

# Filtered Audio Demo

## Max Kamenetsky

In this demo you'll listen to a 10 second segment of music, alternating with various filtered versions of it. You should try to relate what you hear to the frequency response, impulse and step responses, and snapshots of the input and output signals.

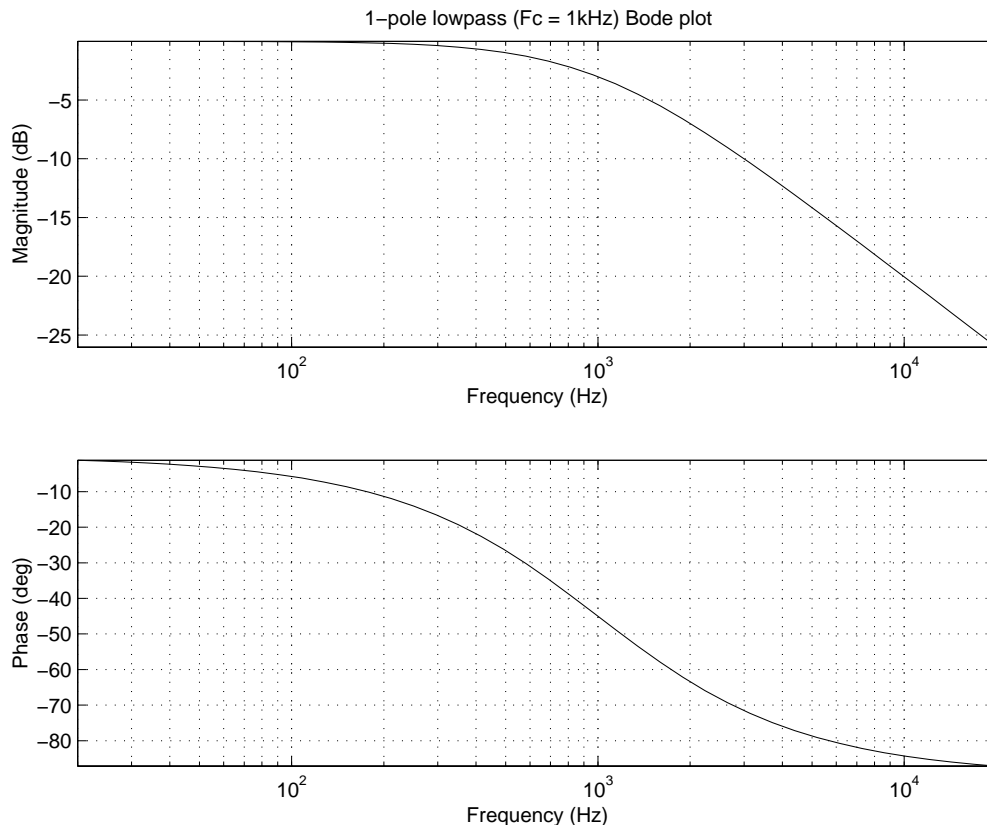
### First order lowpass filter

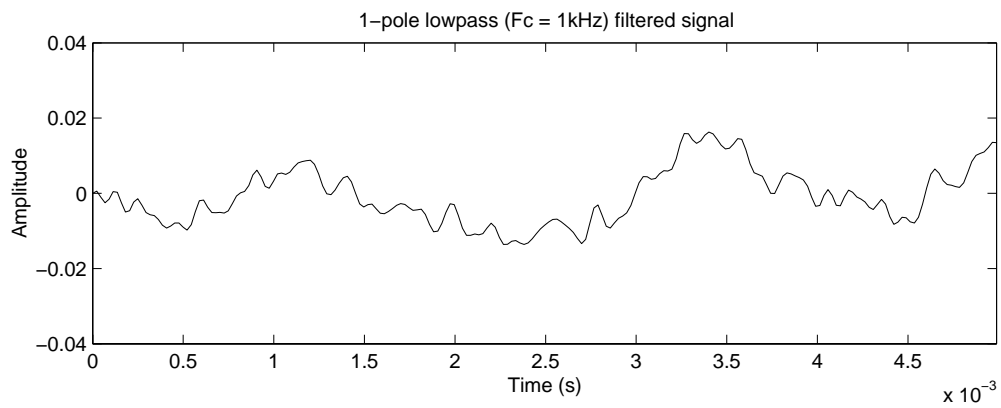
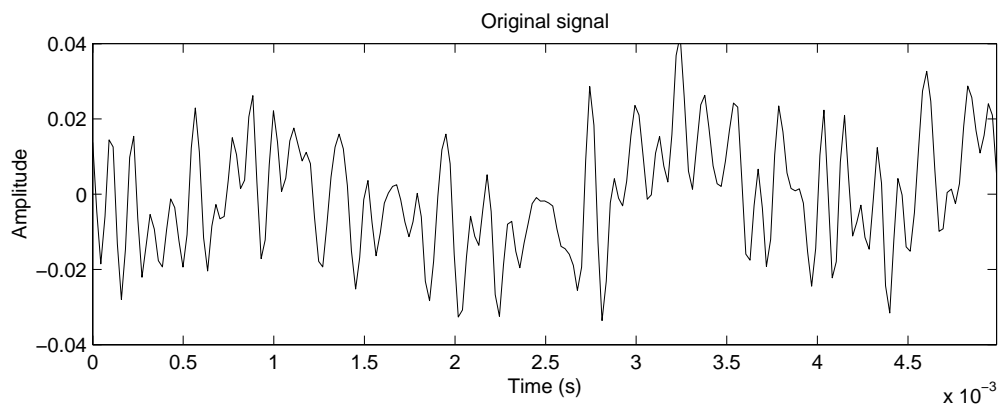
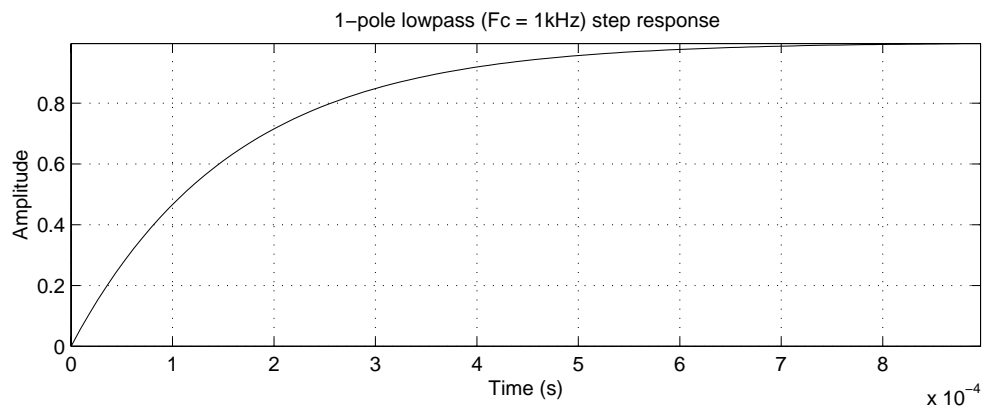
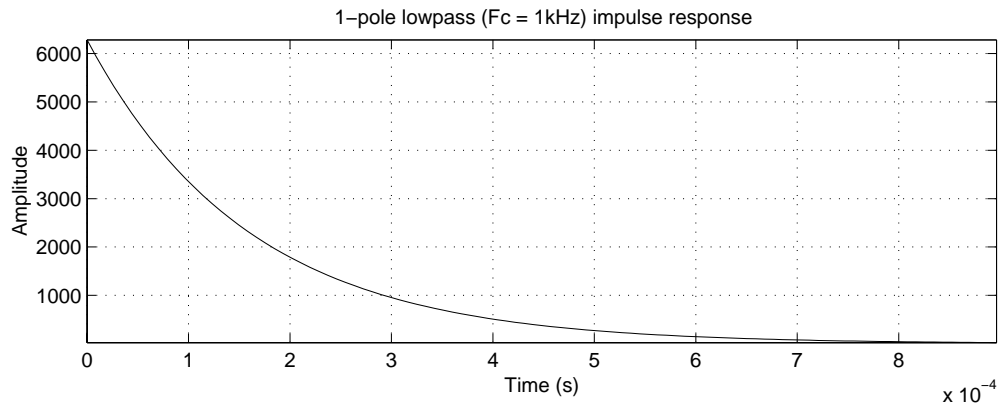
The first filter is a first order lowpass with cutoff frequency 1kHz, with transfer function

