

# Filter Design Results

Generated by: <http://www-users.cs.york.ac.uk/~fisher/mkfilter>

## Summary

You specified the following parameters:

filtertype = Butterworth  
passtype = Bandpass  
ripple =  
order = 1  
samplerate = 8000  
corner1 = 2000  
corner2 = 2500  
adzero =  
logmin =

## Results

Command line: /www/usr/fisher/helpers/mkfilter -Bu -Bp -o 1 -a 2.5000000000e-01 3.1250000000e-01  
raw alpha1 = 0.2500000000  
raw alpha2 = 0.3125000000  
warped alpha1 = 0.3183098862  
warped alpha2 = 0.4763844100  
gain at dc : mag = 0.000000000e+00  
gain at centre: mag = 6.026183138e+00 phase = 0.0062352617 pi  
gain at hf : mag = 0.000000000e+00

S-plane zeros:  
0.0000000000 + j 0.0000000000

S-plane poles:  
-0.4966057627 + j 2.3957891742  
-0.4966057627 + j -2.3957891742

Z-plane zeros:  
1.0000000000 + j 0.0000000000  
-1.0000000000 + j 0.0000000000

Z-plane poles:  
-0.1659106810 + j 0.8004075736  
-0.1659106810 + j -0.8004075736

Recurrence relation:  
$$y[n] = (-1 * x[n-2])$$
$$+ (0 * x[n-1])$$
$$+ (1 * x[n-0])$$
  
$$+ (-0.6681786379 * y[n-2])$$
$$+ (-0.3318213621 * y[n-1])$$

## Ansi ``C" Code

```
/* Digital filter designed by mkfilter/mkshape/gencode   A.J. Fisher
   Command line: /www/usr/fisher/helpers/mkfilter -Bu -Bp -o 1 -a 2.5000000000e-01 3.1250000000e-01 -l */

#define NZEROS 2
#define NPOLES 2
#define GAIN    6.026183138e+00

static float xv[NZEROS+1], yv[NPOLES+1];

static void filterloop()
{ for (;;)
  { xv[0] = xv[1]; xv[1] = xv[2];
    xv[2] = next input value / GAIN;
    yv[0] = yv[1]; yv[1] = yv[2];
    yv[2] = (xv[2] - xv[0])
            + ( -0.6681786379 * yv[0]) + ( -0.3318213621 * yv[1]);
    next output value = yv[2];
  }
}
```

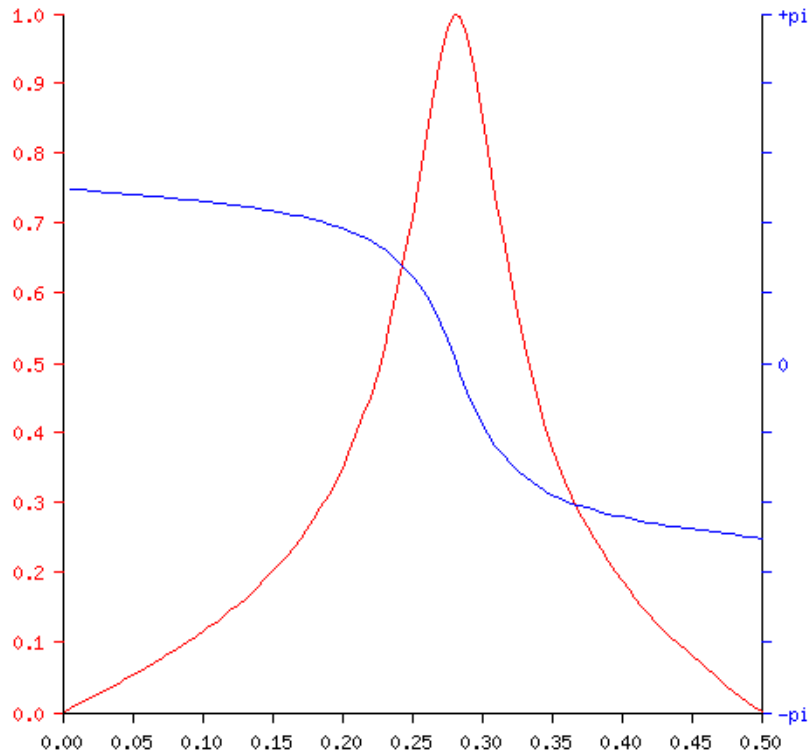
Download code and/or coefficients:

TERSE

VERBOSE

## Magnitude (red) and phase (blue) vs. frequency

- x axis: frequency, as a fraction of the sampling rate (i.e. 0.5 represents the Nyquist frequency, which is 4000 Hz)
- y axis (red): magnitude (linear, normalized)
- y axis (blue): phase

