Subject: Re: How to perform the 1-D signal filter?
Posted by David Fanning on Fri, 01 Feb 2008 14:16:26 GMT
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Wox writes:

```
> Example below filters in time or frequency domain:
>
>
> ; Time domain
> freq1=2.
> freq2=3.
> freq3=4.
> dtime=0.1
> ntime=1000
>
> time=dtime*findgen(ntime)
>  signal=sin(2*!pi*freq1*time)+sin(2*!pi*freq2*time)+sin(2*!pi *freq3*time)
>
> ; Time domain Filter
> f_low = 0
> f_high = 2.5
> timefilter = DIGITAL_FILTER(f_low*2*dtime, f_high*2*dtime, 50.,40)
> signal=convol(signal,timefilter)
>
> ; Frequency domain
> nfreq=ntime/2+1
>
> freq=findgen(nfreq)/(dtime*ntime)
> fsignal=fft(signal)
>
> ; Frequency domain filter (instead of time domain filter)
> if n_elements(timefilter) eq 0 then begin
>  steep=20.
>  freqfilter= 1./(1.+(freq/f_high)^steep)
>  fsignal*=freqfilter
> endif
>
>  plot,freq,abs(fsignal[0:nfreq-1])^2,xtitle='frequency',ytitl e='spectrum'
```

Wonderful example, but I'm trying to understand this whole
subject. Do you think you could flush this out with a little
explanation of what you are doing and why you choose the terms
you use, etc.? What kind of frequency filter are you constructing
here? I don't necessarily see it doing any filtering of the signal,
at least if I pass it the original signal, rather than the signal
that had already been filtered in the time domain, as written
in your example.

Cheers,

Confused

--
David Fanning, Ph.D.
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Coyote's Guide to IDL Programming: http://www.dfanning.com/
Sepore ma de ni thui. ("Perhaps thou speakest truth.")