You must complete all sections. Label all graphs. Show your work!

Section A: Short questions concerning Signals & Systems

A1. Give the sound pressure levels (in dB SPL) of the following sound pressures in Pa:

(a) 10 μPa (b) 40 μPa (c) 1 Pa (**3 marks**)

(a) -6 dB SPL, (b) 6 dB SPL (c) 94 dB SPL

A2. Give the sound pressures (in Pa or μPa) represented by the following levels in dB SPL:

(a) -20 dB SPL (b) 100 dB SPL (c) 0 dB SPL (**3 marks**)

(a) 2 μPa (b) 2 Pa (c) 20 μPa

A3. What is the difference between a peak value and an rms value for a wave? (**2 marks**)

peak is instantaneous maximum, whilst the rms is averaged over the whole wave (better reflects the energy in the wave)

A4. Please give brief answers to each of the following questions (one sentence) (**12 marks**)

a) At what frequency intervals are the harmonics of a periodic complex waveform spaced?

b) What is an impulse, and what is important about its spectrum?

c) What is an impulse response, and what is its significance?

d) What two main properties characterise a bandpass filter?

e) What makes sinusoids special in the analysis of LTI systems?

f) How can you calculate the frequency response of a cascade of two LTI systems from the individual frequency responses?

a) *whole number multiples of the fundamental frequency*

b) an imaginary signal that is infinitesimally narrow and infinitely high; contains all frequencies

c) the output of a system to an impulse; its spectrum is the same as the frequency response of the system

d) center frequency and bandwidth

e) sinusoids can be added together to make any wave; a sine wave through an LTI system can only be changed in amplitude and phase – no other way

f) add them together if the y-axis is on dB scales; multiply if they are linear scales

(2 marks each)

Section B: Signals & Systems

B1: Suppose you had an unknown complex periodic electrical wave with a period of 5 ms, and you wanted to determine the level of each of its harmonics up to a frequency of 5 kHz.

a) Specify the characteristics of a filter-bank that would be appropriate for doing this analysis. (5 marks)

b) Compare this filter-bank analysis to how the basilar membrane analyzes sound. What are the similarities? (5 marks)

c) What are the differences? (10 marks)

(Total 20 points)

Answer:

a) Ordinary filter-bank needs to consist of band-pass filters with evenly spaced collection of band-pass filters with a bandwidth less than 200 Hz so that single harmonics can be extracted. (5 marks)

b) Similarities: 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. (5 marks)

c) Differences: BM filters have bandwidths increasing with increasing frequency spaced on a quasi-logarithmic scale. BM filters are nonlinear (increase bandwidth and decrease gain with increasing level) (10 marks)

B2: Suppose you had a system that multiplies each input amplitude value by a factor of 4.

a) What change in dB does this increase correspond to? (1 marks)

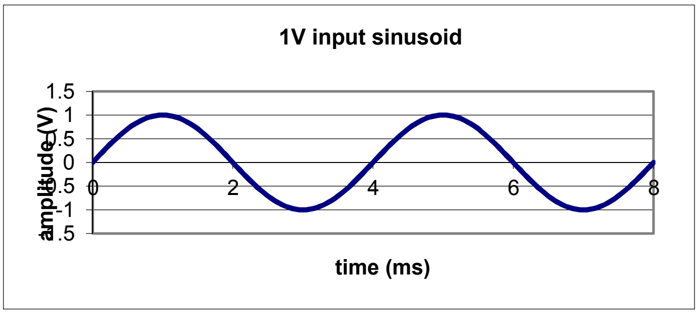
b) Draw input and output waveforms for 2 cycles of an input wave which is a sinusoid of peak amplitude 1 V and frequency of 250 Hz. (5 marks)

