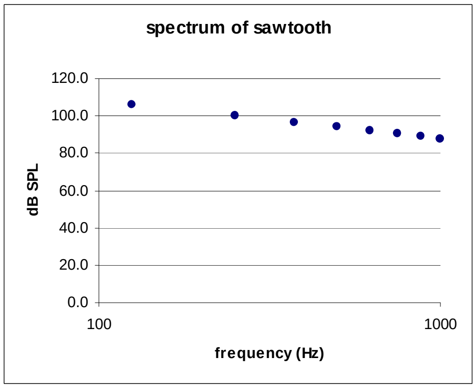
AUDLS001: May 2006

Final exam questions & answers

1) Consider a wave which consists of the first 8 harmonics of a sawtooth wave whose fundamental period is 8 ms, and whose fundamental component has a level of 4 Pa. Give its amplitude spectrum in a table and also draw its spectrum (on dB SPL and logarithmic frequency scales). What further information do you need to synthesise the time waveform. What does a spectrum (in general) tell you? **(10 points)**

harmonic	frequency	level (Pa)	dB SPL
1	125	4.00	106.0
2	250	2.00	100.0
3	375	1.33	96.5
4	500	1.00	94.0
5	625	0.80	92.0
6	750	0.67	90.5
7	875	0.57	89.1
8	1000	0.50	88.0



3 points: table of correct values (Pa or dB SPL)
3 points: drawn spectrum: lose 1 point for misuse of log x axis
1 point: synthesis requires the phase spectrum too
3 points: A spectrum tells you the frequency, amplitude and phase of the sinewaves that must be added together to obtain a particular wave.

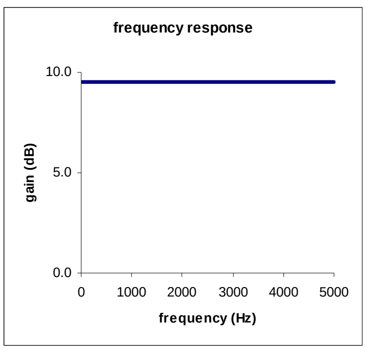
2) Each line of the following table indicates the spectral components in a complex periodic waveform. For each combination, calculate the fundamental frequency and fundamental period. Note that frequencies are given in hertz and periods in ms, and your answers should also be expressed in these units. **(10 points= 1 point/answer.)**

					Answer: frequency (Hz)	Answer: period (ms)
a)	frequencies	50	100	150	50	20.00
b)	periods	1	1.25	5	200	5.00
c)	frequencies	2100	2240	2590	70	14.29
d)	periods	2	1	0.5	500	2.00
e)	frequencies	2000	2700	3000	100	10.00

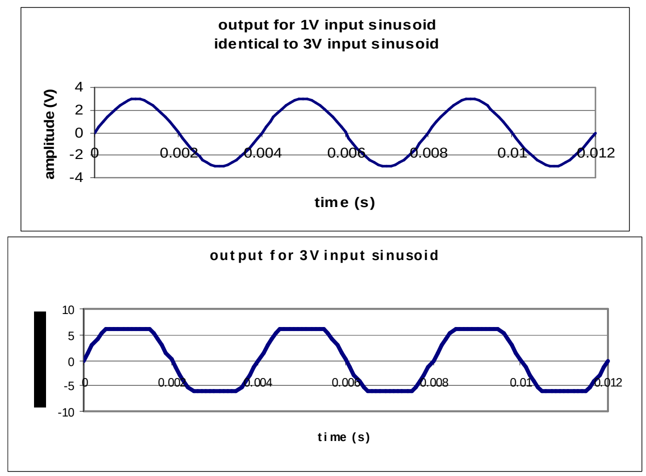
Allow ½ point if the frequency and period match, even if the fundamental implied is wrong.

