Holy City Audio Forum

Former Home of SpinCAD Designer
Skip to content

Search… Search
Advanced search

[Moderator Control Panel]

A fully flexible filter on the FV-1?

Post a reply

Search this topic… Search

5 posts • Page **1** of **1**

- Edit post (./posting.php?mode=edit&f=42&p=1615)
- Delete post (./posting.php?mode=delete&f=42&p=1615)
- Report this post (./report.php?f=42&p=1615)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=1615&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=1615)

A fully flexible filter on the FV-1? (#p1615)

by **Digital Larry** » Tue Jun 10, 2014 5:10 pm

One of my forum members gave me the encouraging remark that he thought SpinCAD Designer would be useful "eventually". Hey I feel your pain, man! There are two technical aspects to developing this program. The first is making it functional, and as intuitive as possible to use. The other one is coming up with blocks which suit the desires of people who want to design effects on the FV-1. The example that the person offered was a resonant filter with independently adjustable frequency, Q, and gain.

I went off to the audio EQ cookbook (http://www.musicdsp.org/files/Audio-EQ-Cookbook.txt) web page, where the equations to calculate 2nd order filter coefficients from f0 and Q are given. And they are fairly straightforward, with the exception that you need to use the cosine function, which the FV-1 does not support.

The notes at the Spin website about filters are pretty vague. Some examples are given, and even some equations for calcuating coefficients, but as far as calculating them on the fly, that's where it gets a bit sticky.

My initial thoughts about an approach for doing this fall into two categories:

1) Use Taylor series expansions to approximate the cosine function.
2) Use piecewise linear interpolation to calculate the coefficients. For this you'd just need to calculate the parameter values at each endpoint and then use SOF to draw a straight line between them.

Both of these would seem to have their challenges, mostly with regards to use of code space.

Top

- Edit post (./posting.php?mode=edit&f=42&p=1642)
- Delete post (./posting.php?mode=delete&f=42&p=1642)
- Report this post (./report.php?f=42&p=1642)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=1642&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=1642)

Re: A fully flexible filter on the FV-1? (#p1642)

by **Digital Larry** » Fri Jun 20, 2014 6:04 am

I decided to take another look at the Spin Knowledge base section about filters yesterday. I personally find this section exceedingly frustrating, because while Keith Barr went to a fair amount of trouble to draw filter structures and explain their pluses and minuses, he did not include any corresponding FV-1 code representing what he was talking about! You have to piece it together using these structures and parts of the code samples found elsewhere on the Spin web site.

One thing Keith DID mention was the fact that fixed structure filters are hard to control in real time because the coefficients require elaborate calculation which is somewhat beyond the FV-1's instruction set.

However, the state-variable structure apparently lends itself more easily to taking a single input parameter for frequency and another for the Q. I'll have to look into this more closely.

Top

- Edit post (./posting.php?mode=edit&f=42&p=1643)
- Delete post (./posting.php?mode=delete&f=42&p=1643)
- Report this post (./report.php?f=42&p=1643)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=1643&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=1643)

Re: A fully flexible filter on the FV-1? (#p1643)

by **Digital Larry** » Fri Jun 20, 2014 6:19 am

From the musicdsp.org (http://www.musicdsp.org/archive.php?classid=3#142) website.

```
State Variable Filter (Chamberlin version)
```

References : Hal Chamberlin, "Musical Applications of Microprocessors," 2nd Ed, Hayden Book Company 1985. pp 490-492.
Code :

```
//Input/Output
I - input sample
L - lowpass output sample
B - bandpass output sample
H - highpass output sample
N - notch output sample
F1 - Frequency control parameter
Q1 - Q control parameter
D1 - delay associated with bandpass output
D2 - delay associated with low-pass output

// parameters:
Q1 = 1/Q
// where Q1 goes from 2 to 0, ie Q goes from .5 to infinity

// 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 )

// algorithm
// loop
L = D2 + F1 * D1
H = I - L - Q1*D1
B = F1 * H + D1
N = H + L

// store delays
D1 = B
D2 = L

// outputs
L,H,B,N
```

Top

- Edit post (./posting.php?mode=edit&f=42&p=1645)
- Delete post (./posting.php?mode=delete&f=42&p=1645)
- Report this post (./report.php?f=42&p=1645)
- Warn user (./mcp.php?i=warn&mode=warn_post&f=42&p=1645&sid=d6d1dcc4501596550704606791e62b4c)
- Information (./mcp.php?i=main&mode=post_details&f=42&p=1645&sid=d6d1dcc4501596550704606791e62b4c)
- Reply with quote (./posting.php?mode=quote&f=42&p=1645)