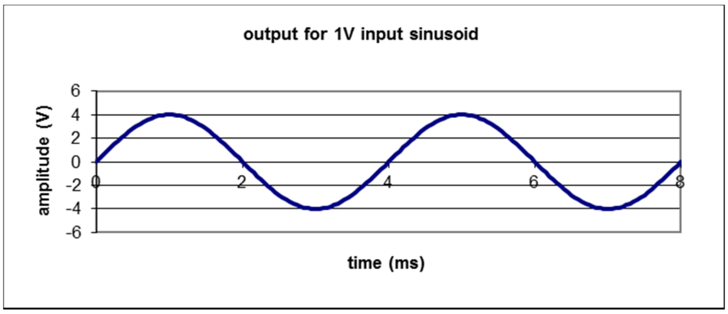
c) Given what you've been told about the response of LTI systems to sinusoids, do you think this system is LTI? Provide a one sentence explanation for your answer. (2 marks)

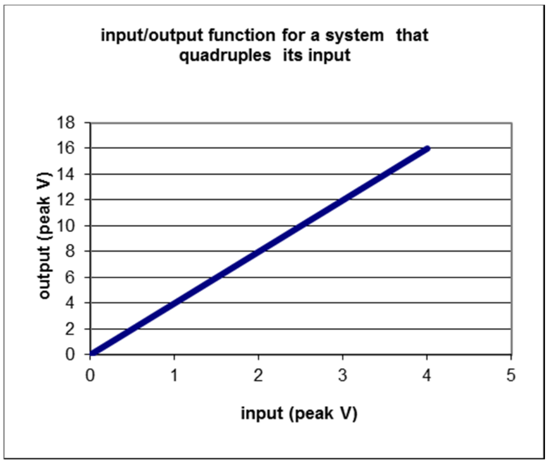
d) Is this system homogeneous)? Provide a one sentence explanation for your answer and sketch the input/output function on a graph.(5 marks)

e) Where could such a system occur in daily life? (2 marks)

(Total 15 marks**)**







Factor of 4 in amplitude is a change of +12 dB (1 marks)

graphs of input and output waves (5 marks)

System is LTI because a sinusoidal input always results in a sinusoidal output of the same frequency. (2 marks)

Homogeneity: yes. Proportionally equal changes are applied to each amplitude value thus input/output function showing a straight line going through the origin. (5 marks)

System could for example be an amplifier in a stereo system that quadruples the output amplitude of a particular waveform. (2 marks)

B3

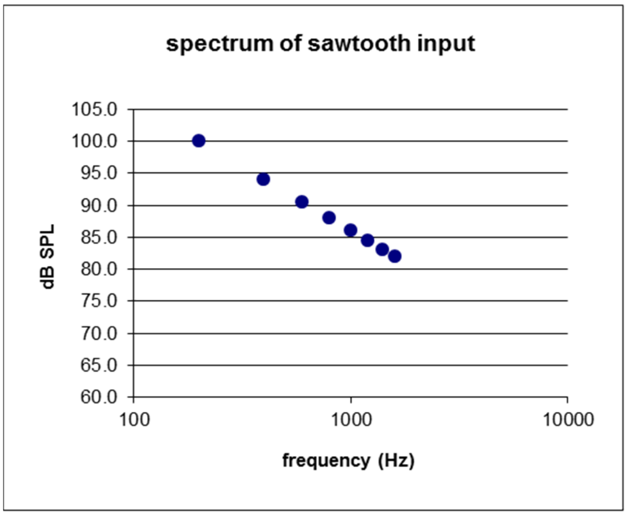
Consider a sound wave which consists of the first 8 harmonics of a sawtooth wave whose fundamental period is 5 ms, and whose fundamental component has a level of 2 Pa.

