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.

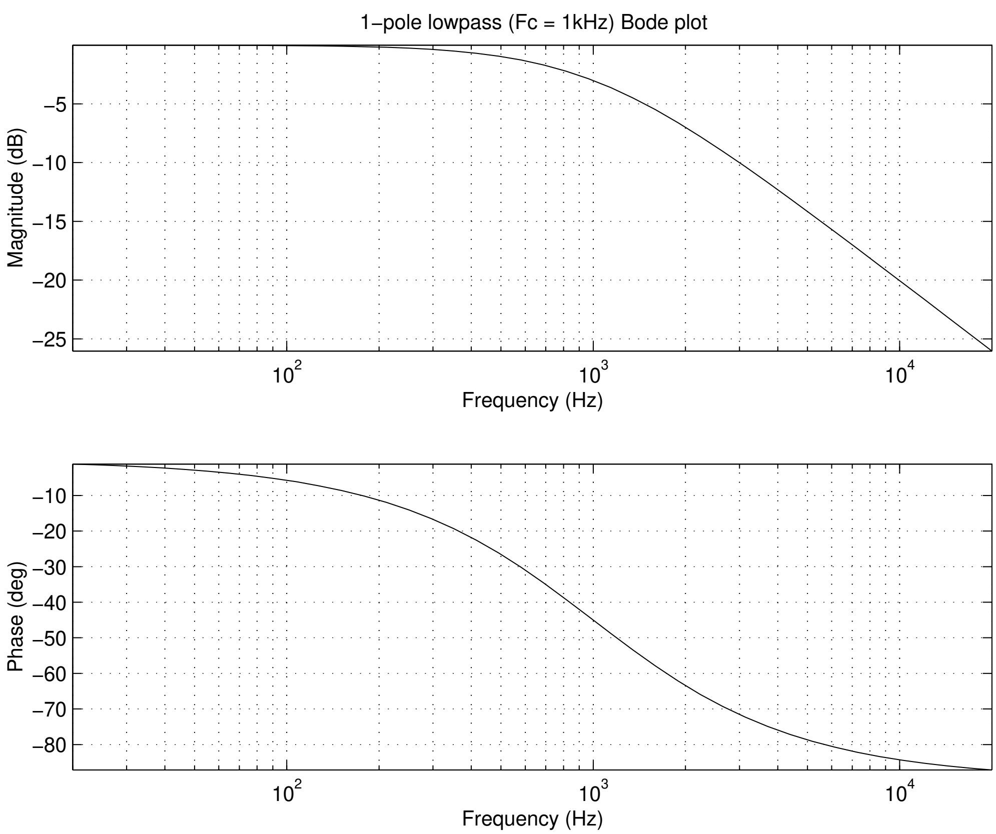
First order lowpass filter

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

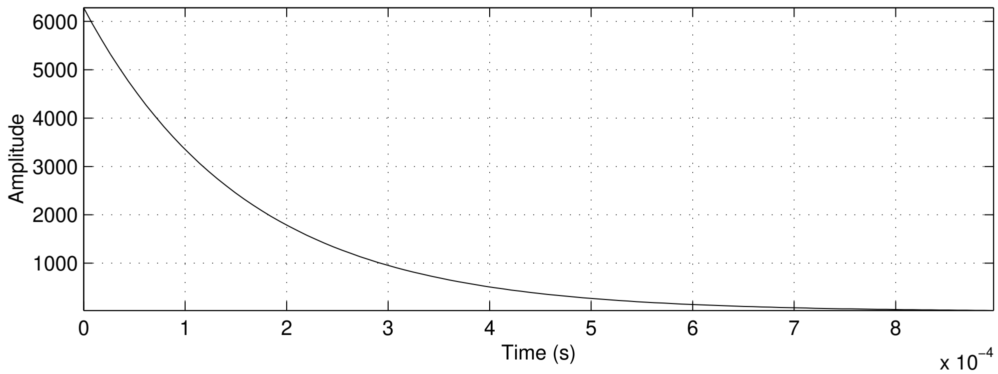
$$H(s) = \frac{\omega_c}{s + \omega_c} = \frac{1}{1 + s/\omega_c},$$

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

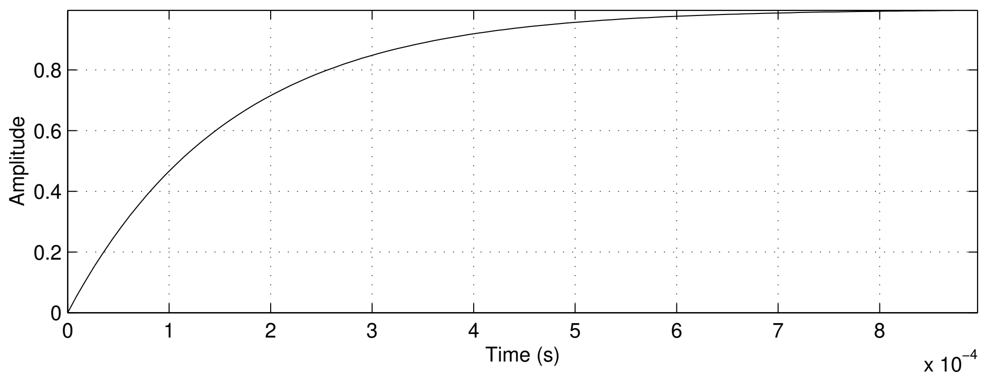
Note that it passes frequencies under around 500Hz or so, but attenuates high frequencies. Since it attenuates high frequencies the filtered segment will sound a bit muffled. (A higher order lowpass filter, with a sharper cutoff characteristic, would sound much more muffled.) The impulse response shows 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.



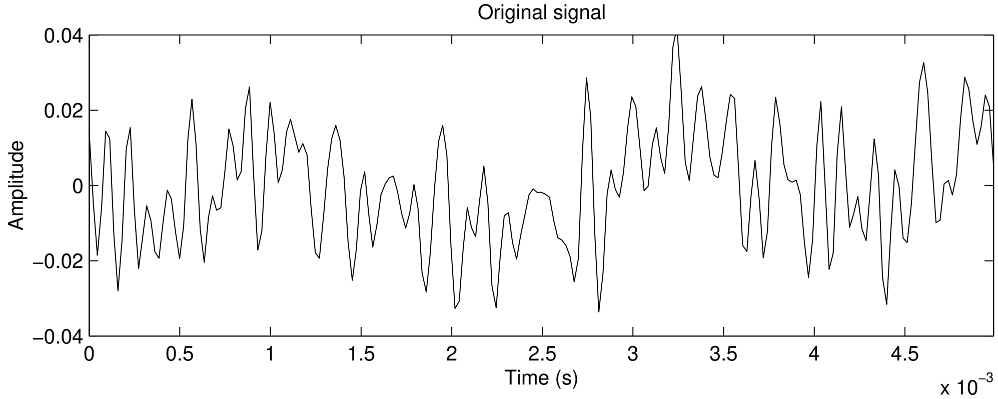
1−pole lowpass (Fc = 1kHz) impulse response



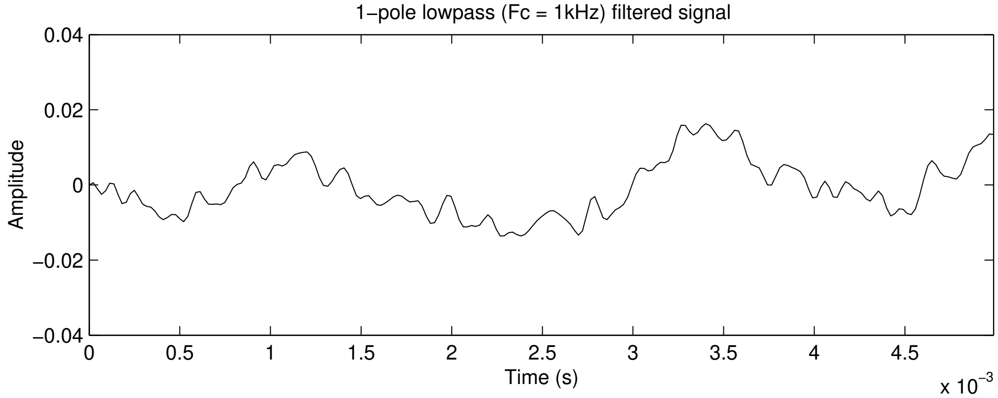
1−pole lowpass (Fc = 1kHz) step response



