# Answers to Exercises

Chapter 6, exercise 1A:

The amplitude response is shown below, on both linear and logarithmic frequency scales, and with the y axis specified in dB and in linear units. Strictly speaking, the plots using a linear amplitude measure, volts, are not amplitude responses, as they show the output voltage for a particular input voltage, not the *ratio* between the two.

Clearly, this is a band-pass filter, with a centre frequency of 1100 Hz and a bandwidth of 200 Hz (ranging from 1000-1200 Hz). Therefore, the Q value of this filter is (p 88):

$F_{cf}/Bw = 1100/200 = 5.5$
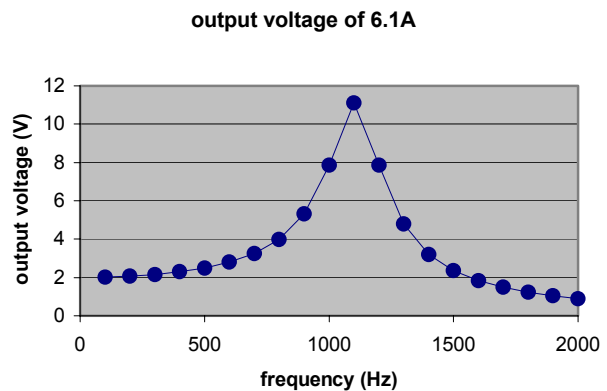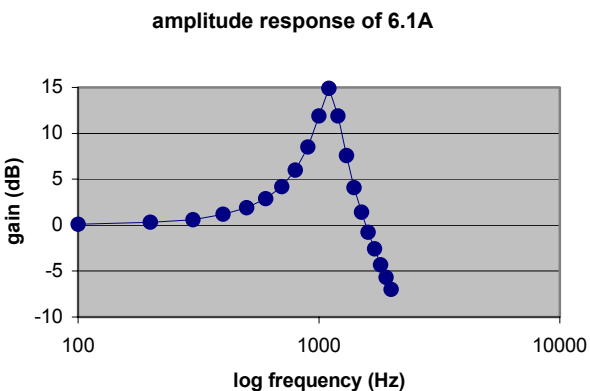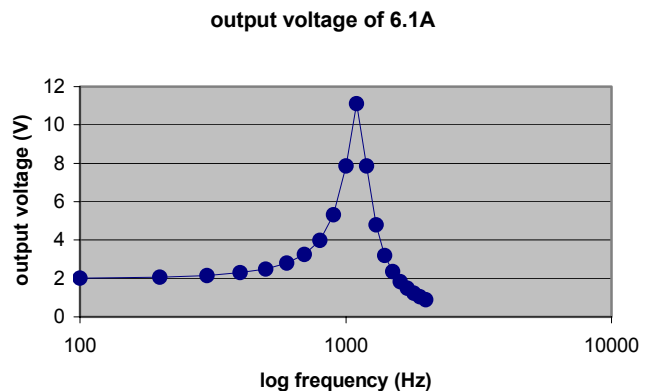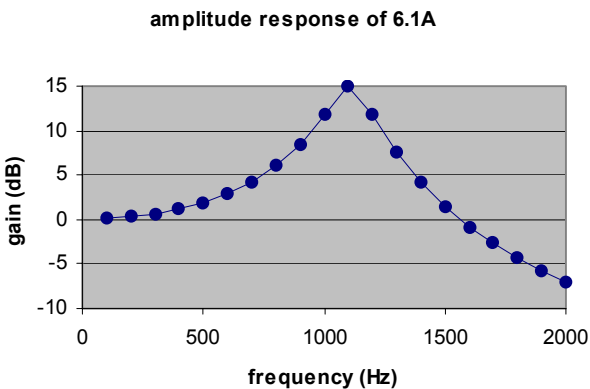
The filter slopes look more linear on log frequency scales, so we calculate roll-offs on that scale. For the upper frequency side, we can estimate the slope from about 1.4 kHz to 2 kHz because this part of the curve is fairly linear. This frequency range spans about ½ of an octave, as the number of octaves between any two frequencies is given by:

$$log_2(f_2/ f_1) = log_2(2000/1400) = 0.515$$

The difference in the gain is –7-4.1=-11.1 dB, so the slope of the roll-off is –11.1/0.515 = -21.6 dB/octave.