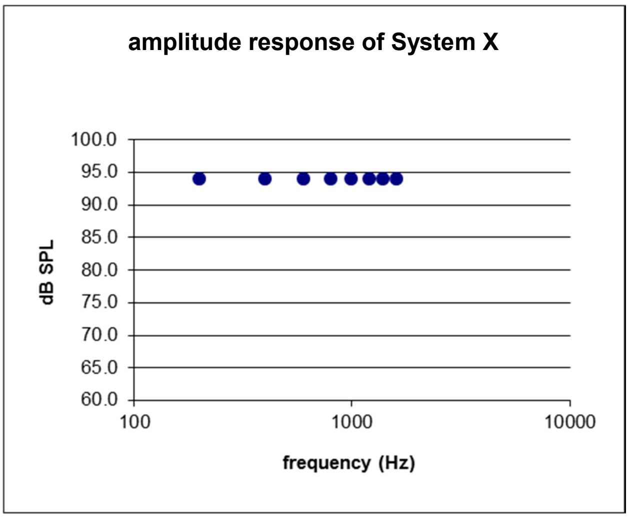
a) Draw its spectrum on a graph with dB SPL and logarithmic frequency scales over the frequency range 200 Hz to 3.2 kHz. *(5* marks*)*

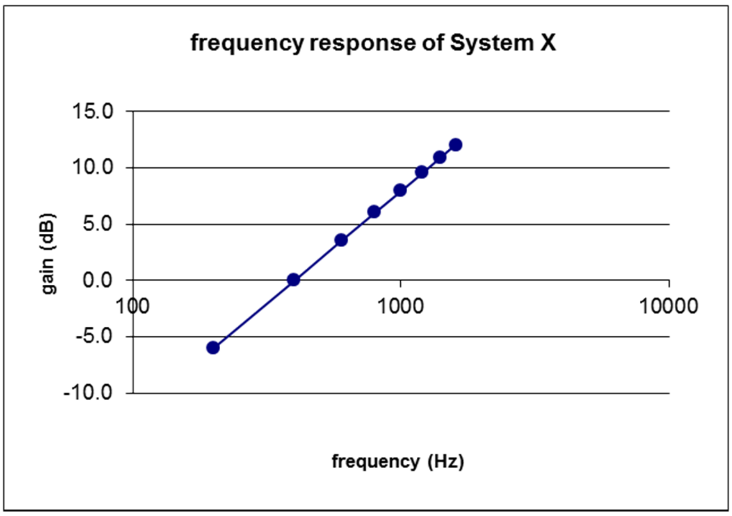
b) This sound wave is then put through 'System X' which results in the output wave having a spectrum in which all components are at an equal amplitude of 1 Pa. Draw this spectrum on a graph with dB SPL and logarithmic frequency scales over the frequency range 200 Hz to 3.2 kHz. *(3* marks*)*

c) Over the same frequency range, and again using dB and logarithmic frequency scales, draw a graph representing the amplitude response of 'System X'. *(7* marks*)*

(Total 15 **marks*)***







a) graph 5 points

b) graph 3 points

c) graph 7 points (also note that points can be given for a correct frequency response for wrongly defined initial spectra)

Section C: Outer Ear

Basic acoustics (15 Marks)

1) For the following short questions assume the speed of sound is c = 340 m/s
   a) How long does sound take to travel a distance of 34 cm? (1 MARK)
   b) What is the wavelength of a 1 kHz sound wave? (1 MARK)
   c) Distances can be measured using time (e.g. a light year). What is the distance between your ears measured in terms of the travel time for a sound coming directly from your right? Assume the head is 17cm wide and provide your answer in micro-seconds. (1 MARK)

