Acoustic Echo Cancellation

Mini Project 1 of Array and Sensor Signal Processing 2012

Last revised: October 8, 2015

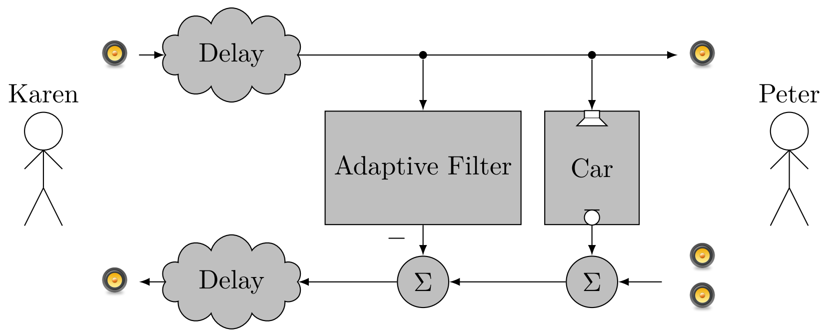

Figure 1: Overview over the problem of acoustic echo cancellation.

Karen makes a phone call to her friend Peter who is driving his car. Due to legal reasons, Peter has installed a hands free setup in his car based on the build-in loudspeaker of the car and an installed microphone. In this setup, Karen's voice is convolved with the impulse response of the acoustical path between the build-in loudspeakers of the car and the installed microphone. Therefore, Karen receives not only Peter's voice and the car sound, but also her own voice in a delayed and down-sampled version. Karen thinks that this acoustic echo is very annoying, and your task is to attenuate it. The problem is illustrated in figure 1.

Q1: Rewrite the problem in figure 1 as the block diagram in figure 3.2 of the lecture notes (ignore the delays). Map the signals and blocks in figure 1 to $u(n)$, $d(n)$, $y(n)$, $e(n)$, $v(n)$, $z(n)$, $\boldsymbol{w}_o$, and $\boldsymbol{w}(n)$. Describe in words what the adaptive filter should do.

The acoustical transfer function between the build-in loudspeakers of the car and the installed microphone depends on the reverberation within the car. To simulate this reverberation, use Schroeder's artificial reverberation algorithm. An implementation of it may be found in the enclosed Matlab function `schroeder_reverb.m`, and you can read more about the Schroeder's artificial reverberator in the enclosed pdf-file `reverb_intro.pdf`. If you want to make the problem a bit easier for yourself (and lower the computation time), you can resample the signals to a common sampling frequency of 8 kHz, say.

Q2: For a filter length of 2000, a reverberation time of 300 ms, a pre-delay of 0 s, a reverberation level of 1, a corner frequency of 1000 Hz, and $v(n)$ being a white Gaussian noise signal so that $d(n)$ has a signal-to-noise ratio (SNR) of 30 dB (you can use `awgn()` in Matlab), design the adaptive filter so that Karen's echo is attenuated.

Q3: Same as in Q2, but now you should let $v(n)$ be car noise and/or Peter's voice. Experiment with the reverberation time, and the mixing levels between $z(n)$ and $v(n)$. How does the length of the filter depend on the reverberation time?

Q4: Sketch a diagram for an adaptive filter with two or more subbands. What are the benefits and disadvantages of such an adaptive filterbank?

Documentation of Results

We expect that you make a slideshow presentation on your findings. The presentation should be approximately 15 minutes and cover the answers to the four questions above. You are free to select and use your favourite adaptive filter algorithm(s), and you are also allowed to use Matlab's fast implementation of it/them. The presentation should be sent to `jkn@es.aau.dk` no later than October 18, 2015. In the end of November, we will have a small workshop where we present the results of the mini project for each other.