverberation techniques

References : Posted by Sean Costello

Notes :
* Parallel comb filters, followed by series allpass filters. This was the original design by Schroeder, and was extended by Moorer. Has a VERY metallic sound for sharp transients.

* Several allpass filters in serie (also proposed by Schroeder). Also suffers from metallic sound.

* 2nd-order comb and allpass filters (described by Moorer). Not supposed to give much of an advantage over first order sections.

* Nested allpass filters, where an allpass filter will replace the delay line in another allpass filter. Pioneered by Gardner. Haven't heard the results.

* Strange allpass amp delay line based structure in Jon Dattorro article (JAES). Four allpass filters are used as an input to a cool "figure-8" feedback loop, where four allpass reverberators are used in series with
a few delay lines. Outputs derived from various taps in structure. Supposedly based on a Lexicon reverb design. Modulating delay lines are used in some of the allpass structures to "spread out" the eigentones.

* Feedback Delay Networks. Pioneered by Puckette/Stautner, with Jot conducting extensive recent research. Sound VERY good, based on initial experiments. Modulating delay lines and feedback matrixes used to spread out eigentones.

* Waveguide-based reverbs, where the reverb structure is based upon the junction of many waveguides. Julius Smith developed these. Recently, these have been shown to be essentially equivalent to the feedback delay network reverbs. Also sound very nice. Modulating delay lines and scattering values used to spread out eigentones.

* Convolution-based reverbs, where the sound to be reverbed is convolved with the impulse response of a room, or with exponentially-decaying white noise. Supposedly the best sound, but very computationally expensive, and not very flexible.

* FIR-based reverbs. Essentially the same as convolution. Probably not used, but shorter FIR filters are probably used in combination with many of the above techniques, to provide early reflections.

Saturation

Type : Waveshaper
References : Posted by Bram

Notes :
when the input is below a certain threshold (t) these functions return the input, if it goes over that threshold, they return a soft shaped saturation.
Neigther claims to be fast ;-)

Code :

```
float saturate(float x, float t)
{
    if(fabs(x)<t)
        return x
    else
    {
        if(x > 0.f);
            return t + (1.f-t)*tanh((x-t)/(1-t));
        else
            return -(t + (1.f-t)*tanh((-x-t)/(1-t)));
}
}

or

float sigmoid(x)
{
    if(fabs(x)<1)
        return x*(1.5f - 0.5f*x*x);
    else
        return x > 0.f ? 1.f : -1.f;
}

float saturate(float x, float t)
{
    if(abs(x)<t)
        return x
    else
    {
        if(x > 0.f);
            return t + (1.f-t)*sigmoid((x-t)/((1-t)*1.5f));
        else
            return -(t + (1.f-t)*sigmoid((-x-t)/((1-t)*1.5f)));
    }
}
```

SawSin

Type : Oscillator shape
References : Posted by Alexander Kritov
Code :

```
double sawsin(double x)
{
   double t = fmod(x/(2*M_PI),(double)1.0);
   if (t>0.5)
       return -sin(x);
   if (t<=0.5)
       return (double)2.0*t-1.0;
}
```

Simple peak follower

Type : amplitude analysis
References : Posted by Phil Burk

Notes :
This simple peak follower will give track the peaks of a signal. It will rise rapidly when the input is rising, and then decay exponentially when the input drops. It can be used to drive VU meters, or used in an automatic gain control circuit.

Code :

```
// halfLife = time in seconds for output to decay to half value after an
impulse

static float output = 0.0;

float scalar = pow( 0.5, 1.0/(halfLife * sampleRate)));

if( input < 0.0 )
  input = -input;  /* Absolute value. */

if ( input >= output )
{
   /* When we hit a peak, ride the peak to the top. */
   output = input;
}
else
{
   /* Exponential decay of output when signal is low. */
   output = output * scalar;
   /*
   ** When current gets close to 0.0, set current to 0.0 to prevent FP
underflow
   ** which can cause a severe performance degradation due to a flood
   ** of interrupts.
   */
   if( output < VERY_SMALL_FLOAT ) output = 0.0;
}
```

Sin, Cos, Tan approximation

References : Posted by jan[AT]rpgfan[DOT]demon[DOT]co[DOT]uk
Linked file : approx.h (this linked file is included below)

Notes :
Code for approximation of cos, sin, tan and inv sin, etc.
Surprisingly accurate and very usable.

Code :