3) Almost all systems *clip* signals that are too large to be handled by them, so any system can only be considered to be linear time-invariant (LTI) over some limited range of levels. Suppose you had a system that acted as a perfect amplifier with a gain of 3. However, the magnitude of the output voltage is strictly limited to 12 V peak-to-peak (so the minimum voltage is -6 V and the maximum 6 V). **(a)** Draw the frequency response of the system, on dB and linear frequency scales (0-5 kHz), assuming the input voltage to be 1 V. **(b)** Draw input and output waveforms for 3 cycles of a 250 Hz sinusoid when: (i) the peak voltage of the input is 1 V, and (ii) the peak voltage of the input is 3 V. In terms of a general description, what kinds of output waves do you obtain in the two cases (aperiodic, simple, complex, periodic)? **(c)** Make your best guess as to what the spectrum of the output to a 3 V input sinusoid would look like, and draw it (using linear scales on both axes). **(d)** Draw the input/output function of the system for a 250 Hz sinusoid for peak voltages ranging from 0 V to 5 V. **(e)** What's the easiest way to show that this system is not LTI for input voltages ranging from 0-5 V? Is it time-invariant? Is it homogeneous? Give reasons for your answers. **(20 points)**

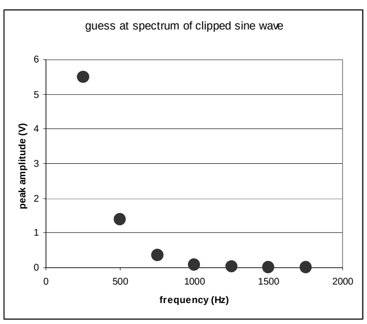
a)



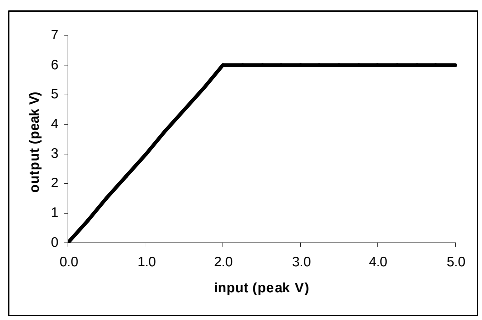
b)



c)



d)



a) 3 points

b) 4 points: sinewave output: simple periodic wave

clipped sinewave: complex periodic wave

c) 5 points: spectrum of clipped sine wave must have harmonics at multiples of 250 Hz, with decreasing amplitude; appropriately labeled axes

d) 3 points

e) 5 points: not LTI: An input sinusoid results in an output that is not a sinusoid, as seen above right. Or that the input/output function is not a straight line.

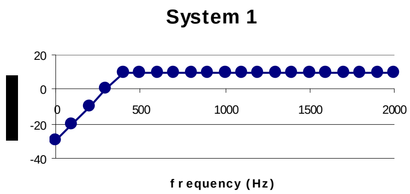
Homogeneous? no; as input/output function shows

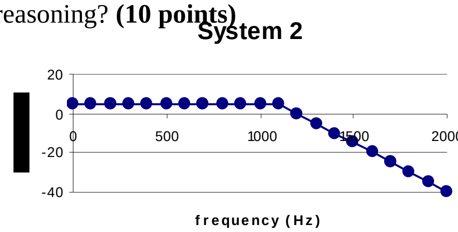
Time-invariant? yes; mention effect of delay

4) 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 how the notion of a filter bank can be used to understand peripheral auditory function. What properties would the filter bank need to have in order to best mimic the functioning of the inner ear? **(20 points) (2 points/point + 4 for extras/quality)**

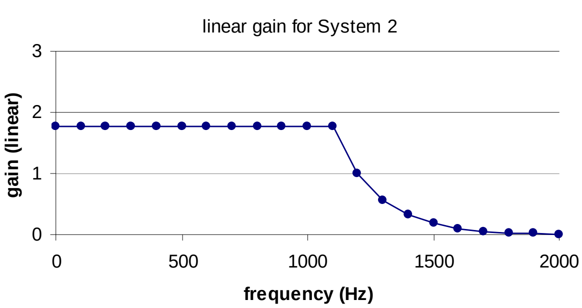
A filter bank is a collection or set of band-pass filters
whose centre frequency varies over some range
As the basilar membrane (BM) does a kind of frequency analysis, its function can be compared to a filter bank.
each auditory nerve fibre responding to the acoustic world through a single bandpass filter.
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)