$$H(s) = \frac{\omega_c}{s + \omega_c} = \frac{1}{1 + s/\omega_c},$$

where  $\omega_c = 2\pi \cdot 1000$ .

Note that it passes frequencies under around 500Hz or so, but attenuates high frequencies. Since it attenuates high frequencies the filtered segment will sound a bit muffled. (A higher order lowpass filter, with a sharper cutoff characteristic, would sound much more muffled.) The impulse response shows that this filter smooths out the input, giving a sort of averaging over a few milliseconds. You can see that the filtered signal is a smoothed version of the original signal.





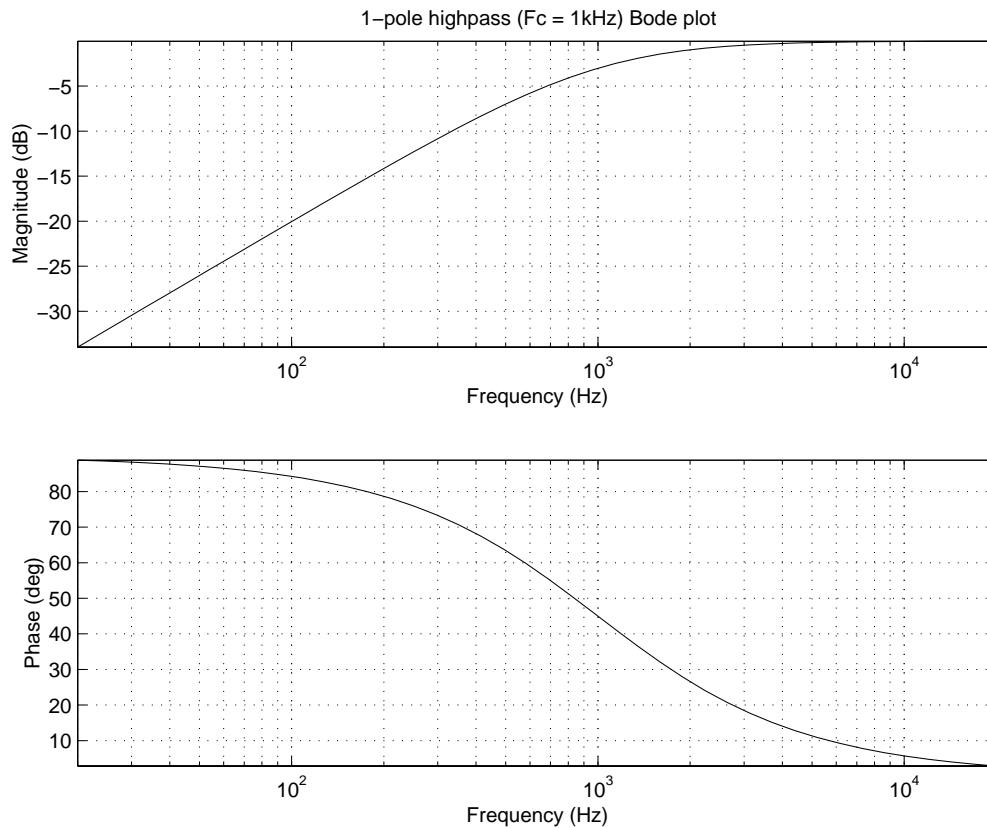
# First order highpass filter

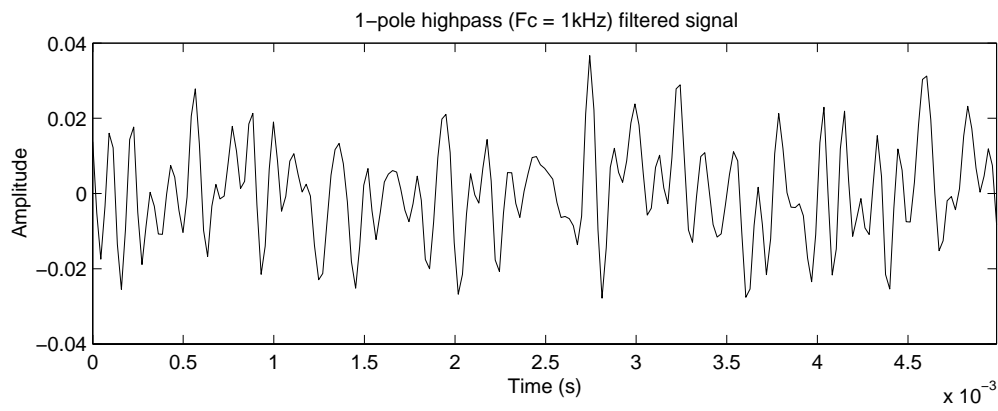