Linked files

```
//------------------------------------------------------------------------
Real Math::FastSin0 (Real fAngle)
{
    Real fASqr = fAngle*fAngle;
    Real fResult = 7.61e-03f;
    fResult *= fASqr;
    fResult -= 1.6605e-01f;
    fResult *= fASqr;
    fResult += 1.0f;
    fResult *= fAngle;
    return fResult;
}
//------------------------------------------------------------------------
Real Math::FastSin1 (Real fAngle)
{
    Real fASqr = fAngle*fAngle;
    Real fResult = -2.39e-08f;
    fResult *= fASqr;
    fResult += 2.7526e-06f;
    fResult *= fASqr;
    fResult -= 1.98409e-04f;
    fResult *= fASqr;
    fResult += 8.3333315e-03f;
    fResult *= fASqr;
    fResult -= 1.666666664e-01f;
    fResult *= fASqr;
    fResult += 1.0f;
    fResult *= fAngle;
    return fResult;
}
//------------------------------------------------------------------------
Real Math::FastCos0 (Real fAngle)
```

```
{
    Real fASqr = fAngle*fAngle;
    Real fResult = 3.705e-02f;
    fResult *= fASqr;
    fResult -= 4.967e-01f;
    fResult *= fASqr;
    fResult += 1.0f;
    return fResult;
}
//----------------------------------------------------------------------
Real Math::FastCos1 (Real fAngle)
{
    Real fASqr = fAngle*fAngle;
    Real fResult = -2.605e-07f;
    fResult *= fASqr;
    fResult += 2.47609e-05f;
    fResult *= fASqr;
    fResult -= 1.3888397e-03f;
    fResult *= fASqr;
    fResult += 4.16666418e-02f;
    fResult *= fASqr;
    fResult -= 4.999999963e-01f;
    fResult *= fASqr;
    fResult += 1.0f;
    return fResult;
}
//----------------------------------------------------------------------
Real Math::FastTan0 (Real fAngle)
{
    Real fASqr = fAngle*fAngle;
    Real fResult = 2.033e-01f;
    fResult *= fASqr;
    fResult += 3.1755e-01f;
    fResult *= fASqr;
    fResult += 1.0f;
    fResult *= fAngle;
    return fResult;
}
//----------------------------------------------------------------------
Real Math::FastTan1 (Real fAngle)
{
    Real fASqr = fAngle*fAngle;
    Real fResult = 9.5168091e-03f;
    fResult *= fASqr;
    fResult += 2.900525e-03f;
    fResult *= fASqr;
```

```
        fResult += 2.45650893e-02f;
        fResult *= fASqr;
        fResult += 5.33740603e-02f;
        fResult *= fASqr;
        fResult += 1.333923995e-01f;
        fResult *= fASqr;
        fResult += 3.333314036e-01f;
        fResult *= fASqr;
        fResult += 1.0f;
        fResult *= fAngle;
        return fResult;
    }
    //-----------------------------------------------------------------------
    Real Math::FastInvSin (Real fValue)
    {
        Real fRoot = Math::Sqrt(1.0f-fValue);
        Real fResult = -0.0187293f;
        fResult *= fValue;
        fResult += 0.0742610f;
        fResult *= fValue;
        fResult -= 0.2121144f;
        fResult *= fValue;
        fResult += 1.5707288f;
        fResult = HALF_PI - fRoot*fResult;
        return fResult;
    }
    //-----------------------------------------------------------------------
    Real Math::FastInvCos (Real fValue)
    {
        Real fRoot = Math::Sqrt(1.0f-fValue);
        Real fResult = -0.0187293f;
        fResult *= fValue;
        fResult += 0.0742610f;
        fResult *= fValue;
        fResult -= 0.2121144f;
        fResult *= fValue;
        fResult += 1.5707288f;
        fResult *= fRoot;
        return fResult;
    }
    //-----------------------------------------------------------------------
    Real Math::FastInvTan0 (Real fValue)
    {
        Real fVSqr = fValue*fValue;
        Real fResult = 0.0208351f;
        fResult *= fVSqr;
```

```
    fResult -= 0.085133f;
    fResult *= fVSqr;
    fResult += 0.180141f;
    fResult *= fVSqr;
    fResult -= 0.3302995f;
    fResult *= fVSqr;
    fResult += 0.999866f;
    fResult *= fValue;
    return fResult;
}
//----------------------------------------------------------------------
Real Math::FastInvTan1 (Real fValue)
{
    Real fVSqr = fValue*fValue;
    Real fResult = 0.0028662257f;
    fResult *= fVSqr;
    fResult -= 0.0161657367f;
    fResult *= fVSqr;
    fResult += 0.0429096138f;
    fResult *= fVSqr;
    fResult -= 0.0752896400f;
    fResult *= fVSqr;
    fResult += 0.1065626393f;
    fResult *= fVSqr;
    fResult -= 0.1420889944f;
    fResult *= fVSqr;
    fResult += 0.1999355085f;
    fResult *= fVSqr;
    fResult -= 0.3333314528f;
    fResult *= fVSqr;
    fResult += 1.0f;
    fResult *= fValue;
    return fResult;
}
//----------------------------------------------------------------------
```

Sine calculation

Type : waveform generation, Taylor approximation of sin()
References : Posted by Phil Burk

Notes :
Code from JSyn for a sine wave generator based on a Taylor Expansion. It is not as efficient as the filter methods, but it has linear frequency control and is, therefore, suitable for FM or other time varying applications where accurate frequency is needed. The sine generated is accurate to at least 16 bits.

Code :
```
for(i=0; i < nSamples ; i++)
{
  //Generate sawtooth phasor to provide phase for sine generation
  IncrementWrapPhase(phase, freqPtr[i]);
  //Wrap phase back into region where results are more accurate

  if(phase > 0.5)
    yp = 1.0 - phase;
  else
  {
    if(phase < -0.5)
       yp = -1.0 - phase;
    else
        yp = phase;
  }

  x = yp * PI;
  x2 = x*x;

  //Taylor expansion out to x**9/9! factored  into multiply-adds
  fastsin = x*(x2*(x2*(x2*(x2*(1.0/362880.0)
            - (1.0/5040.0))
            + (1.0/120.0))
            - (1.0/6.0))
            + 1.0);

  outPtr[i] = fastsin * amplPtr[i];
}
```

smsPitchScale Source Code

Type : Pitch Scaling (often incorrectly referred to as "Pitch Shifting") using the Fourier transform
References : Posted by sms[AT]dspdimension.com
Linked file : http://www.dspdimension.com
Code :

```
See above web site
```

Soft saturation

Type : waveshaper
References : Posted by Bram de Jong

Notes :
This only works for positive values of x. a should be in the range 0..1

Code :

```
x < a:
  f(x) = x
x > a:
  f(x) = a + (x-a)/(1+((x-a)/(1-a))^2)
x > 1:
  f(x) = (a+1)/2
```

Square Waves

Type : waveform generation
References : Posted by Sean Costello

Notes :
One way to do a square wave:

You need two buzz generators (see Dodge & Jerse, or the Csound source code, for implementation details). One of the buzz generators runs at the desired square wave frequency, while the second buzz generator is exactly one octave above this pitch. Subtract the higher octave buzz generator's output from the lower buzz generator's output - the result should be a signal with all odd harmonics, all at equal amplitude. Filter the resultant signal (maybe integrate it). Voila, a bandlimited square wave! Well, I think it should work...

