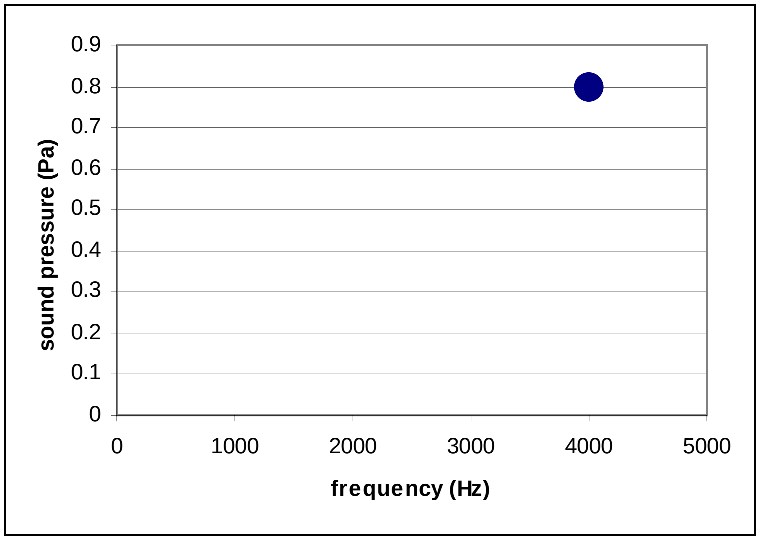
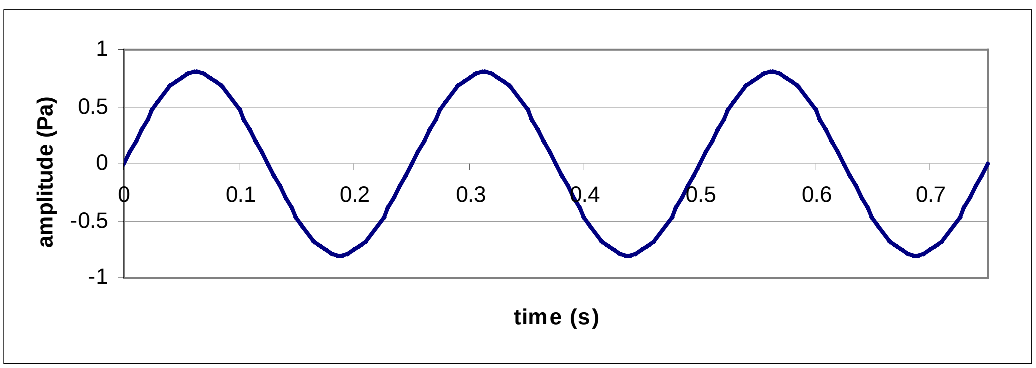
A102 Signals and Systems for Hearing and Speech: Final exam answers

1) Take two sinusoids of 4 kHz, both with a phase of 0°. One has a peak level of 0.8 Pa while the other has a peak level of 0.5 Pa. Draw the spectrum of the larger sinusoid (on linear scales) and 3 cycles of its waveform. What are the peak levels of the two sinusoids in dB SPL? What would be their peak level (in dB SPL) if added? What if one had a phase of 180° while the other remained at 0°? What level would the sum of these two sinusoids be, in Pa and dB SPL? **(10 points)**





	Pa	dB SPL
larger	0.8	92.0
smaller	0.5	88.0
sum (in phase)	1.3	96.3
sum (out of phase)	0.3	83.5

2) A complex periodic waveform is made up from the following components. For each combination, calculate the fundamental frequency and fundamental period. **(15 points)**

