Please answer all of the following questions.

1) Consider 8 harmonics of a sawtooth wave which has a fundamental period of 10 ms and a fundamental component with a level of 120 dB SPL. Give its amplitude values in a table (both in Pa and in dB SPL) and also draw its amplitude spectrum (on dB SPL and logarithmic frequency scales) *and* phase spectrum (as degrees or radians on a logarithmic frequency scale). (**10 points**)

| harmonic	frequency	level (Pa)	level (µPa)	dB SPL
1	100	20.00	20000000.00	120.0
2	200	10.00	10000000.00	114.0
3	300	6.67	6666666.67	110.5
4	400	5.00	5000000.00	108.0
5	500	4.00	4000000.00	106.0
6	600	3.33	3333333.33	104.4
7	700	2.86	2857142.86	103.1
8	800	2.50	2500000.00	101.9
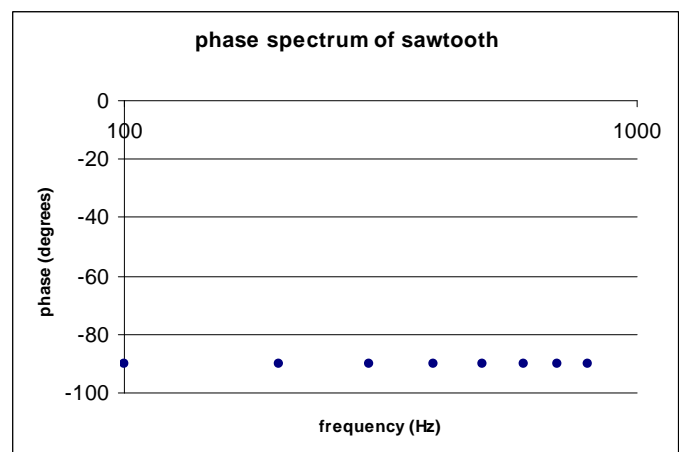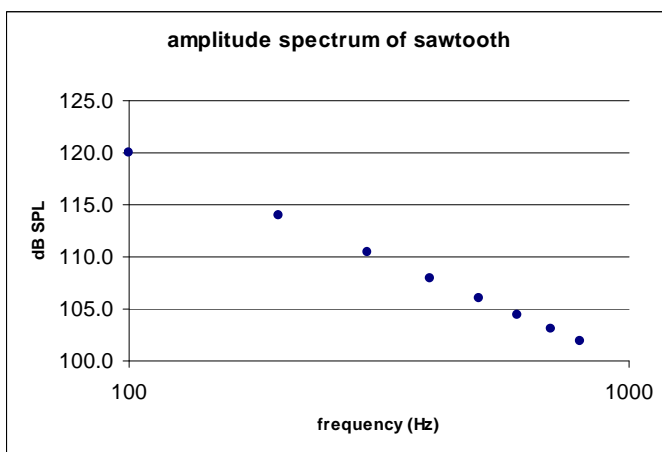
2) Draw 3 cycles of the sawtooth wave in question 1, and then a full-wave and a half-wave rectified version of it. (Remember: full-wave rectification = take absolute amplitude values, half-wave rectification = set negative amplitudes to 0). Which of the three waveforms will have the highest, and which the lowest RMS energy? Why? (**5 points**)

Full-wave rectified waveform results in a positive triangular waveform.

Half-wave rectified waveform results in a zero amplitude for all previously negative sections.

RMS between original sawtooth and full-wave rectified version are equal.

RMS is lower for the half-wave rectified version.