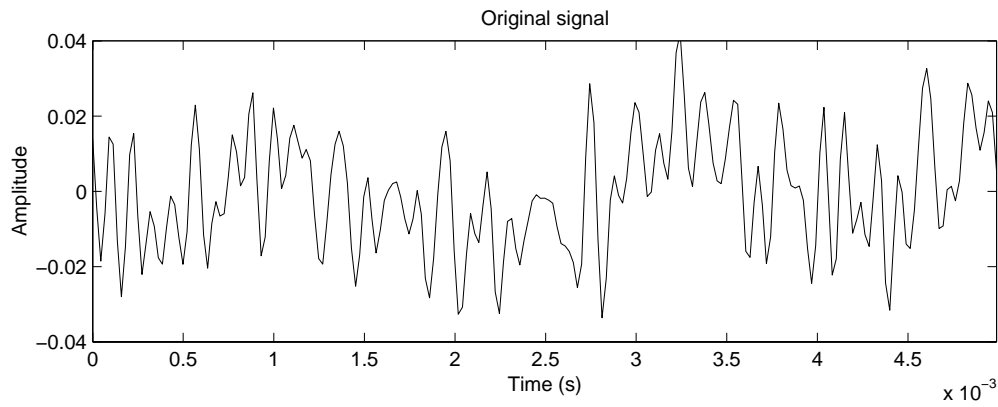
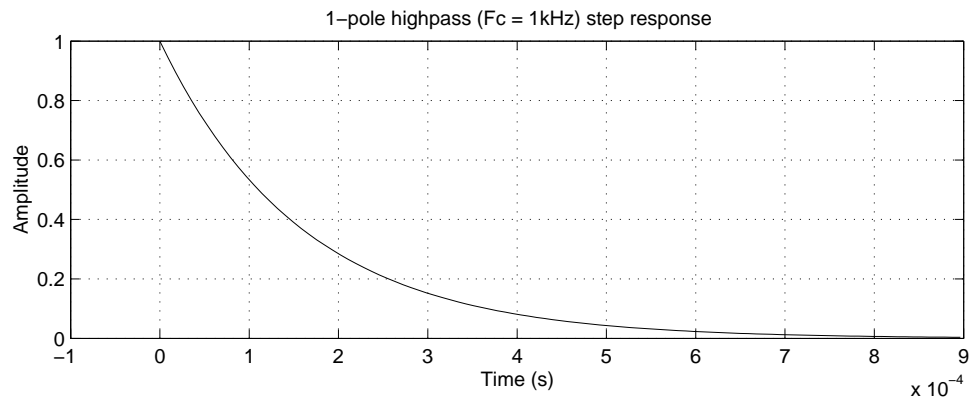
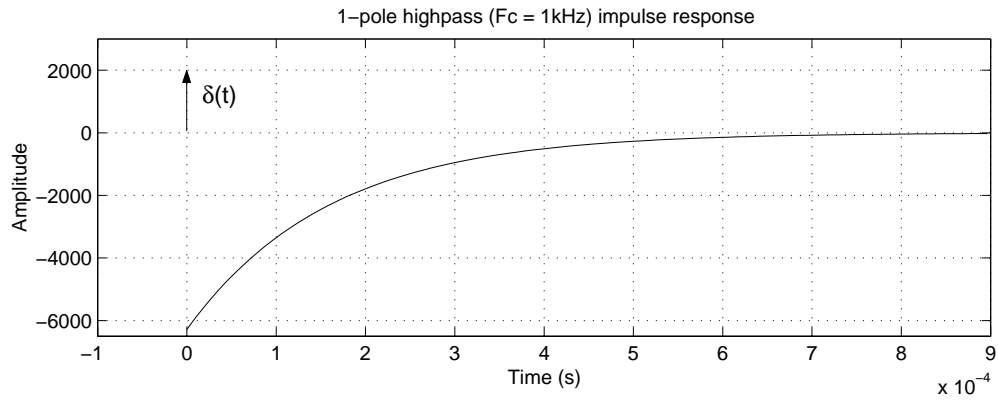
The transfer function is

$$H(s) = \frac{s}{s + \omega_c} = \frac{s/\omega_c}{1 + s/\omega_c}$$

where  $\omega_c = 2\pi \cdot 1000$ .

This transfer function attenuates low frequencies, but lets frequencies above 2kHz or so pass. You can hear that the bass is quite reduced. You can also see from the signal waveforms that the fast wiggles in the signal come through, but the slower variations are quite reduced.





## Low Q bandpass filter

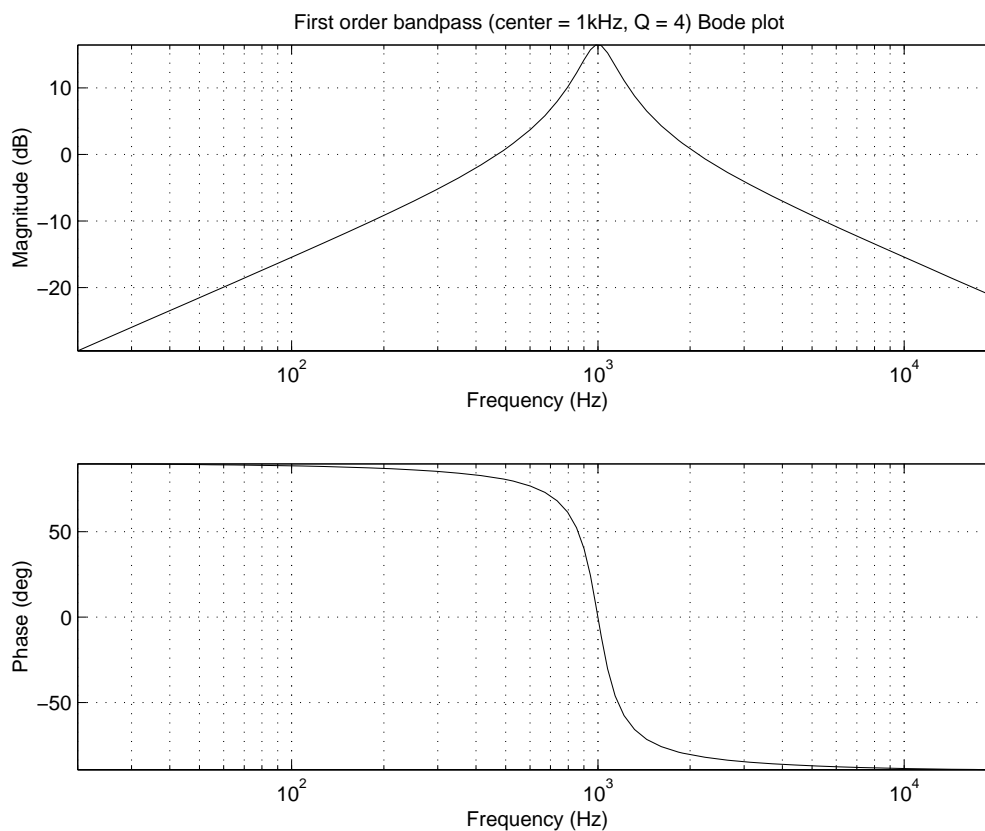
This is a second order bandpass filter, with center frequency 1 kHz and  $Q = 4$ . Transfer function:

