BSc Audiology
Year 1

AUDL1001 – Signals and Systems for Hearing and Speech

2007

EXAMINATION

Time allowed: 3 hours

There is one section in this examination.

Candidates should answer **ALL SIX** questions.

- Questions ONE to FOUR are worth 10 marks each.
- Questions FIVE and SIX are worth 20 marks each.

Answer Booklets

Candidates must write all answers in the Answer Booklets provided.

Candidates requiring additional answer booklets should contact an invigilator.

Any answers written on the Examination Paper will **not** be marked.

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Question 1

a) Suppose a hearing aid amplifies the amplitude of a sound outside the ear by a factor of ten (i.e. so that the sound pressure in the ear is ten times higher than it is outside the ear). What would be the dB SPL level inside the ear given an external sound level of 48 dB SPL? What are the input and output levels in Pa?

b) Volker normally has a threshold for a 1 kHz sinusoid of 30 μPa. After visiting a disco with loud music, Volker's threshold of hearing for a 1 kHz tone is temporarily raised by 17 dB. What is his new threshold of hearing in dB SPL?

c) The normal threshold of hearing at 125 Hz is 632.5 μPa. Stuart has a threshold that is 7 dB better than average at this frequency.

  i) What sound pressure (in Pa or μPa) is the least intense intense Stuart can hear at 125 Hz?
  ii) What is the normal average threshold in dB SPL at 125 Hz?

(10 marks)

a) factor of 10 = 20 dB, hence 68 dB SPL; 5023.8 and 50237.7 μPa. (0.005024 and 0.050238 Pa)

b) 30 μPa = 3.5 dB SPL so new threshold= 20.5 dB SPL

c) Stuart's threshold is -7 dB re 632.5 μPa = 282.5 μPa; 632.5 μPa = 30 dB SPL

Question 2

A complex periodic waveform is made up of sinusoidal components with the following frequencies or periods. For each combination, calculate the fundamental frequency and fundamental period.

a) frequencies = 120, 240 and 480 Hz

b) periods = 1 ms, 3 ms, 9 ms

c) frequencies = 150, 200 and 400 Hz

d) periods = 7.5 ms, 5 ms, 2.5 ms

e) frequencies = 1100, 1110 and 1500 Hz

(10 marks)

Please turn over.

120 Hz and 8.3 ms; 111.1 Hz and 9 ms; 50 Hz and 20 ms; 66.67 Hz and 15 ms; 10 Hz and 100 ms

Question 3

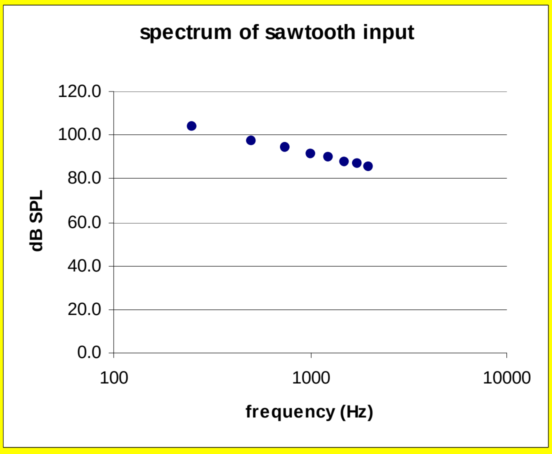
Consider a wave which consists of the first 8 harmonics of a sawtooth wave whose fundamental period is 4 ms, and whose fundamental component has a level of 3 Pa.

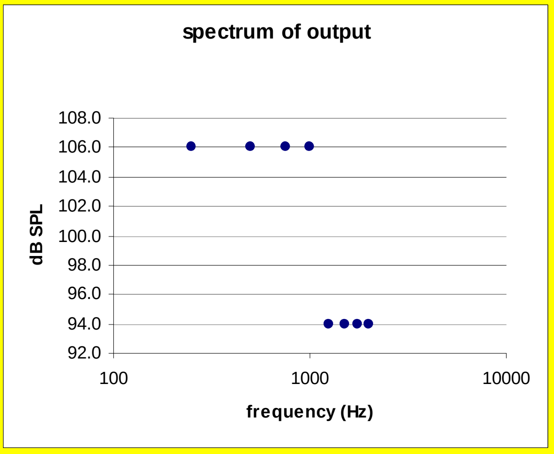
a) Draw its spectrum (on dB SPL and logarithmic frequency scales over the frequency range 125 Hz to 2 kHz).