The one question I have with the above technique is whether it produces a waveform that truly resembles a square wave in the time domain. Even if the number of harmonics, and the relative ratio of the harmonics, is identical to an "ideal" bandwidth-limited square wave, it may have an entirely different waveshape. No big deal, unless the signal is processed by a nonlinearity, in which case the results of the nonlinear processing will be far different than the processing of a waveform that has a similar shape to a square wave.

State variable

Type : 12db resonant low, high or bandpass
References : Effect Deisgn Part 1, Jon Dattorro, J. Audio Eng. Soc., Vol 45, No. 9, 1997 September

Notes :
Digital approximation of Chamberlin two-pole low pass. Easy to calculate coefficients, easy to process algorithm.

Code :
```
cutoff = cutoff freq in Hz
fs = sampling frequency //(e.g. 44100Hz)
f = 2 sin (pi * cutoff / fs) //[approximately]
q = resonance/bandwidth [0 < q <= 1]  most res: q=1, less: q=0
low = lowpass output
high = highpass output
band = bandpass output
notch = notch output

scale = q

low=high=band=0;

//--beginloop
low = low + f * band;
high = scale * input - low - q*band;
band = f * high + band;
notch = high + low;
//--endloop
```

State Variable Filter (Double Sampled, Stable)

Type : 2 Pole Low, High, Band, Notch and Peaking
References : Posted by Andrew Simper

Notes :
Thanks to Laurent de Soras for the stability limit
and Steffan Diedrichsen for the correct notch output.

Code :

```
input  = input buffer;
output = output buffer;
fs     = sampling frequency;
fc     = cutoff frequency normally something like:
         440.0*pow(2.0, (midi_note - 69.0)/12.0);
res    = resonance 0 to 1;
drive  = internal distortion 0 to 0.1
freq   = MIN(0.25, 2.0*sin(PI*fc/(fs*2)));  // the fs*2 is because it's
double sampled
damp   = MIN(2.0*(1.0 - pow(res, 0.25)), MIN(2.0, 2.0/freq - freq*0.5));
notch  = notch output
low    = low pass output
high   = high pass output
band   = band pass output
peak   = peaking output = low - high
--
double sampled svf loop:
for (i=0; i<numSamples; i++)
{
  in    = input[i];
  notch = in - damp*band;
  low   = low + freq*band;
  high  = notch - low;
  band  = freq*high + band - drive*band*band*band;
  out   = 0.5*(notch or low or high or band or peak);
  notch = in - damp*band;
  low   = low + freq*band;
  high  = notch - low;
  band  = freq*high + band - drive*band*band*band;
  out  += 0.5*(same out as above);
  output[i] = out;
}
```

Time domain convolution with O(n^log2(3))

References : Wilfried Welti

Notes :
[Quoted from Wilfrieds mail...]
I found last weekend that it is possible to do convolution in time domain (no complex numbers, 100% exact result with int) with O(n^log2(3)) (about O(n^1.58)).

Due to smaller overhead compared to FFT-based convolution, it should be the fastest algorithm for medium sized FIR's.
Though, it's slower as FFT-based convolution for large n.

It's pretty easy:

Let's say we have two finite signals of length 2n, which we want convolve : A and B. Now we split both signals into parts of size n, so we get A = A1 + A2, and B = B1 +B2.

Now we can write:

(1) A*B = (A1+A2)*(B1+B2) = A1*B1 + A2*B1 + A1*B2 + A2*B2

where * means convolution.

This we knew already: We can split a convolution into four convolutions of halved size.

Things become interesting when we start shifting blocks in time:

Be z a signal which has the value 1 at x=1 and zero elsewhere. Convoluting a signal X with z is equivalent to shifting X by one rightwards. When I define z^n as n-fold convolution of z with itself, like: z^1 = z, z^2 = z*z, z^0 = z shifted leftwards by 1 = impulse at x=0, and so on, I can use it to shift signals:

X * z^n means shifting the signal X by the value n rightwards.
X * z^-n means shifting the signal X by the value n leftwards.

Now we look at the following term:

(2) (A1 + A2 * z^-n) * (B1 + B2 * z^-n)

This is a convolution of two blocks of size n: We shift A2 by n leftwards so it completely overlaps A1, then we add them.
We do the same thing with B1 and B2. Then we convolute the two resulting blocks.

now let's transform this term:

(3) (A1 + A2 * z^-n) * (B1 + B2 * z^-n)

= A1*B1 + A1*B2*z^-n + A2*z^-n*B1 + A2*z^ n*B2*z^-n
= A1*B1 + (A1*B2 + A2*B1)*z^-n + A2*B2*z^-2n

(4) (A1 + A2 * z^-n) * (B1 + B2 * z^-n) - A1*B1 - A2*B2*z^-2n

= (A1*B2 + A2*B1)*z^-n

Now we convolute both sides of the equation (4) by z^n:

(5) (A1 + A2 * z^-n)*(B1 + B2 * z^-n)*z^n - A1*B1*z^n - A2*B2*z^-n

= (A1*B2 + A2*B1)

Now we see that the right part of equation (5) appears within equation (1), so we can replace this appearance by the left part of eq (5).

(6) A*B = (A1+A2)*(B1+B2) = A1*B1 + A2*B1 + A1*B2 + A2*B2

= A1*B1
+ (A1 + A2 * z^-n)*(B1 + B2 * z^-n)*z^n - A1*B1*z^n - A2*B2*z^-n
+ A2*B2

Voila!