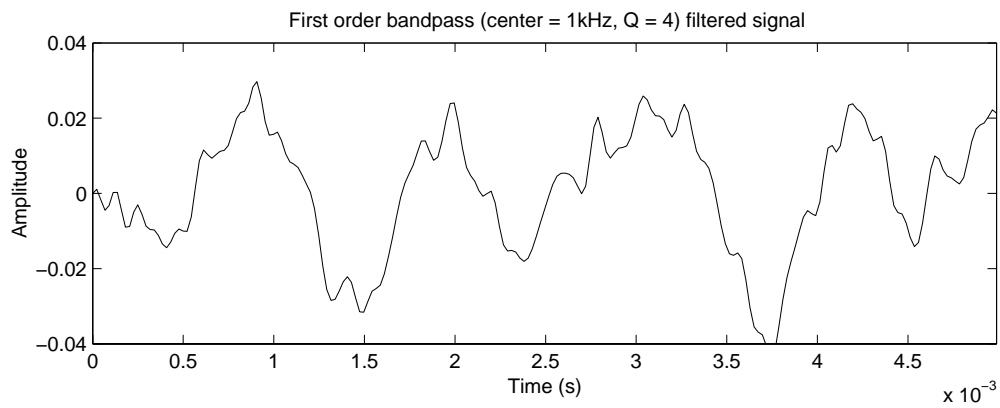
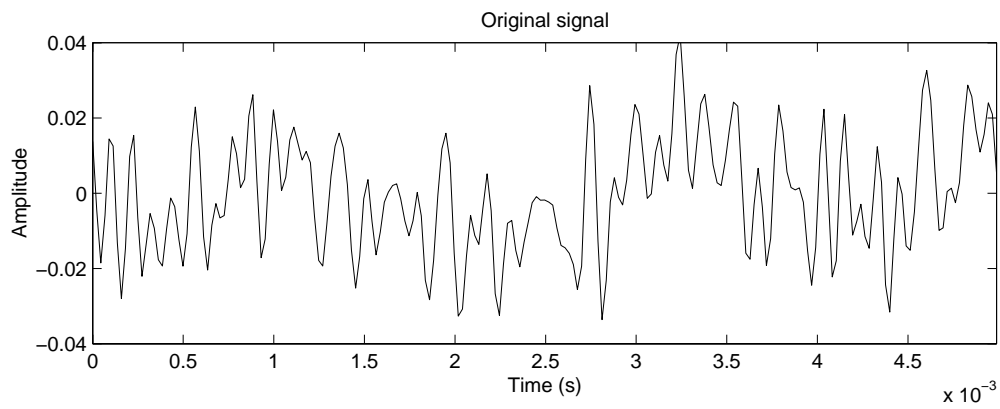
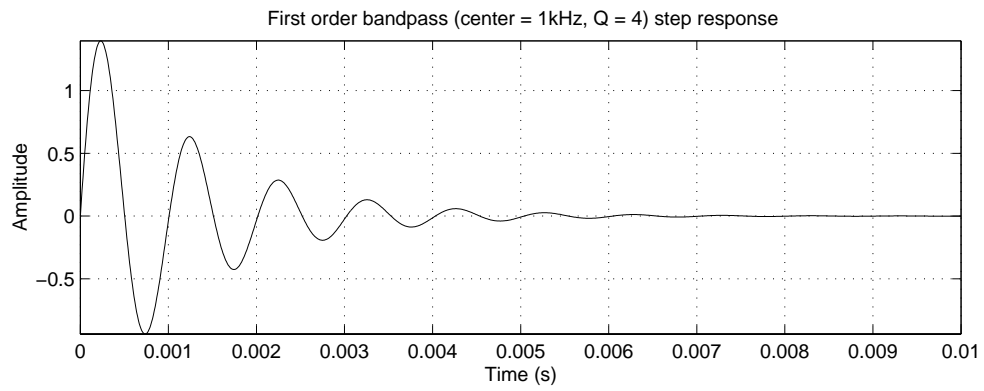
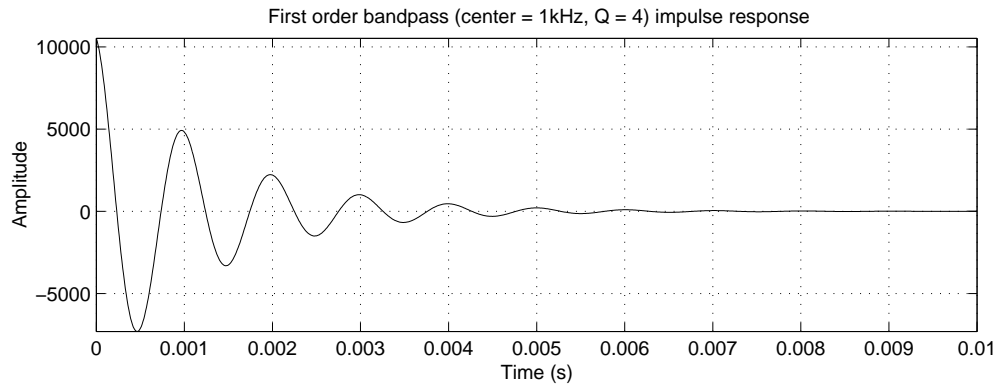
$$H(s) = K \frac{s}{(Q/\omega_c)s^2 + s + Q\omega_c}$$

where  $Q = 4$ ,  $\omega_c = 2\pi \cdot 1000$ , and  $K = 6.7 \approx 16.5$  dB is a scale factor to increase the volume level.

Here frequencies between 500Hz and 2kHz or so are passed, and both low and high frequencies are attenuated. The acoustic effect is something like listening through a tube (which has resonances — you'll learn about that in EE141!). You can also hear when a note (or a harmonic) gets near 1kHz.

The impulse response is oscillatory, and you can see the effect in the plot of the filtered signal. Here both the fast wiggles and the slow undulations are attenuated.