2) 'Binaural summation' is a term used to describe an apparent increase in loudness for cochlear implant (CI) users wearing two CIs vs. those wearing just one. This increase in loudness is accompanied by a measureable increase in speech intelligibility in noise. As a young audiologist you are asked to help investigate the mechanism of binaural summation using new CI processing strategy for a binaural cochlear implant. The engineers have made a system that simply adds the two signals (in analog) from the two microphones (one from the left and one from the right) and then produces the single added output. The hope is that this might produce results similar to those seen in patients. Assume this system is in a test environment with a speech signal coming directly from the front (speech signal is identical in both ears) and noise signal that is not correlated between the two ears (different signals in both ears).
   a) How much larger would the combined *speech* signal level be at the system output (in dB) relative to the output level produced by the same signal from just one microphone? NOTE: output levels are always rms. (2 MARKS)
   b) How much larger would the combined *noise* signal level be at the system output (in dB) relative to the output level produced by the same signal from just one microphone? (You may assume the noise signal at the 2 microphones have different waveforms but the same rms levels). (2 MARKS)
   c) Considering the above might there be a benefit from this system under this signal in noise condition? What signal to noise ratio (SNR) advantage might you expect? Give a number to justify your answer. If you cannot give a numerical answer describe how to calculate the SNR for half marks. (3 MARKS)

3) You are consulted to help design the stage at a rock concert and you measure the average (LAeq) sound emanating from the stage during a typical set to be 114 dBA SPL at 4 meters. To protect the audience from over exposure, you are instructed to erect a fence where the average sound level will be 94 dBA. How far should you put the fence from the stage? Show your work as partial credit will be awarded. (5 MARKS)

Basic Acoustics Answers:

a) i) 1 ms, ii) 0.34 m, iii) 500 us

b) i) Identical speech signal sounds therefore a 6dB increase in signal level for two identical inputs vs. one input. ii) non-identical noise signal inputs of equal level will generate a 3dB increase in noise level from the 2 inputs vs. one input. iii) The SNR advantage would be (6-3 dB) or 3 dB.

c) Most will remember that a doubling of distance is equivalent to a reduction of sound pressure level of 6 dB SPL providing free field conditions are met (outside). This requires that pressure is inversely proportional to distance P = 1/r. Therefore if we need to reduce the pressure from the initial pressure of 2e-5*10^(114/20) to a final pressure of 2e-5*10(94/20) which is a factor of 10 we must move the fence back by a factor of 10 to 40 meters from the stage. This can also be solved by seeing that the reduction of 20 dB must conform to: 20*log10(r2/r1) and solve for r2 (since it is 20 dB we know the ratio must be a factor of 10). Half credit will be given for those who can only remember there is a 6dB loss for doubling of distance and come up with a number in the range of 32 meters to 64 meters.

Standing Waves (8 Marks)

4) Standing waves can build up in the ear canal. Industrial noise-induced hearing loss (NIHL) is usually diagnosed from survey data and from a 'noise notch' in the audiogram at 4 kHz. Because audiometric spacing is sparse we do not know the exact most common frequency for such a notch. It is interesting that this is also near the 'most sensitive' region of human hearing.

a) Consider the (only partially correct) hypothesis that this notch and the sensitivity of human hearing is completely due to the 'ear canal resonance'. Given that the average adult ear canal is 25mm long, calculate the exact frequency you would expect to measure at the maximum depth of the NIHL notch. Show your work. If you make any assumptions about the values of constants you use for your calculation state that clearly. HINT: trust yourself. (3 MARKS)

b) The uptake of MP3 players and the reduction of industrial noise at least for developing countries might shift the cause of common NIHL from industrial noise to earphone noise in the future. What might the difference mean to the shape of the audiograms used to diagnose NIHL in the future? (3 MARKS)

c) Why could measuring and diagnosing NIHL caused by earphone wearing be potentially more difficult than diagnosing NIHL due to normal industrial noise? HINT: think about how pure tone audiometry is measured and the topic of this section. (2 MARKS)