We have constructed the convolution of A*B with only three convolutions of halved size. (Since the convolutions with z^n and z^-n are only shifts
of blocks with size n, they of course need only n operations for processing :)

This can be used to construct an easy recursive algorithm of Order O(n^log2(3))

Code :

```
void convolution(value* in1, value* in2, value* out, value* buffer, int
size)
{
  value* temp1 = buffer;
  value* temp2 = buffer + size/2;
  int i;

  // clear output.
  for (i=0; i<size*2; i++) out[i] = 0;

  // Break condition for recursion: 1x1 convolution is multiplication.
```

```
  if (size == 1)
  {
    out[0] = in1[0] * in2[0];
    return;
  }

  // first calculate (A1 + A2 * z^-n)*(B1 + B2 * z^-n)*z^n

  signal_add(in1, in1+size/2, temp1, size/2);
  signal_add(in2, in2+size/2, temp2, size/2);
  convolution(temp1, temp2, out+size/2, buffer+size, size/2);

  // then add A1*B1 and substract A1*B1*z^n

  convolution(in1, in2, temp1, buffer+size, size/2);
  signal_add_to(out, temp1, size);
  signal_sub_from(out+size/2, temp1, size);

  // then add A2*B2 and substract A2*B2*z^-n

  convolution(in1+size/2, in2+size/2, temp1, buffer+size, size/2);
  signal_add_to(out+size, temp1, size);
  signal_sub_from(out+size/2, temp1, size);
}

"value" may be a suitable type like int or float.
Parameter "size" is the size of the input signals and must be a power of 2.
out and buffer must point to arrays of size 2*n.

Just to be complete, the helper functions:

void signal_add(value* in1, value* in2, value* out, int size)
{
  int i;
  for (i=0; i<size; i++) out[i] = in1[i] + in2[i];
}

void signal_sub_from(value* out, value* in, int size)
{
  int i;
  for (i=0; i<size; i++) out[i] -= in[i];
}

void signal_add_to(value* out, value* in, int size)
{
  int i;
  for (i=0; i<size; i++) out[i] += in[i];
}
```

Time domain convolution with O(n^log2(3))

References : Posted by Magnus Jonsson

Notes :
[see other code by Wilfried Welti too!]

Code :

```
void mul_brute(float *r, float *a, float *b, int w)
{
    for (int i = 0; i < w+w; i++)
        r[i] = 0;
    for (int i = 0; i < w; i++)
    {
        float *rr = r+i;
        float ai = a[i];
        for (int j = 0; j < w; j++)
            rr[j] += ai*b[j];
    }
}

// tmp must be of length 2*w
void mul_knuth(float *r, float *a, float *b, int w, float *tmp)
{
    if (w < 30)
    {
        mul_brute(r, a, b, w);
    }
    else
    {
        int m = w>>1;

        for (int i = 0; i < m; i++)
        {
            r[i  ] = a[m+i]-a[i  ];
            r[i+m] = b[i  ]-b[m+i];
        }

        mul_knuth(tmp, r  , r+m, m, tmp+w);
        mul_knuth(r  , a  , b  , m, tmp+w);
        mul_knuth(r+w, a+m, b+m, m, tmp+w);

        for (int i = 0; i < m; i++)
        {
            float bla = r[m+i]+r[w+i];
            r[m+i] = bla+r[i    ]+tmp[i  ];
            r[w+i] = bla+r[w+m+i]+tmp[i+m];
```

```
            }
        }
    }
```

tone detection with Goertzel

Type : Goertzel
References : Posted by espenr@ii.uib.no
Linked file : http://www.ii.uib.no/~espenr/tonedetect.zip

Notes :
Goertzel is basically DFT of parts of a spectrum not the total spectrum as you normally do with FFT. So if you just want to check out the power for some frequencies this could be better. Is good for DTFM detection I've heard.

The WNk isn't calculated 100% correctly, but it seems to work so ;) Yeah and the code is C++ so you might have to do some small adjustment to compile it as C.