Original signal



1−pole lowpass (Fc = 1kHz) filtered signal



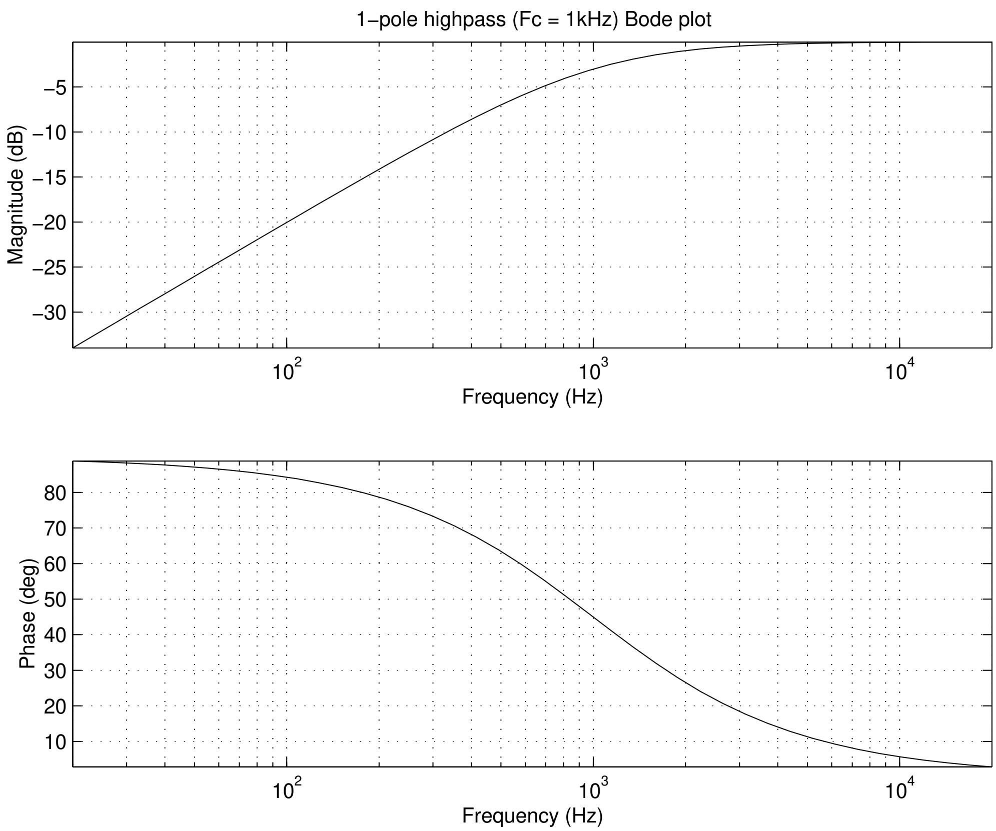
First order highpass filter

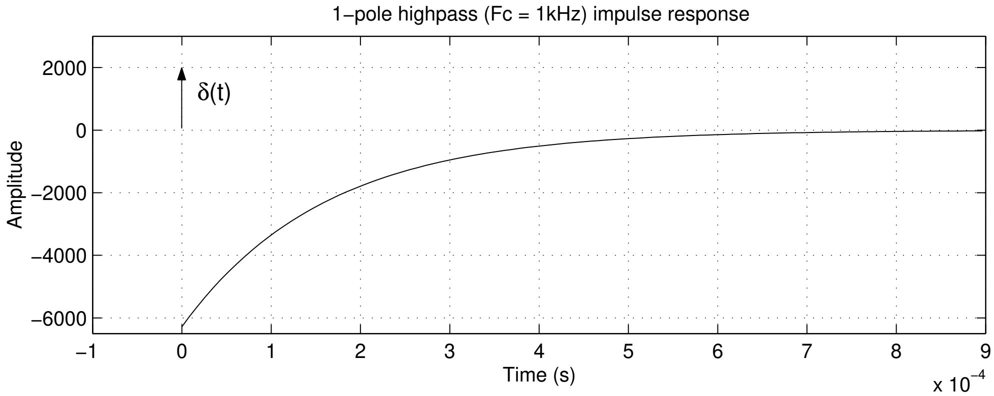
The transfer function is

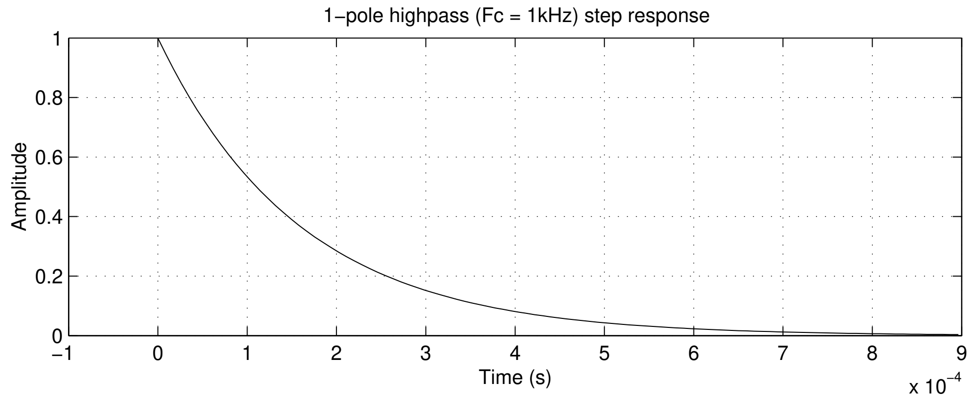
$$H(s) = \frac{s}{s+\omega_c} = \frac{s/\omega_c}{1+s/\omega_c}$$

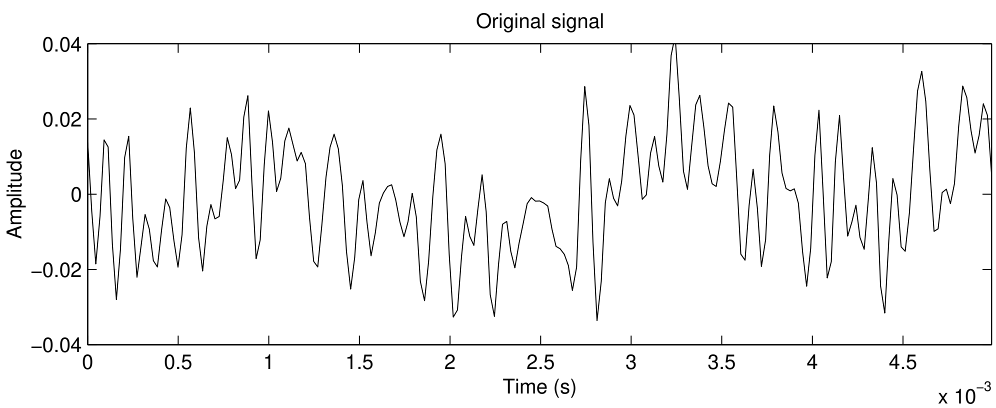
where $\omega_c = 2\pi \cdot 1000$.

