Signal Processing

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Based on

Signal Processing with Free Software : Practical Experiments
F. Auger

filter (1)

```
: y = filter (b, a, x)
: [y, sf] = filter (b, a, x, si)
: [y, sf] = filter (b, a, x, [], dim)
: [y, sf] = filter (b, a, x, si, dim)
```

https://octave.sourceforge.io/octave/function/filter.html

filter (2)

Apply a 1-D digital filter to the data x.

filter returns the solution to the following linear, time-invariant difference equation:

$$\sum_{k=0}^{N} a(k+1)\, y(n-k) = \sum_{k=0}^{M} b(k+1)\, x(n-k) \qquad for \;\; 1 \le n \le length(x)$$

where N=length(a)-1 and M=length(b)-1.

$$\boldsymbol{a} = [a(1), a(2), \cdots, a(N+1)]$$
$$\boldsymbol{b} = [b(1), b(2), \cdots, b(M+1)]$$
$$length(\boldsymbol{a}) = N+1$$
$$length(\boldsymbol{b}) = M+1$$

$$\boldsymbol{x} = [x(1), x(2), \cdots, x(L+1)]$$
$$length(\boldsymbol{x}) = L+1$$

$$1 \le n \le L+1$$

https://octave.sourceforge.io/octave/function/filter.html

filter (3)

The result is calculated over the **first** non-singleton dimension of x or over **dim** if supplied.

$$x(1),\quad x(2),\quad x(3),\quad \cdots \quad,\quad x(K)$$

$$x(1),\quad x(2),\quad x(3),\quad \cdots \quad,\quad x(K)$$

$$\begin{array}{cccccc}
x(1,1), & x(1,2), & x(1,3), & \cdots & , & x(1,K) \\
x(2,1), & x(2,2), & x(2,3), & \cdots & , & x(2,K) \\
x(3,1), & x(3,2), & x(3,3), & \cdots & , & x(3,K) \\
 & & & & & \\
x(L,1), & x(L,2), & x(L,3), & \cdots & , & x(L,K) \\
 & & & & & \\
x(:,1), & x(:,2), & x(:,3), & \cdots & , & x(:,K)
\end{array}$$

$$\begin{array}{cccccc}
y(1,1), & y(1,2), & y(1,3), & \cdots & , & y(1,K) \\
y(2,1), & y(2,2), & y(2,3), & \cdots & , & y(2,K) \\
y(3,1), & y(3,2), & y(3,3), & \cdots & , & y(3,K) \\
 & & & & & \\
y(L,1), & y(L,2), & y(L,3), & \cdots & , & y(L,K) \\
 & & & & & \\
y(:,1), & y(:,2), & y(:,3), & \cdots & , & y(:,K)
\end{array}$$

https://octave.sourceforge.io/octave/function/filter.html

filter (4)

$$\sum_{k=0}^{N} a(k+1)\,y(n-k) \;=\; \sum_{k=0}^{M} b(k+1)\,x(n-k) \qquad \mathit{for}\;\; 1 \le n \le \mathit{length}(x)$$

$$a(1)\,y(n) + \sum_{k=1}^{N} a(k+1)\,y(n-k) \;=\; \sum_{k=0}^{M} b(k+1)\,x(n-k)$$

$$a(1)\,y(n) \;=\; -\sum_{k=1}^{N} a(k+1)\,y(n-k) + \sum_{k=0}^{M} b(k+1)\,x(n-k)$$

$$y(n) \;=\; -\sum_{k=1}^{N} \frac{a(k+1)}{a(1)}\,y(n-k) + \sum_{k=0}^{M} \frac{b(k+1)}{a(1)}\,x(n-k)$$

$$y(n) \;=\; -\sum_{k=1}^{N} c(k+1)\,y(n-k) + \sum_{k=0}^{M} d(k+1)\,x(n-k) \qquad \mathit{for}\, 1 \le n \le \mathit{length}(x)$$

where c = a/a(1) and d = b/a(1).

https://octave.sourceforge.io/octave/function/filter.html

filter (5)

si : the initial state of the system
sf : the final state

the state vector is a column vector
whose length is equal to the length of
the longest coefficient vector – 1

No **si** is presented, the zero initial state.

in terms of the z transform,
y is the result of passing the discrete-time signal **x**
through a system characterized
by the following rational system function:

$$H(z) = \frac{\sum_{k=0}^{M} d(k+1) z^{-k}}{1 + \sum_{k=1}^{N} c(k+1) z^{-k}}$$

https://octave.sourceforge.io/octave/function/filter.html

freqz (1)

: [**h**, **w**] = **freqz** (**b**, **a**, n, "whole")
: [**h**, **w**] = **freqz** (**b**)
: [**h**, **w**] = **freqz** (**b**, **a**)
: [**h**, **w**] = **freqz** (**b**, **a**, n)
: **h** = **freqz** (**b**, **a**, **w**)
: [**h**, **w**] = **freqz** (…, Fs)
: **freqz** (…)

https://octave.sourceforge.io/octave/function/freqz.html

freqz (2)

Return the complex frequency response **h**
of the rational **IIR** filter
with the numerator coefficients **b** and
the denominator coefficients **a**

The response is evaluated
at **n** angular frequencies between **0** and **2*pi**.