1

sawtooth

full-wave rectified sawtooth

half-wave rectified sawtooth
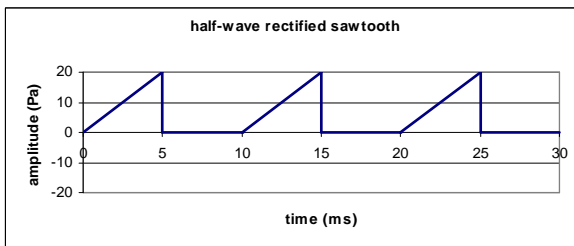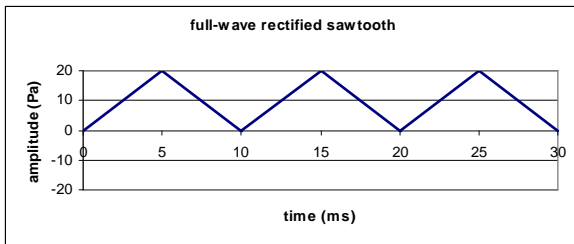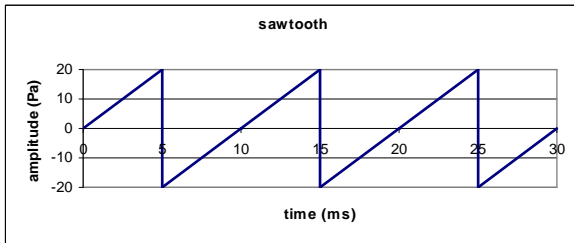
3) Each line of the following table indicates the spectral components in a complex periodic waveform. For each combination, calculate the fundamental frequency and fundamental period. Note that frequencies are given in hertz and periods in ms, and your answers should also be expressed in these units. **(5 points)**

| a)	frequencies	500	1000	1100
b)	periods	0.5	1	4
c)	frequencies	10	15	30
d)	periods	7	3	1
e)	frequencies	100	1010	2000

Answers:

a) 100 Hz = 10 ms

b) 4 ms = 250 Hz

c) 5 Hz = 200 ms

d) 21 ms = 47.6 Hz

e) 10 Hz = 100 ms

4) Suppose you had a system that multiplies each input amplitude value by 2. What change in dB does this correspond to? Draw input and output waveforms for 3 cycles of an input wave which is a sinusoid of peak amplitude 2 V and frequency of 400 Hz. Given what you've been told about the response of LTI systems to sinusoids, do you think this system is LTI? Why? Is this system homogeneous (sketch the input/output function)? Time-invariant? Give reasons for your answers. Where could such a system occur in daily life? (**10 points**)
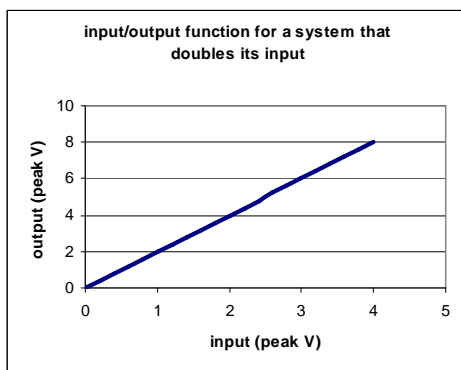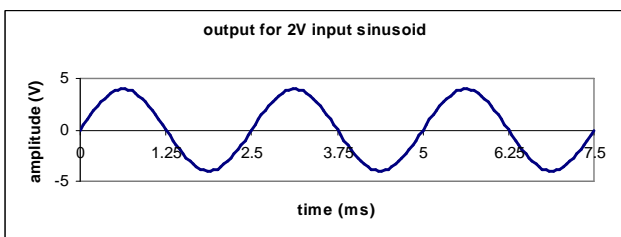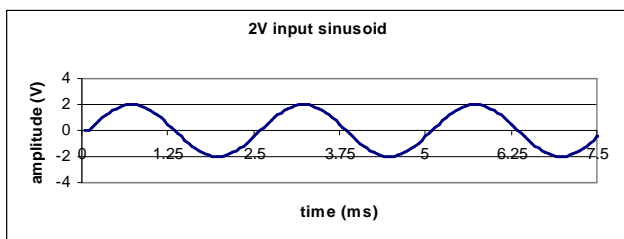
Doubling in amplitude is a change of +6 dB.

System is LTI because a sinusoidal input always results in a sinusoidal output of the same frequency.

Homogeneity: yes. Proportionally equal changes are applied to each amplitude value thus input/output function showing a straight line going through the origin.