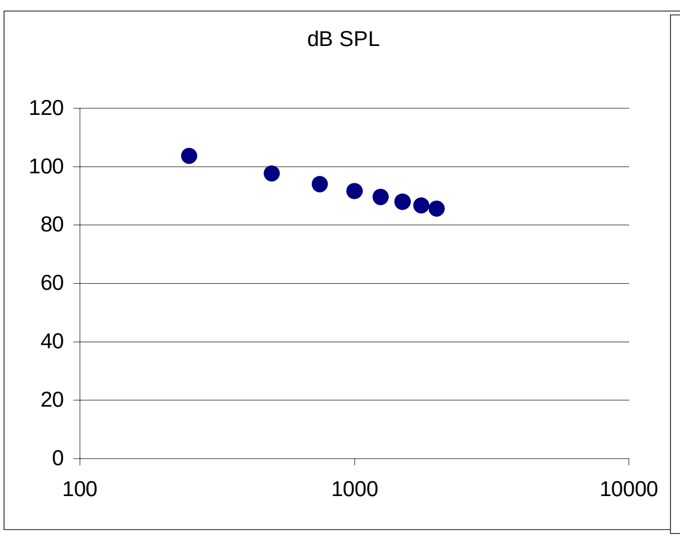
						answer	
a)	frequencies	200	400	600		200 Hz	5 ms
b)	periods	2.5	5	15		66.67 Hz	15 ms
c)	frequencies	250	300	600		50 Hz	20 ms
d)	periods	2.5	2	1		100 Hz	10 ms
e)	frequencies	1000	1040	2000		40 Hz	25 ms

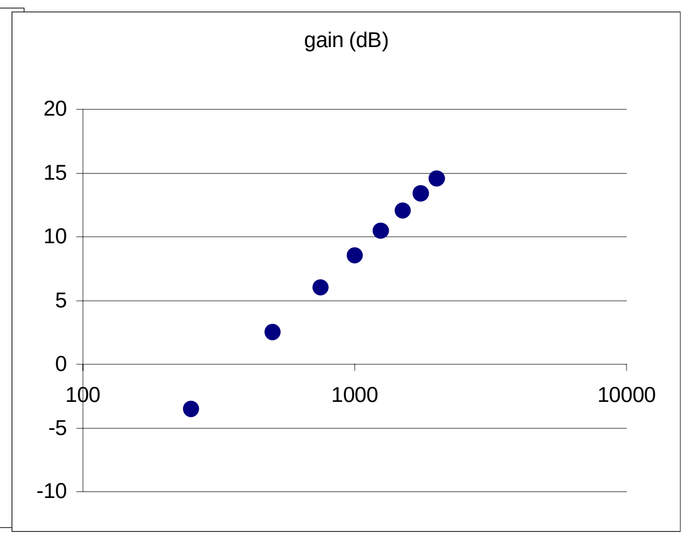
3) Consider a wave which consists of the first 16 harmonics of a sawtooth wave whose fundamental period is 4 ms, and whose fundamental component has a level of 3 Pa. **(20 points)**

   a) Draw its spectrum (on dB SPL and logarithmic frequency scales over the frequency range 125 Hz to 2 kHz).

   b) This wave is then put through a 'System X' which results in the output wave having a spectrum in which all components are of equal amplitude at 2 Pa. Draw this spectrum on dB SPL and logarithmic frequency scales over the frequency range 125 Hz to 2 kHz.

   c) Over the same frequency range, and again using dB and logarithmic frequency scales, draw the amplitude response of 'System X'.





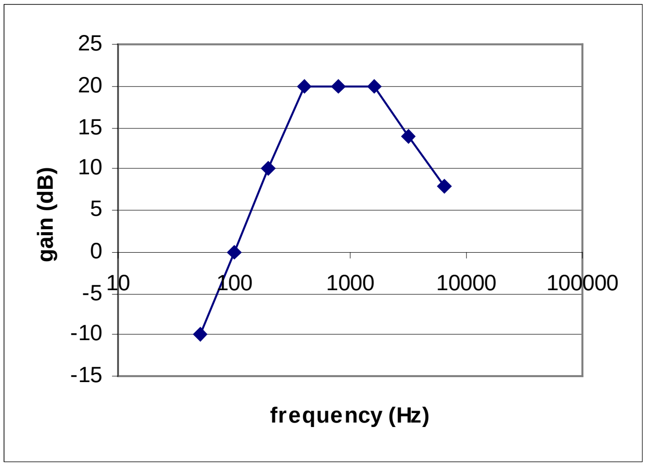
Note: x-axis to go from 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2kHz. Section b) essentially the same graph as Section a) but all points at the same level of 100 dB SPL.