Chapter 6, exercise 2: The high- and low-pass filters in cascade result in a band-pass filter because the cutoff of the low-pass filter is higher than the cut-off of the high-pass filter.
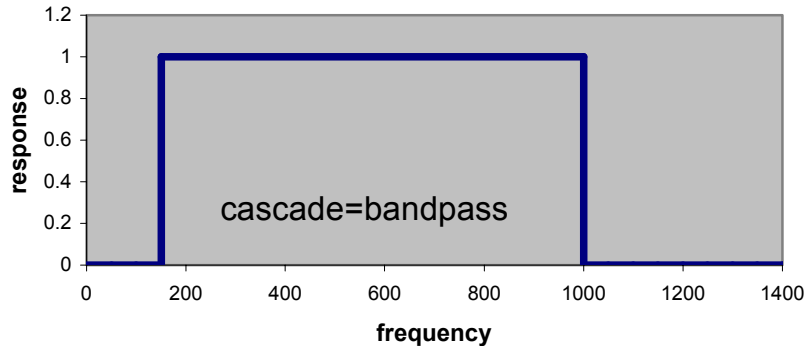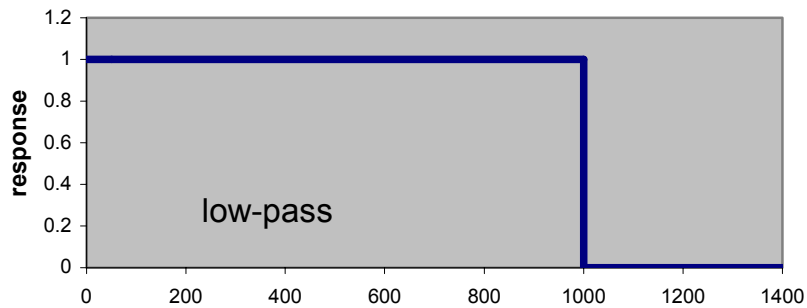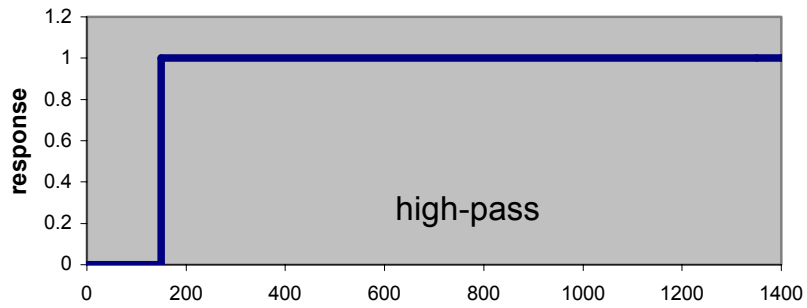
Chapter 6, exercise 3: This question refers to the situation described on pp 86-88. As long as the loudspeaker gave out a reasonable level of sound for all the frequencies used in measuring the frequency response, it wouldn't matter if the level of the output varied from frequency to frequency. Calculating the amplitude response requires the ratio of the output over the input amplitude. For an LTI system, homogeneity means any change in the input level will be mirrored in the output level, so the ratio will not change. The essential thing here is that the two microphones be identical in their amplitude responses. Otherwise, any differences in the microphones will be wrongly included in the calculation of the system response.

Chapter 6, exercise 4: This question addresses the issue of how to calculate the frequency response of a cascade of two LTI systems, but from 'the other way around'. Suppose you have an acoustic system whose frequency response, *A*, has been measured with a probe microphone which is known to have a frequency response of *P*. (For example, suppose you were interested in the acoustic effects

of the ear canal. The input to the system would be the sound pressure level at the entrance to the ear canal, whereas the output would be the sound pressure level near the eardrum. Assume you have a loudspeaker that is known to have a completely flat response when measured at the entrance to the ear canal.) The sound presented to the ear now undergoes two transformations, first by the acoustic system and then by the probe microphone. The amplitude part of the frequency response of these two systems together, O can then be found by:

$$A + P = O,$$

because the frequency responses given are on dB scales. In this situation, you know O and P and are trying to determine A, so:

$$A = O - P.$$

This calculation needs to be done at each frequency, so the result becomes:
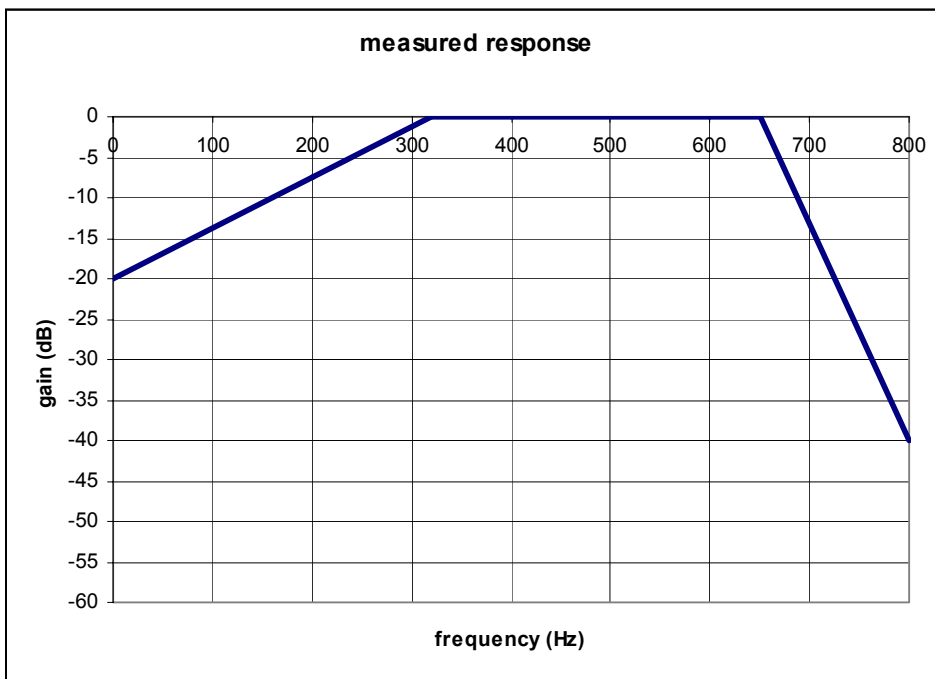
Consider this further. If you had two identical probe mikes, or could move the probe to measure the input and output, would you need to correct the measured frequency response for the properties of the probe? Why or why not?