Subject: Re: How to perform the 1-D signal filter?
Posted by Wox on Fri, 01 Feb 2008 16:57:33 GMT
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On Fri, 1 Feb 2008 06:54:42 -0800 (PST), "duxiyu@gmail.com"
<duxiyu@gmail.com> wrote:

> In the example, you set A=50 and Nterms=40 and I do not know where
> their values come from and how I get them.

The Nterms=40 determines the width of the filter: width=2.Nterms+1

The A=50 has something to do with the Gibbs phenomena, i.e. the wiggles you get when you Fourier expand a function with a finite number of terms (as opposed to infinite). The wiggles are visible at discontinuities.

I don't know anything about the Kaiser-Bessel filter (I think that's the one used here). So maybe someone else can explain this.

> Seondly, I want to know the relation between the filter in time and
> frequency domain.
> if the following command is excuted, I think newfsignal is exactly
> equal to fft(newsignal), isn't it?
> "newsignal=convol(signal,timefilter)
> fsignal=fft(signal)
> newfsignal=fsignal*freqfilter"

You are absolutly right. That's why in the code comments I wrote: "; Frequency domain filter (instead of time domain filter)". This means you can choose in which domain you are filtering, the time or frequency(fourier) domain. You could even have a filter in the time domain, take the fourier transform and use that to apply the filter in the frequency domain. Keep in mind however that every fft introduces some errors.

> And I also do not konw why you set steep=20.

If you plot the lowpass filter (in the fourier domain):

```
steep=20.
freqfilter= 1./(1.+(freq/f_high)^steep)
plot,freq,freqfilter
```

Do this for different "steep" and you will see that the edge becomes sharper with higher "steep".