Code :

```
/** Tone detect by Goertzel algorithm
*
* This program basically searches for tones (sines) in a sample and reports
the different dB it finds for
* different frequencies. Can easily be extended with some thresholding to
report true/false on detection.
* I'm far from certain goertzel it implemented 100% correct, but it works
:)
*
* Hint, the SAMPLERATE, BUFFERSIZE, FREQUENCY, NOISE and SIGNALVOLUME all
affects the outcome of the reported dB. Tweak
* em to find the settings best for your application. Also, seems to be
pretty sensitive to noise (whitenoise anyway) which
* is a bit sad. Also I don't know if the goertzel really likes float values
for the frequency ... And using 44100 as
* samplerate for detecting 6000 Hz tone is kinda silly I know :)
*
* Written by: Espen Riskedal, espenr@ii.uib.no, july-2002
*/

#include <iostream>
#include <cmath>
#include <cstdlib>

using std::rand;
// math stuff
using std::cos;
using std::abs;
using std::exp;
using std::log10;
// iostream stuff
```

```
using std::cout;
using std::endl;

#define PI 3.14159265358979323844
// change the defines if you want to
#define SAMPLERATE 44100
#define BUFFERSIZE 8820
#define FREQUENCY 6000
#define NOISE 0.05
#define SIGNALVOLUME 0.8

/**  The Goertzel algorithm computes the k-th DFT coefficient of the input
signal using a second-order filter.
*   http://ptolemy.eecs.berkeley.edu/papers/96/dtmf_ict/www/node3.html.
*   Basiclly it just does a DFT of the frequency we want to check, and none
of the others (FFT calculates for all frequencies).
*/
float goertzel(float *x, int N, float frequency, int samplerate) {
    float Skn, Skn1, Skn2;
    Skn = Skn1 = Skn2 = 0;

    for (int i=0; i<N; i++) {
Skn2 = Skn1;
Skn1 = Skn;
Skn = 2*cos(2*PI*frequency/samplerate)*Skn1 - Skn2 + x[i];
    }

    float WNk = exp(-2*PI*frequency/samplerate); // this one ignores
complex stuff
    //float WNk = exp(-2*j*PI*k/N);
    return (Skn - WNk*Skn1);
}

/** Generates a tone of the specified frequency
*  Gotten from: http://groups.google.com/groups?hl=en&lr=&ie=UTF-8&oe=UTF-
8&safe=off&selm=3c641e%243jn%40uicsl.csl.uiuc.edu
*/
float *makeTone(int samplerate, float frequency, int length, float
gain=1.0) {
    //y(n) = 2 * cos(A) * y(n-1) - y(n-2)
    //A= (frequency of interest) * 2 * PI / (sampling frequency)
    //A is in radians.
    // frequency of interest MUST be <= 1/2 the sampling frequency.
    float *tone = new float[length];
    float A = frequency*2*PI/samplerate;

    for (int i=0; i<length; i++) {
if (i > 1) tone[i]= 2*cos(A)*tone[i-1] - tone[i-2];
else if (i > 0) tone[i] = 2*cos(A)*tone[i-1] - (cos(A));
```

```
else tone[i] = 2*cos(A)*cos(A) - cos(2*A);
    }

    for (int i=0; i<length; i++) tone[i] = tone[i]*gain;

    return tone;
}

/** adds whitenoise to a sample */
void *addNoise(float *sample, int length, float gain=1.0) {
    for (int i=0; i<length; i++) sample[i] += (2*(rand()/(float)RAND_MAX)-
1)*gain;
}

/** returns the signal power/dB */
float power(float value) {
    return 20*log10(abs(value));
}

int main(int argc, const char* argv) {
    cout << "Samplerate: " << SAMPLERATE << "Hz\n";
    cout << "Buffersize: " << BUFFERSIZE << " samples\n";
    cout << "Correct frequency is: " << FREQUENCY << "Hz\n";
    cout << " - signal volume: " << SIGNALVOLUME*100 << "%\n";
    cout << " - white noise: " << NOISE*100 << "%\n";

    float *tone = makeTone(SAMPLERATE, FREQUENCY, BUFFERSIZE,
SIGNALVOLUME);
    addNoise(tone, BUFFERSIZE,NOISE);

    int stepsize = FREQUENCY/5;

    for (int i=0; i<10; i++) {
int freq = stepsize*i;
cout << "Trying freq: " << freq << "Hz  ->  dB: " << power(goertzel(tone,
BUFFERSIZE, freq, SAMPLERATE)) << endl;
    }
    delete tone;

    return 0;
}
```

Variable-hardness clipping function

References : Posted by Laurent de Soras
Linked file : laurent.gif (this linked file is included below)

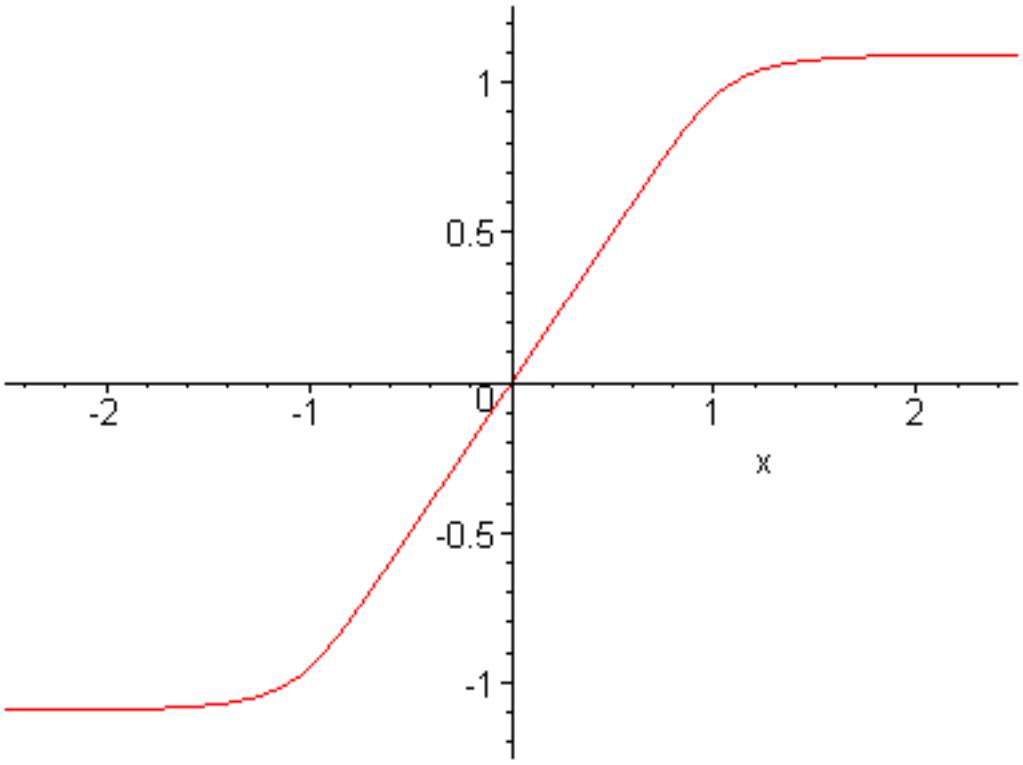
Notes :
k >= 1 is the "clipping hardness". 1 gives a smooth clipping, and a high value gives hardclipping.