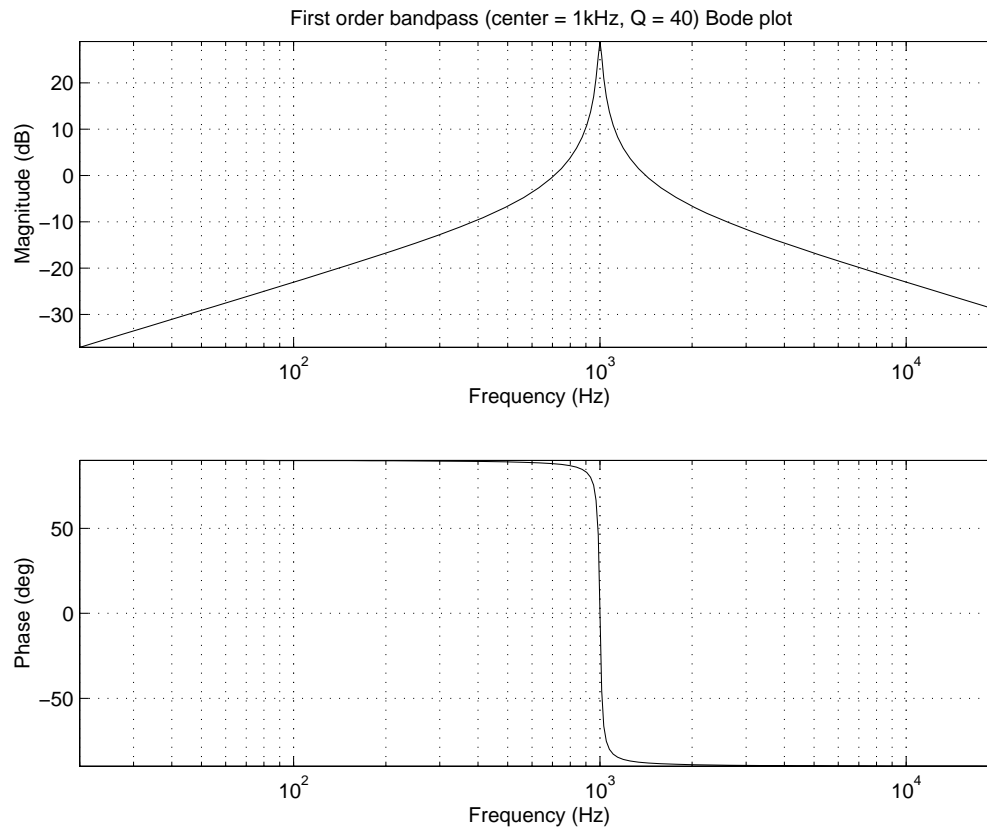
## High Q bandpass filter

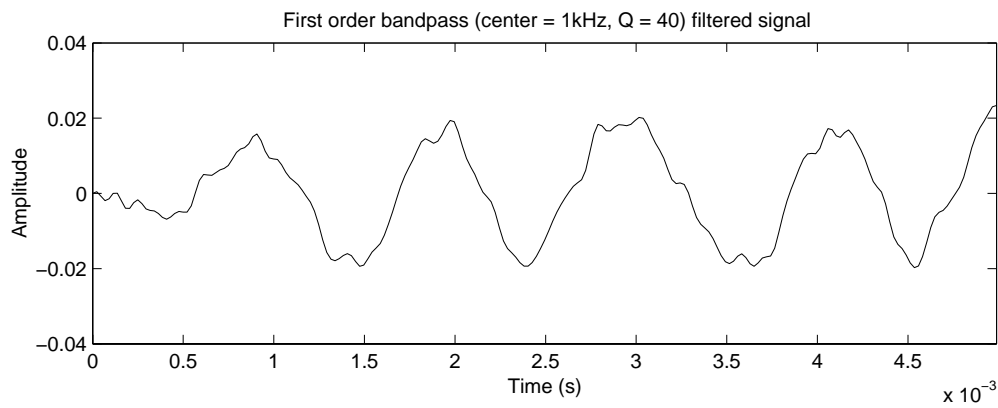
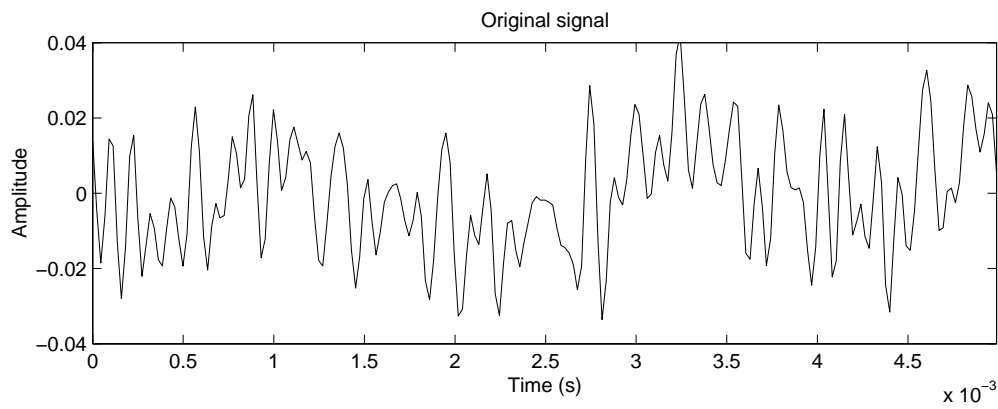
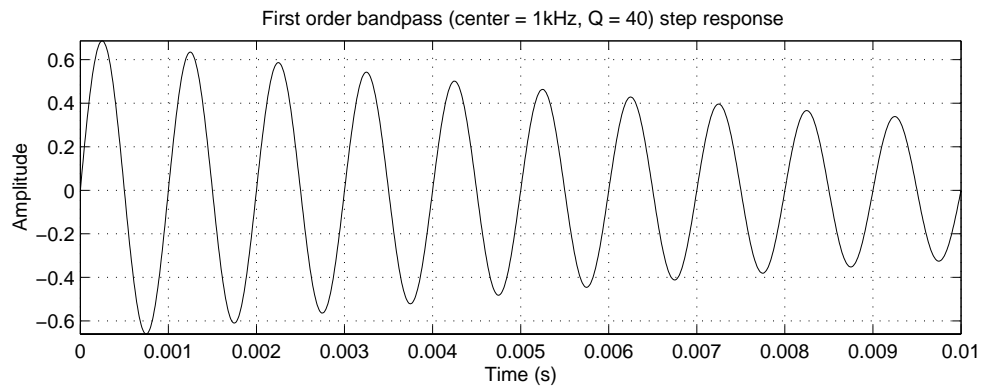
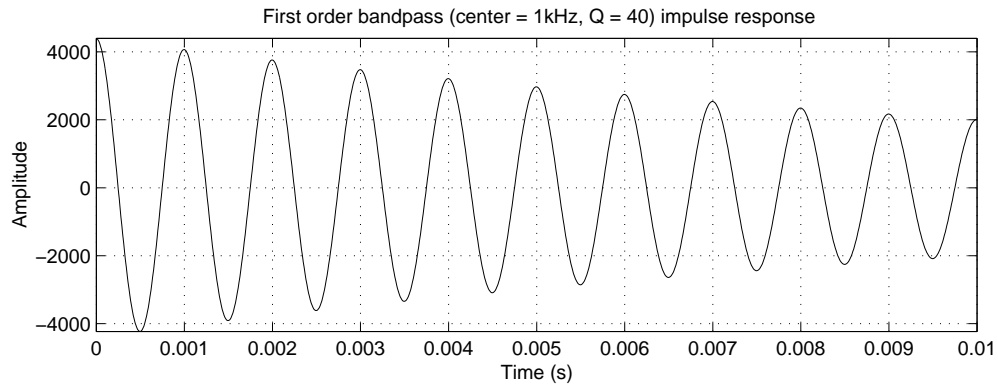
Second order bandpass filter with center frequency 1kHz and  $Q = 40$ . Transfer function:

