Adaptive Array Signal Processing: Assignment

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Final assignment (assignment B)

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Chapter 1

B: Beamforming in hearing aids

The main goal of this assignment is to use your knowledge about adaptive array signal processing, your creativity, and your tools from the previous chapters to come up with a creative solution for a wireless binaural hearing aid.
The total amount of credits for this assignment is 20. The amount of credits for each of the sub assignment is denoted in text between square brackets [ ].

1.1 Preview

1.1.1 Background

Modern hearing aids are capable of performing a variety of advanced digital signal processing tasks. Improving the intelligibility and quality of speech in noise through beamforming is arguably the most sought after feature among hearing aid users. In this assignment, you will investigate various aspects of beamforming in modern hearing aids. Most hearing aids

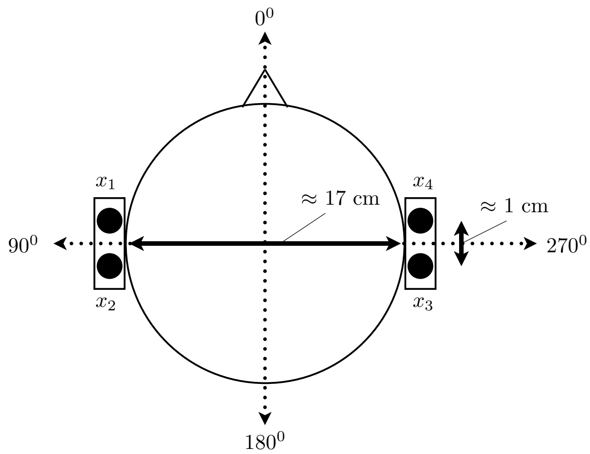

Figure 1.1: *A schematic depiction of hearing aids mounted on a user's ears. The two microphones in a single hearing aid are arranged in an end-fire configuration, and the left-right microphones are in a broadside configuration.*

contain two closely spaced omnidirectional microphones in an end-fire configuration (see setup of the placement of four microphones x_1, x_2, x_3 and x_4 as depicted in Fig.1.1), with an inter-element spacing of $0.8 - 1$ cm. As human hearing is binaural by nature, it is intuitive to expect an improved experience by using a hearing aid for each ear, and the number of such fittings has increased significantly. The larger spacing (≈ 17 cm) between microphones in binaural systems (two hearing aids) provides more flexibility for tasks such as beamforming. The desired source is commonly assumed to be in front of the user, as is typical in a normal conversation. Interfering sources can be located anywhere.

1.1.2 Broadband GSC

In this assignment each group will design a broadband adaptive Generalized Sidelobe Canceller (GSC) from which the global setup is depicted in Fig.1.2. In order to keep the figure

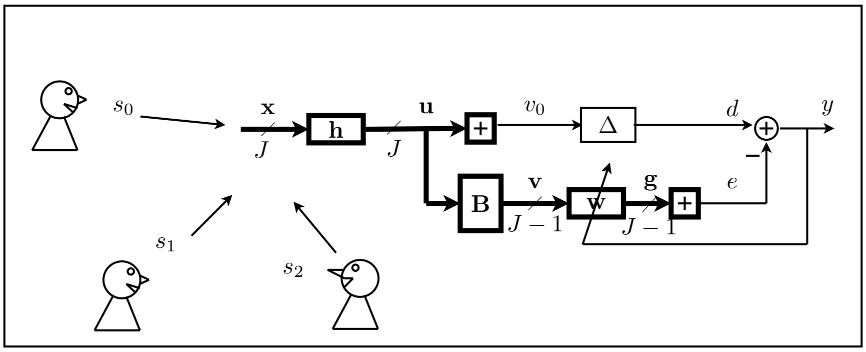

Figure 1.2: *Adaptive GSC structure.*

as simple as possible, only symbols (with no time indices) are represented. Furthermore we will assume that the setup takes place in 2D (two dimensional case). The desired speech signal $s_0(t)$ is corrupted by undesired signals $s_1(t)$ and $s_2(t)$. All signals $s_i(t)$ are broadband signals. These signals arrive at an array consisting of J microphones. The data is generated in such a way that the signals $s_i(t)$ are in the far field, there are no reflections and the microphones have ideal omnidirectional characteristics. Thus, the microphones pick up only delayed versions of the broadband signals. Each microphone ideally converts the measured analog signal to the discrete-time signal samples $x_j[nT_s]$, with $T_s = 1/f_s$:

$$x_j[nT_s] = \sum_{i=0}^{2} s_i(t)|_{t=(n-\tau_{i,j})T_s} \text{ for } j = 0, 1, \cdots, J-1 \tag{1.1}$$

with time index $n \in \mathbb{Z}$ an integer number. The delay $\tau_{i,j} \in \mathbb{R}$ is not restricted to integer numbers. Each of the J measured sensor signals $x_j[nT_s]$ is stacked into a length N vector

$$\underline{\mathbf{x}}_j[nT_s] = (x_j[nT_s], x_j[(n-1)T_s], \cdots, x_j[(n-(N-1))T_s])^t \tag{1.2}$$