The output value **w** is a vector of the frequencies.

h : the frequency response vector
w : the frequency vector

https://octave.sourceforge.io/octave/function/freqz.html

freqz (3)

If **a** is omitted, the denominator is assumed to be **1**
(this corresponds to a simple **FIR** filter).

If n is omitted, a value of 512 is assumed.
For fastest computation, n should factor
into a small number of small primes.

If the fourth argument, "**whole**", is omitted
the response is evaluated
at frequencies between **0** and **pi**.

https://octave.sourceforge.io/octave/function/freqz.html

freqz (4)

freqz (**b**, **a**, **w**)

Evaluate the response
at the specific frequencies in the vector **w**.
The values for **w** are measured in radians.

freqz (...)

Plot the magnitude and phase response
of **h** rather than returning them.

https://octave.sourceforge.io/octave/function/freqz.html

freqz (5)

[…] = **freqz** (…, Fs)

Return frequencies in Hz instead of radians
assuming a sampling rate Fs.
If you are evaluating the response
at specific frequencies **w**,
those frequencies should be requested
in Hz rather than radians.

[**h**, **w**] = **freqz** (**b**, **a**, n, "whole", Fs)
[**h**, **w**] = **freqz** (**b**, Fs)
[**h**, **w**] = **freqz** (**b**, **a**, Fs)
[**h**, **w**] = **freqz** (**b**, **a**, n, Fs)
h = **freqz** (**b**, **a**, **w**, Fs)

https://octave.sourceforge.io/octave/function/freqz.html

freqz_plot

: **freqz_plot** (**w**, **h**)
: **freqz_plot** (**w**, **h**, freq_norm)

Plot the magnitude and phase response of **h**.

If the optional freq_norm argument is true,
the frequency vector **w** is in units of normalized radians.
If freq_norm is false, or not given,
then **w** is measured in Hertz.

https://octave.sourceforge.io/octave/function/freqz_plot.html

conv

: **conv** (**a**, **b**)
: **conv** (**a**, **b**, shape)

Convolve two vectors **a** and **b**.

The output convolution is a vector
with length equal to length (**a**) + length (**b**) - 1.
When **a** and **b** are the coefficient vectors of two polynomials,
the convolution represents
the coefficient vector of the product polynomial.

The optional shape argument may be
shape = "full"
Return the full convolution. (default)
shape = "same"
Return the central part of the convolution with the length(**a**).

https://octave.sourceforge.io/octave/function/conv.html

fftconv

: **fftconv** (**x**, **y**)
: **fftconv** (**x**, **y**, n)

Convolve two vectors using the FFT for computation.

c = **fftconv** (**x**, **y**) returns
a vector of length equal to length(**x**) + length(**y**) - 1

If **x** and **y** are the coefficient vectors of two polynomials,
the returned value is the coefficient vector of the product polynomial.

The computation uses the FFT
by calling the function **fftfilt**.

If the optional argument n is specified,
an n-point FFT is used.

https://octave.sourceforge.io/octave/function/fftconv.html

deconv

: **deconv** (**y**, **a**)

Deconvolve two vectors.

[**b**, **r**] = **deconv** (**y**, **a**) solves for **b** and **r** such that **y** = **conv** (**a**, **b**) + **r**.

If **y** and **a** are polynomial coefficient vectors,
b will contain the coefficients of the polynomial quotient and
r will be a remainder polynomial of lowest order.

https://octave.sourceforge.io/octave/function/deconv.html

Low Pass filter

```
t = 0: 1/100 : 1;
x = sin(2 * pi * t);
x  = (x > 0);
x = (x – 0.5) * 2;
xd = [x 0 0 0];
for i=1 : length(x)
     y(i) = xd(i) + xd(i+1) + xd(i+2) + xd(i+3)) / 4;
endfor
hold
plot(t, x)
plot(t, y, ‘m—’);
```

DSP for sound engineers (in Korean), J.W. Chae

Low Pass filter

```
ir = zeros(1, 44100);
ir(1:2) = 0.5;
irfft = abs(fft(ir));
irfft = irfft(1: 22050);
plot(irfft)
```

```
ir = zeros(1, 44100);
ir(1:3) = 0.333;
irfft = abs(fft(ir));
irfft = irfft(1: 22050);
plot(irfft)
```

```
ir = zeros(1, 44100);
ir(1:4) = 0.25;
irfft = abs(fft(ir));
irfft = irfft(1: 22050);
plot(irfft)
```

DSP for sound engineers (in Korean), J.W. Chae

Low Pass filter

```
h0=0.36281; h1= 0.28920; h2 = 0.12082;
sys = zeros(1, 44100);
sys(1)=h2; sys(2) = h1; sys(3)=h0; sys(4)=h1; sys(5)=h2;
sysft = abs( fft(sys) );
sysft = sysft(1: 44100/2);
plot(sysft)
```

DSP for sound engineers (in Korean), J.W. Chae

High Pass filter

```
ir=zeros(1, 44100);
Ir(1)=0.5;
Ir(2)=-0.5;
irfft=abs(fft(ir));
Irfft = irfft(1: 22050);
plot(irfft);
```