Don't set k too high, because the formula use the pow() function, which use exp() and would overflow easily. 100 seems to be a reasonable value for "hardclipping"

Code :

```
f (x) = sign (x) * pow (atan (pow (abs (x), k)), (1 / k));
```

Linked files



Various Biquad filters

References : JAES, Vol. 31, No. 11, 1983 November
Linked file : filters003.txt (this linked file is included below)

Notes :
(see linkfile)
Filters included are:
presence
shelvelowpass
2polebp
peaknotch
peaknotch2

Linked files

```
/*
 * Presence and Shelve filters as given in
 *   James A. Moorer
 *   The manifold joys of conformal mapping:
 *   applications to digital filtering in the studio
 *   JAES, Vol. 31, No. 11, 1983 November
 */

#define SPN MINDOUBLE

double bw2angle(a,bw)
double a,bw;
{
  double T,d,sn,cs,mag,delta,theta,tmp,a2,a4,asnd;

  T = tan(2.0*PI*bw);
  a2 = a*a;
  a4 = a2*a2;
  d = 2.0*a2*T;
  sn = (1.0 + a4)*T;
  cs = (1.0 - a4);
  mag = sqrt(sn*sn + cs*cs);
  d /= mag;
  delta = atan2(sn,cs);
  asnd = asin(d);
  theta = 0.5*(PI - asnd - delta);
  tmp = 0.5*(asnd-delta);
  if ((tmp > 0.0) && (tmp < theta)) theta = tmp;
  return(theta/(2.0*PI));
}

void presence(cf,boost,bw,a0,a1,a2,b1,b2)
```

```
double cf,boost,bw,*a0,*a1,*a2,*b1,*b2;
{
  double a,A,F,xfmbw,C,tmp,alphan,alphad,b0,recipb0,asq,F2,a2plus1,ma2plus1;

  a = tan(PI*(cf-0.25));
  asq = a*a;
  A = pow(10.0,boost/20.0);
  if ((boost < 6.0) && (boost > -6.0)) F = sqrt(A);
  else if (A > 1.0) F = A/sqrt(2.0);
  else F = A*sqrt(2.0);
  xfmbw = bw2angle(a,bw);

  C = 1.0/tan(2.0*PI*xfmbw);
  F2 = F*F;
  tmp = A*A - F2;
  if (fabs(tmp) <= SPN) alphad = C;
  else alphad = sqrt(C*C*(F2-1.0)/tmp);
  alphan = A*alphad;

  a2plus1 = 1.0 + asq;
  ma2plus1 = 1.0 - asq;
  *a0 = a2plus1 + alphan*ma2plus1;
  *a1 = 4.0*a;
  *a2 = a2plus1 - alphan*ma2plus1;

  b0 = a2plus1 + alphad*ma2plus1;
  *b2 = a2plus1 - alphad*ma2plus1;

  recipb0 = 1.0/b0;
  *a0 *= recipb0;
  *a1 *= recipb0;
  *a2 *= recipb0;
  *b1 = *a1;
  *b2 *= recipb0;
}

void shelve(cf,boost,a0,a1,a2,b1,b2)
double cf,boost,*a0,*a1,*a2,*b1,*b2;
{
  double a,A,F,tmp,b0,recipb0,asq,F2,gamma2,siggam2,gam2p1;
  double gamman,gammad,ta0,ta1,ta2,tb0,tb1,tb2,aa1,ab1;

  a = tan(PI*(cf-0.25));
  asq = a*a;
  A = pow(10.0,boost/20.0);
  if ((boost < 6.0) && (boost > -6.0)) F = sqrt(A);
  else if (A > 1.0) F = A/sqrt(2.0);
  else F = A*sqrt(2.0);

  F2 = F*F;
```

```
  tmp = A*A - F2;
  if (fabs(tmp) <= SPN) gammad = 1.0;
  else gammad = pow((F2-1.0)/tmp,0.25);
  gamman = sqrt(A)*gammad;

  gamma2 = gamman*gamman;
  gam2p1 = 1.0 + gamma2;
  siggam2 = 2.0*sqrt(2.0)/2.0*gamman;
  ta0 = gam2p1 + siggam2;
  ta1 = -2.0*(1.0 - gamma2);
  ta2 = gam2p1 - siggam2;

  gamma2 = gammad*gammad;
  gam2p1 = 1.0 + gamma2;
  siggam2 = 2.0*sqrt(2.0)/2.0*gammad;
  tb0 = gam2p1 + siggam2;
  tb1 = -2.0*(1.0 - gamma2);
  tb2 = gam2p1 - siggam2;

  aa1 = a*ta1;
  *a0 = ta0 + aa1 + asq*ta2;
  *a1 = 2.0*a*(ta0+ta2)+(1.0+asq)*ta1;
  *a2 = asq*ta0 + aa1 + ta2;

  ab1 = a*tb1;
  b0 = tb0 + ab1 + asq*tb2;
  *b1 = 2.0*a*(tb0+tb2)+(1.0+asq)*tb1;
  *b2 = asq*tb0 + ab1 + tb2;

  recipb0 = 1.0/b0;
  *a0 *= recipb0;
  *a1 *= recipb0;
  *a2 *= recipb0;
  *b1 *= recipb0;
  *b2 *= recipb0;
}

void initfilter(f)
filter *f;
{
  f->x1 = 0.0;
  f->x2 = 0.0;
  f->y1 = 0.0;
  f->y2 = 0.0;
  f->y = 0.0;
}

void setfilter_presence(f,freq,boost,bw)
filter *f;
double freq,boost,bw;
```

```
{
  presence(freq/(double)SR,boost,bw/(double)SR,
           &f->cx,&f->cx1,&f->cx2,&f->cy1,&f->cy2);
  f->cy1 = -f->cy1;
  f->cy2 = -f->cy2;
}

void setfilter_shelve(f,freq,boost)
filter *f;
double freq,boost;
{
  shelve(freq/(double)SR,boost,
   &f->cx,&f->cx1,&f->cx2,&f->cy1,&f->cy2);
  f->cy1 = -f->cy1;
  f->cy2 = -f->cy2;
}

void setfilter_shelvelowpass(f,freq,boost)
filter *f;
double freq,boost;
{
  double gain;

  gain = pow(10.0,boost/20.0);
  shelve(freq/(double)SR,boost,
   &f->cx,&f->cx1,&f->cx2,&f->cy1,&f->cy2);
  f->cx /= gain;
  f->cx1 /= gain;
  f->cx2 /= gain;
  f->cy1 = -f->cy1;
  f->cy2 = -f->cy2;
}

/*
 * As in ''An introduction to digital filter theory'' by Julius O. Smith
 * and in Moore's book; I use the normalized version in Moore's book.
 */
void setfilter_2polebp(f,freq,R)
filter *f;
double freq,R;
{
  double theta;

  theta = 2.0*PI*freq/(double)SR;
  f->cx = 1.0-R;
  f->cx1 = 0.0;
  f->cx2 = -(1.0-R)*R;
  f->cy1 = 2.0*R*cos(theta);
  f->cy2 = -R*R;
}
```

```
/*
 * As in
 *   Stanley A. White
 *   Design of a digital biquadratic peaking or notch filter
 *   for digital audio equalization
 *   JAES, Vol. 34, No. 6, 1986 June
 */
void setfilter_peaknotch(f,freq,M,bw)
filter *f;
double freq,M,bw;
{
  double w0,p,om,ta,d;

  w0 = 2.0*PI*freq;
  if ((1.0/sqrt(2.0) < M) && (M < sqrt(2.0))) {
    fprintf(stderr,"peaknotch filter: 1/sqrt(2) < M < sqrt(2)\n");
    exit(-1);
  }
  if (M <= 1.0/sqrt(2.0)) p = sqrt(1.0-2.0*M*M);
  if (sqrt(2.0) <= M) p = sqrt(M*M-2.0);
  om = 2.0*PI*bw;
  ta = tan(om/((double)SR*2.0));
  d = p+ta;
  f->cx = (p+M*ta)/d;
  f->cx1 = -2.0*p*cos(w0/(double)SR)/d;
  f->cx2 = (p-M*ta)/d;
  f->cy1 = 2.0*p*cos(w0/(double)SR)/d;
  f->cy2 = -(p-ta)/d;
}

/*
 * Some JAES's article on ladder filter.
 * freq (Hz), gdb (dB), bw (Hz)
 */
void setfilter_peaknotch2(f,freq,gdb,bw)
filter *f;
double freq,gdb,bw;
{
  double k,w,bwr,abw,gain;

  k = pow(10.0,gdb/20.0);
  w = 2.0*PI*freq/(double)SR;
  bwr = 2.0*PI*bw/(double)SR;
  abw = (1.0-tan(bwr/2.0))/(1.0+tan(bwr/2.0));
  gain = 0.5*(1.0+k+abw-k*abw);
  f->cx = 1.0*gain;
  f->cx1 = gain*(-2.0*cos(w)*(1.0+abw))/(1.0+k+abw-k*abw);
  f->cx2 = gain*(abw+k*abw+1.0-k)/(abw-k*abw+1.0+k);
  f->cy1 = 2.0*cos(w)/(1.0+tan(bwr/2.0));
```

```
  f->cy2 = -abw;
}

double applyfilter(f,x)
filter *f;
double x;
{
  f->x = x;
  f->y = f->cx * f->x + f->cx1 * f->x1 + f->cx2 * f->x2
    + f->cy1 * f->y1 + f->cy2 * f->y2;
  f->x2 = f->x1;
  f->x1 = f->x;
  f->y2 = f->y1;
  f->y1 = f->y;
  return(f->y);
}
```