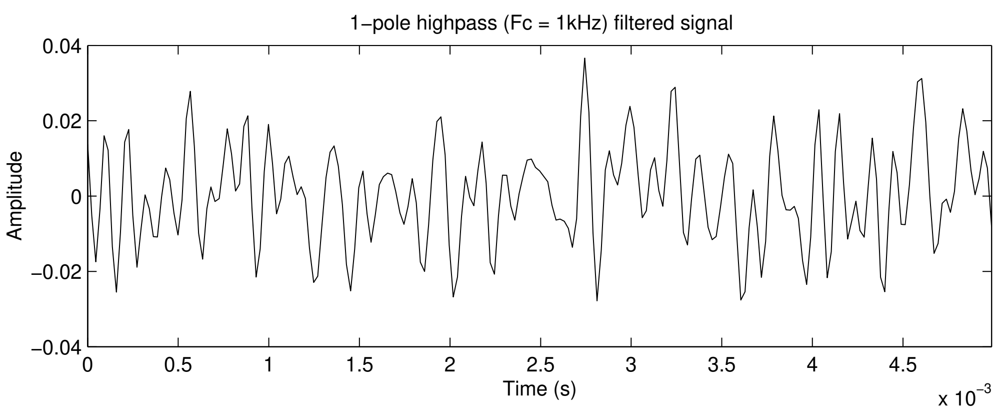
This transfer function attenuates low frequencies, but lets frequencies above 2kHz or so pass. You can hear that the bass is quite reduced. You can also see from the signal waveforms that the fast wiggles in the signal come through, but the slower variations are quite reduced.




1−pole highpass (Fc = 1kHz) impulse response
δ(t)
Amplitude
Time (s)
x 10−4


1−pole highpass (Fc = 1kHz) step response
Amplitude
Time (s)
x 10−4


Original signal
Amplitude
Time (s)
x 10−3


1−pole highpass (Fc = 1kHz) filtered signal
Amplitude
Time (s)
x 10−3

Low Q bandpass filter

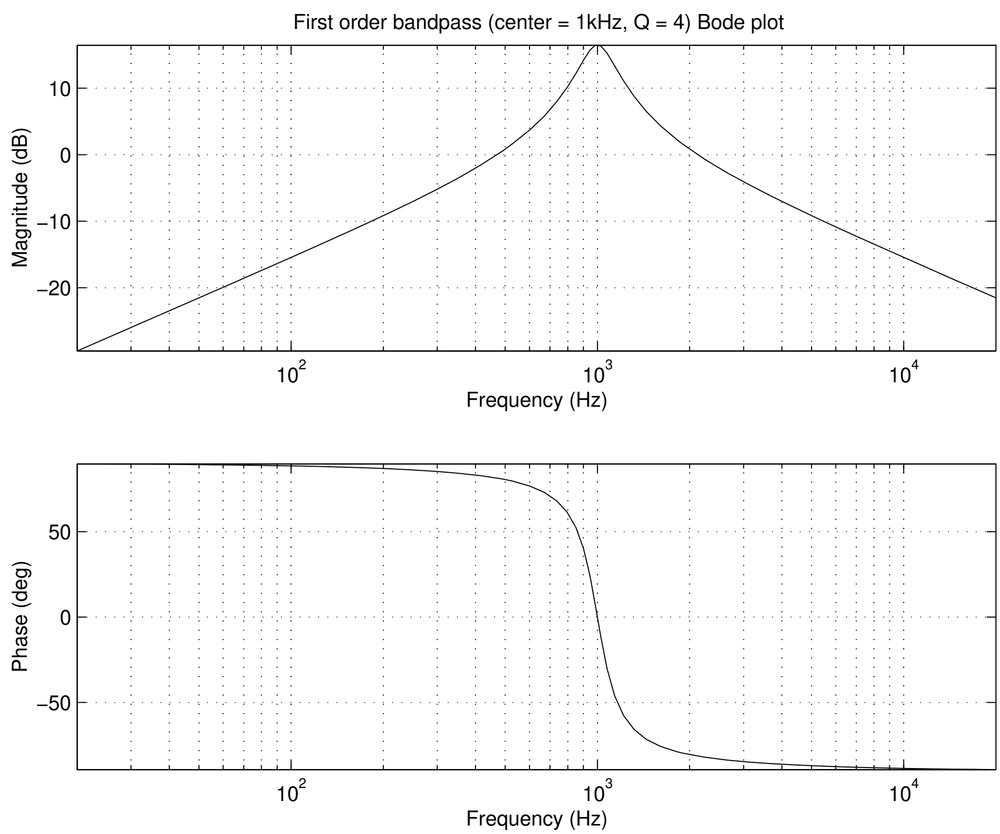
This is a second order bandpass filter, with center frequency 1 kHz and $Q = 4$. Transfer function:

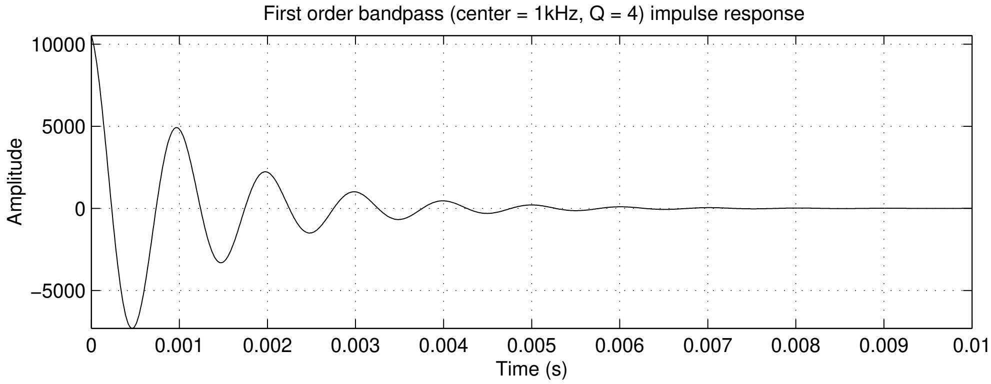
$$H(s) = K\frac{s}{(Q/\omega_c)s^2 + s + Q\omega_c}$$

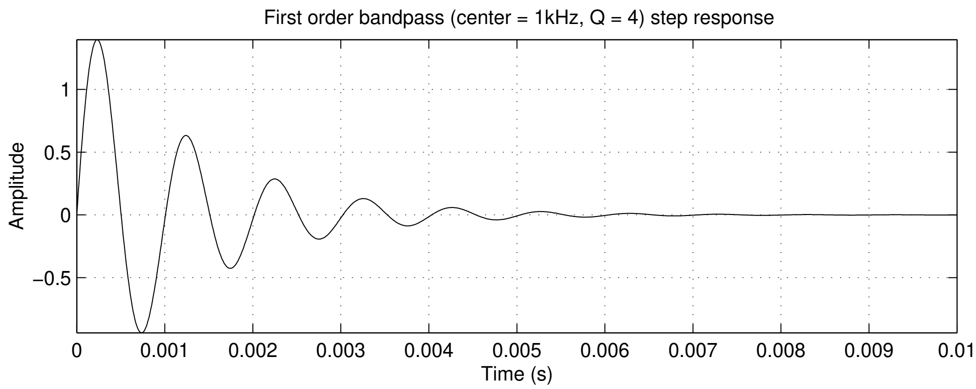
where $Q = 4$, $\omega_c = 2\pi \cdot 1000$, and $K = 6.7 \approx 16.5$ dB is a scale factor to increase the volume level.