$$H(s) = K \frac{s}{(Q/\omega_c)s^2 + s + Q\omega_c}$$

where  $Q = 40$ ,  $\omega_c = 2\pi \cdot 1000$ , and  $K = 28 \approx 29$  dB is a scale factor to increase the volume level.

Here the frequencies near 1kHz are strongly emphasized, which is quite annoying. The impulse response is quite oscillatory, which you can also see in the plots of the filtered signal.







# Allpass filter

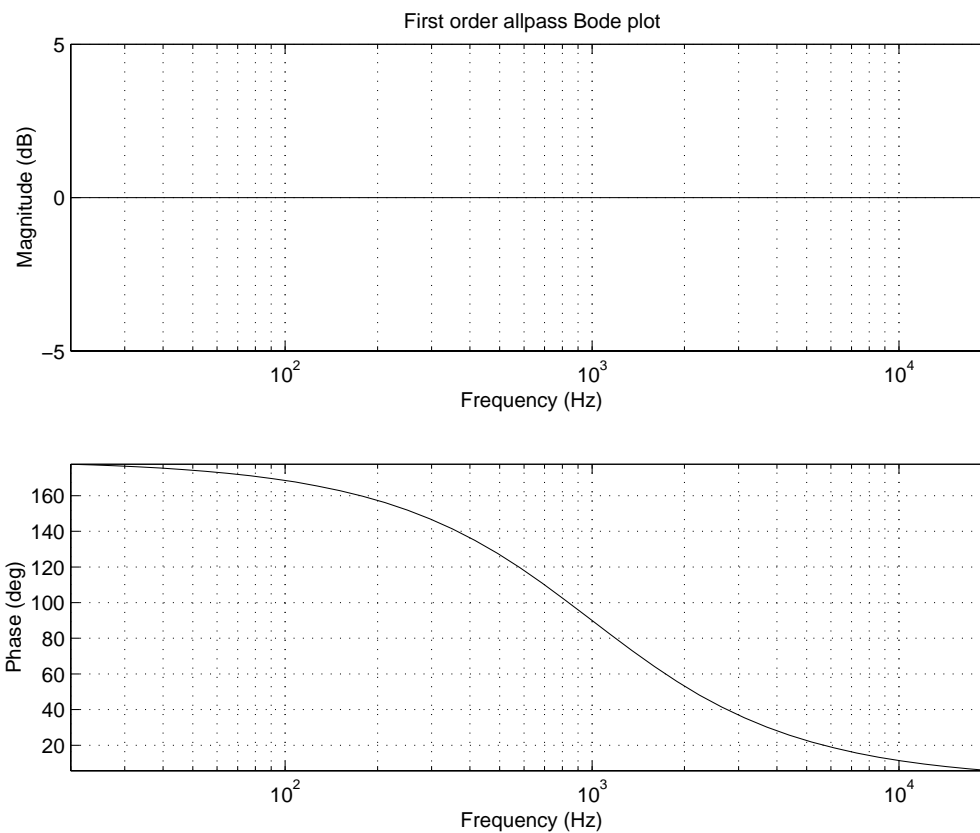
First order allpass filter with 90° phase shift at 1kHz. Transfer function:

