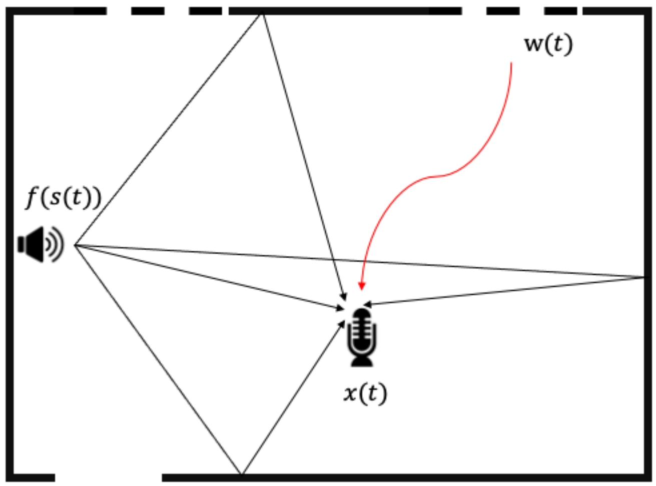
28 October, 2021

Advanced Digital Signal Processing 2021/2022
WS #1: System Identification

1 Audio signal recovery



Let us consider the scenario depicted in the figure, where a microphone is recording an audio signal $x(t)$ in a room. The input signal $s(t)$ is first amplified and transferred to a loudspeaker and then it is played. Assuming that the microphone is distortion-less, the recorded audio signal can be expressed as

$$x(t) = h(t) * f(s(t)) + w(t) \tag{1}$$

where $h(t)$ is the impulse response of the environment, i.e, the reverberations of the environment, $f(.)$ is the distortion produced by the audio amplifier, and $w(t)$ is the background noise. The goal of the WS is to recover the input signal $s(t)$ by removing the distortion effects $f(.)$ and the reverberations of the environment $h(t)$. In particular:

(a) considering the recorded signal **x1** in **data_1.mat**, obtained assuming $h(t) = \delta(t)$ (e.g., microphone close to the loudspeaker), estimate the distortion parameters using the discrete-time model:

$$x[n] = f(s([n])) + w[n] \tag{2}$$

where $w[n] \sim \mathcal{N}(0, \sigma_w^2)$, with $\sigma_w^2 = -20dB$. The distortion function $f(.)$ is a polynomial of order p such that $f(s[n]) = a_1\, s[n] + a_2\, s^2[n] + \cdots + a_p\, s^p[n]$. Determine the coefficients of the distortion function $\{a_1, a_2, \ldots, a_p\}$ and restore the clean signal $s[n]$.

(b) considering the recorded signal **x2** in **data_2.mat**, obtained assuming a perfect restoring of the audio signal $s(t)$, estimate (and compensate) the environment reverberations $h(t)$ using the discrete-time model

$$x[n] = h[n] * s[n] + w[n]; \tag{3}$$

where $w[n] \sim \mathcal{N}(0, \sigma_w^2)$, with $\sigma_w^2 = -34dB$.

(c) using the distortion function $\hat{f}$ estimated in (a) and the estimated system $\hat{h}[n]$ in (b), compute through Monte-Carlo simulations the mean square error (MSE) for signal recovery, distortion recovery, and system identification, assuming the received signal according to the system model in (1) and for $\sigma_w^2 = [-34dB : 2dB : 10dB]$.