The first step of the GSC structure is to manipulate the measured broadband signals in such a way that the desired signal is time-aligned in all sensor branches. This is achieved by a set of J length N FIR filters $\underline{\mathbf{h}}_j$. The result is a new set of J beam-steered signals $u_j[nT_s]$. Thus for $j = 0, 1, \cdots, J-1$ we have:

$$\begin{aligned} u_j[nT_s] &= \sum_{i=0}^{N-1} x_j[(n-i)T_s]h_{j,i} = \underline{\mathbf{x}}_j^t[nT_s] \cdot \underline{\mathbf{h}}_j \\ \underline{\mathbf{h}}_j &= (h_{j,0}, h_{j,1}, \cdots, h_{j,N-1})^t \end{aligned} \tag{1.3}$$

A delay-and-sum beamformer is constructed in the upper branch of the GSC structure by adding the J signals $u_j[nT_s]$. This results in the signal v_0, which is defined as:

$$v_0[nT_s] = \frac{1}{J}\sum_{j=0}^{J-1} u_j[nT_s] \tag{1.4}$$

The $(J-1) \times J$ blocking matrix $\mathbf{B}$ in the lower branch combines the J beam-steered signals $u_j[nT_s]$ in such a way that the $J-1$ resulting signals $v_j[nT_s]$ do not contain the desired signal. Now each of these $J-1$ signals can be filtered by a length M adaptive FIR filter $\underline{\mathbf{w}}_p[nT_s]$. The result is a new set of $J-1$ signals $g_p[nT_s]$. Thus for $p = 1, 2, \cdots, J-1$ we have:

$$\begin{aligned}
g_p[nT_s] &= \sum_{i=0}^{M-1} v_p[(n-i)T_s] w_{p,i}[nT_s] = \underline{\mathbf{v}}_p^t[nT_s] \cdot \underline{\mathbf{w}}_p[nT_s] \\
\underline{\mathbf{v}}_p[nT_s] &= (v_p[(n)T_s], v_p[(n-1)T_s], \cdots, v_p[(n-(M-1))T_s])^t \\
\underline{\mathbf{w}}_p[nT_s] &= (w_{p,0}[nT_s], \cdots, w_{p,M-1}[nT_s])^t
\end{aligned} \tag{1.5}$$

These $J-1$ signals are summed, resulting in the signal e, which is defined as:

$$e[nT_s] = \frac{1}{J-1}\sum_{p=1}^{J-1} g_p[nT_s]. \tag{1.6}$$

This signal is an estimate of the undesired signal component, which is left in the upper branch after the broadband Delay-and-Sum Beamformer (DSB). For causality reasons an extra delay Δ is needed in the upper branch. So the output v_0 of the DSB is delayed by Δ, whicht results in signal d. By subtracting signal e from the upper branch signal d we obtain the output signal y.

1.2 Scenario 1: Analyse hearing aid array configuration *[4 credits]*

Report about the following steps by using the setup as described en depicted in the preview section:

a) Consider a desired source located at 0 degrees and an interferer located at 135 degrees. Use beamsteering to design a beamformer for this scenario using:

   i) An array consisting of the two microphones x_1 and x_2 on the left hearing aid

   ii) An array consisting of the first mic x_1 on the left hearing aid and the first mic x_4 on the right hearing aid

   iii) An array consisting of all four microphones x_1, x_2, x_3 and x_4.

   In all cases, plot the beampattern at least at 500 Hz, 1 kHz, 2 kHz and 4 kHz, and describe/interpret the similarities/differences. Use the inter-sensor spacing provided in Fig.1.1.

b) Repeat the tasks in part a), with the only difference being that the interferer is located at 45 degrees. Again, comment on the beampatterns at the frequencies mentioned above, and on the differences/similarities with the behavior in part a).