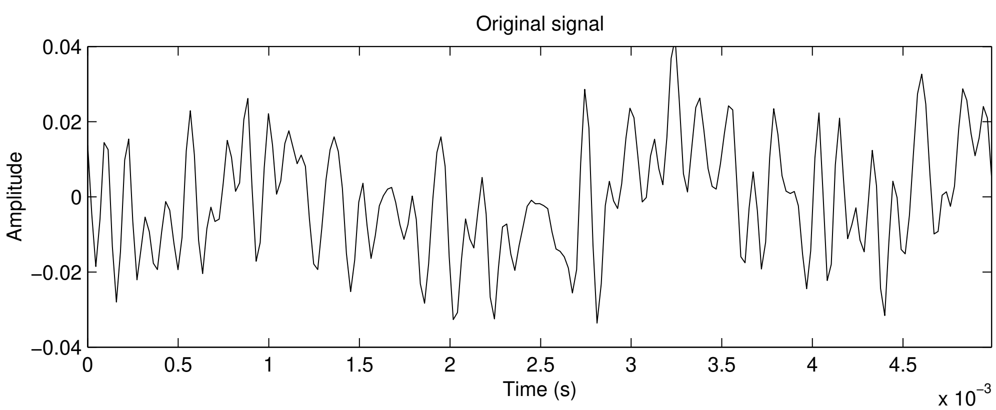
Here frequencies between 500Hz and 2kHz or so are passed, and both low and high frequencies are attenuated. The acoustic effect is something like listening through a tube (which has resonances — you'll learn about that in EE141!). You can also hear when a note (or a harmonic) gets near 1kHz.

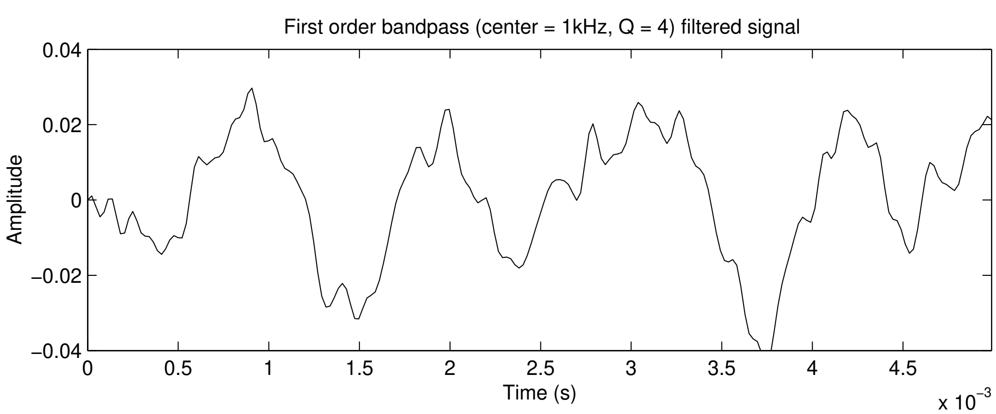
The impulse response is oscillatory, and you can see the effect in the plot of the filtered signal. Here both the fast wiggles and the slow undulatations are attentuated.




First order bandpass (center = 1kHz, Q = 4) impulse response
Amplitude
Time (s)


First order bandpass (center = 1kHz, Q = 4) step response
Amplitude
Time (s)


Original signal
Amplitude
Time (s)
x 10^-3


First order bandpass (center = 1kHz, Q = 4) filtered signal
Amplitude
Time (s)
x 10^-3

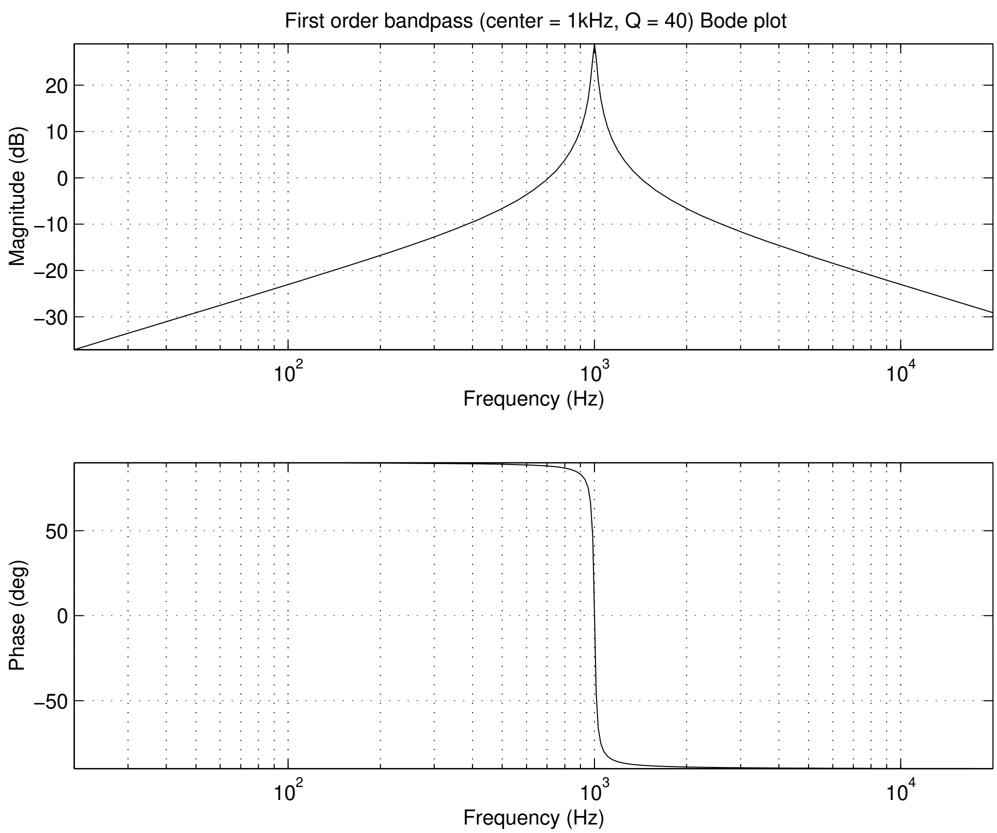
High Q bandpass filter

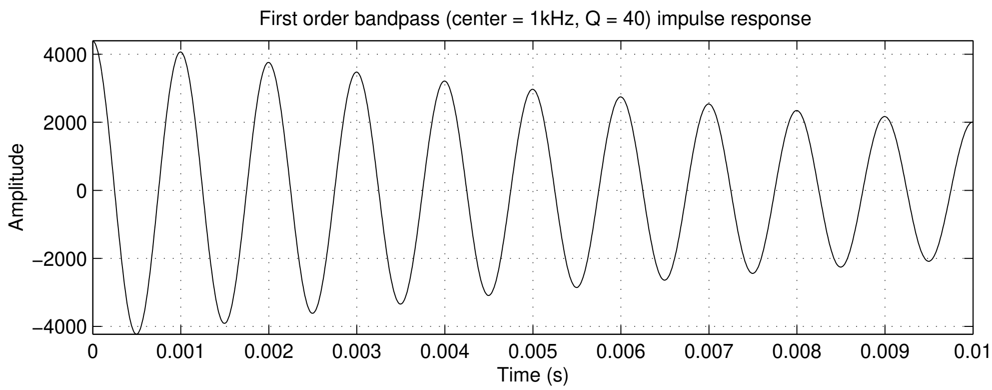
Second order bandpass filter with center frequency 1kHz and $Q = 40$. Transfer function:

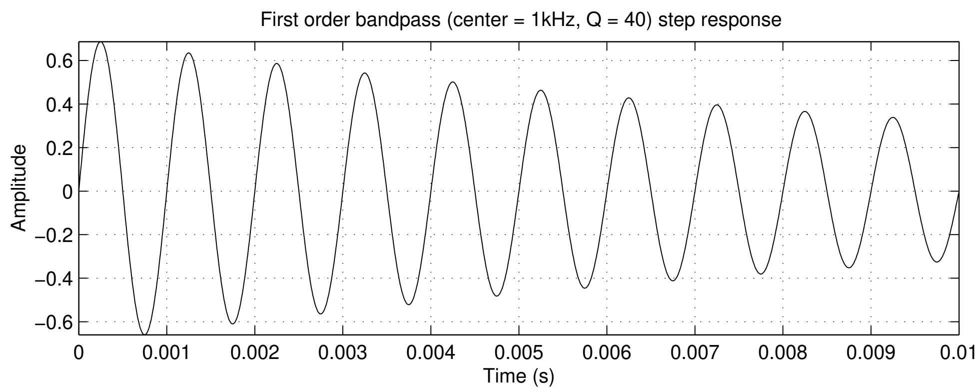
$$H(s) = K \frac{s}{(Q/\omega_c)s^2 + s + Q\omega_c}$$

