

Intuitive DSP- FIR Filter implementation in MS-Excel

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This tutorial is intended to teach DSP basics in a much more intuitive way which would set a path to implement filters for practical applications. For detailed explanation on DSP terms, Design and implementation refer any standard text book.

What you need?

1. Microsoft Excel
2. Scilab or Matlab, or any other computing software.

Scilab closely resembles to Matlab and it is free. This tutorial is based on Scilab.

Introduction:

A filter in DSP is defined by a set of numbers (coefficients).

Working with Filters has two steps: Filter design and Filter implementation. Filter designing means finding the filter coefficients. Implementing means using the filter coefficients in a software (or programming language) for filtering operation. Here Scilab will be used for Filter designing and Excel for Filter implementation.

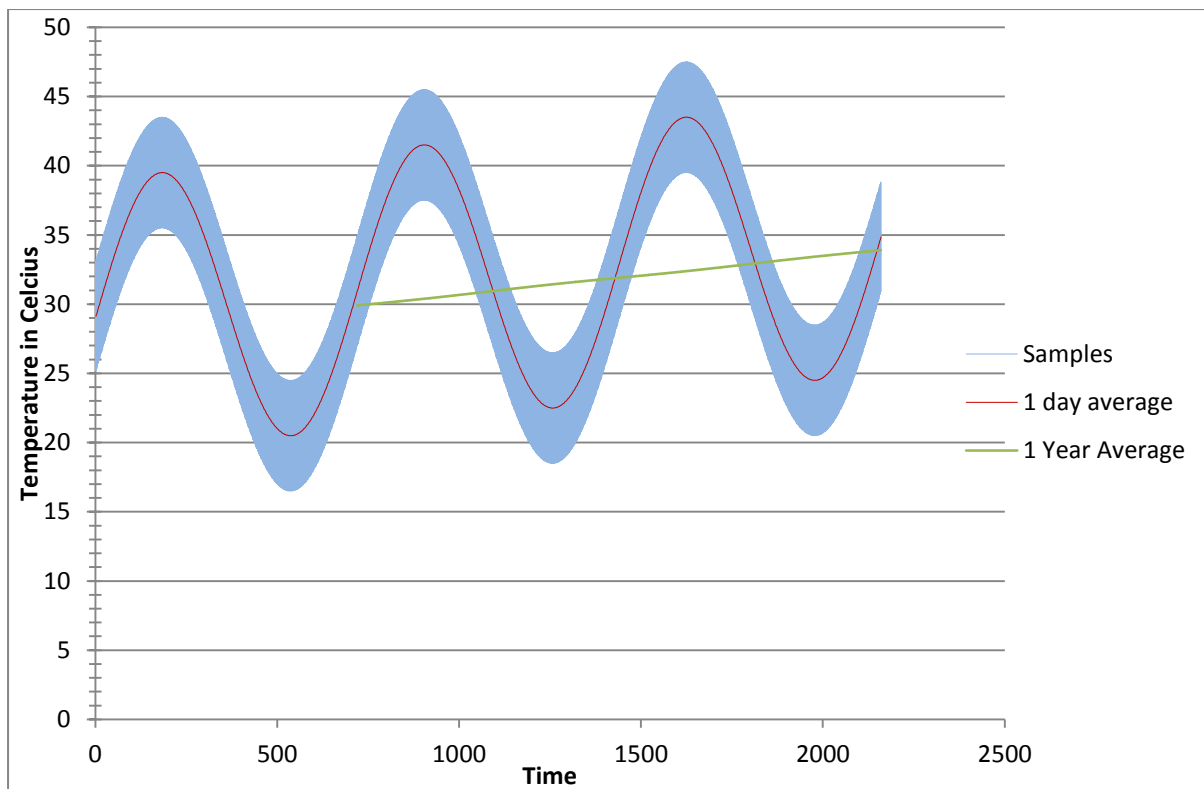
Intuitive approach to FIR Low pass filter:

An FIR low pass filter can simply be considered as an averaging operation. Let's take this example

Imagine that the temperature of a region was increased by 6 deg Celsius during a span of 3 years due to global warming. We collected 2 temperature samples a day (morning and evening) for 3 years (total 2190 samples).

Now let's analyse the temperature data. There will be large temperature difference between morning and evening data. Similarly seasonal variation of temperature will also be there.

To filter out the variations that occur between morning and evening just take average of two samples. So every sample will be modified as the average of that sample and its previous sample. This is called a moving average filter. If you want, you can modify the value as the average of that day and its previous 7 days. This will filter out any variations that may occur within every week.