Note: *In all cases use the geometrical center of the (sub) array as the origin.*

1.3 Fractional delay *[1 credit]*

In general the delays of the broadband setup can have any non-integer value. A way to cope with such non-integer delays is to use fractional delays. The general scheme of a fractional delay filter is given in Fig. 1.3. In this figure the fractional delay equals $z^{-i/L}$, with

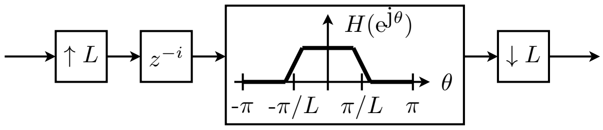

Figure 1.3: *General scheme of fractional delay .*

both i and L integer values. An efficient implementation of a fractional delay filter is given in the Matlab programma `delay.m`. The function `h=delay(d, fd, L, dmax)` returns the fractional delay impulse response `h` of the delay $z^{-(d+\frac{fd}{L})}$, where the parameters `d`, `fd`, and `L` are all integers. The parameter `dmax` is the maximum delay in integer, i.e., `ceil(i/L)`, which is used to obtain delay filters with the same lengths and synchronization.
In order to become familiar with the delay programme you must report about the following steps:

a) Derive an analytical expression for the impulse response $d[n]$ of the delay τ by using the inverse Fourier transform of the frequency response

$$D(e^{j\theta}) = e^{-j\tau\theta}. \tag{1.7}$$

b) Make in Matlab a figure containing the `plots` of the resulting analytical expressions of the impulse responses $d_0[n]$ and $d_1[n]$ for the delays $\tau_0 = 3$ and $\tau_1 = 5.15$.

c) Generate the impulse responses $\hat{d}_0[n]$ and $\hat{d}_1[n]$, of the delays $\tau_0 = 3$ and $\tau_1 = 5, 15$ by using the fractional delay programme `delay.m`. In order to compare these results with the above derived theoretical results, use the Matlab function `stem` to plot these results into the previous figure. Explain how you cope with causality.

1.4 Scenario 2: GSC applied to monaural hearing aid *[6 credits]*

You are provided a data set **DatasetAssignBs2.mat** consisting of speech signals observed in a set-up similar to the one shown in Fig.1.1 and Fig.1.2. [1] The desired source s_0 is near 0 degrees but the locations of interfering source(s) are unknown and may vary. In this scenario you should make use of a monaural hearing aid consisting of an array with two microphones x_1 and x_2 from the left hearing aid. Report about the following steps:

a) Implement the delay and sum beamformer that forms the upper part of a GSC. Listen to the output and describe the results.

b) Design and calculate a blocking matrix **B** that blocks the desired source. Verify your result.

[1] **Note**: *The file in the data set contains the four measured signals $x_1, \cdots, x_4$ as row vectors in the matrix x. For scenario 2, only x_1 and x_2 have to be used, which are the first row vectors of matrix x. The sample frequency is also included.*

c) Explain which adaptive filter algorithm you would choose in the lower branch to suppress the interference and noise in the upper branch. Implement this an adaptive filter. **Note**: *You may need to add a delay or a filter in the upper branch for causality.*

d) Store the final filter coefficients in a matrix. Based on these coefficients and the delay filters, calculate the transfer function in the frequency domain from the signals $\mathbf{x}$ to signal y. Use the multitone beamformer tool from Assignment A2 to plot the multitone beampattern. Compare your results with the results of scenario 1.

1.5 Scenario 3: GSC applied to binaural hearing aids *[5 credits]*

In this scenario you should make use of a binaural hearing aid consisting of an array with two microphones x_1 and x_2 from the left hearing aid and two microphones x_3 and x_4 from the right hearing aid. Perform the same steps as in the monaural case (scenario 2) and compare your results with the results of scenario 1 and 2.

1.6 Scenario 4: Wireless binaural hearing aid *[4 credit]*

It is intuitive to expect that using four microphones will result in better performance than using two microphones. Using all four microphones however requires an exchange of signals between the left and right hearing aids. A wired link between the two devices is cumbersome and unacceptable from a user-design perspective, thus necessitating a wireless link. Wireless transmission of data is power intensive, and to preserve battery life, it becomes important to limit the amount of data exchanged over the link.

Can you "describe" an intelligent scheme[2] that limits the amount of data exchanged between the devices? Use your creativity and the conclusions from scenario 1,2 and 3.

[2]You do not have to implement this scheme yourself. Give a clear picture and a description.