
Subject: How to perform the 1-D signal filter?

Posted by duxiyu@gmail.com on Fri, 01 Feb 2008 10:20:52 GMT

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Dear all,

Here I give a signal example and hope someone can show me how to perform 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 Convolve, but I do not know how to select the parameters for Signal_Filter.

Du

Subject: Re: How to perform the 1-D signal filter?

Posted by [Kenneth P. Bowman](#) on Fri, 01 Feb 2008 23:27:02 GMT

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In article <MPG.220d13322fc9fa3c98a25f@news.frii.com>, David Fanning <news@dfanning.com> wrote:

> 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

The math is there is you want it, but it is not essential, I suppose, to using a digital filter. The programming examples are as straightforward as is possible when using FFTs. ;-)

Ken

Subject: Re: How to perform the 1-D signal filter?
Posted by [jdu](#) on Sat, 02 Feb 2008 19:27:47 GMT
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There is another question about the filter example.
The code of example is below:

```
; 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_1=convol(signal,timefilter)
fsignal_1=fft(signal_1)

; Frequency domain
nfreq=ntime/2+1

freq=findgen(nfreq)/(dtime*ntime)
fsignal=fft(signal)

; Frequency domain filter (instead of time domain filter)
steep=100.d
y=1./(1.+(freq/f_high)^steep)
freqfilter=[y[0:nfreq-2],reverse(y[0:nfreq-2])]
fsignal_2=fsignal*freqfilter
```

```
signal_2=fft(fsignal_2,1)
```

```
window,/free
```

```
plot,freq,abs(fsignal[0:nfreq-1])^2,title='Original  
Data',xtitle='frequency',ytitle='spectrum'
```

```
window,/free
```

```
plot,freq,abs(fsignal_1[0:nfreq-1])^2,title='Filter in Time  
domain',xtitle='frequency',ytitle='spectrum'
```

```
window,/free
```

```
plot,freq,abs(fsignal_2[0:nfreq-1])^2,title='Filter in Frequency  
domain',xtitle='frequency',ytitle='spectrum'
```

I found that the the value of the peak in the second figure is lower than those in first and third figure. Why does it happen?
And the filtered signal_1 is not equal to the real part of signal_2.
Does it mean that the filter in time domain is not exactly equal to it in frequency domain?

Du

Subject: Re: How to perform the 1-D signal filter?
Posted by [Wox](#) on Mon, 04 Feb 2008 09:28:58 GMT
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On Sat, 2 Feb 2008 11:27:47 -0800 (PST), jdu.ustc@gmail.com wrote:

> I found that the the value of the peak in the second figure is lower
> than those in first and third figure. Why does it happen?
> And the filtered signal_1 is not equal to the real part of signal_2.
> Does it mean that the filter in time domain is not exactly equal to it
> in frequency domain?

Well, I'm not sure. If you execute the code below, you will see that the two filters are similar. Maybe this is just the result of the intrinsic approximation of time domain filtering?

```
; 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)
```

```
; Frequency domain
```

```

nfreq=ntime/2+1
freq=findgen(nfreq)/(dtime*ntime)
fsignal=fft(signal)

; 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)

; Time domain Filter in Frequency domain
ntime2=n_elements(timefilter)
nfreq2=ntime2/2+1
freq2=findgen(nfreq2)/(dtime*ntime2)
ftimefilter=fft(timefilter,1); == ntime2*fft(timefilter)

; Frequency domain filter (instead of time domain filter)
steep=50.
freqfilter= 1./(1.+(freq/f_high)^steep)
fsignal*=freqfilter

; Plot
window,0
plot,freq,abs(fsignal[0:nfreq-1])^2,xtitle='frequency',ytitle='spectrum',title='Time
domain filtered'
window,1
plot,freq,abs((fft(signal))[0:nfreq-1])^2,xtitle='frequency',ytitle='spectrum',title='Freq
domain filtered'
window,2
plot,freq,abs(freqfilter)^2,yrange=[0,1.1],ystyle=1,xtitle=' frequency',ytitle='spectrum',title='Filters'
oplot,freq2,abs((ftimefilter)[0:nfreq2-1])^2,col=128

```

Subject: Re: How to perform the 1-D signal filter?
 Posted by [jdu](#) on Mon, 04 Feb 2008 16:48:08 GMT

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> On Sat, 2 Feb 2008 11:27:47 -0800 (PST), jdu.u...@gmail.com wrote:
 >> I found that the the value of the peak in the second figure is lower
 >> than those in first and third figure. Why does it happen?
 >> And the filtered signal_1 is not equal to the real part of signal_2.
 >> Does it mean that the filter in time domain is not exactly equal to it
 >> in frequency domain?
 >
 > Well, I'm not sure. If you execute the code below, you will see that
 > the two filters are similar. Maybe this is just the result of the
 > intrinsic approximation of time domain filtering?

```

>
> ; 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)
>
> ; Frequency domain
> nfreq=ntime/2+1
> freq=findgen(nfreq)/(dtime*ntime)
> fsignal=fft(signal)
>
> ; 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)
>
> ; Time domain Filter in Frequency domain
> ntime2=n_elements(timefilter)
> nfreq2=ntime2/2+1
> freq2=findgen(nfreq2)/(dtime*ntime2)
> ftimefilter=fft(timefilter,1); == ntime2*fft(timefilter)
>
> ; Frequency domain filter (instead of time domain filter)
> steep=50.
> freqfilter= 1./(1.+(freq/f_high)^steep)
> fsignal*=freqfilter
>
> ; Plot
> window,0
> plot,freq,abs(fsignal[0:nfreq-1])^2,xtitle='frequency',ytitle='spectrum',title='Time
> domain filtered'
> window,1
> plot,freq,abs((fft(signal))[0:nfreq-1])^2,xtitle='frequency',ytitle='spectrum',title='Freq
> domain filtered'
> window,2
> plot,freq,abs(freqfilter)^2,yrange=[0,1.1],ystyle=1,xtitle='
frequency',ytitle='spectrum',title='Filters'
> oplot,freq2,abs((ftimefilter)[0:nfreq2-1])^2,col=128

```

In this example, $\max(\text{abs}(\text{fsignal}[0:\text{nfreq}-1])^2) = 0.249995$ and $\max(\text{abs}((\text{fft}(\text{signal}))[0:\text{nfreq}-1])^2) = 0.211242$.

The peak of spectra in window 0 is not equal to it in window 1.

Does this difference between them result from the arithmetical errors

of the filters?

Is there any methods to eliminate it?

```
> ftimefilter=fft(timefilter,1); == ntime2*fft(timefilter)
```

Besides, I don't know why you use the inverse transform instead of the forward transform in IDL to caculate ftimefilter.

Du