(Notice that the 1 year moving average starts after 1 year because it needs previous 1 year samples for computing moving average)

Instead of just averaging the samples we can have weighted average also, by adding the samples after multiplying with a set of coefficients. This is called weighted moving average. Now the output of this filter will be more dependent on the coefficients. FIR Low Pass Filter is a special case of this. To find this coefficients we have to do the math (refer any DSP textbook for this), or use software like Scilab. The coefficients can be designed to filter out variations corresponding to any time period/frequency. More the number of samples used for averaging better is the response.

Let's implement a filter in Excel

MS Excel is more than good enough to demonstrate a Digital Filter but to design this we need Scilab or Matlab.

Step 1: Designing of filter in Scilab

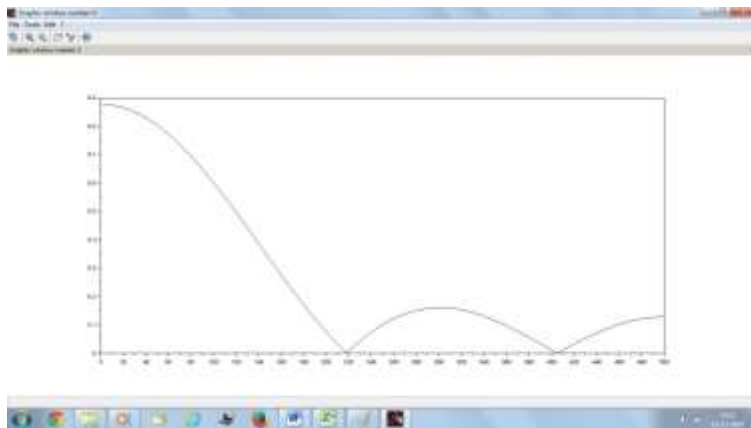
Install Scilab and open it.

Now we are going to design a 5th order FIR lowpass filter with cutoff frequency=1/10th of sampling frequency.

Type this in the console of Scilab and press enter:

