Digital Filtering

This is a very brief overview of the principles underpinning digital filtering.
Consider a time varying voltage which has been captured and stored digitally in a computer. It can be plotted on a voltage vs. time graph (fig1a), or with some clever mathematics, which you will be taught in year two, as a voltage vs. frequency graph (fig1b). You have already seen, on the oscilloscopes, a composite signal formed from of a sinusoidal wave and a DC-offset. On a frequency voltage plot this would show as two spikes – one at 0hz with an amplitude related to the DC-offset size, and one at the frequency of the sinusoidal wave, with an amplitude proportional to it's amplitude. In the same way a more complex wave, e.g. an ECG trace is mathematically equivalent to the sum of a large set of sinusoidal waves of differing amplitudes.
If we look at fig 1a, it looks like a DC-offset and a sign wave 'sitting on top of' something of lower frequency that might be interesting, but it's hard to see because of the higher frequency component. If we look at fig1b we can see a DC spike, a spike at 50Hz and some lower amplitude 'stuff' in-between. We want to delete the spike at 50Hz, do the reverse of the cleaver mathematics we did before and hay-presto we end up with a nice clean signal trace (fig1c). And that, effectively, is what software based digital filtering does, multiplies the frequency components we want to keep by 1 and the ones we want to get rid of by 0. If we want to reduce, but not completely remove a frequency we multiply it by a number between 0 and 1.

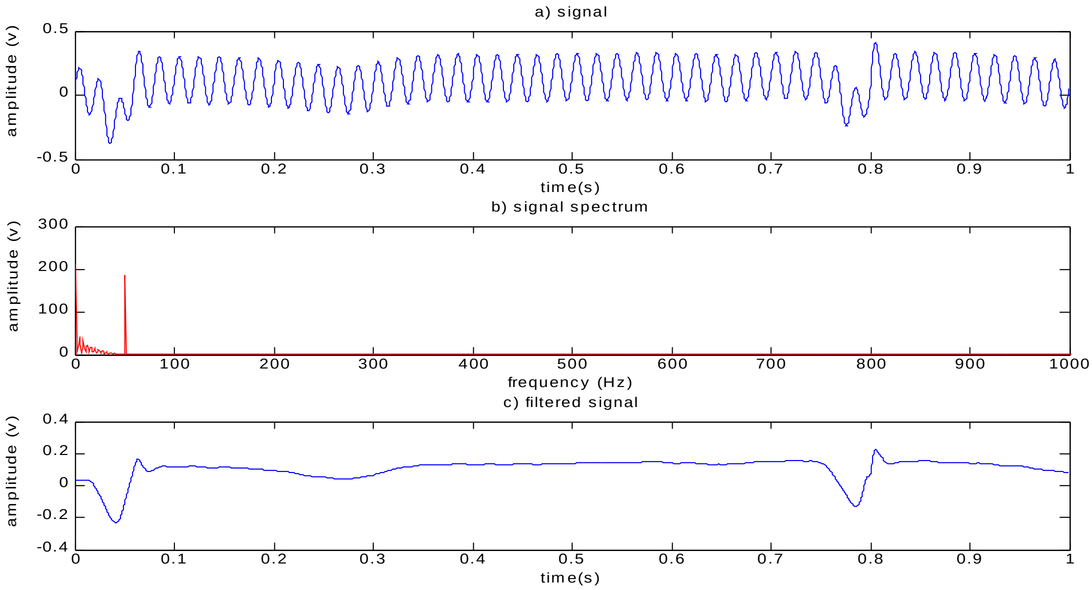

Fig1

Notes:

1. Digital filters are dependant on the data being accurately recorded and digitized in the first place.
2. The phase of each frequency component can also be adjusted.
3. Whilst the process described here illustrates the concepts, it is not how all digital filtering is implemented, sometimes mathematical 'tricks' are used to improve computational efficiency.