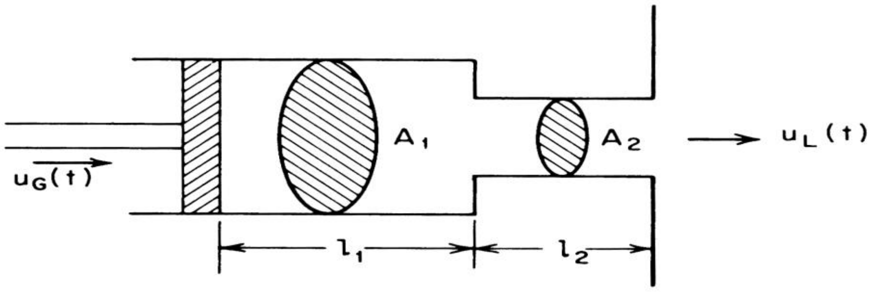
Homework 4 (ECE6255 Spring 2010) Grade=4/100

1. Consider an ideal lossless tube model for the production of vowels consisting of two sections as shown. Assume that the terminations at the glottis and lips are completely lossless. For the above conditions the system function of the tube model will be obtained from the equation on Slide 53 in Lecture 8, by substituting the two boundary reflection coefficients with $r_G = r_L = 1$, and $r_1 = (A_2 - A_1)/(A_2 + A_1)$ at the intersection of the tube sections.



   a) Show that the poles of the system are on the $j\Omega$ axis and are located at the values of Ω satisfying the following equations:

$$\cos\Omega(\tau_1 + \tau_2) + r_1 \cos\Omega(\tau_2 - \tau_1) = 0$$

   or equivalently,

$$(A_1 / A_2)\tan\Omega\tau_2 = \cot\Omega\tau_1$$

   with $\tau_1 = l_1 / c$ and $\tau_2 = l_2 / c$;

   The values of Ω that satisfy equations derived in Part (a) are the formant frequencies of the lossless tube model. By judicious choice of the parameters of A_1, l_1, A_2 and l_2, we can approximate the above equations to obtain the formant frequencies. The following Table gives parameters for several assumed vowel configurations. Solve for the formant frequencies for each case. Use *c* = 35000 cm/sec as the velocity of sound. (Note the nonlinear equations have to be solved iteratively, graphically or by a program.)

Vowel	l_1	A_1	l_2	A_2
/i/	9 cm	8 cm^2	6 cm	1 cm^2
/ae/	4 cm	1 cm^2	13 cm	8 cm^2
/a/	9 cm	1 cm^2	8 cm	7 cm^2
/Λ/	17 cm	6 cm^2	0	6 cm^2

2. Work out Problem 4.14 in Quatieri.

3. Work out Problem 7.5 in Quatieri.

4. Work out Problem 7.6 in Quatieri.

5. A proposed digital filter bank pitch detector consists of a bank of digital bandpass filters with lower and upper cutoff frequencies given as

$$F_k = 2^{k-1} F_1 \text{ and } F_{k+1} = 2^k F_1 \ , \ k = 1, 2, ..., M \ ;$$

This choice of cutoff frequencies gives the filter bank the property that if the input is periodic with fundamental frequency, F_0, such that $F_k < F_0 < F_{k+1}$, then the filter outputs of bands 1 to *k - 1* will have little energy, the output of band *k* will contain the fundamental frequency, and the bands *k+1* to *M* will contain 1 or more harmonics. Thus, by following each filter output with a detector which can detect pure tones, a good indication of pitch can be obtained.

a) Determine F_1 and *M* such that this method would work for pitch frequencies ranging from 50 Hz to 800 Hz;
b) Sketch the required frequency response of each of the *M* bandpass filters;
c) Can you suggest simple ways to implement the tone detector required at the output of each filter?
d) What type of problems would you anticipate in implementing this method using non-ideal bandpass filters?

What would happen if the input speech were band-limited from 300Hz to 3,300Hz, e.g. telephone band? Can you suggest improvement in these cases?

6. Write a MATLAB program to speed up a speech file by a factor of 2-to-1. Use the method of overlap-add to analyze the STFT of the signal (using a rectangular window of length 512 samples with 256 sample overlap between frames), throw out every other frame, and re-synthesize the speeded-up speech. Plot the original speech file and the speeded up speech file, and plot the narrow band spectrograms of both the original and the speeded up speech files. Use the speech file 1.wav and 5.wav to test your program.