Standing Wave Answers

a) The relationship between frequency and length is determined by 'c = L*f' where L is the length of the wave. We know that the first mode of a standing wave in a tube with one end closed is one quarter of the wavelength (L/4). This means that the wavelength of the frequency we seek is 100mm. So assuming we are at STP and the speed of sound is 343m/s then f = (343m/s)/(100e-3) = 3.43 kHz
b) When headphones are worn, the ear canal is no longer open at one end. It would be better modeled as a double ended tube. This would mean that the ear canal resonance would no longer be at 3.43 kHz as described above but would be at twice this frequency (assuming the termination at the earphone is like a rigid wall and the same for the stapes). We might begin to see NIHL as 6 kHz or 8 kHz notches.
c) When measuring a pure tone audiogram the ear canal is coupled to a headphone.. This measurement situation is like a tube closed at both ends and therefore could present standing waves in the ear canal at the same frequencies we might expect NIHL from standing waves from earphones (the higher ones). NIHL could become over-diagnosed as false positives could be registered.. Extra Note: A small FM modulation 'warble' could be introduced into 'pure tone' audiometry to avoid this problem.

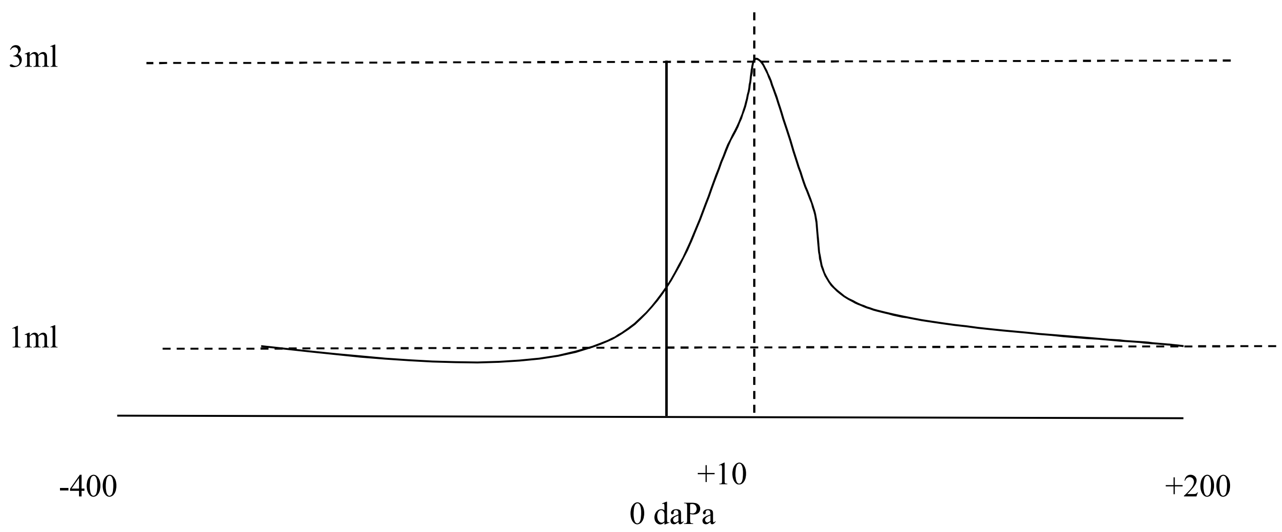
Section D: Middle Ear

1) Describe the fundamental problem our middle ear solves for listening to very low level sounds and how it solves it. Use the word 'impedance' in your answer and describe what structures of the middle ear are involved in solving the problem (5 MARKS).

2) When the acoustic reflex is activated and the tensor tympani and stapedius muscle are contracting, would this mainly affect the mass or spring constant of the middle ear system? How would this parameter change and what would you expect to happen to the frequency of resonance of the system and why? Your answer should include the equation for relating the mass and spring constant of the middle ear to its resonant frequency. (3 MARKS)

3) The middle ear can be modeled as a mass-spring system. The ossicles are the masses (which need to be moved) and the annular ligament of the stapes footplate (and other squishy parts of the middle ear) are the 'spring'. If the spring constant was measured at resonance to be 100 N/m and the mass of the bones is 2.5 mg, what would you measure as the resonant frequency (in Hz)? Show your work. (3 MARKS)

4) Tympanometers are used clinically to determine the functioning of the middle ear. The following graph is a sketch that shows a result of a test from a tympanometer. (4 MARKS TOTAL)