Waveform generator using MinBLEPS

References : Posted by locke[AT]rpgfan.demon.co.uk
Linked file : MinBLEPS.zip

Notes :
C code and project file for MSVC6 for a bandwidth-limited saw/square (with PWM) generator using MinBLEPS.

This code is based on Eli's MATLAB MinBLEP code and uses his original minblep.mat file. Instead of keeping a list of all active MinBLEPS, the output of each MinBLEP is stored in a buffer, in which all consequent MinBLEPS and the waveform output are added together. This optimization makes it fast enough to be used realtime.

Produces slight aliasing when sweeping high frequencies. I don't know wether Eli's original code does the same, because I don't have MATLAB. Any help would be appreciated.

The project name is 'hardsync', because it's easy to generate hardsync using MinBLEPS.

Code :

WaveShaper

Type : waveshaper
References : Posted by Bram de Jong

Notes :
where x (in [-1..1] will be distorted and a is a distortion parameter that goes from 1 to infinity
The equation is valid for positive and negativ values.
If a is 1, it results in a slight distortion and with bigger a's the signal get's more funky.

A good thing about the shaper is that feeding it with bigger-than-one
values, doesn't create strange fx. The maximum this function will reach is
1.2 for a=1.

Code :

```
f(x,a) = x*(abs(x) + a)/(x^2 + (a-1)*abs(x) + 1)
```

Waveshaper

Type : waveshaper
References : Posted by Jon Watte

Notes :
A favourite of mine is using a sin() function instead.
This will have the "unfortunate" side effect of removing
odd harmonics if you take it to the extreme: a triangle
wave gets mapped to a pure sine wave.
This will work with a going from .1 or so to a= 5 and bigger!
The mathematical limits for a = 0 actually turns it into a linear
function at that point, but unfortunately FPUs aren't that good
with calculus :-) Once a goes above 1, you start getting clipping
in addition to the "soft" wave shaping. It starts getting into
more of an effect and less of a mastering tool, though :-)

Seeing as this is just various forms of wave shaping, you
could do it all with a look-up table, too. In my version, that would
get rid of the somewhat-expensive sin() function.

