Notes on SF-2 SuperFilter Settings from the Alembic Club

-Paul Boulet Draft Ver. 1.0 5/23/2003

SF-2 Superfilter

The SF-2 Super Filter contains two tunable active filters of exceptional dynamic range. The filters can be configured individually as low-pass, band-pass, and high-pass. In the stereo mode, two channels are provided, each with a two channel mixer for combining the filtered and unfiltered signals. In an alternate mono mode, a three channel mixer combines the two filtered and one unfiltered signal paths, together with an input gain control. An instrument preamplifier with front panel input jack is also available for stand-alone use.

Circuit detail

The SF-2 filter circuit is a resistively tuned two-pole universal active filters, which yields 12 dB/Oct ultimate slopes in the high-pass and low-pass modes, and 6 dB/Oct ultimate slopes in the band-pass mode. The unconventional filter design was optimized for wide dynamic range, and achieves 20 dB better signal to noise performance compared with standard circuits. This increased dynamic range makes use practical with guitar level signals as well as line level signals. Additionally, the absolute polarity of the signals in the pass band of the filter in all three modes of operation is preserved, allowing the additive mixing of the unfiltered original signal with the filtered output without phase cancellation. A control calibrated in Reciprocal Damping Ratio units adjusts the response of the filter in the vicinity of cutoff, controlling the size of the resonant peak at the tuned frequency.

In the stereo mode, each channel is provided with a two channel mixer. The direct, or unfiltered, signal is adjustable from zero level to unity gain. The filtered signal is adjustable from zero to a gain of 3 times or 10 dB. The additional gain in the filtered section is provided for those settings when the filter is used to bring out softer spectral components of the signal. As mentioned above, in the passband of the filter, the polarity is preserved so that in combination with the direct signal, the resultant will always be additive, no matter whether in low- band- or high-pass mode. The amplitude of the filtered signal in the passband of the filter is also held constant when the damping ratio control is adjusted. The result is an easy to adjust unit, each control logically responding to its designated function.

In the optional mono mode, the output is derived from a three channel mixer, one channel for the direct or unfiltered signal, and a channel for each of the two filter circuits. As in the previous paragraph, the direct signal adjusts to a maximum of unity gain, and each filter can be mixed up to three times (10 dB) gain, needed when enhancing weak spectral components. The left-most control on the front panel serves as an input gain control (or master gain control) and provides unity gain when fully clockwise when the signal is connected to the mono line input jack on the rear panel.

An instrument preamplifier with 1 MegOhm input impedance is provided with an input jack on the front panel. The instrument preamplifier is automatically connected thru 'normal' springs when no jack is inserted in the line level input on the rear panel. A jumper block inside the unit provides signal amplification of 0 dB, 10 dB, and 20 dB for the preamplifier circuit by selection of feedback resistors. The unit is set for 10 dB amplification when shipped from the factory.

The Super Filter has many applications. As an effector for guitar and bass, an unlimited field of both dramatic and subtle spectral effects are available. And when the direct volume control is set fully counterclockwise, the filtered signal alone directs the ear's attention to the natural-sounding filter effect, unlike an equalizer where the input signal is always supplied to the output at unity gain.

Interesting vocal effects including band limiting can be dialed in. And the opposite of the last effect can be used to extend the frequency response of closed box speaker enclosures. The Super Filter is a very useful tool.

Michael DeVincenzo (jlpicard) Posted on Monday, November 04, 2002 - 9:06 am:

Rob, Absolutely! I believe that the Superfilter is the most musically useful tone shaping device I have ever come across, far better than any equalizer. I use it mainly to compensate for varying room acoustics which I'm sure you know, can wreak havoc with your sound. I play mainly with my Europa which is mono so I run the SF-2 in mono through the effects loop in my Eden. The first channel I will set as a lowpass filter and vary the frequency control from 100 hz to maybe 350-400 hz depending upon whether the room is boomy or thinning out the sound and I need more punch or definition. I try to make sure that the low end character of my sound is well defined and tight before I add in any high end color to my sound. I don't like to rely on having to boost highs or upper mids in order to cut through a mix. The Superfilter works very well for this. I use the second channel in a more traditional tone control approach to dial in a nice crisp "alembic" snap for my slap tone. I set the filter type to highpass an run the frequency knob all the way up which I believe is around 6.5k (not sure, I tend to forget about the numbers silk-screened on the front panel and just let my ears guide me) The primary character of my sound comes mainly from the onboard controls of the Europa which I change often depending on the song, much to the continued annoyance of the sound man! These settings are just my preference but I must warn you, the possibilities in this unit are endless! You can tweak on this thing for a long time. Also, check out the Preamp page on the Alembic website for more info on what makes this unit unique. Then go get one!
Mike

Posted on Tuesday, November 05, 2002 - 12:24 pm:

What we're really talking about here is the whole Alembic approach to tone shaping. We tend to perceive the overall tone of our basses (or anything really)at a frequency or set of frequencies that have the greatest amplitude. We'll call this the resonant frequency. Herein lies the beauty of a low pass filter. As you turn the frequency knob clockwise on the SF-2, (or any Alembic with a filter control)
you move that resonant freq. higher, the result being the bass sounds first middier, then brighter. So with the movement of one knob you can dial thru all the possibilities! Adding the Q switch just increases the amplitude of the resonant frequency while narrowing the the range of affected frequencies to either side.(visually it looks like a sharp peak rather than a gentle bump in the EQ curve)
Now,add the second frequency control and Q switch and you can dial in anything from the most nasaly Jaco mid to a total "smiley face" response curve with ultra lows and glass shattering (and eardrum shattering) highs! Impressive control from two knobs and two switches! And I haven't even covered the fact that you can vary the overall gain of each channel of the SF-2 and change each filter from low pass to band pass or high pass. Tweaker Heaven or pinpoint tone shaping power. It's all in how you use it. I hope that helped rather than confuse. I've omitted a few things along the way to better communicate and shorten the length of this long winded post. Mike

Bob Novy (bob) Posted on Wednesday, January 15, 2003 - 11:13 pm:

Well, it's not quite the same as having an onboard SF-2, for at least the following reasons:

1) The standard onboard filters are low-pass only, while the SF-2 also offers band and high pass.

2) The standard frequency control only goes down to 350 Hz on the instrument, while the SF-2 will go all the way down to 45 Hz (the upper limit is 6K for both).

3) The SF-2 allows you to blend the direct and filtered signals, whereas the instrument signal always goes through the filter. With a 12 dB/octave rolloff in the filter, you can't really get much back beyond the Q frequency by boosting treble (as I recall, maximum treble boost is 6 dB).

The standard onboard cutoff at 350 makes reasonable sense for a low pass filter which is always in the circuit. If my math is right, 350 is approximately the fundamental of an F played at the 10th fret on the G string, so you're already starting to "lose" notes if you set it that low. However, it can be useful to do so on the SF-2, even when set to low-pass, because there you have the option of mixing the direct and filtered signals, and of course a lower frequency setting can be interesting in band or high pass mode.

So you might say that the standard Series II electronics are "like" having one channel of an SF-2 per pickup, except they won't filter as low, won't do band or high pass, and won't give you the option to mix in the unfiltered signal (but they're still pretty amazing, of course).

Aside from all that, separate bass/treble controls for each pickup adds a lot of flexibility, without requiring a stereo preamp. Unless you really needed a tube sound or something, you could probably plug this thing straight into a power amp (with a gain control) and be very happy.

And that is a pretty piece of wood...