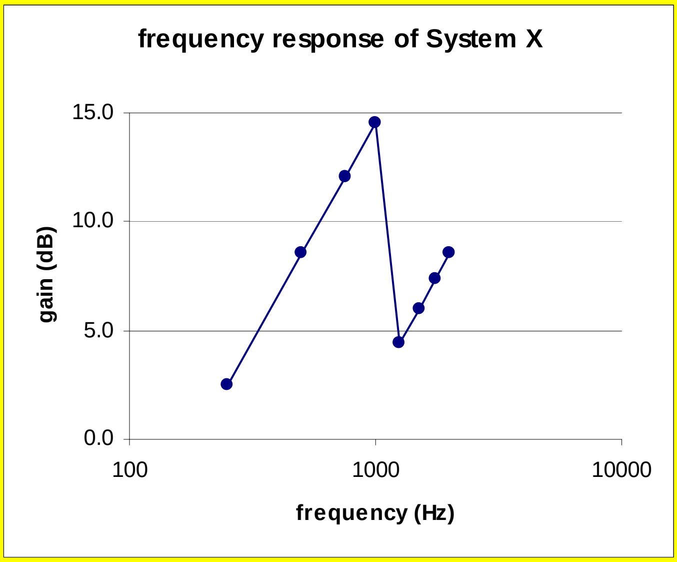
b) This wave is then put through a ‘System X’ which results in the output wave having a spectrum in which the first 4 components are at an equal amplitude of 4 Pa and the second four components at an equal amplitude of 1 Pa. Draw this spectrum on dB SPL and logarithmic frequency scales over the frequency 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’.

(10 marks)







Question 4

It is often said that the function of the basilar membrane can be likened to that of a filter bank.

a) Describe what a filter bank is, and how the notion of a filter bank can be used to understand peripheral auditory function.

b) What properties would the filter bank need to have in order to best mimic the functioning of the inner ear?

c) In what ways are these properties different from the filter bank used to make ordinary spectrograms?

(20 marks)

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 responds to the acoustic world through a single bandpass filter.
filter bank 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)
spectrogram has a filter bank with:
LTI filters of …