4) Suppose that a pair of ear plugs reduced the amplitude of sound outside the ear by a factor of four (i.e. so that the sound pressure in the ear was one-quarter of what it was outside the ear), what would be the dB SPL level inside the ear given an external noise level of 106 dB SPL? How many Pa does this correspond to? **(10 Points)**

original	one quarter	result (dB SPL)	result (Pa)
106	-12.0412	93.96	0.997631

5) Sketch, on dB and logarithmic frequency scales (over a frequency range of 50 Hz – 12.8 kHz), a bandpass filter which has a gain of 20 dB in the passband which ranges from 800 Hz to 3.2 kHz, and which rolls off at 10 dB/octave on the low frequency side, and 6 dB/octave on the high frequency side. **(10 Points)**

frequency	gain (dB)
50	-20
100	-10
200	0
400	10
800	20
1600	20
3200	20
6400	14
12800	8



6) Give three examples of the way in which the notion of a bandpass filter can be useful in characterizing the functioning of the auditory periphery. Be sure to specify the input and

output signals, and the units in which they would be measured, for each system you name **(15 points)**

Ear canal: input = sound at ear canal entrance; output = sound at tympanic membrane; both measure in pascals

Middle ear: input = movement of tympanic membrane (meters) or sound pressure at TM (Pa); output = movement of stapes (m) or pressure in cochlear fluids (Pa)

Basilar membrane: input = movement of stapes (m) or pressure in cochlear fluids (Pa); output = movement of BM (m)

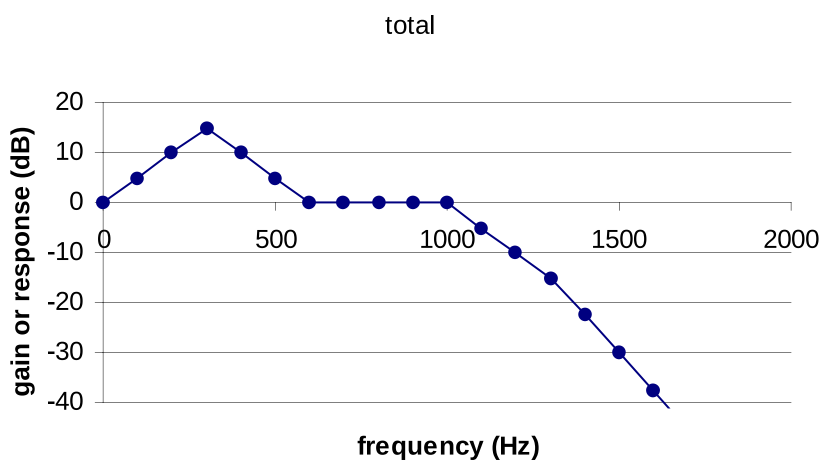
7) Suppose you were told that someone had a threshold that was 7 dB better than the average normal-hearing listener at 125 Hz, and that the normal threshold of hearing at this frequency was 632.5 µPa. What sound pressure (in Pa or µPa) would be the least intense that person could hear at 125 Hz? What is the normal average threshold in dB SPL at 125 Hz? **(5 points)**

632.5 µPa = 30 dB SPL

7 dB better means a threshold of 23 dB SPL = 282.5 µPa or 0.0002825 Pa

8) Consider a cascade of three systems with the following amplitude responses. Draw the amplitude response of the complete cascade. What would be the amplitude response of the cascade if the position of Systems A and B were reversed? **(15 points)**

It doesn't matter if the order of the systems is reversed.