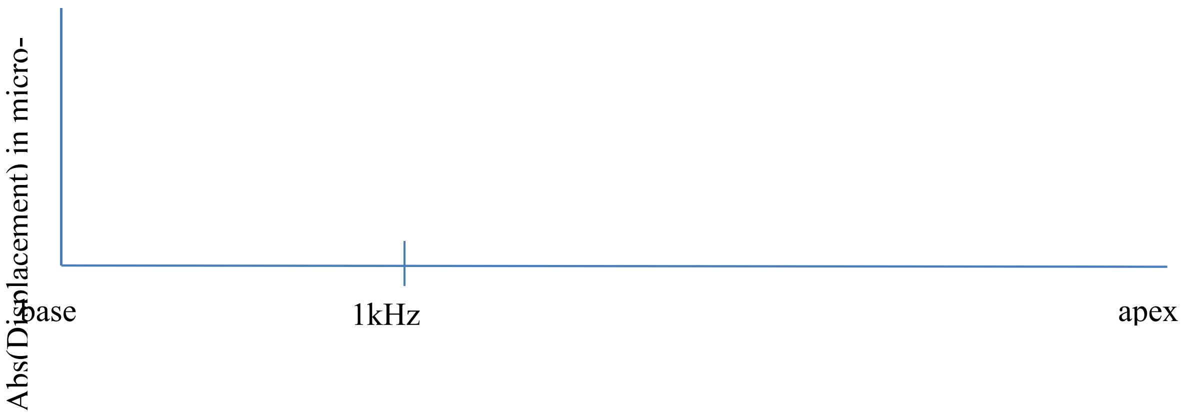
a) What is the pressure behind the tympanic membrane of the middle ear? (1 MARK).
b) What is the ear canal volume? (1 MARK)
c) Copy this graph onto your exam paper, and over the top of it, sketch a qualitative graph that describes how loud the test tone would appear to the patient during the test (use 0ml for no sound and 3ml for maximum loudness). (2 MARKS)
d) If the tympanic membrane had a hole in it, sketch and describe what the graph would probably look like, explain your answer. (1 MARK)

Middle Ear Topics Answers

1) The middle ear must perform impedance matching. Because the cochlea is filled with fluid that naturally has a higher mass (density) than air we need a greater pressure input to move it otherwise the sound energy would be reflected (just as it is reflected off the surface of water). In order to push the sound energy into the ear we need to increase the pressure. This is mainly achieved by the surface area ratio of TM to the oval window (this lets you gather many light air particles to collective push on a smaller number of fluid particles). The bones of the ear also add a level action and there is a buckling contribution. All these combine to give a pressure increase necessary to move the heavy cochlear fluids.
2) Muscle contractions would mostly affect the spring constants, making the system stiffer. By the equation for mechanical resonant frequency w0=sqrt(k/M) where k is the spring constant. If this constant increases the resonant frequency also increases. (10 Marks)
3) Using the formula w0=sqrt(k/M) we can solve this equation. Remember w = 2*pi*f0 so. f0 = 1/(2*pi) * sqrt(k/M); solving algebra gives f0 = 1000 Hz approximately. Or 1006.6 Hz.
4)
   a) 10 daPa
   b) 1 ml
   c) The graph would be the horizontal reflection of the current graph – 0 where the graph is a maximum and maximum where the graph is a minimum
   d) The graph would be flat and registering at a high admittance (> 3ml). The ear would be behaving like a large cavity.

Section E: Inner Ear: Otoacoustic Emissions

1) The way that DPOAES are generated is thought to be more complicated then low level TEOAEs. Replicate the axes below in your exam book, and then draw the envelope of the traveling waves on the basilar membrane for the F1 and F2 frequencies, assuming that F1 is 1000 Hz and F2 is 1,200 Hz. Mark clearly where the distortion product OAEs are generated and separately where they are amplified. (5 MARKS).