equal bandwidth
spaced linearly

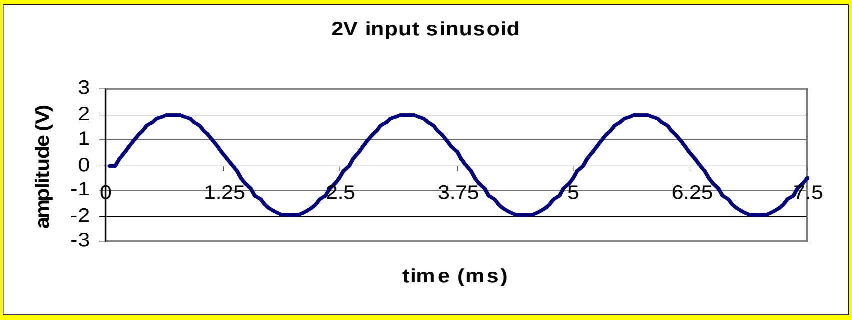
Question 5

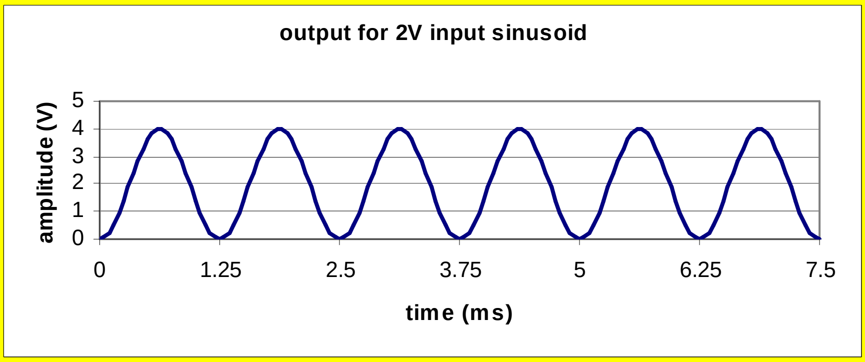
Suppose you had a system that squares each amplitude value of a waveform.

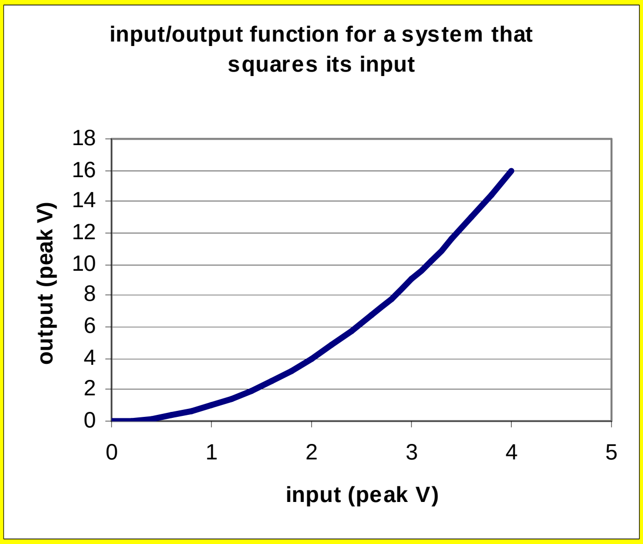
a) Draw input and output waveforms (3 cycles) for a sinusoid of peak amplitude 2 V and frequency of 400 Hz. What is the simplest way of knowing that this is not an LTI system?

b) Plot the input/output function (using peak amplitude) for a 400 Hz sinusoid over the range 0-4V. Is this system homogeneous? Time-invariant? Give reasons for your answers.

(20 marks)







Simplest: An input sinusoid results in an output that is not a sinusoid at the same frequency.
(Squaring a sinusoid does not result in a sinusoid of the same frequency; in fact, it results in a sinusoid at twice the input frequency.)

Homogeneous: no; show input/output function

Time-invariant: yes; mention effect of delay

Please turn over.

Question 6

In internet radio it is important to transfer the smallest amount of data possible.

a) Calculate how much data (in kbytes/s) needs to be transmitted for a hifi signal (mono, not stereo) that is sampled at 44100 samples/second and 32 bit-quantisation.

b) What two things can be done easily in order to bring down the amount of data?

c) What problems do you run into when you decrease the data rate too much?

(20 marks)

a) Data size of the signal:

44100 x 32 bits = 1411200 bits

1411200 bit/ 8 = 176400 bytes = 176.4 kb

b) Size can be decreased by reducing the number of quantisation levels and by reducing the number of samples per second (downsampling):

1) Number of quantization levels:

- **32 bit is far higher than required. 16 bit has a dynamic range of about 96 dB which covers the dynamic range of human hearing well enough.**
- **Too much decrease: The less quantisation levels are used below 16 bits the higher is the chance that quantization noise will be introduced. This is likely to become audible from about 8 bit.**

2) Downsampling:

- **44100 samples/second truly represents acoustic frequencies up to 22050 Hz (Nyquist frequency). This is higher than the highest frequency average human listeners can perceive. Downsampling to 32000 samples/second (highest represented frequency 16 kHz) would still cover most of the average human frequency perception and thus hardly affect the signal.**
- **Too much decrease: Higher frequencies get lost which is well audible. Signal sounds low pass filtered. However downsampling to 7000 samples/second is still perfectly intelligible as we can see in telephone signals.**

END OF PAPER