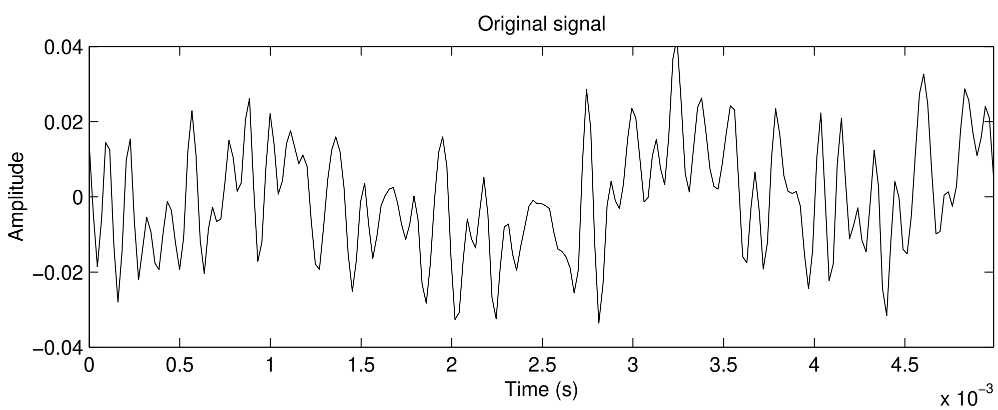
where $Q = 40$, $\omega_c = 2\pi \cdot 1000$, and $K = 28 \approx 29$ dB is a scale factor to increase the volume level.

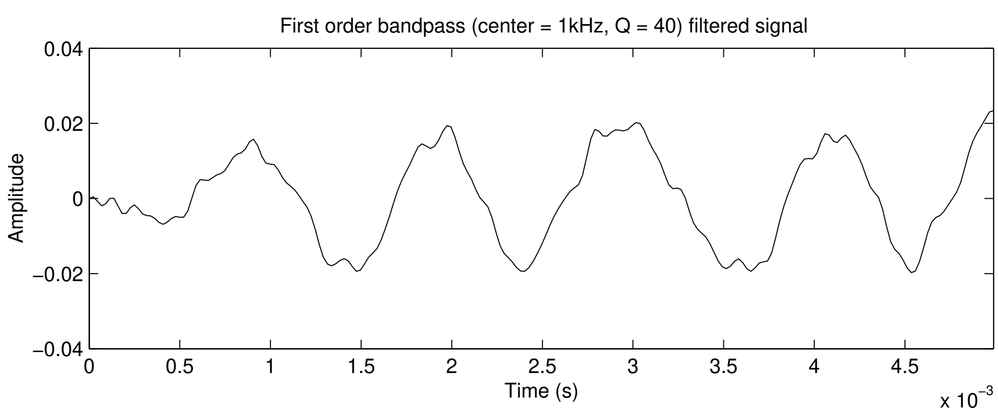
Here the frequencies near 1kH are strongly emphasized, which is quite annoying. The impulse response is quite oscillatory, which you can also see in the plots of the filtered signal.




First order bandpass (center = 1kHz, Q = 40) impulse response
Amplitude
Time (s)


First order bandpass (center = 1kHz, Q = 40) step response
Amplitude
Time (s)


Original signal
Amplitude
Time (s)
x 10^-3


First order bandpass (center = 1kHz, Q = 40) filtered signal
Amplitude
Time (s)
x 10^-3

Allpass filter

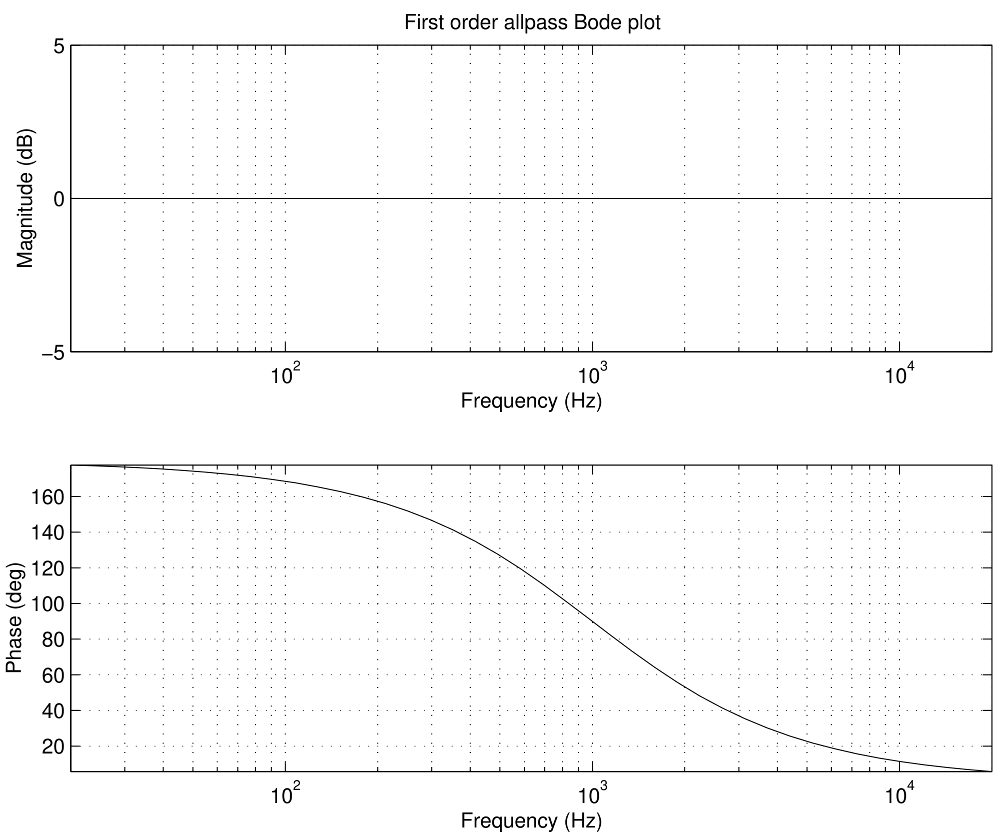
First order allpass filter with 90° phase shift at 1kHz. Transfer function:

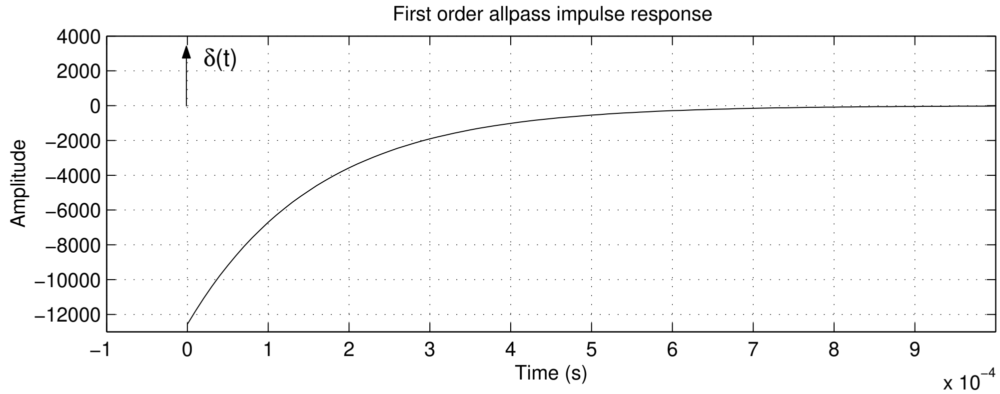
$$H(s) = \frac{s - \omega_c}{s + \omega_c} = \frac{s/\omega_c - 1}{s/\omega_c + 1},$$