2) What is the expected frequency of the human DPOAE generated from the stimulus used above? (2 MARKS)

3) Name the remaining 3 types of OAES and describe how they are measured. Extra points will be awarded for describing how they are generated within the cochlea. (6 MARKS)

4) OAEs using low level stimuli can be very informative about a particularly important cochlear function that will only be intact in a healthy cochlea. What is this function and what structures are responsible for it? (2 MARKS)

Inner Ear Answers:

a) A BM shaped wave at 1000 Hz, a BM shaped traveling wave at 1200 Hz. The region of overlap below the 1200 Hz wave is the area of interaction this is the site of DPOAE generation. However , the site of amplification is at 800 Hz.

b) This would be 2F1 – F2 = 800 Hz.

c) 1. Spontaneous OAEs (SOAES) are measured without any evoking stimulus. They can be measured by placing a microphone in the ear canal and then looking at the spectrum of the recorded signal. These are generated by the method of coherent reflection which involves amplification and the summation of micro-reflections off of cochlear microstructures.

2. Click evoked OAEs (CEOAES) are measured by introducing short duration (80us) low level click (84 dBpe) into the ear canal. The ear's response is a delayed 'echo' of the signal which can be separated in time from the stimulus. Extra: Analysis of the response shows that high frequencies return first and lower frequencies return later showing a phenomenon known as dispersion.

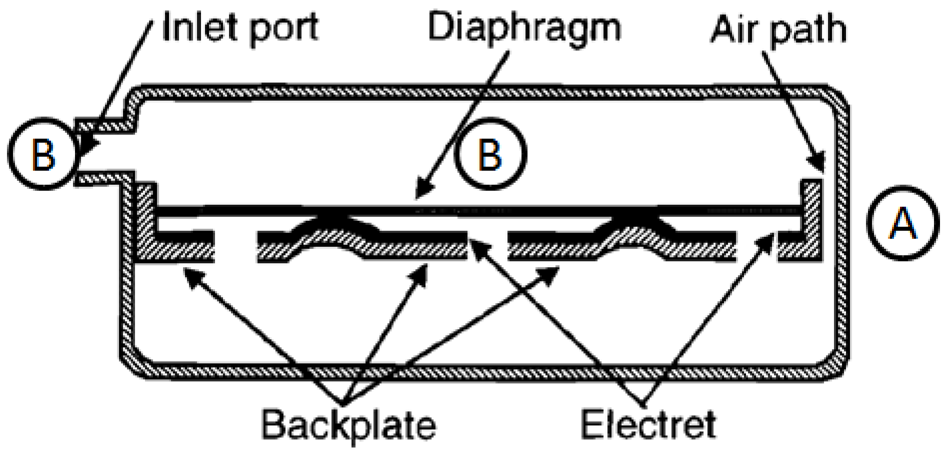
3. Stimulus frequency OAES (SFOAEs). These are evoked using a continuous signal tone and the ear responds by producing a sinusoid at the same frequency as the stimulus. Because the response happens at the same time and at the same frequency as the input, SFOAEs cannot be separated from the stimulus in the ear canal unless the stimulus is swept in frequency and an oscillating pattern emerges that cannot be due to the stimulus, or the emission component of the recorded signal is changed while observing the SFOAE for changes (for example by evoking medial olivocochlear efferent reflex, MOCR, using contralateral stimulation).

d) All emissions that are evoked by low-level signals require non-linear processes. Several non-linearities are possible that can generate emissions but the main necessary component is amplification. This is most obvious with spontaneous emissions where the ear is spontaneously generating noise which cannot be done without an energy consumption amplification process. But in all cases amplification is necessary to observe the OAEs, without amplification, the ear's emission response levels would either not exist (SOAES) or would be too small to measure in the ear canal (DPOAEs). The main structure responsible for amplification in the cochlea is the outer hair cells. So in a basic sense the existence of OAEs tells you about the health of cochlea and in particular the functioning of the outer hair cells. However no convincing research has proved that the quantity or level of the measured emission can tell you about the quantity or level of health of the OHCs. There are just too many noise and internal OAE generating variations which affect those measured levels. This prevents OAEs from being more than binary diagnostic at the moment.