DSP for sound engineers (in Korean), J.W. Chae

High Pass filter

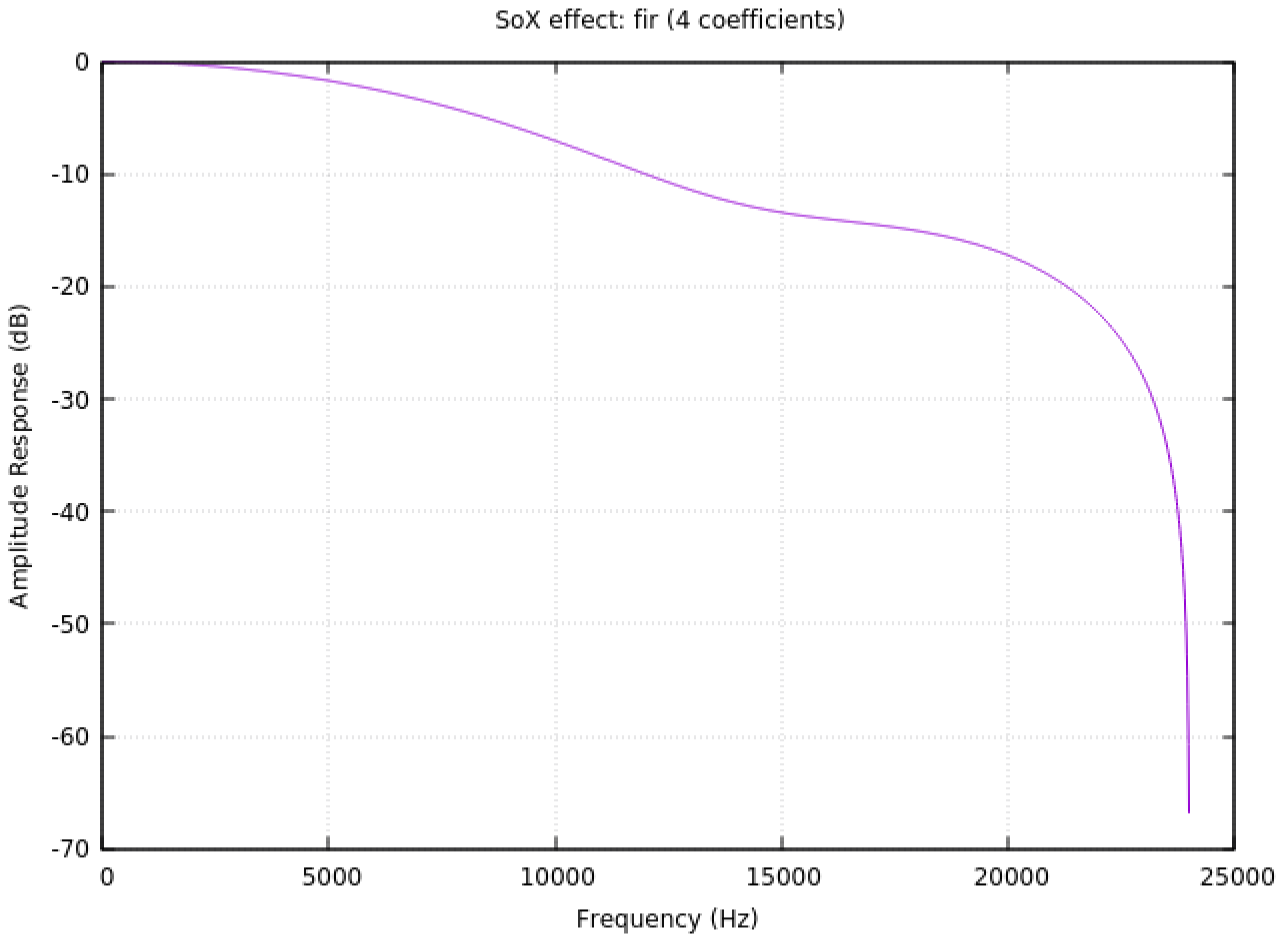
```
h0=0.63719; h1 = 0.28920; h2 = 0.12082;
sys = zeros(1, 44100);
sys(1) = h2; sys(2) =- h1; sys(3) = h0; sys(4)=-h1; sys(5)=h2;
sysft = abs(fft(sys));
sysft = sysft(1: 44100/2);
plot(sysft);
```

DSP for sound engineers (in Korean), J.W. Chae

--plot gnuplot | octave

```
sox --plot gnuplot s6s.wav -n fir 0.1 0.2 0.4 0.3          >fir1.plt
sox --plot gnuplot s6s.wav -n fir coeff.txt                >fir2.plt
sox --plot gnuplot s6s.wav -n biquad .6 .2 .4 1 -1.5 .6 >fir3.plt
sox --plot gnuplot s6s.wav -n fir 0.2 0.2 0.2 0.2 0.2      >fir4.plt
```

--plot gnuplot | octave



References

[1] F. Auger, Signal Processing with Free Software : Practical Experiments