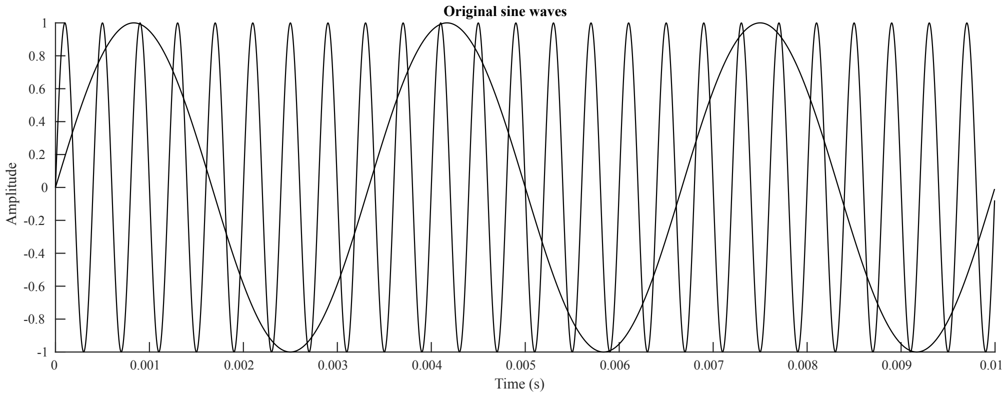
Filtering in MATLAB

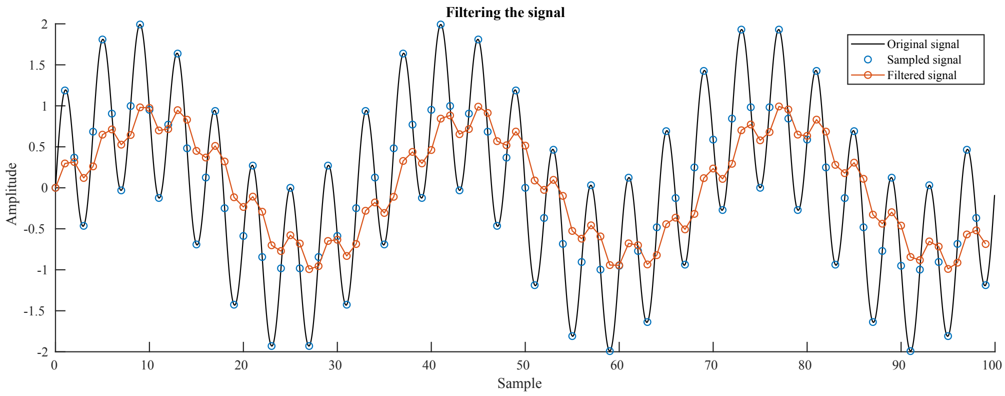
Pieter P

We can use MATLAB to visualize the effects of the filter. The scripts used can be found at the bottom of the page.

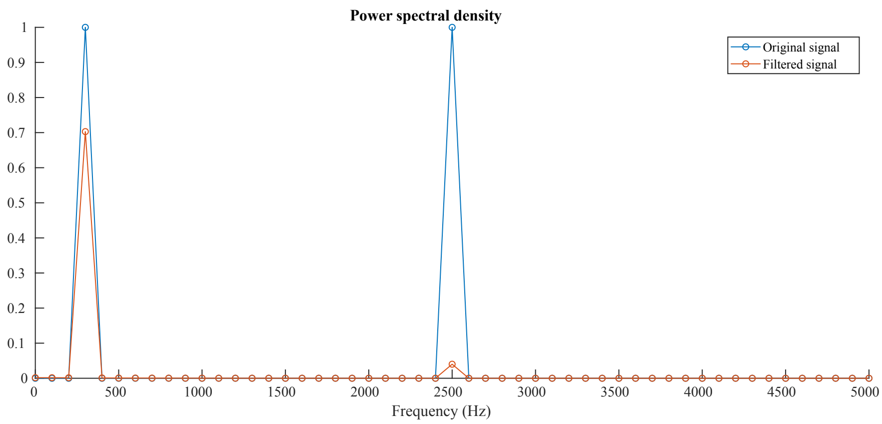
First, we generate a test signal that consists of two sine waves.



Then we apply the filter to it and plot the result. You can clearly see how the high-frequency sine wave is attenuated. Also note the phase shift between the original and the filtered signal: the red curve is delayed slightly, it is shifted to the right.



Finally, we can apply a fast fourier transform to inspect the frequency content.




```
Attenuation of first sine wave (30 Hz) = -1.53 dB
Attenuation of second sine wave (250 Hz) = -13.97 dB
```

You can hear the difference for yourself:

Original Filtered

Audio

It can be used on music as well. Try it out using the buttons below (you can switch between the original and filtered audio while the music is playing).

