SGN-21006 Advanced Signal Processing

Exercise 5: NLMS

Introduction to Adaptive noice canceling

Consider an adaptive filtering application known as adaptive noise canceling (ANC). ANC effectively analyzes the effects of a certain environment (a room) on the recording of audio data. The setup is presented in Fig. 1a. The room contains two sound sources $c(t)$ and $v(t)$. The source $c(t)$ is considered to be a clear speech sound without any perturbations. The source $v(t)$ is an unwanted sound (noise) and our goal is to remove it. The trick is, that we do not observe exactly the noise $v(t)$ since the environment corrupts the sound traveling from the source $v(t)$ by the time it is recorded as the output sequence $s(t)$. The output $s(t)$ is, thus, a superposition of the clear source $c(t)$ and the noise that passed through the room. In reality, such a scenario is present for example in our mobile phones which typically contain two microphones: one at the top observing the noise and one at the bottom observing our speech and the corrupted noise. Note that in a real application, the clear sound $c(t)$ is not available! We are, however, also using a simulated environment where $c(t)$ is available and can be used for error analysis.

Your task is to find an adaptive noise canceler, Fig. 1b. An FIR filter with coefficients that can change in time is used to cancel the noise.

The application is based on the important observation that the signals $v(t)$ and $c(t)$ are not correlated i.e. the speech sound is not travelling to the microphone that records the noise. When this assumption is true, we have

$$s(t) = v_0(t) + c(t), \tag{1}$$

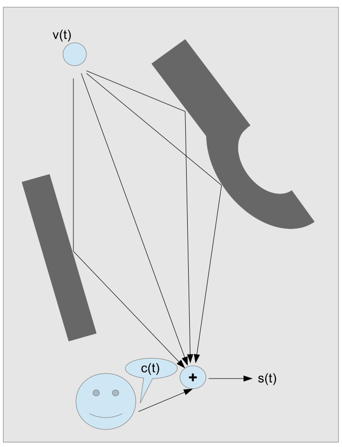
where $v_0(t)$ is the perturbed version of $v(t)$. If we compute the error we, it results that

$$e(t) = s(t) - \sum_{i=1}^{n} w_i(t)v(t-i) = c(t) + v_0(t) - \sum_{i=1}^{n} w_i(t)v(t-i). \tag{2}$$

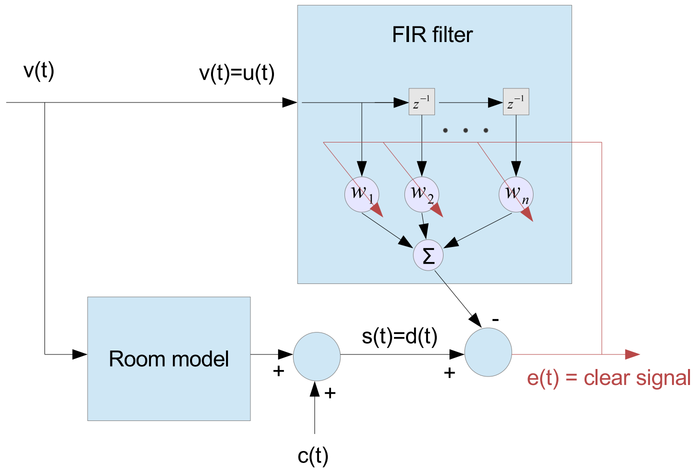
The goal is to use an adptive algorithm to minimize the square error. This translates to finding the coefficients $w_i(t)$ for each time instant t such that $E\{e(t)^2\}$ is minimized. Under our hypothesis ($v(t)$ and $c(t)$ are uncorrelated) it results that

$$E\{e(t)^2\} = E\{c(t)^2\} + E\{(v_0(t) - \sum_{i=1}^{n} w_i(t)v(t-i))^2\}. \tag{3}$$

From (3), since $E\{c(t)^2\}$ does not depend on the coefficients $w_i(t)$, it results that $E\{e(t)^2\}$ is minimized if $E\{(v_0(t) - \sum_{i=1}^{n} w_i(t)v(t-i))^2\}$ attains its minimum. In such case, we can conclude that statistically $e(t)$ will be close to $c(t)$ and thus that the output error approximates the clear signal $c(t)$.



(a) Audio recording setup.



(b) ANC block diagram.

Figure 1: Adaptive noise canceling.

Test signals

For testing ANC you are given genarated signals.

There are two different rooms: one constant (imagine a person standing still in a quiet room) and one varying (the person is moving or something is happening around him).

- `clear_speech.wav` - the clear speech source $c(t)$.
- `noise_source.wav` - a random noise source $\nu(t)$.
- `speech_and_noise_through_room_1.wav` - the measured signal $s(t)$. This corresponds to the signal $\nu(t)$ distorted by passing it through an FIR filter of length 200, the filter coefficients are constant.
- `speech_and_noise_through_room_2.wav` - the measured signal $s(t)$. This corresponds to the signal $\nu(t)$ distorted by passing it through an FIR filter of length 200, the filter coefficients are changing in time.

Tasks

1. a) Implement the NLMS algorithm. (Hint: See previous exercise and e.g. slide 22 of lecture 5). What is the purpose of the α parameter?

b) Use your algorithm with given signals. If you have headphones, listen to all the signals. How does NLMS perform?

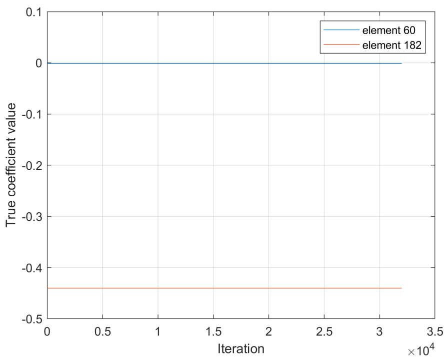
2. a) Write a line of code that computes how big is the error. We define it as average square error (ASE) for the second half of the signal (i.e. for the stationary regime):

$$ASE = \frac{\|\mathbf{c}(N_s/2 : N_s) - \mathbf{e}(N_s/2 : N_s)\|_2^2}{\|\mathbf{c}(N_s/2 : N_s)\|_2^2},$$

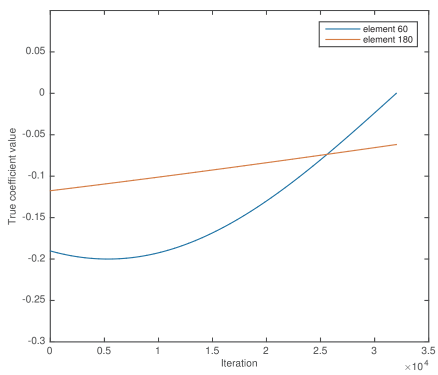
where N_s is the length of the vector (number of samples), $\mathbf{c}$ and $\mathbf{e}$ are vectors containing the values of $c(t)$ and $e(t)$ for all t.

b) Vary μ and compute ASE for the different room models. What is the best μ for each data set? Is it different for different inputs? Why?

3. Save the evolution of the coefficients in the NLMS function. Select two entries with indices 60 and 182 from room1 and plot how they change in time. Do the same for room2 for with indices 60 and 180. Which of the room models requires an adaptive filtering algorithm? You can see the true coefficient values below. Is NLMS able to adapt?



(a) Room 1



(b) Room 2

Figure 2: The true coefficient values.