1. Write a MATLAB program to read in a speech file with a sampling rate of F_S=16kHz and filter it to bandwidths of 6, 4, and 2kHz. Save the filtered speech to files using the MATLAB **wavwrite** command. Listen to each of the resulting files and describe the effect of lowpass filtering on speech intelligibility and quality. (Use the speech file **test_16k.wav** to test your program and generate the filtered speech files.)

2. Use MATLAB to compute and plot the vocal tract log magnitude spectrum and mark the locations of the formants for a two-tube model of the vocal tract. The model is shown in the following figure,

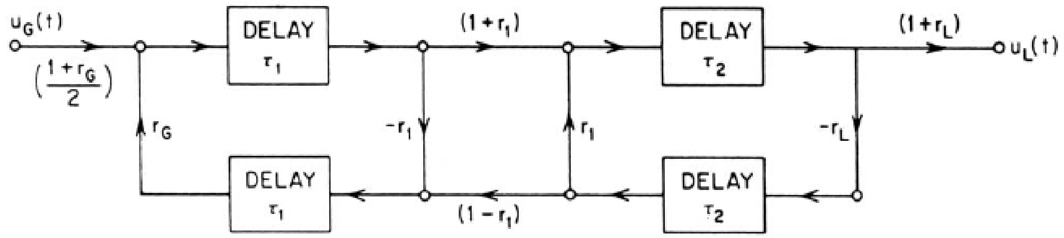

Fig. 3.37 Complete flow diagram of a two-tube model.

and the frequency response of this model is given as

$$V_a(\Omega) = \frac{U_L(\Omega)}{U_G(\Omega)}$$

$$= \frac{0.5(1+r_G)(1+r_L)(1+r_1)e^{-j\Omega(\tau_1+\tau_2)}}{1+r_1 r_G e^{-j\Omega 2\tau_1} + r_1 r_L e^{-j\Omega 2\tau_2} + r_L r_G e^{-j\Omega 2(\tau_1+\tau_2)}}$$

Your MATLAB code should accept the input lengths (l_1 and l_2 in cm) and areas(A_1 and A_2 in cm^2) of a two-tube model of the vocal tract, along with the reflection coefficients at the glottis (r_G) and at the lips (r_L). Test your code on the following examples:

(1) $l_1 = 10, A_1 = 1, l_2 = 7.5, A_2 = 1, r_G = 0.7, r_L = 0.7$

(2) $l_1 = 15.5, A_1 = 8, l_2 = 2, A_2 = 1, r_G = 0.7, r_L = 0.7$

(3) $l_1 = 9.5, A_1 = 1, l_2 = 8, A_2 = 8, r_G = 0.7, r_L = 0.7$

(4) $l_1 = 8.75, A_1 = 8, l_2 = 8.75, A_2 = 1, r_G = 0.7, r_L = 0.7$

You can use a value of $c = 35{,}000$ cm/sec as the speed of sound. What happens to the log magnitude spectral plots if both r_G and r_L are set to 1.0 (rather than 0.7)?