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[c]=ffilt("lp",5,0.1); [a,b]=frmag(c,500); plot (a)
```

Now you will see the Frequency Vs magnitude plot of the newly designed filter in a new window.



“ffilt” is the function used to design FIR filter. “lp” refers to lowpass filter. “5” refers to 5th order filter. “0.1” refers to 1/10th of sampling frequency.

“frmag” is the function used for obtaining magnitude plot of a filter. ‘x’ is the filter of which magnitude plot is needed. ‘500’ refers to number of data points.

The variable ‘x’ has the filter parameters. To see it, either go to variable browser or double click on ‘x’ or type x on console, and press enter. You will get the filter parameters as:

0.151365, 0.187098, 0.2, 0.187098, 0.151365

Step 2: Implementing filter in Excel

Open Excel.

Make a table like this.

Q15					
	A	B	C	D	E
1	Serial no.	Angle in degree	Angle in radian(x)	sin(x)	sin(x)+sin(4x)
2	1	0	0	0	0
3	2	20	0.348889	0.341854	1.326538439
4	3	40	0.697778	0.642516	0.985866564
5	4	60	1.046667	0.86576	0.000798157
6	5	80	1.395556	0.984685	0.339730594
7	6	100	1.744444	0.984961	1.625033395
8	7	120	2.093333	0.866556	1.734696925
9	8	140	2.442222	0.643736	0.306376179
10	9	160	2.791111	0.34335	-0.642425172
11	10	180	3.14	0.001593	-0.004777918
12	11	200	3.488889	-0.34036	0.643197216
13	12	220	3.837778	-0.6413	-0.291968792
14	13	240	4.186667	-0.86496	-1.72670882
15	14	260	4.535556	-0.98441	-1.634215105
16	15	280	4.884444	-0.98523	-0.350070147
17	16	300	5.233333	-0.86735	0.003935771
18	17	320	5.582222	-0.64495	-0.976309909
19	18	340	5.931111	-0.34485	-1.331671489

Column A is serial number. Column B is Angle in degrees. Degree data should be till 720 degrees so that we get two full sine wave cycles. Convert this to radian in Column C.

SUM			
A	B	C	D
Serial no.	Angle in degree	Angle in radian(x)	sin(x)
1	0	=B2*3.14/180	
2	20	0.348889	0.341854

Get the sine of Column C in Column D.

SUM			
A	B	C	D
Serial no.	Angle in degree	Angle in radian(x)	sin(x)
1	0	0	=SIN(C2)
2	20	0.348889	0.341854
3	40	0.697778	0.642516
4	60	1.046667	0.86576

Get Sin(x) +Sin (4x) in Column E.

A	B	C	D	E	F
Serial no.	Angle in degree	Angle in radian(x)	sin(x)	sin(x)+sin(4x)	Filter output
1	0	0	0	=SIN(C2)+SIN(4*C2)	
2	20	0.348889	0.341854	1.326538439	
3	40	0.697778	0.642516	0.985866564	

Now we have to fill Column E.

The output of a FIR lowpass filter is defined as

$$y[n]=c_0 x[n] + c_1 x[n-1] + ...+b_N x[n-N]$$

Here, $c_0=0.151365$, $c_1=0.187098$, $c_2=0.2$, $c_3=0.187098$, $c_4=0.151365$

Also, $N=5$ (fifth order filter). So we can start with 5th sample.

$$y[5]= 0.151365* x[5] + 0.187098* x[4] + 0.2* x[3] + 0.187098* x[2] + 0.151365* x[1]$$

$$y[6]= 0.151365* x[6] + 0.187098* x[5] + 0.2* x[4] + 0.187098* x[3] + 0.151365* x[2]$$

Etc.