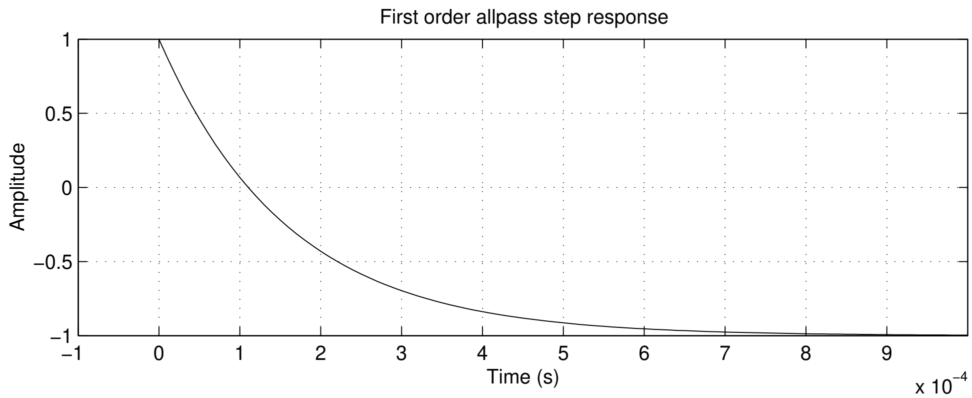
where $\omega_c = 2\pi \cdot 1000$.

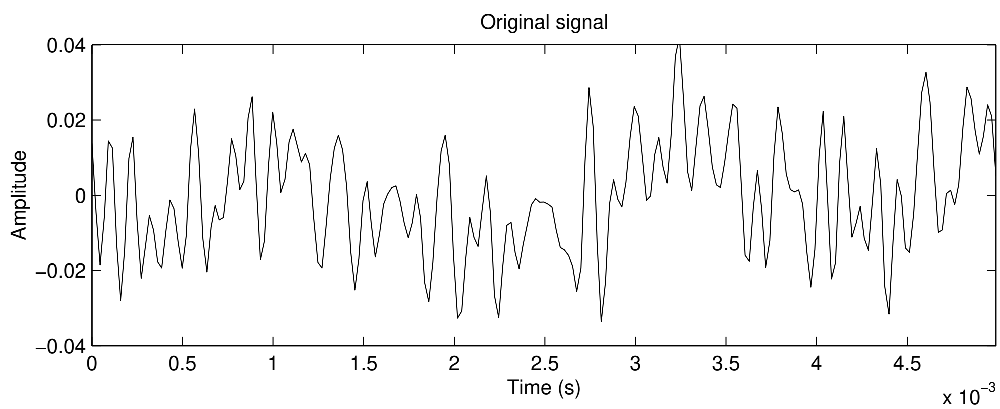
This filter passes all frequencies with unity gain. It does, however, shift the phase of the signal.

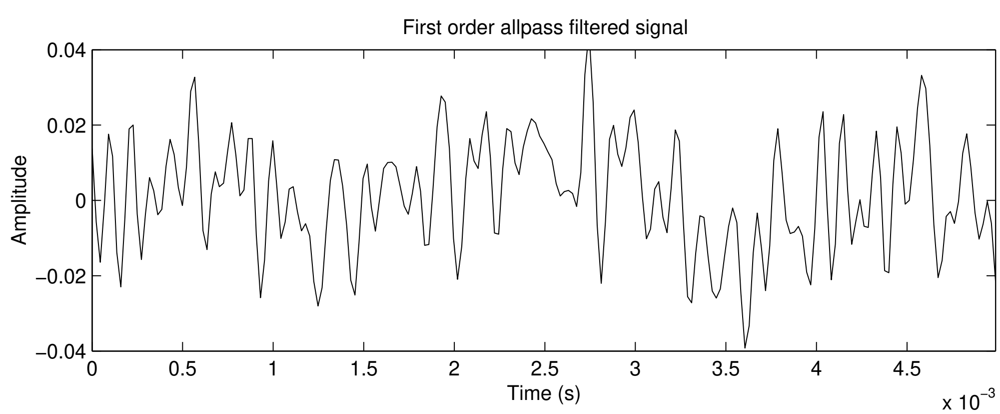
You really can't hear any difference at all, since the ear is pretty insensitive to (moderate) phase shift. The filtered signal looks quite a bit like the original signal, but is not the same.




First order allpass impulse response
Amplitude
Time (s)
δ(t)
4000
2000
0
−2000
−4000
−6000
−8000
−10000
−12000
−1 0 1 2 3 4 5 6 7 8 9
x 10−4


First order allpass step response
Amplitude
Time (s)
1
0.5
0
−0.5
−1
−1 0 1 2 3 4 5 6 7 8 9
x 10−4


Original signal
Amplitude
Time (s)
0.04
0.02
0
−0.02
−0.04
0 0.5 1 1.5 2 2.5 3 3.5 4 4.5
x 10−3


First order allpass filtered signal
Amplitude
Time (s)
0.04
0.02
0
−0.02
−0.04
0 0.5 1 1.5 2 2.5 3 3.5 4 4.5
x 10−3

