Holy City Audio Forum

Former Home of SpinCAD Designer
Skip to content

Search... Search
Advanced search

[Moderator Control Panel]

2P SVP fixed coefficients

Post a reply

Search this topic... Search

3 posts • Page **1** of **1**

- Edit post (./posting.php?mode=edit&f=42&p=2421)
- Delete post (./posting.php?mode=delete&f=42&p=2421)
- Report this post (./report.php?f=42&p=2421)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=2421&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=2421)

2P SVP fixed coefficients (#p2421)

by **Digital Larry** » Thu Jun 11, 2015 7:46 pm

http://www.musicdsp.com/ (http://www.musicdsp.com/)

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
```

From the control panel for the 2P SVF fixed block

```
   public double getFreq() {
      return f0;
   }

   public void setFreq(double f) {
      f0 = f;
      setCoefficients();
   }

   public void setQ(double value) {
      q0 = value;
      setCoefficients();
   }

   public double getQ() {
      return q0;
   }

   public void setCoefficients() {
      q1 = 1.0/q0;
      fZ = Math.sin(2 * Math.PI * f0/getSamplerate());
   }
```

Top

- Edit post (./posting.php?mode=edit&f=42&p=2422)
- Delete post (./posting.php?mode=delete&f=42&p=2422)
- Report this post (./report.php?f=42&p=2422)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=2422&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=2422)

Re: 2P SVP fixed coefficients (#p2422)

by **Digital Larry** » Thu Jun 11, 2015 7:49 pm

DSP reference shows:

f = 2 sin (pi * cutoff / fs) //[approximately]

SpinCAD:

fZ = Math.sin(2 * Math.PI * f0/getSamplerate());

Ah so the **two** is in a different place. I need a tie breaker!

Top

- Edit post (./posting.php?mode=edit&f=42&p=2423)
- Delete post (./posting.php?mode=delete&f=42&p=2423)
- Report this post (./report.php?f=42&p=2423)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=2423&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=2423)

Re: 2P SVP fixed coefficients (#p2423)

by **Digital Larry** » Thu Jun 11, 2015 7:54 pm

Also from musicdsp.org:

```
// simple frequency tuning with error towards nyquist
// F is the filter's center frequency, and Fs is the sampling rate
F1 = 2*pi*F/Fs

// ideal tuning:
F1 = 2 * sin( pi * F / Fs )
```

I think this is just an approximation for low frequencies, since for x close to 0, x is about sin(x).

Many things point back to Hal Chamberlain (I still have his Musical Applications of Microprocessors around here somewhere):

http://www.earlevel.com/main/2003/03/02 ... le-filter/ (http://www.earlevel.com/main/2003/03/02/the-digital-state-variable-filter/)

I'm guessing that the expression should be:

F1 = 2 * sin(pi * F / Fs)

rather than what's in there now:

F1 = sin(2 * pi * F / Fs)

Somebody else give me their opinion. This has obviously been in there for a long time so I have not been thinking about that recently.

Top

Display posts from previous:	All posts	Sort by	Post time	Ascending	Go

Post a reply

3 posts • Page **1** of **1**

Return to Developer's Corner

Jump to:	Developer's Corner	Go
Quick-mod tools:	Lock topic	Go

Who is online

Users browsing this forum: Digital Larry and 0 guests