Re: A fully flexible filter on the FV-1? (#p1645)

by **disasterarea** » Sun Jun 22, 2014 5:00 am

> ***Digital Larry wrote:***
> I decided to take a look at the Spin Knowledge base section about filters yesterday. I personally find this section exceedingly frustrating, because while Keith Barr went to a fair amount of trouble to draw filter structures and explain their pluses and minuses, he did not include any corresponding FV-1 code representing what he was talking about! You have to piece it together using these structures and parts of the code samples found elsewhere on the Spin web site.
>
> One thing Keith DID mention was the fact that fixed structure filters are hard to control in real time because the coefficients require elaborate calculation which is somewhat beyond the FV-1's instruction set.
>
> However, the state-variable structure apparently lends itself more easily to taking a single input parameter for frequency and another for the Q. I'll have to look into this more closely.

Keith knew exactly what went where, so he focused on the "why" rather than the "how." And I agree that this section is super frustrating! But I disagree with your other forum member, SpinCAD Designer is useful NOW. Unless that was me that said it, in which case I retract that statement and apologize.

Filtering with the FV-1 is fun and it's got a ton of power to do 2- and 4-pole filters. Have you checked out the "Disco Reverb" algorithm? The filtering in that guy is intense, four pole HP and LP filters plus a great simple reverb that does "infinite" if you dime the reverb time control.

http://www.spinsemi.com/get_spn.php?spn=dance_ir_h_l.spn&prodnum=SPN1001 (http://www.spinsemi.com/get_spn.php?spn=dance_ir_h_l.spn&

prodnum=SPN1001)

Re: A fully flexible filter on the FV-1? (#p1647)

by **Digital Larry** » Sun Jun 22, 2014 5:43 am

Haha, no it wasn't you, but don't worry one way or the other - your comments a few months ago are what got me back into active development. Unfortunately for FV-1 fans my concerns recently are making the infrastructure for creating blocks and control panels easier. Developing a new CAD block and control panel in Java is REALLY tedious. I'm getting close to getting checkboxes working, and one more thing, what is it - a ListBox to choose one of several options, and I'm pretty close to done on that front. And then I REALLY want to focus more on the blocks.

Back to the filter discussion... I know I had read this section of the Knowledge base several times but it only sunk in on me last week what Keith was trying to say. Digging deep into the recesses of my memories of the DSP classes I took more than 30 years ago, all I remembered about filter structures was the classical layout of delay elements, multipliers and summers. I'm not even sure I studied the state variable structure though I recall it from analog classes.

The two frequency-variable filters I have already included in SpinCAD Designer were taken from some of the example code at the Spin site, including the Disco Reverb you mentioned. I have a 4-pole low pass and 2-pole high pass. As I recall, the 4-pole filter is just two identical 2-pole structures back to back. Now we get into the section where Keith talks about "musical" results being more important.

Normally, a Butterworth (maximally flat) type filter results when you have the poles equally-spaced angularly-speaking (in the left side of the s-plane, for you analog filter freaks). This is fine for a 2-pole section. But if you stack two identical 2-pole sections back to back, their poles stack on top of each other and it's no longer a Butterworth type filter. But how's it sound? Should I worry about technical accuracy, or should I simply move on to allowing people to lay some WHOMP on their guitars? This is what Keith was getting at. In a way, it's a slight cop-out on the (theoretical) shortcomings of the FV-1 instruction set (if he had implemented SIN and COS table lookup, we wouldn't be having this discussion). On the other hand, once I finally got the point of what he was saying, I decided to stop worrying and love the state-variable filter structure.

There is a "q" parameter hiding in the filter code, though to date I have simply left it as-is. Combined with the info from the audio DSP cookbook, I have a better idea of how this could be manipulated by a pot or control panel.

One of the typical-structure filters in the DSP cookbook includes ways to calculate coefficients such that midband gain is always 0 dB. That's one of the challenges. There are a few of the fixed filters in SpinCAD Designer where you can adjust the resonance, and they definitely start to distort if you don't turn the input level down accordingly. I have not seen a discussion of the SVF where you can control the midband gain and Q together with a single parameter. But maybe it's out there. let me know if you find something.

I also read something somewhere sometime about the "Moog" filter structure being a series of one-pole filters back to back, with a global feedback around the entire thing to allow resonance to take place. Gotta get feedback loops working again and give that one a try!

Who is online

Users browsing this forum: Digital Larry and 0 guests

Administration Control Panel