Time-invariance: yes. Any delay in the input signal would be matched by an equal delay in the output.

System could for example be an amplifier in a stereo system that doubles the output amplitude of a particular waveform.

3

5) It is often said that the function of the basilar membrane can be likened to that of a filter bank. Describe what a filter bank is and specify what its properties should be to most faithfully reflect the processing in the inner ear. **(20 points)**

> A filter bank is a collection or set of band-pass filters
> whose centre frequency varies over some range
> As the basilar membrane (BM) does a kind of frequency analysis,
> its function can be compared to a filter bank.
> each auditory nerve fibre responding to the acoustic world through a single bandpass filter.
> Filterbank should span 20 Hz-20 kHz
> have bandwidths increasing with increasing frequency)
> spaced on a quasi-logarithmic scale.
> be nonlinear (increase bandwidth and decrease gain with increasing level)

6) In order to reduce data, telephone systems typically band-pass filter their signals. Consider a given system that filters between 300 and 3500 Hz and attenuates the signal by 20 dB in its pass-band and by 50 dB outside the pass-band. It rolls off linearly over 100 Hz at each side of the filter. **(20 points)**

a) Draw a frequency response of that filter on dB amplitude and linear frequency scales up to 5 kHz.

b) Draw input and output spectra of the following three signals passed through this system. You are free to choose the input level in each case:

- A sinewave at 200 Hz

- A sinewave at 1 kHz

- An impulse

- A periodic train of impulses with a period of 0.002 s.

4
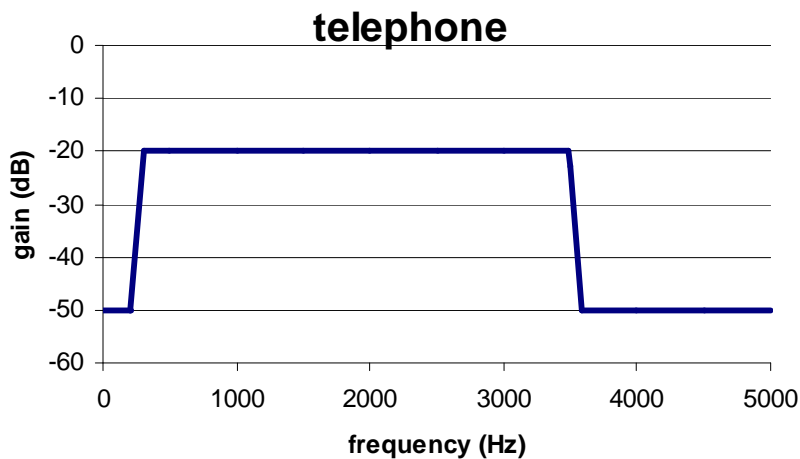
telephone

Draw correct frequency response of the telephone system.

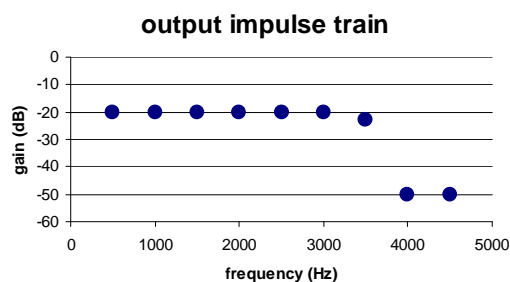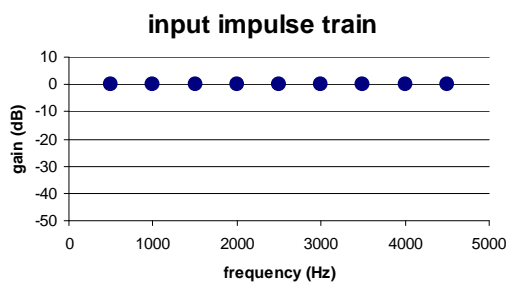Sketch four input spectra of the above specified waveforms.