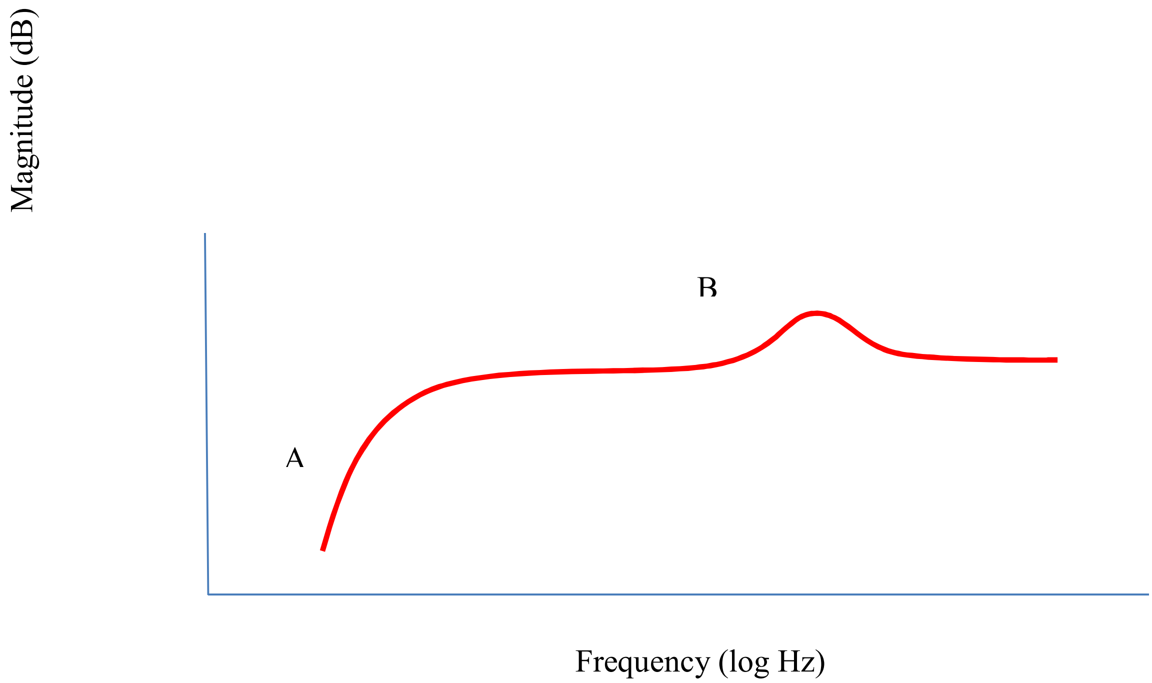
This section was only included in the exam for GS08, not for GAV1.

Section F: Directional microphones

Below is a schematic of a typical omnidirectional microphone. The two letters label parts that correspond to elements that create two different effects on the frequency response of the microphone. Note that the letter 'B' denotes two parts that act together. Plot what you would expect the frequency response curve for this microphone to be, labeling the regions affected by the structures labeled 'A' and 'B' on the graph. Explain how the structures 'A' and 'B' create this response (you can assume the device has an 'electrically flat' response, i.e. that any deviations from a flat response are entirely due to the acoustics of A and B). (8 MARKS)



Answers to Special Topics



A. An intentional low frequency cut is introduced in hearing aid mics to eliminate low frequency noise by the addition of a small tube between the front and back of the diaphragm through which only low frequency sounds can pass (hint: think of a straw)
B. An unintentional peak of ~5dB between 4-5kHz is an artifact of the **acoustic 'Helmholtz Resonator'** created by the mass of air in the inlet port and the volume of air in front and behind the diaphram (hint: think blowing across a jug)