Exercises:

1. We have an analysis system based on LPC with an input signal s[n]. We want to compute the preduction error signal e[n]. This signal has the form: e[n] = s[n] + 0.25s[n-2]

   a. Find the coefficients α_k, and the function corresponding to the prediction error filter A(z), also known as inverse filter

   b. Find all the parameters from the system function H(z), this means, G and a_k, knowing that H(z) = 1.25, for $z = e^{j0}$

   c. Find the frequency or frequencies of the formants, if we work with a sampling frequency of 8 kHz.

   Remember that the system function H(z) of our LPC speech model, and its frequency response $H(e^{jw})$ are:

$$H(z) = \frac{S(z)}{E(z)} = \frac{G}{1 - \sum_{k=1}^{p} a_k z^{-k}}. \qquad H(e^{j\omega}) = \frac{G}{1 - \sum_{k=1}^{p} \alpha_k e^{-j\omega k}}$$

2. We decided to use the autocorrelation method to compute the parameters of an LPC system. We use the following system of equations:

$$\boxed{\sum_{k=1}^{p} \alpha_k R_n(|i-k|) = R_n(i), \quad 1 \le i \le p}$$

   Which can be written as the following matrix equation:

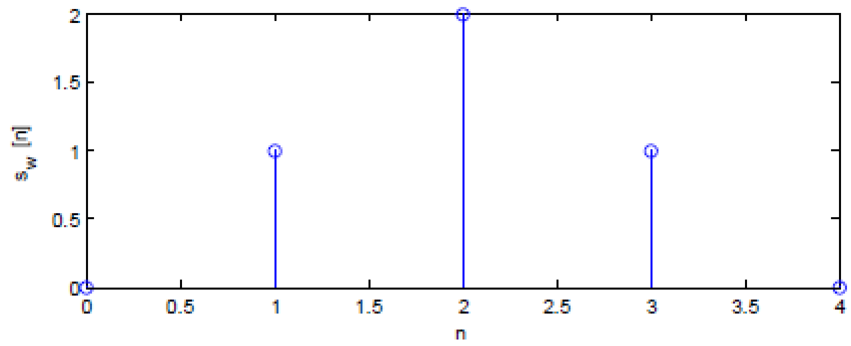
$$\begin{bmatrix} \phi[0] & \phi[1] & \cdots & \phi[p-1] \\ \phi[1] & \phi[0] & \cdots & \phi[p-2] \\ \cdots & \cdots & \cdots & \cdots \\ \phi[p-1] & \phi[p-2] & \cdots & \phi[0] \end{bmatrix} \begin{bmatrix} \alpha_1 \\ \alpha_2 \\ \cdots \\ \alpha_p \end{bmatrix} = \begin{bmatrix} \phi[1] \\ \phi[2] \\ \cdots \\ \phi[p] \end{bmatrix}$$