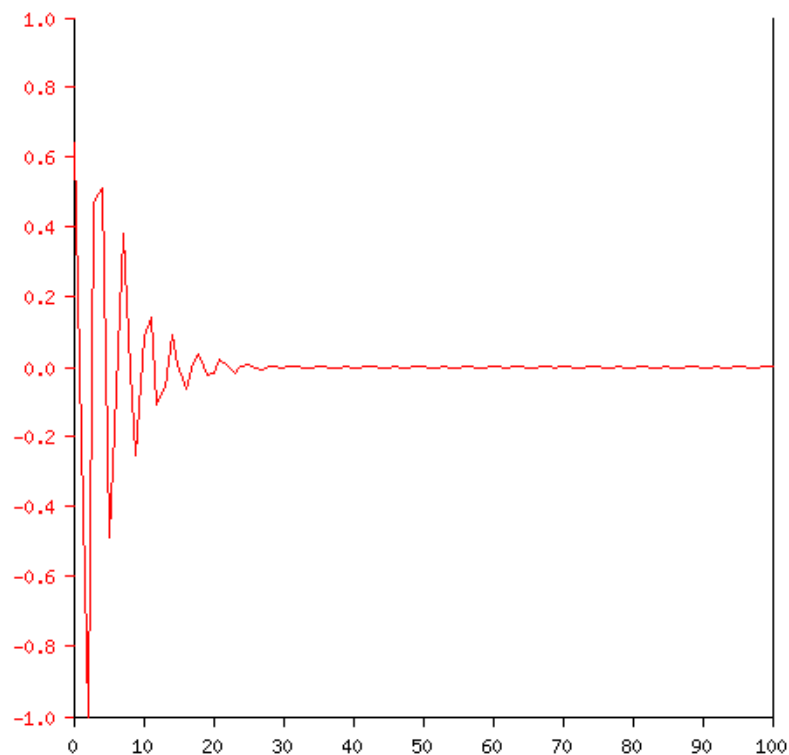


For an expanded view, enter frequency limits (as a fraction of the sampling rate) here:

Lower limit:  Upper limit:

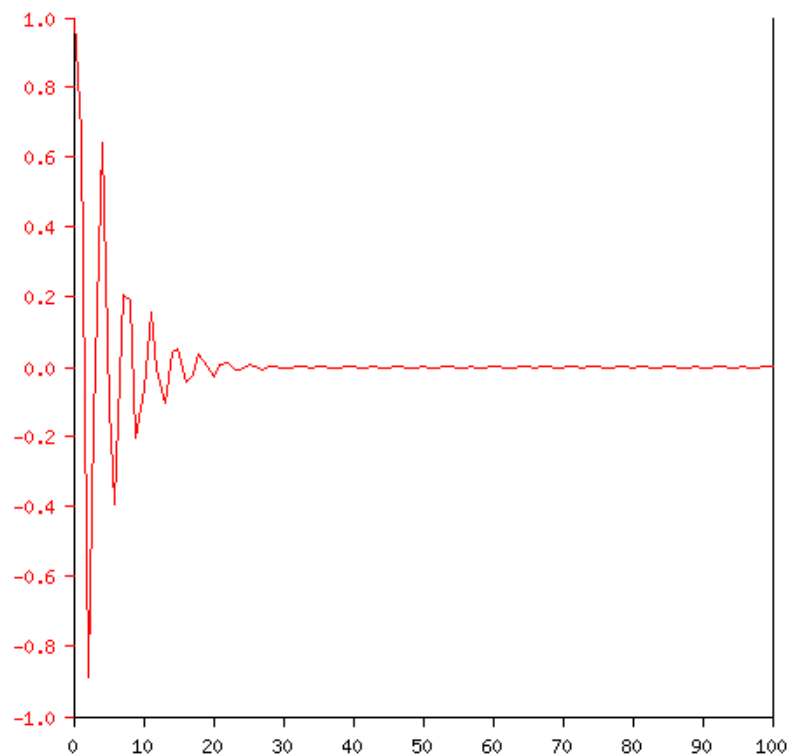
## Impulse response

- x axis: time, in samples (i.e. 8000 represents 1 second)
- y axis (red): filter response (linear, normalized)



## Step response

- x axis: time, in samples (i.e. 8000 represents 1 second)
- y axis (red): filter response (linear, normalized)



For a view on a different scale, enter upper time limit (integer number of samples) here:

Upper limit:

---

[Tony Fisher](#) [fisher@minster.york.ac.uk](mailto:fisher@minster.york.ac.uk)