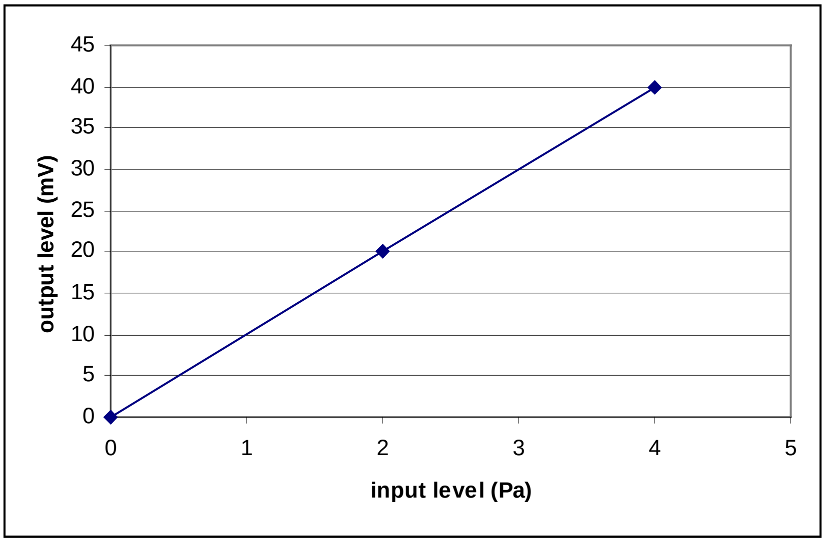


9) Draw the input and output spectra of white noise passed through the first system in the cascade of question 8. **(10 points)**

Input spectrum must be a continuous flat line; output spectrum identical to frequency response of 1st system. Must be labeled dB or some appropriate linear measure by frequency.

10) Imagine you had a microphone that could be considered a perfect LTI system. Draw its input/output function (on linear scales) for a 1 kHz sinusoid, assuming that a sound pressure level of 100 dB SPL leads to an output level of 20 mV. **(10 points)**

100 dB SPL = 2 Pa. Graph must be a straight line with the given point.



11) After digitising a signal, you find that its spectrum of the digital signal only contains frequencies up to 5000 Hz. What is the most likely reason for that? **(10 points)**

The sound card is sampling at 10 kHz, so the anti-aliasing low-pass filter is set to 5 kHz.

12) Suppose you had a system that *half-wave rectified* the input, which is the same as setting all negative values of the wave to zero, but leaving the positive values untouched.

Here are sample input and output waveforms for a sinusoid of peak amplitude 1 V and frequency of 200 Hz:

What's the simplest way of knowing that this is not an LTI system? Is this system homogeneous? Additive? Time-invariant? Give reasons for your answers. **(20 points)**

An input sinusoid resulted in a non-sinusoidal output.

Homogeneous: yes
Additive: no
Time-invariant: yes

13) Draw frequency domain diagrams to show how a narrow band-pass filter can be used to extract a single harmonic from a complex periodic input signal of fundamental period 10 ms (for example, a sawtooth). How would you decide what bandwidth to use for the band-pass filter? **(20 Points)**

Need:

Sensible graph of input spectrum (many harmonics at multiples of 100 Hz)

Sensible graph of a bandpass filter whose bandwidth is less than 100 Hz

Sensible graph of output spectrum (a single harmonics at some multiple of 100 Hz)

14) A sound card is a device within, or inserted into a computer, that changes analogue signals into digital ones (and back). One crucial function of a sound card is to filter a signal at its input. What is this filtering process called? What type of filter is used? Sketch a possible frequency response for this filter. Which frequencies are filtered and why? Make sure you explain the Nyquist frequency in this respect. **(15 Points)**

Anti-aliasing filtering. A low pass filter. Need sensible graph of a lowpass filter.

Only frequencies less than half the sampling frequency can be represented accurately.

Higher frequencies than this are aliased into lower frequencies.

The Nyquist rate is twice the highest frequency in the signal, and is the minimum rate needed for accurate representation.

15) A compressor-limiter is a tool used widely in speech and music amplification systems to normalise the amplitudes, i.e. attenuate high amplitudes and boost low amplitudes. Without a compressor-limiter we would always need a volume control at hand when watching TV to increase volume when people speak with a low-voice and decrease it in a loud passage. Explain whether a compressor-limiter is a linear time-invariant system. **(15 Points)**

It is not LTI because it is not homogeneous. The gain of the system changes with the amplitude of the input.