Sketch output spectra for each signal:

sinewave at 200 Hz: 50 dB reduction`

sinewave at 1 kHz: 20 dB reduction

impulse: output spectrum has the same shape as the frequency response of the system

impulse train: harmonics spaced at 500 Hz, apply filtering of system.

7) For a project you are making digital recordings using a sampling rate of 44100 samples/second and a quantization depth of 16 bits. Now you realize that your recording is too large to fit on your disc. What two things can you do in order to decrease the size of the recording, *not* using a compression technique like MP3? How may each of these affect the sound quality of the signal and what kind of errors would you possibly introduce? Make sure you explain how the dynamic range and the bandwidth of your sampled signal changes as an effect of your data reduction. (**20 points**)

> Reduce either by downsampling (e.g. 16 k) or by using fewer quantization levels (e.g. 12 bit)
>
> Sound quality: downsampling reduces the amount of higher frequencies in the signal (higher than Nyquist frequency, i.e. higher than half the sampling rate). Using fewer quantisation levels reduces the dynamic range and the resolution of the amplitude axis. Dynamic range can be calculated by the formula 20*log (quantisation levels/1). Magnitude in- and decreases with in- and decrease in number of quantisation levels. Absolute number of quantisation levels is calculated: 2^number of bit.
>
> Errors introduced by reducing quantisation levels is mainly quantisation noise (eplain). Downsampling is a way of low-pass filtering to the signal.

8) Sketch (with linear frequency scales spanning 0-2 kHz) a low-pass filter with a cut-off at 500 Hz, a high-pass filter with a cut-off at 1000 Hz and a band-pass filter with a pass-band between 500 and 1000 Hz. The pass-band in each filter has a gain of 40 dB and the stop-band sections attenuate the signal by 40 dB. The slopes are steep.

Now consider you play white noise at a level of 96 dB re 1 V through all the filters in a cascade (first the low-pass, then the band-pass, then the high-pass filter). Draw the input and output spectra of the white noise passing through this cascade. Does the sequence of filters have an effect on the output spectrum of the noise? Why or why not? (**10 points**)

lowpass

highpass

bandpass
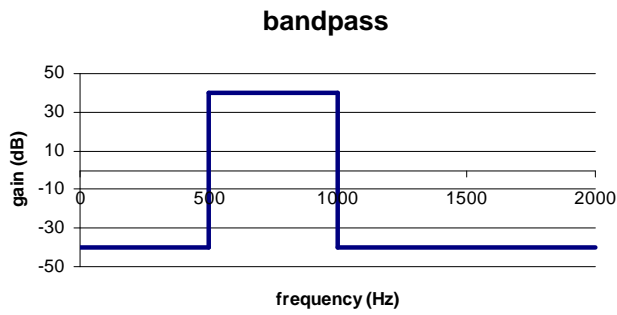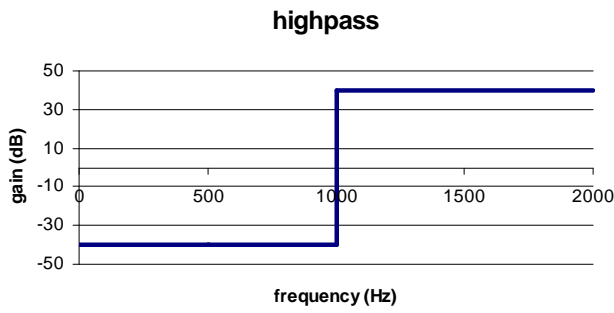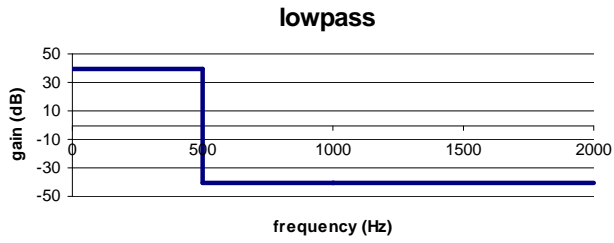
The sequence in the cascade has no effect, because both multiplication and addition are commutative.

white noise input spectrium

white noise output spectrium