Subject: Re: How to perform the 1-D signal filter?
Posted by Wox on Fri, 01 Feb 2008 13:27:59 GMT
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On Fri, 1 Feb 2008 02:20:52 -0800 (PST), "duxiyu@gmail.com"
<duxiyu@gmail.com> wrote:

> Dear all,
>
> Here I give a signal example and hope somone can show me how to
> perferm the frequency filter on it.
>
> ;creat a signal data with two peaks in frequency domain at 2 and 3 Hz.
> t=findgen(1000)/10.
> data=sin(2*!pi*2*t)+sin(2*!pi*3*t)
>
> freq=findgen(501)/100.
> v=fft(data)
> plot,freq,abs(v[0:500])^2,xtitle='frequency',ytitle='spectru m'
>
>
> I want to filter the signal with the frequency higher than 2.5 Hz. How
> do I do this?
>
> I have read the help files about Digital_Filter and Convol, but I do
> not know how to select the parameters for Signal_Filter.
>
> Du
>
>

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'
```

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)
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> signal=convol(signal,timefilter)
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> ; Frequency domain
> nfreq=ntime/2+1
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```

```
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>  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.
Fanning Software Consulting, Inc.
Coyote's Guide to IDL Programming: http://www.dfanning.com/
Sepore ma de ni thui. ("Perhaps thou speakest truth.")

Subject: Re: How to perform the 1-D signal filter?
Posted by duxiyu@gmail.com on Fri, 01 Feb 2008 14:54:42 GMT
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Thank you very much.
But I still have some questions about the parameters in the procedure Digital_Filter.

Result = DIGITAL_FILTER(Flow, Fhigh, A, Nterms [, /DOUBLE])

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.

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

Du

On Feb 1, 9:27 pm, Wox <nom...@hotmail.com> wrote:
> On Fri, 1 Feb 2008 02:20:52 -0800 (PST), "dux...@gmail.com"
>
>
>
> <dux...@gmail.com> wrote:
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>
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>> I have read the help files about Digital_Filter and Convol, but I do
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>
> Example below filters in time or frequency domain:
>
> ; Time domain
> freq1=2.
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> 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'
```

Subject: Re: How to perform the 1-D signal filter?
Posted by Kenneth P. Bowman on Fri, 01 Feb 2008 14:58:38 GMT

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In article
<e9b5822c-2240-4937-ad85-a53f057d9ec0@p69g2000hsa.googlegroups.com>,
"duxiyu@gmail.com" <duxiyu@gmail.com> wrote:

```
> Dear all,
>
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```

> not know how to select the parameters for Signal_Filter.
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> Du

You could look at the chapter on FFTs and digital filtering in my IDL book (http://idl.tamu.edu). I wrote the chapter in part so that *I* could refer to it whenever I need to do a digital filter. :-)

Ken Bowman

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".

Subject: Re: How to perform the 1-D signal filter?
Posted by Wox on Fri, 01 Feb 2008 17:16:30 GMT
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On Fri, 1 Feb 2008 07:16:26 -0700, David Fanning <news@dfanning.com>
wrote:

> 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
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>> timefilter = DIGITAL_FILTER(f_low*2*dtime, f_high*2*dtime, 50.,40)
>> signal=convol(signal,timefilter)
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>> ; Frequency domain
>> nfreq=ntime/2+1
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> Wonderful example, but I'm trying to understand this whole
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> 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

Ok, sorry for the confusion, but I was just illustrating that you can do the same filtering in the frequency domain as in the time domain. You do one or the other, not both at the same time. Btw, convolution in one domain becomes multiplication in the other:

filtered = signal "convol" filter
fft(filtered) = fft(signal) x fft(filter)

But I guess you already knew all this.

The filter used is the Kaiser-Bessel filter. At least I think digital_filter is using this. For the filter I constructed in the fourier domain, I'm not quit sure whether it is really identical to the KB filter, but if you plot it, it looks like a nice lowpass filter to me :-).

I'm just typing this in a hurry... Did I answer your questions?

Subject: Re: How to perform the 1-D signal filter?
Posted by David Fanning on Fri, 01 Feb 2008 18:30:29 GMT

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Kenneth P. Bowman writes:

> You could look at the chapter on FFTs and digital filtering in my IDL
> book (http://idl.tamu.edu). I wrote the chapter in part so that *I*
> could refer to it whenever I need to do a digital filter. :-)

Ah, right. I remember that chapter. Here you are going along in a nice, beginning book and all of a sudden you are surrounded by formidable equations! All I really remember about it was being dazed and thinking, "Boy, that was a big ending."

I'll have another look. :-)

Cheers,

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

Subject: Re: How to perform the 1-D signal filter?
Posted by David Fanning on Sat, 02 Feb 2008 01:07:17 GMT
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Kenneth P. Bowman writes:

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> book (http://idl.tamu.edu). I wrote the chapter in part so that *I*
> could refer to it whenever I need to do a digital filter. :-)

Humm. I found quite a lot of interest in the last three or four chapters in that book, and had a nice little read this afternoon. Unlike Chloe, I have no need to impress anyone anymore, so I'm just going to steal one or two of those ideas, if you don't mind. :-)

Thanks,

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

Subject: Re: How to perform the 1-D signal filter?
Posted by duxiyu@gmail.com on Sat, 02 Feb 2008 15:14:05 GMT

On Feb 2, 1:16 am, Wox <nom...@hotmail.com> wrote:
> On Fri, 1 Feb 2008 07:16:26 -0700, David Fanning <n...@dfanning.com>
> wrote:
>
>
>
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> digital_filter is using this. For the filter I constructed in the
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> the KB filter, but if you plot it, it looks like a nice lowpass filter
> to me :-).
>
> I'm just typing this in a hurry... Did I answer your questions?

Thank you again for your detailed explaination.
Now I am clear about it.

Best regards,
Du

Subject: Re: How to perform the 1-D signal filter?
Posted by duxiyu@gmail.com on Sat, 02 Feb 2008 15:20:22 GMT
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> In article
> < e9b5822c-2240-4937-ad85-a53f057d9...@p69g2000hsa.googlegroup s.com >,
>
>
>
> "dux...@gmail.com" <dux...@gmail.com> wrote:
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> could refer to it whenever I need to do a digital filter. :-)
>
> Ken Bowman

It is a pity that I cannot buy your book in my country.

Du