Original Filtered

Code

Sine Wave Code

```
%% Visualization

close all;      % Close all open figures

alpha = 0.25;   % Filter factor of 1/4

f_s = 10000;     % 10 kHz sample frequency
f_1 = 300;       % First sine wave with a frequency of 300 Hz
f_2 = 2500;      % Second sine wave with a frequency of 2.5 kHz

samples = 100;  % Calculate/plot 100 samples
n = linspace(0,samples-1,samples);  % Generate a vector with sample numbers
t = n / f_s;    % Generate a vector with time

sine_1 = sin(2*pi*f_1*t);   % Calculate the (sampled) sine waves
sine_2 = sin(2*pi*f_2*t);
signal = (sine_1 + sine_2);     % Mix the two sine waves together

b = alpha;              % Coefficients of the numerator of the transfer function
a = [1,-(1-alpha)];     % Coefficients of the denominator of the transfer function
filtered = filter(b,a,signal);   % Filter the signal

oversample_continuous = 20;     % Create a version with ten times more samples
                                % to display the smooth, continuous signal
samples_continuous = oversample_continuous * samples;
n_continuous = linspace(0, samples_continuous-1,samples_continuous) / oversample_continuous;
t_continuous = n_continuous / f_s;
sine_1_continuous = sin(2*pi*f_1*t_continuous);
sine_2_continuous = sin(2*pi*f_2*t_continuous);
signal_continuous = (sine_1_continuous + sine_2_continuous);

% Plot the two original sine waves
figure('pos',[0,0,1200,400]);
hold on;
plot(t_continuous, sine_1_continuous, 'k');
plot(t_continuous, sine_2_continuous, 'k');
title('Original sine waves');
xlabel('Time (s)');
ylabel('Amplitude');

% Plot the continuous signal, the sampled version and the filtered output
figure('pos',[0,0,1200,400]);
hold on;
plot(n_continuous, signal_continuous, 'k');
plot(n, signal,'o');
plot(n, filtered,'-o');
title('Filtering the signal');
xlabel('Sample');
ylabel('Amplitude');
legend('Original signal','Sampled signal','Filtered signal');

% Apply a fast fourier transform and plot the spectra of the
% original signal and of the filtered output
figure('pos',[0,0,1000,400]);
hold on;
f = linspace(0,samples-1,samples)*f_s/samples;
original_spectrum = (abs(fft(signal))*2/samples).^2;
filtered_spectrum = (abs(fft(filtered))*2/samples).^2;
plot(f(1:1+samples/2),original_spectrum(1:1+samples/2),'-o');
plot(f(1:1+samples/2),filtered_spectrum(1:1+samples/2),'-o');
title('Power spectral density');
xlabel('Frequency (Hz)');
legend('Original signal','Filtered signal');

% Calculate the attenuation of the two sine waves
f_1_index = f_1*samples/f_s+1;
A_1 = filtered_spectrum(f_1_index) / original_spectrum(f_1_index);
A_1_dB = 10*log10(A_1);
fprintf('Attenuation of first sine wave (%.0f Hz) = %.02f dB\n', f_1, A_1_dB);

f_2_index = f_2*samples/f_s+1;
A_2 = filtered_spectrum(f_2_index) / original_spectrum(f_2_index);
A_2_dB = 10*log10(A_2);
fprintf('Attenuation of second sine wave (%.0f Hz) = %.02f dB\n', f_2, A_2_dB);

% Open the filter visualization tool
fvtool(b,a,'Fs',f_s);

%% WAV export

samples = f_s*2;    % 2 seconds of audio
n = linspace(0,samples-1,samples);  % Generate a vector with sample numbers
t = n / f_s;        % Generate a vector with time

sine_1 = sin(2*pi*f_1*t);   % Calculate the (sampled) sine waves
sine_2 = sin(2*pi*f_2*t);
signal = (sine_1 + sine_2)/2; % Mix the two sine waves together

filtered = filter(alpha,[1,-(1-alpha)],signal);   % Filter the signal

audiowrite('original.wav',signal,f_s);    % Export as audio
audiowrite('filtered.wav',filtered,f_s);
```

Audio Code

```
[signal,f_s] = audioread('telegraph_road_original.wav');

alpha = 0.25;   % Filter factor of 1/4

b = alpha;            % Coefficients of the numerator of the transfer function
a = [1,-(1-alpha)];   % Coefficients of the denominator of the transfer function
filtered = filter(b,a,signal);   % Filter the signal

audiowrite('telegraph_road_filtered.wav',filtered,f_s);
```