To do these in a single step type this and press enter. Select the cell double click on bottom right corner (this is a basic Excel operation).

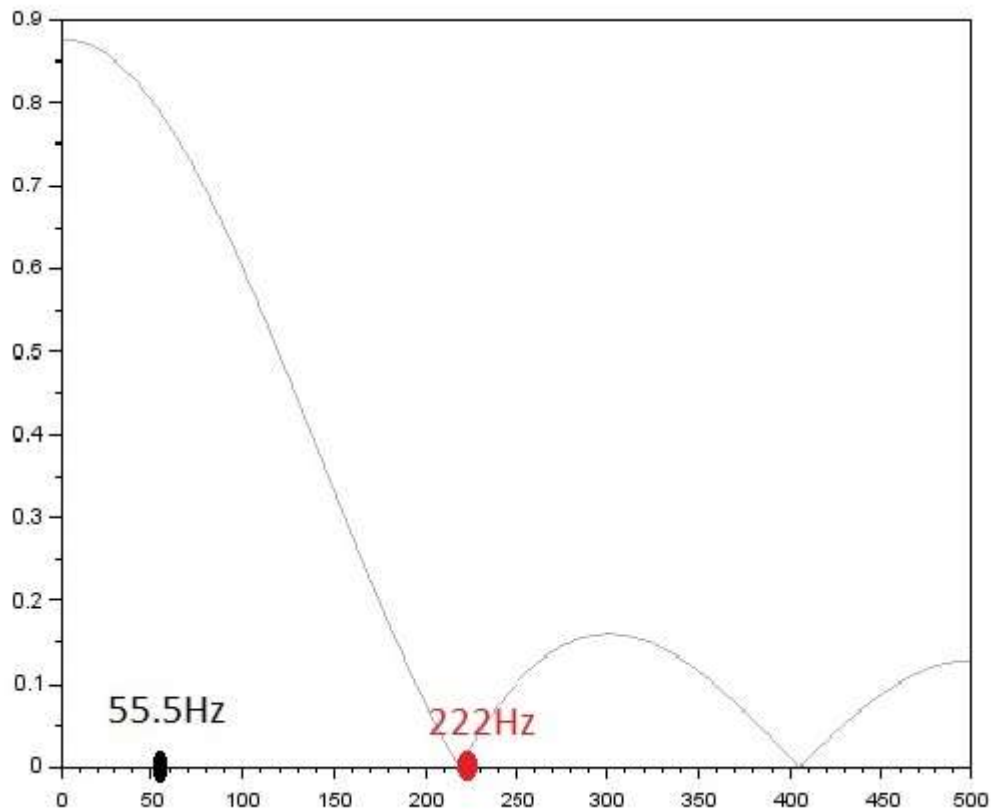
SUM										
X ✓ f _x =(E6*0.151365+E5*0.187098+E4*0.2+E3*0.187098+E2*0.151365)										
	A	B	C	D	E	F	G	H	I	J
1	Serial no.	Angle in degree	Angle in radian(x)	sin(x)	sin(x)+sin(4x)	Filter output				
2	1	0	0	0	0					
3	2	20	0.348889	0.341854	1.326538439					
4	3	40	0.697778	0.642516	0.985866564					
5	4	60	1.046667	0.86576	0.000798157					
6	5	80	1.395556	0.984685	0.339730594	=(E6*0.151365+E5*0.187098+E4*0.2+E3*0.187098+E2*0.151365)				
7	6	100	1.744444	0.984961	1.625033395					
8	7	120	2.093333	0.866556	1.734696925					
9	8	140	2.442222	0.643736	0.306376179					
10	9	160	2.791111	0.34335	-0.642425172					
11	10	180	3.14	0.001593	-0.004777918					
12	11	200	3.488889	-0.34036	0.643197216					

Then it will look like this.

O19							
X ✓ f _x							
	A	B	C	D	E	F	G
1	Serial no.	Angle in degree	Angle in radian(x)	sin(x)	sin(x)+sin(4x)	Filter output	
2	1	0	0	0	0		
3	2	20	0.348889	0.341854	1.326538439		
4	3	40	0.697778	0.642516	0.985866564		
5	4	60	1.046667	0.86576	0.000798157		
6	5	80	1.395556	0.984685	0.339730594	0.496939	
7	6	100	1.744444	0.984961	1.625033395	0.694941	
8	7	120	2.093333	0.866556	1.734696925	0.783934	
9	8	140	2.442222	0.643736	0.306376179	0.759623	
10	9	160	2.791111	0.34335	-0.642425172	0.662485	
11	10	180	3.14	0.001593	-0.004777918	0.510887	
12	11	200	3.488889	-0.34036	0.643197216	0.287873	
13	12	220	3.837778	-0.6413	-0.291968792	0.00137	
14	13	240	4.186667	-0.86496	-1.72670882	-0.28549	
15	14	260	4.535556	-0.98441	-1.634215105	-0.5092	
16	15	280	4.884444	-0.98523	-0.350070147	-0.66136	
17	16	300	5.233333	-0.86735	0.003935771	-0.759	
18	17	320	5.582222	-0.64495	-0.976309909	-0.78418	
19	18	340	5.931111	-0.34485	-1.331671489	-0.69631	
20	19	360	6.28	-0.00319	-0.015926186	-0.49908	
21	20	380	6.628889	0.338859	1.32124207	-0.25139	

Assume a sampling frequency of 1 KHz (sampling interval=1mS). So it can sample up to 500 KHz sine wave information without distortion (Nyquist Theorem). That is why we made the magnitude plot for 500 points.

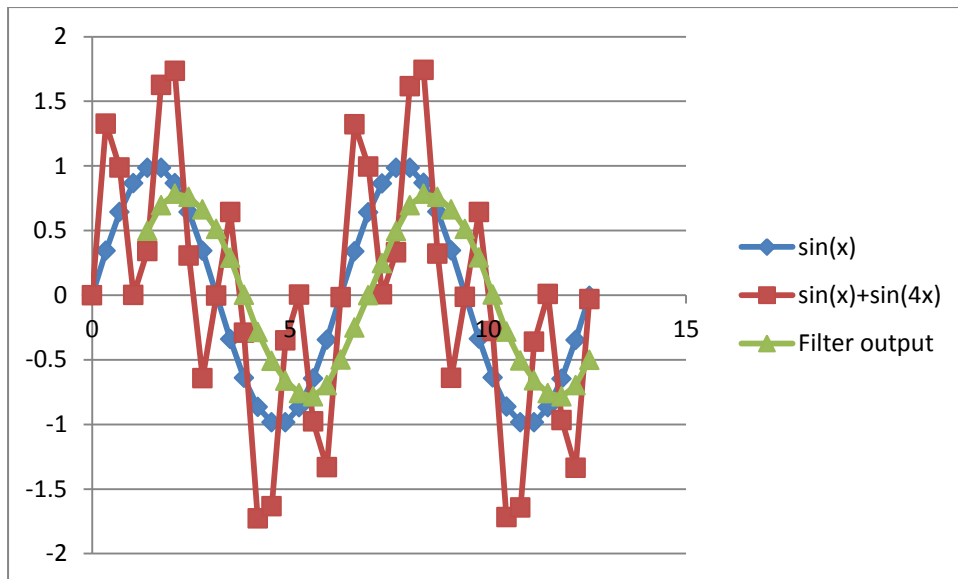
The time period of our $\sin(x)$ is 18 samples=18mS (see the table). So the frequency of $\sin(x)$ is 55.5Hz. Then frequency of $\sin(4x)$ is 222Hz.



We can see that 55.5Hz is in pass band of filter and 222Hz is in stop band.

Now plot the data obtained from excel table.

Select Column C, D, E and F. Go to Insert->Scatter->select any type.



Blue line is showing pure sine wave. Red line shows $\sin(x) + \sin(4x)$. Green line shows our filter input.

We can see that Filter output contains only 55.5Hz component, the 222Hz component is filtered off. The small phase shift is expected.