Signals and Systems in Bioengineering

Master on Biomedical Engineering

1st Semester 2020/2021

Adaptive Filtering

Lab Session 5

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This work is about adaptive filtering using the canonical structure displayed in Fig.1

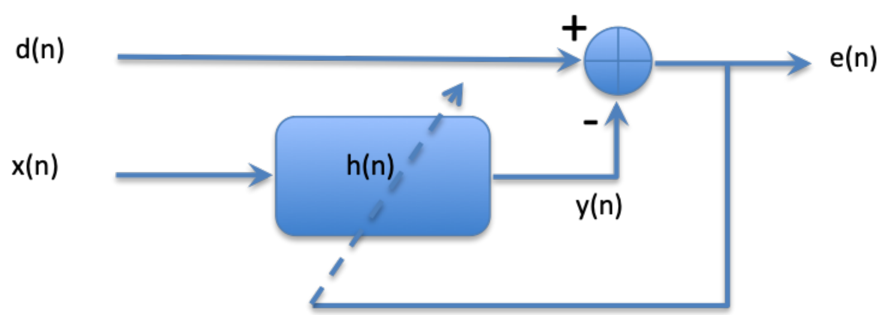

Fig. 1: Canonical adaptive filter topology

I. Synthetic data

1) Write a MatLab script/function to filter a noisy signal $d(n)$ given a signal correlated with the noise, $x(n)$, according the scheme displayed in Figure 1,

$$[e, y] = adaptFilter(d, x, p) \tag{1}$$

where p is the order of the FIR filter $h(n)$.

2) Generate a step function $s(n) = u(t(n) - 0.5)$ with a sampling frequency, $f_s = 4000$ Hz and duration of $T = 1$ second where $t(n) = n/f_s$.

   Test the function (1) by making $d(n) = s(n)$ and $x(n) = 1$. **Comment** the results, namely at the transition region.

3) Generate a Gaussian white noise signal, with the same dimensions of $s(n)$, $\eta(n)$, and filter it (using the MatLab function *filter*) with a low-pass 10 order Butterworth filter, $g(n)$, with a cut-off frequency of $\pi/3$ rad/sample,

$$\epsilon(n) = g(n) * \eta(n) \tag{2}$$

   and make a noisy version of $s(n)$ corrupting it with *Additive White Gaission Noise* (AWGN) according with

$$d(n) = s(n) + \epsilon(n) \tag{3}$$

   a) Display $\eta(n)$ and $\epsilon(n)$ and **comment** the differences.
   b) Display together $s(n)$ and $d(n)$.

4) Use (1), represented in Fig.1, to filter $d(n)$ by making $x(n) = \eta(n)$ and $p = 50$.

   a) Represent $e(n)$ and $s(n)$ together in the same figure and

   b) Compare the impulse responses $g(\tau)$ and $h(\tau)$ estimated along the process at evenly distributed times separated by 0.1 seconds, $t_k = 0.1k$ s.

      Compute and display $J(n) = \|g(\tau) - h_n(\tau)\|$ where $h_n(\tau)$ is the estimated FIR filter $h(\tau)$ at each discrete time point. **Comment**.

   c) Present and **comment** the results, namely, the ability of the system to estimate the $g(n)$.

5) Repeat 4b) using the LMS algorithm for several values of the parameter μ. **Comment**, namely the ability of the algorithm to adapt to the discontinuity at $t = 0.5$ seconds.

II. REAL DATA

Use the LMS algorithm to separate the voices from the background music in both pairs of audio data. Assure the sampling rate is the same in both audio data and confirm they are aligned (if you need you can down sample the files and use only part of them to reduce the computational burden of the process).

1) ”musica-voz.wav” and ”musica.wav”.
2) ”HouseCuddy.mp3” and the original ”RSHouse.mp4” (you can downloaded it from the YouTube, here).

Comment the results and the parameters used to tune the algorithm.

III. ENCRYPTION

Load the image House8000.png and creatively find hidden information but ...As the philosopher Jagger once said: You Cant Always Get What You Want ....

Fig. 2: You Cant Always Get What You Want