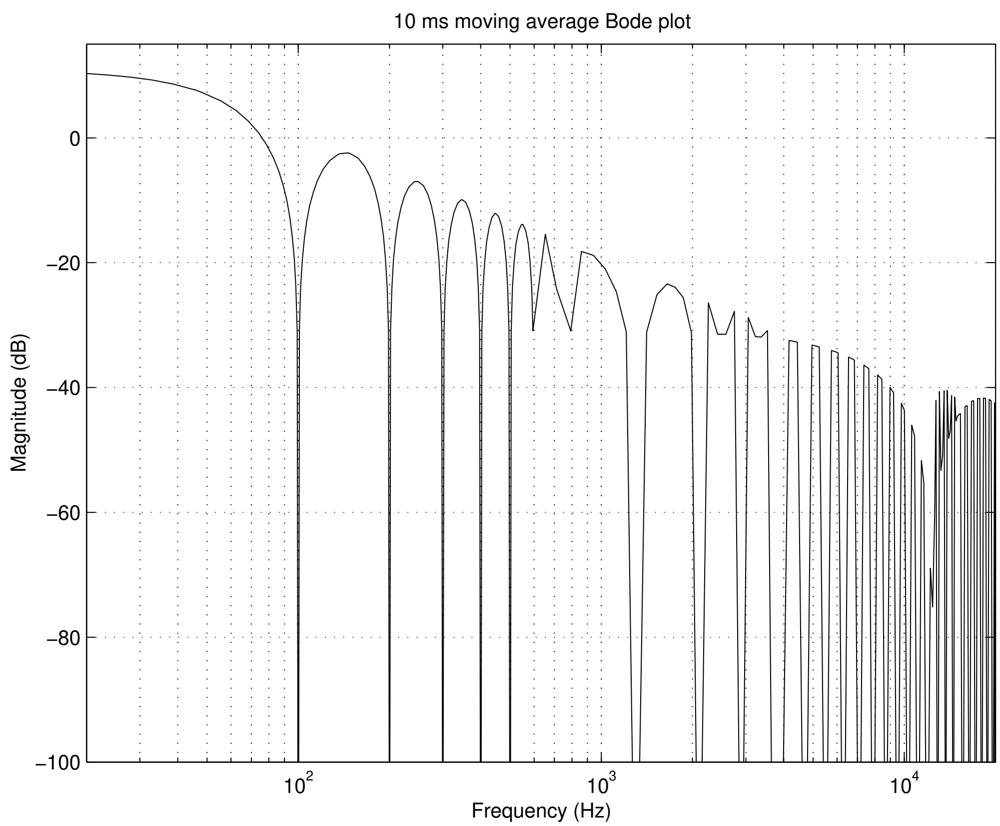
Moving average filter

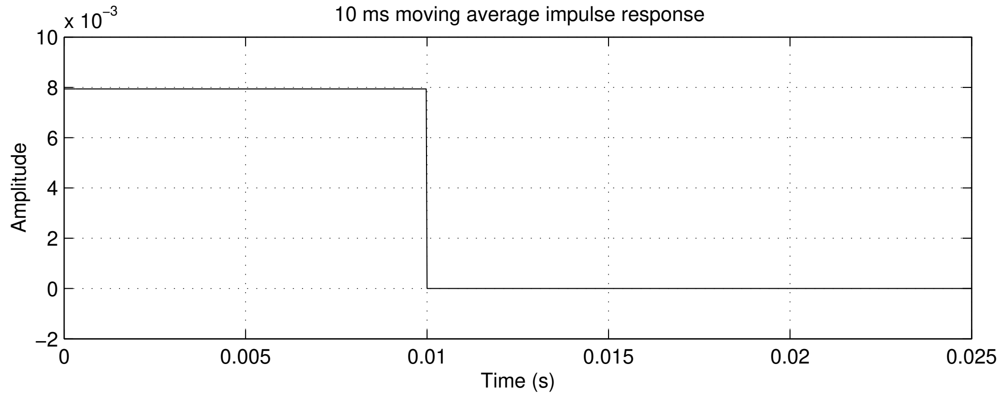
This filter is a 10ms moving average filter. Impulse response:

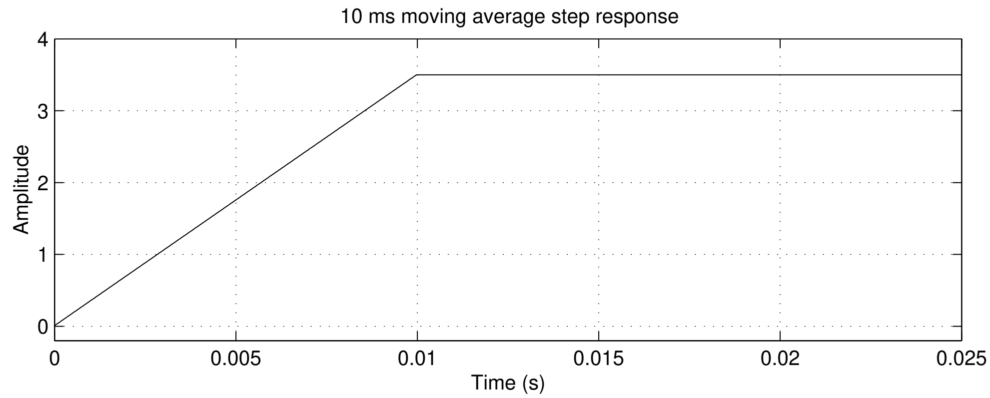
$$h(t) = \begin{cases} K(1/0.1), & 0 \leq t < 0.01 \\ 0, & t \geq 0.01 \end{cases}$$

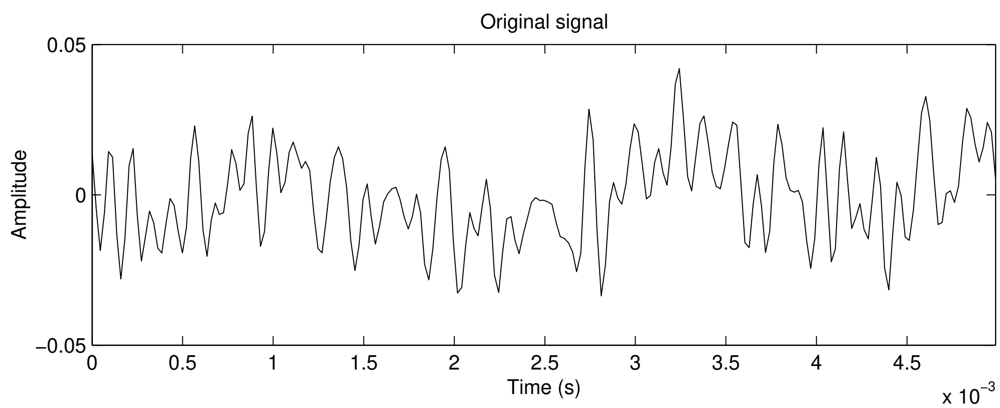
The DC gain is K, which we take to be $K = 3.5 \approx 10.9$dB to increase the volume level.

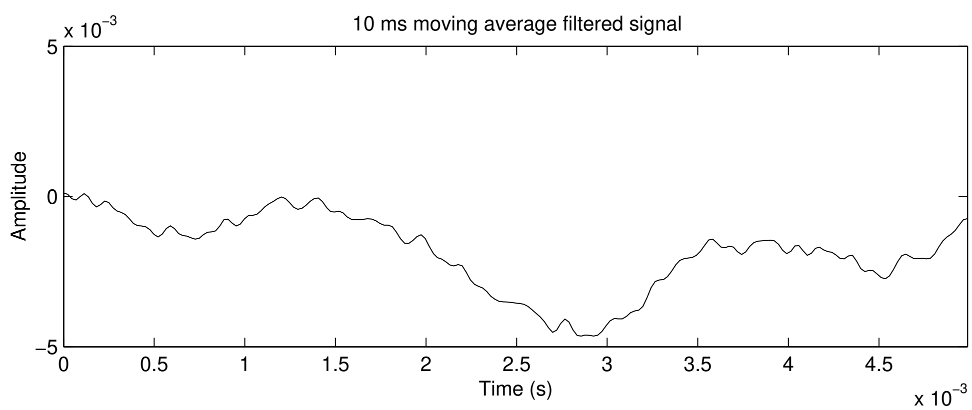
In this case the filtered signal output is exactly (K times the) average of the input signal over the last 10ms. Here the filtered signal sounds very muffled; the high frequencies are strongly attenuated.




10 ms moving average impulse response
x 10^-3
10
8
6
4
2
0
−2
Amplitude
0
0.005
0.01
0.015
0.02
0.025
Time (s)


10 ms moving average step response
4
3
2
1
0
Amplitude
0
0.005
0.01
0.015
0.02
0.025
Time (s)


Original signal
0.05
0
−0.05
Amplitude
0
0.5
1
1.5
2
2.5
3
3.5
4
4.5
Time (s)
x 10^-3

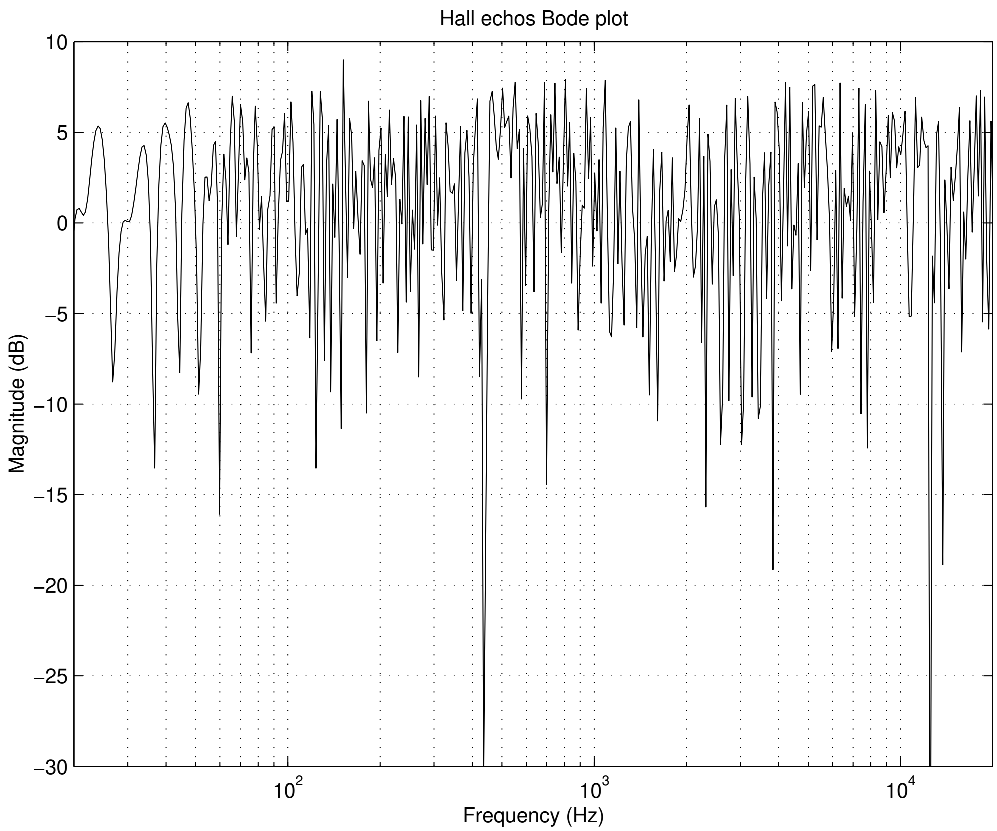

10 ms moving average filtered signal
x 10^-3
5
0
−5
Amplitude
0
0.5
1
1.5
2
2.5
3
3.5
4
4.5
Time (s)
x 10^-3