Code :
```
(input: a == "overdrive amount")

z = M_PI * a;
s = 1/sin(z)
b = 1/a

if (x > b)
  f(x) = 1
else
  f(x) = sin(z*x)*s
```

Waveshaper

References : Posted by Partice Tarrabia and Bram de Jong

Notes :
amount should be in [-1..1[Plot it and stand back in astonishment! ;)

Code :
```
x = input in [-1..1]
y = output
k = 2*amount/(1-amount);

f(x) = (1+k)*x/(1+k*abs(x))
```

Waveshaper (simple description)

Type : Polynomial; Distortion
References : Posted by Jon Watte

Notes :
> The other question; what's a 'waveshaper' algorithm. Is it simply another
> word for distortion?

A typical "waveshaper" is some function which takes an input sample value X and transforms it to an output sample X'. A typical implementation would be a look-up table of some number of points, and some level of interpolation between those points (say, cubic). When people talk about a wave shaper, this is most often what they mean. Note that a wave shaper, as opposed to a filter, does not have any state. The mapping from X -> X' is stateless.

Some wave shapers are implemented as polynomials, or using other math functions. Hard clipping is a wave shaper implemented using the min() and max() functions (or the three-argument clamp() function, which is the same thing). A very mellow and musical-sounding distortion is implemented using a third-degree polynomial; something like X' = (3/2)X - (1/2)X^3. The nice thing with polynomial wave shapers is that you know that the maximum they will expand bandwidth is their order. Thus, you need to oversample 3x to make sure that a third-degree polynomial is aliasing free. With a lookup table based wave shaper, you don't know this (unless you treat an N-point table as an N-point polynomial :-)

Code :
```
float waveshape_distort( float in ) {
  return 1.5f * in - 0.5f * in *in * in;
}
```

Waveshaper :: Gloubi-boulga

References : Laurent de Soras on IRC

Notes :
Multiply input by gain before processing

Code :
```
const double x = input * 0.686306;
const double a = 1 + exp (sqrt (fabs (x)) * -0.75);
output = (exp (x) - exp (-x * a)) / (exp (x) + exp (-x));
```

Wavetable Synthesis

References : Robert Bristow-Johnson
Linked file : http://www.harmony-central.com/Synth/Articles/Wavetable_101/Wavetable-101.pdf

Notes :
Wavetable sythesis AES paper by RBJ.

Weird synthesis

References : Posted by Andy M00cho

Notes :
(quoted from Andy's mail...)
What I've done in a soft-synth I've been working on is used what I've termed Fooglers, no reason, just liked the name :) Anyway all I've done is use a *VERY* short delay line of 256 samples and then use 2 controllable taps into the delay with High Frequency Damping, and a feedback parameter.

Using a tiny fixed delay size of approx. 4.8ms (really 256 samples/1k memory with floats) means this costs, in terms of cpu consumption practically nothing, and the filter is a real simple 1 pole low-pass filter. Maybe not DSP'litically correct but all I wanted was to avoid the high frequencies trashing the delay line when high feedbacks (99%->99.9%) are used (when the fun starts ;).

I've been getting some really sexy sounds out of this idea, and of course you can have the delay line tuneable if you choose to use fractional taps, but I'm happy with it as it is.. 1 nice simple, yet powerful addition to the base oscillators.

In reality you don't need 2 taps, but I found that using 2 added that extra element of funkiness...

Zoelzer biquad filters

Type : biquad IIR
References : Udo Zoelzer: Digital Audio Signal Processing (John Wiley & Sons, ISBN 0 471 97226 6), Chris Townsend

Notes :
Here's the formulas for the Low Pass, Peaking, and Low Shelf, which should cover the basics. I tried to convert the formulas so they are little more consistent. Also, the Zolzer low pass/shelf formulas didn't have adjustable Q, so I added that for consistency with Roberts formulas as well. I think someone may want to check that I did it right.
---------- Chris Townsend
I mistranscribed the low shelf cut formulas.
Hopefully this is correct. Thanks to James McCartney for noticing.
---------- Chris Townsend

Code :
```
omega = 2*PI*frequency/sample_rate

K=tan(omega/2)
Q=Quality Factor
V=gain

LPF:   b0 =  K^2
       b1 =  2*K^2
       b2 =  K^2
       a0 =  1 + K/Q + K^2
       a1 =  2*(K^2 - 1)
       a2 =  1 - K/Q + K^2

peakingEQ:
      boost:
      b0 =  1 + V*K/Q + K^2
      b1 =  2*(K^2 - 1)
      b2 =  1 - V*K/Q + K^2
      a0 =  1 + K/Q + K^2
      a1 =  2*(K^2 - 1)
      a2 =  1 - K/Q + K^2

      cut:
      b0 =  1 + K/Q + K^2
      b1 =  2*(K^2 - 1)
      b2 =  1 - K/Q + K^2
      a0 =  1 + V*K/Q + K^2
      a1 =  2*(K^2 - 1)
```

```
      a2 =  1 - V*K/Q + K^2

lowShelf:
     boost:
       b0 =  1 + sqrt(2*V)*K + V*K^2
       b1 =  2*(V*K^2 - 1)
       b2 =  1 - sqrt(2*V)*K + V*K^2
       a0 =  1 + K/Q + K^2
       a1 =  2*(K^2 - 1)
       a2 =  1 - K/Q + K^2

     cut:
       b0 =  1 + K/Q + K^2
       b1 =  2*(K^2 - 1)
       b2 =  1 - K/Q + K^2
       a0 =  1 + sqrt(2*V)*K + V*K^2
       a1 =  2*(v*K^2 - 1)
       a2 =  1 - sqrt(2*V)*K + V*K^2
```