```
% Homework 4
% EE 261: FT & Its Applications
% Summer 2020 - Stanford
% Noor Fakih
```

Handel's Hallelujah

```
load handel
load anti-aliasing

% downsample
yhalf = y(1:2:length(y));
audiowrite('downsample_y.wav', yhalf, Fs/2, "BitsPerSample", 16);

% take the fourier transform
Y = fft(y);

% filter and create new signal
Y_filter = Y.*Hs;

% go back into time domain
y_filter = real(ifft(Y_filter));

% downsample
y_filter_half = y_filter(1:2:length(y_filter));

audiowrite('y_filer.wav', y_filter_half, Fs/2, "BitsPerSample", 16);
```

Did you hear any audible difference?

THe downsampled signal is very vlearly distorted as the sound gets louder, making it very unpleasent to listen to. Afte filtering, that problem goes away, and then by downsampling again you are left with a better listening experfience. The user isn't offended by the large amount of distorition crackling at the higher frequencies, but it also isnt as crisp as a sound either agter putting the dignal through the operations. There is definetly more bass in the second signal than the first.I can't tell how exactly this audio affects hiher pitched intruments playing in the background, but they sound slightly different as well.