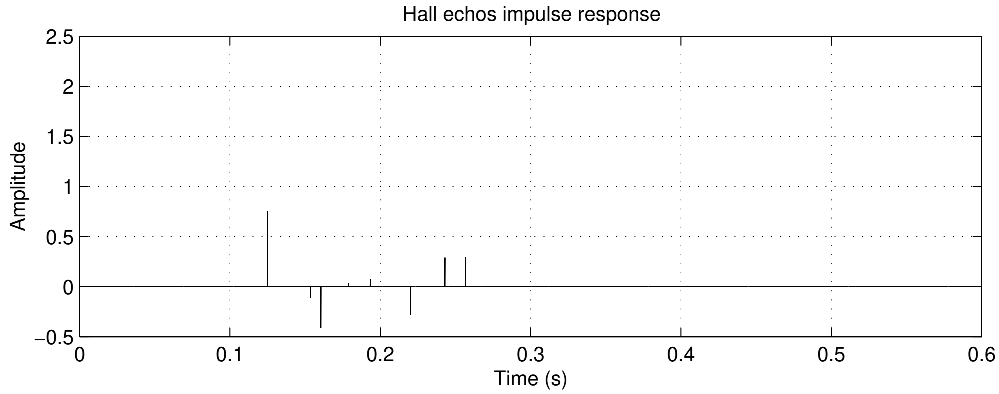
Filter with echos

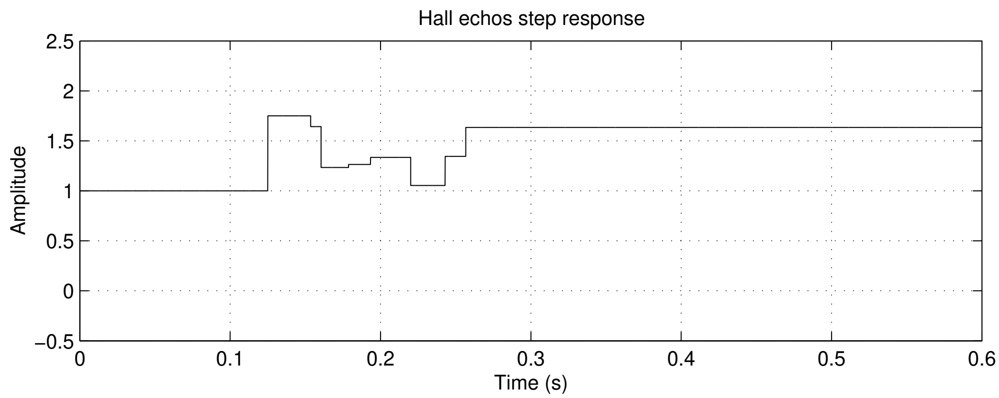
Impulse response:

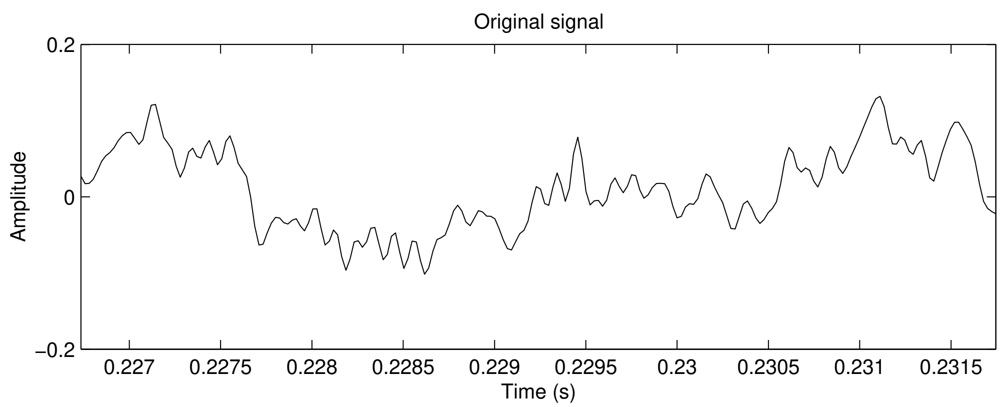
$$
\begin{aligned}
h(t) \;=\; & \delta(t) + 0.75\delta(t-0.125) - 0.1065\delta(t-0.1536) \\
& - 0.4098\delta(t-0.1605) + 0.0308\delta(t-0.1788) + 0.0705\delta(t-0.1934) \\
& - 0.2804\delta(t-0.2201) + 0.2906\delta(t-0.243) + 0.2898\delta(t-0.2567)
\end{aligned}
$$

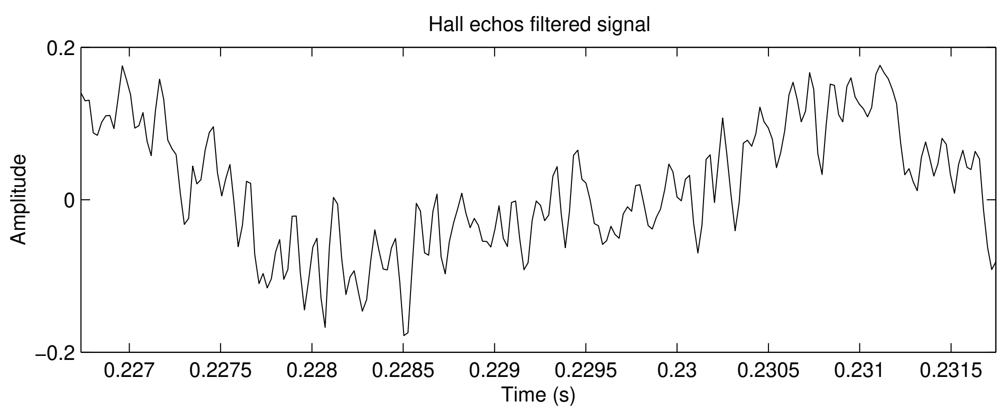
This represents 8 perfect echos. The first one arrives 125ms after the first impulse, and the others come over the next 125ms or so.




Hall echos impulse response
Amplitude
Time (s)
2.5
2
1.5
1
0.5
0
−0.5
0
0.1
0.2
0.3
0.4
0.5
0.6


Hall echos step response
Amplitude
Time (s)
2.5
2
1.5
1
0.5
0
−0.5
0
0.1
0.2
0.3
0.4
0.5
0.6


Original signal
Amplitude
Time (s)
0.2
0
−0.2
0.227
0.2275
0.228
0.2285
0.229
0.2295
0.23
0.2305
0.231
0.2315


Hall echos filtered signal
Amplitude
Time (s)
0.2
0
−0.2
0.227
0.2275
0.228
0.2285
0.229
0.2295
0.23
0.2305
0.231
0.2315