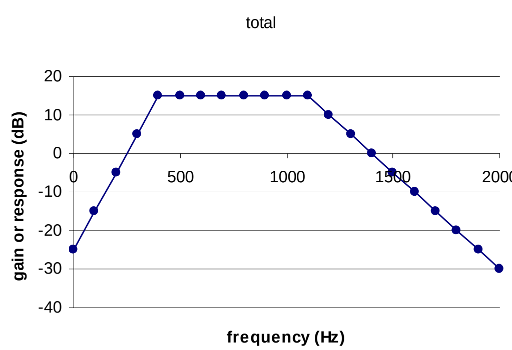
5) Consider the amplitude responses of the following two systems. What kind of a filter does each represent? Draw the amplitude response of System 2 on linear gain and linear frequency scales. Now draw the amplitude response of a cascade of these two systems, System 1 followed by System 2, on dB and linear frequency scales. What kind of a filter results from this arrangement? What would be the amplitude response of the cascade if the position of the two systems were reversed? Explain your reasoning? **(10 points)**





a) *2 points: High pass, low pass.*
b) *3 points: Linear gain of system 2*
c) *2 points: frequency response of Cascade + it's band-pass*
d) *3 points: It doesn't matter if the order of the systems is reversed because the gain values are added and addition doesn't depend on the order.*





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6) Draw input and output spectra of the following three signals passed through System 1 in question 5, over the frequency range 0-2 kHz, on dB and linear frequency scales. Ensure that your labels are accurate! **(10 points)**

a) White noise **b)** An impulse **c)** A periodic train of impulses with a period of 0.01 s.

White noise: Input spectrum must be a continuous flat line; output spectrum identical to frequency response of 1st system. Must be labeled dB SPL or some other appropriate measure.

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

Impulse train: Input spectrum must have harmonics of equal size at 100 Hz; output spectrum also with harmonics but with spectral envelope as in System 1. Must be labeled dB SPL or some other appropriate measure.

1 point/input spectrum; 2 points/output spectrum + 1 point for 'quality'

7) An engineer is trying to design a system for converting from analogue to digital form an electrical signal arising from scalp recordings of EEG. The decision is made to use 12 bits, and a sampling rate of 1000 samples/second. **(20 points)**

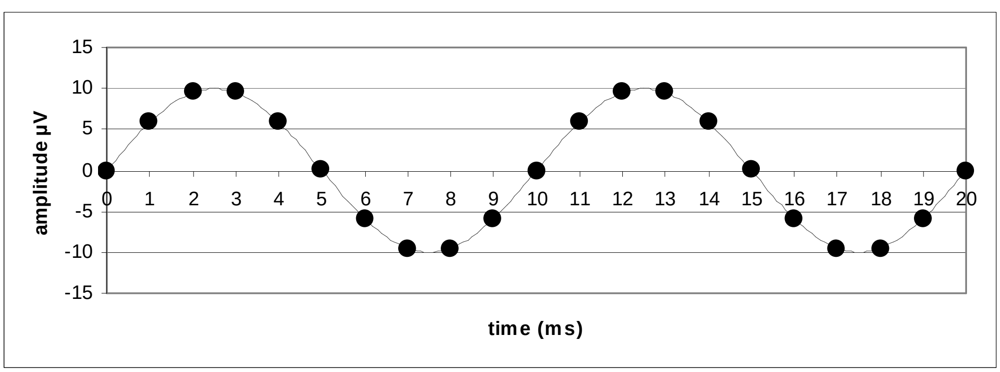
a) What are the limitations in the spectral content of the electrical wave that this system can handle? Explain the Nyquist frequency in this respect. What precautions should the engineer take to ensure the fidelity of his recordings as regards spectral content?

b) What dynamic range can this system handle?

c) Suppose the waveform from a particular electrode is a sinusoid at 100 Hz and a peak voltage of 10 μV. Sketch the first two periods of the digital wave that would be stored by this system. Do not interpolate the samples.

a) 10 points: Only frequencies less than half the sampling frequency can be represented accurately, so here < 500 Hz. The Nyquist frequency, also called the Nyquist limit, is the highest frequency that can be coded at a given sampling rate in order to be able to fully reconstruct the signal, namely half the sampling frequency. Low-pass anti-aliasing filtering at about 500 Hz needs to be implemented.

b) 4 points: 12 bits = 4096 levels = 72.2 dB



c) 6 points