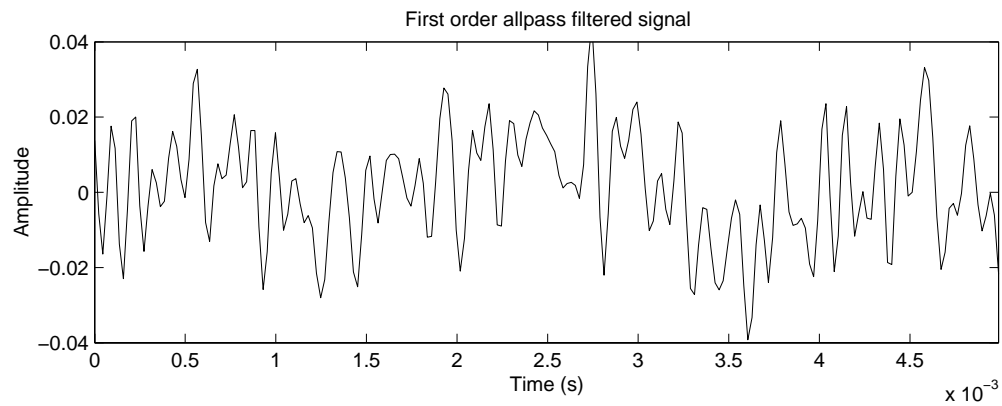
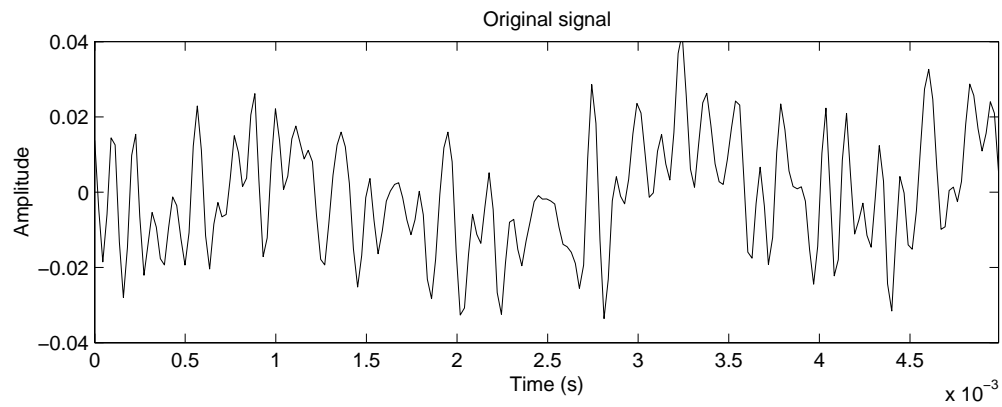
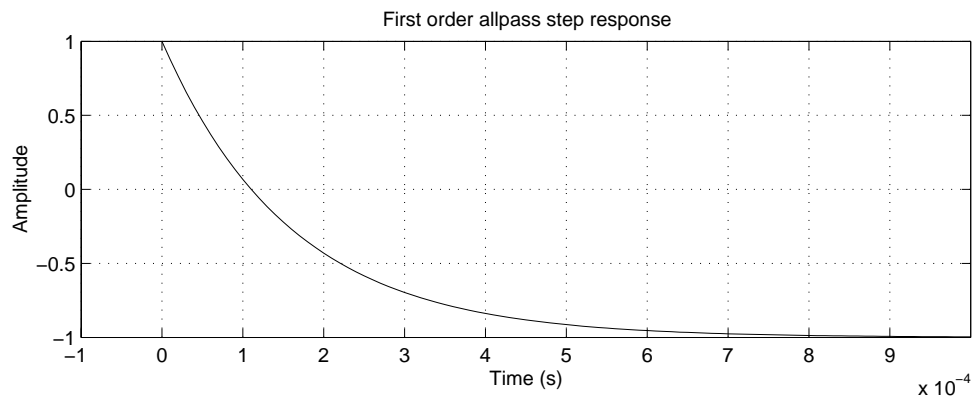
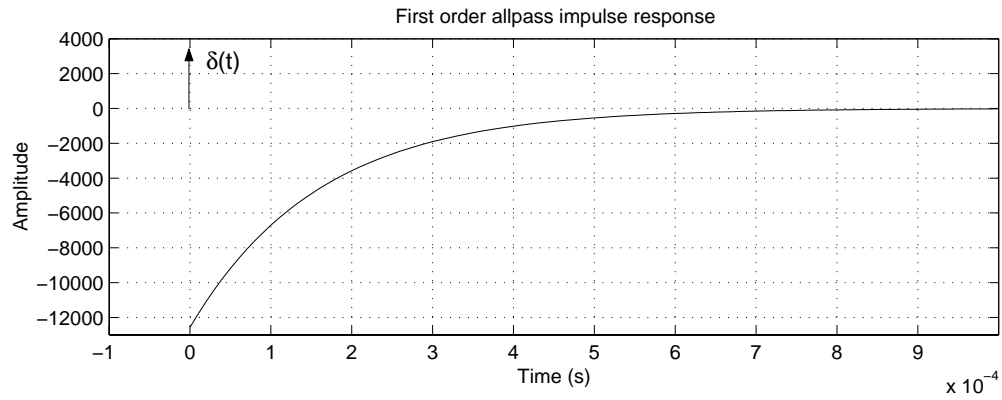
$$H(s) = \frac{s - \omega_c}{s + \omega_c} = \frac{s/\omega_c - 1}{s/\omega_c + 1},$$

where  $\omega_c = 2\pi \cdot 1000$ .

This filter passes all frequencies with unity gain. It does, however, shift the phase of the signal.

You really can't hear any difference at all, since the ear is pretty insensitive to (moderate) phase shift. The filtered signal looks quite a bit like the original signal, but is not the same.