   , where the autocorrelation of a signal is defined as

$$R_n(k) = \sum_{m=0}^{N-1-k} s_n(m) s_n(m+k)$$

$$E_n = \phi_n(0,0) - \sum_{k=1}^{p} \alpha_k \phi_n(0,k)$$

$$= R_n(0) - \sum_{k=1}^{p} \alpha_k R_n(k)$$

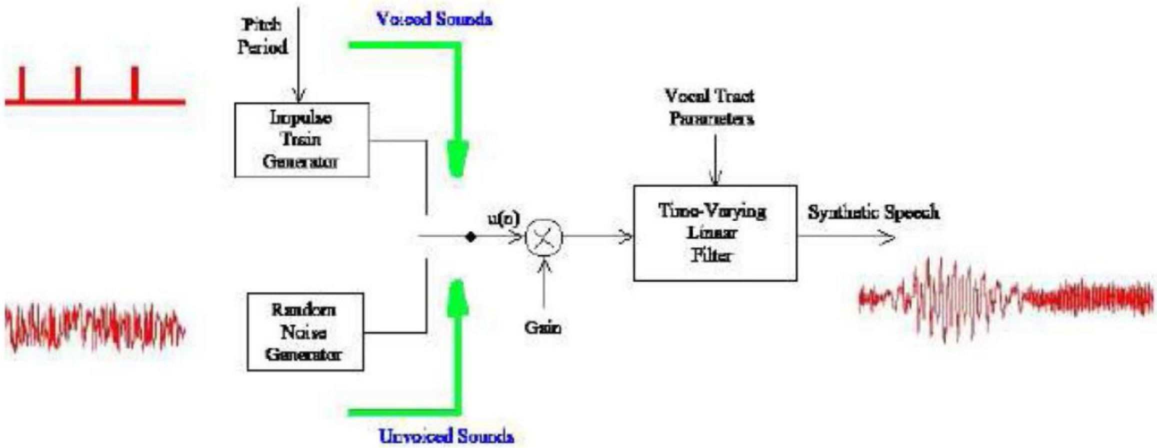


a. The signal $s_w[n]$ presented in the previous Figure, is a signal s[n] which has been windowed with 5 samples. Find out the values (different than 0) of the autocorrelation function $R_n(k)$, for values –Inf<k<+Inf

b. Solve the system of equations to find the coefficients α_k, if p = 2.

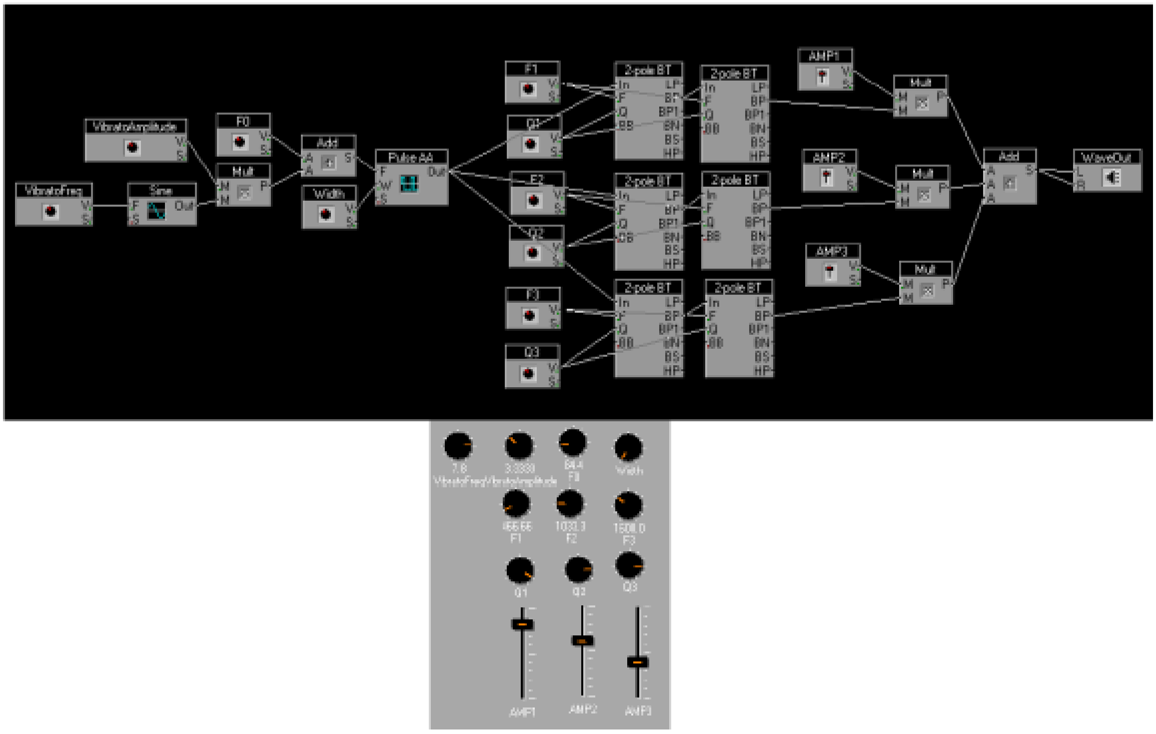
c. Find the parameters of the system function H(z) for the synthesis model

3. The following Figure represents the block diagram of source-filter model



Explain briefly its functioning and explain how does this block diagram relate to the production of human speech.

The following Figures represents a simulation of this model using the software SyncModular.



4. Explain briefly its functioning, and its relation to the previous Figure. Which would be the range for the control parameters?

Materials

1. SyncModular. http://www.sync-modular.org/

2. RTSect. http://www.speech.kth.se/music/downloads/smptool/RTSect.exe

3. Use the file sourcefilter.sme from stud.IP