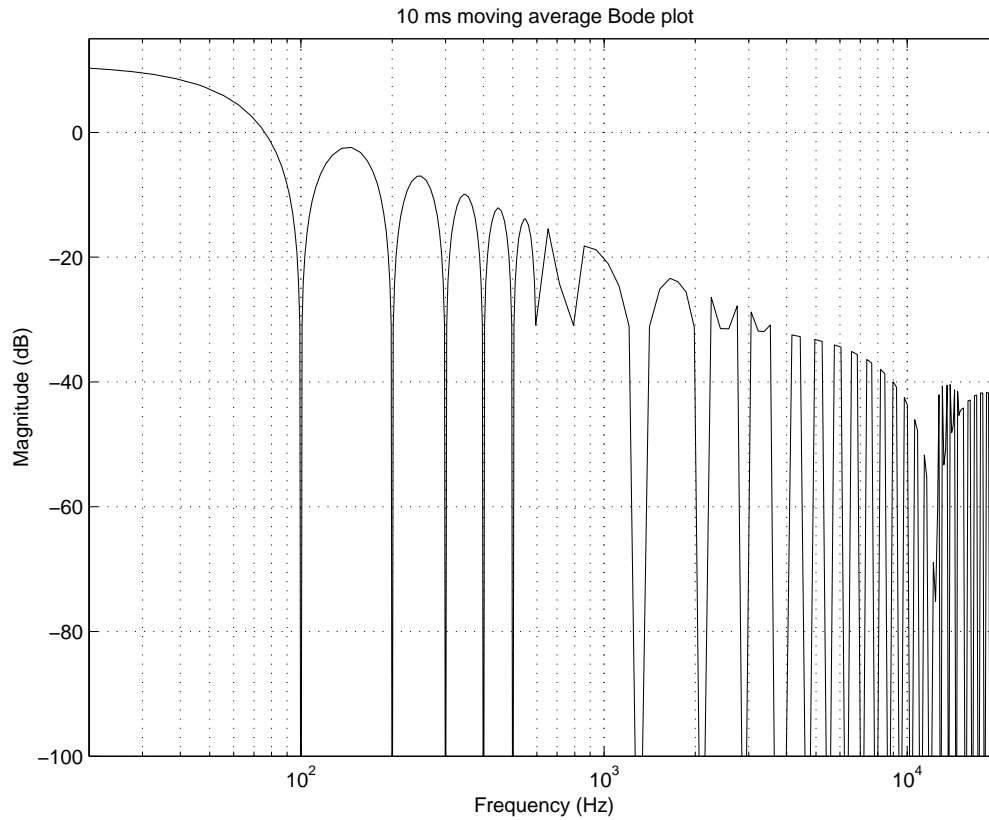
# Moving average filter

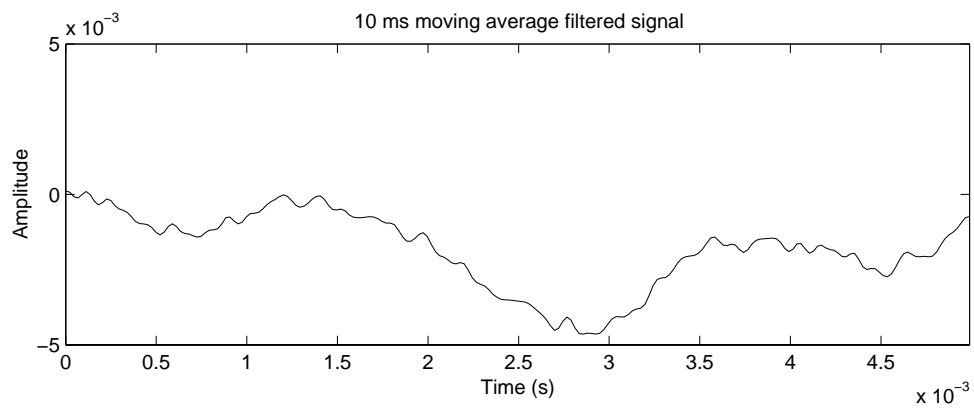
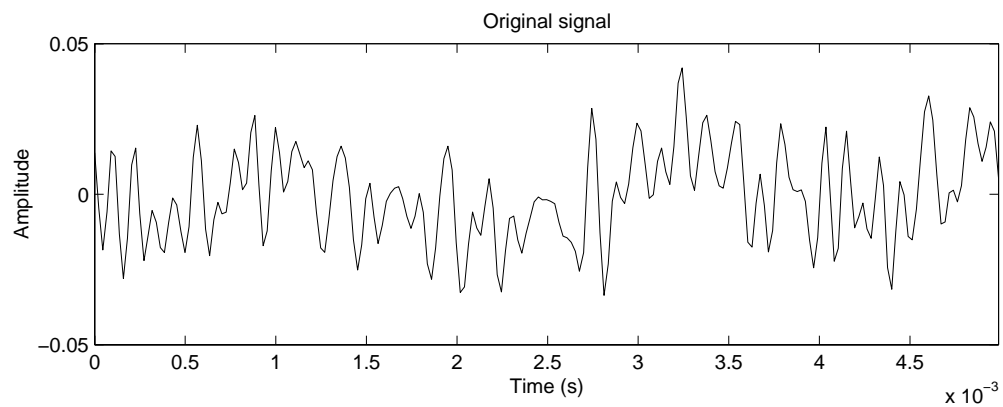
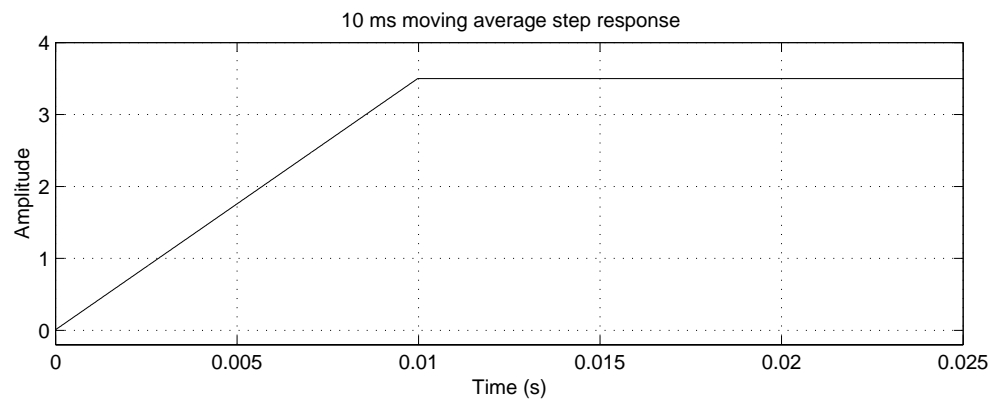
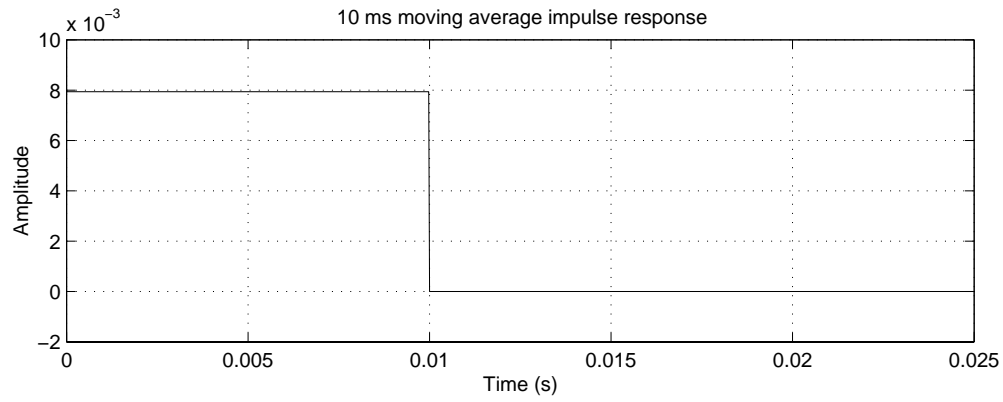
This filter is a 10ms moving average filter. Impulse response:

$$h(t) = \begin{cases} K(1/0.1), & 0 \leq t < 0.01 \\ 0, & t \geq 0.01 \end{cases}$$

The DC gain is  $K$ , which we take to be  $K = 3.5 \approx 10.9\text{dB}$  to increase the volume level.

In this case the filtered signal output is exactly ( $K$  times the) average of the input signal over the last 10ms. Here the filtered signal sounds very muffled; the high frequencies are strongly attenuated.





## Filter with echos

Impulse response:

$$\begin{aligned} h(t) = & \delta(t) + 0.75\delta(t - 0.125) - 0.1065\delta(t - 0.1536) \\ & - 0.4098\delta(t - 0.1605) + 0.0308\delta(t - 0.1788) + 0.0705\delta(t - 0.1934) \\ & - 0.2804\delta(t - 0.2201) + 0.2906\delta(t - 0.243) + 0.2898\delta(t - 0.2567) \end{aligned}$$

This represents 8 perfect echos. The first one arrives 125ms after the first impulse, and the others come over the next 125ms or so.

