DASP3rd Chapter 7 - Exercises

1 Room Impulse Responses

1. How can we measure a room impulse response?

   Solution:

   - Can be measured by impulse excitation and recording of the result.
   - Can be obtained by performing the cross correlation between the driving signal and the systems response. For white noise we have $r_{xx}(m) = \sigma_x^2 \cdot \delta(n)$ and $r_{xy}(m) = r_{xx}(m) * h(n) \Rightarrow r_{xy}(m) \approx \sigma_x^2 \cdot h(n)$.

2. What kind of test signal is necessary?

   Solution:

   - Unit impulse (shotguns, baloons or hand claps)
   - White noise (pseudo noise: MLS - Maximum Length Sequence)
   - Sine sweeps

3. How does the length of the impulse response affect the length of the test signal?

   Solution:

   - The driving signal has to be longer than the impulse response, otherwise aliasing in the periodic cross-correlation occurs

2 First Reflections

For a given sound (voice sound), calculate the delay time of a single first reflection. Write a Matlab program for the following computations.

1. Why do we have to choose this delay time? What coefficient should be used for this delay time?

   Solution:

   - According to Ando, the preferred delay time of a single reflection with the ACF of the signal r_{xx} is determined from $|r_{xx}(t_1)| = 0.1 \cdot r_{xx}(0)$.

2. Write an algorithm which performs the convolution of the input mono signal with two impulse responses which simulate a reflection to the left output $y_L(n)$ and a second reflection to the right output $y_R(n)$. Check the results by listening to the output sound.

   Solution:

   - see Matlab script 'dasp_ex7_2_2'

3. Improve your algorithm to simulate two reflections which can be positioned to any angle inside the stereo mix.

   Solution:

   - see Matlab script 'dasp_ex7_2_3'

3 Comb and Allpass Filters

1. **Comb Filters:** Based on the Schroeder algorithm, draw a signal flow graph for a comb filter consisting of a single delay line of M samples with a feedback loop containing an attenuation factor g.

   (a) Derive the transfer function of the comb filter.

   Solution:
   - $H(z) = \frac{z^{-M}}{1-gz^{-M}} = \frac{1}{z^M-g}$

   (b) Now the attenuation factor g is in the feed-forward path and in the feedback loop no attenuation is applied. Why can we consider the impulse response of this model to be similar to the previous one?

   Solution:
   - see Matlab script 'dasp_ex7_3_1'

   (c) In both cases, how should we choose the gain factor? What will happen if we do not respect that?

   Solution:
   - $|g| < 1$
   - Otherwise, instability occurs (if $|g| > 1$, each echo will be louder than the previous echo, producing a never-ending, growing series of echoes)

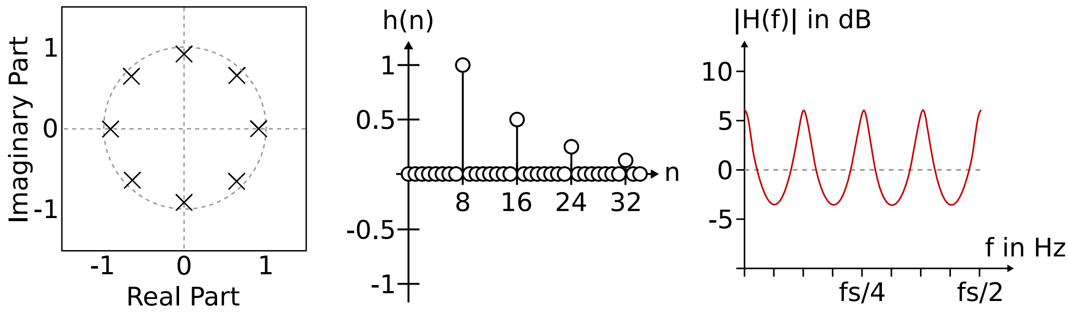
   (d) Calculate the reverberation time of the comb filter for $f_S = 44.1\,\text{kHz}$, $M = 8$, and g specified previously.

   Solution:
   - The reverberation time T_{60} is defined as the time the impulse response takes to decay to -60dB.
   - For every trip around the feedback loop, the sound is attenuated by $20\log_{10}(g)$
   - We have $G = 20\log_{10}(g)$ and $R = 20\log_{10}(r)$, where $r = \frac{h(n)}{h(n-1)}$ is the decay constant per sampling period $\rightarrow R = \frac{G}{M}$ attenuation per sampling period
   - $\frac{-60}{T_{60}} = \frac{R}{T_s}$, we can derive $T_{60} = -60\frac{T_s}{R} = -60\frac{T_s M}{G}$
   - $T_{60} = -60\frac{T_s M}{20\cdot\log_{10}(g)} = \frac{3}{\log_{10}|1/g|}T_s \cdot M = \frac{3}{\log_{10}|\sqrt{2}|}\frac{1}{44100\frac{1}{s}} \cdot 8 = 0.0036s.$

   (e) Make a statement about the filter coefficients, and plot the pole/zero locations and the frequency response of the filter.

   Solution:

- *Example:* $M = 8$, $g = 0.5$



2. **Allpass Filters:** Realize an allpass structure as suggested by Schroeder.

   (a) Why can we expect a better result with allpass filter than with a comb filter? Write a Matlab function for a comb and allpass filter with $M = 8, 16$.

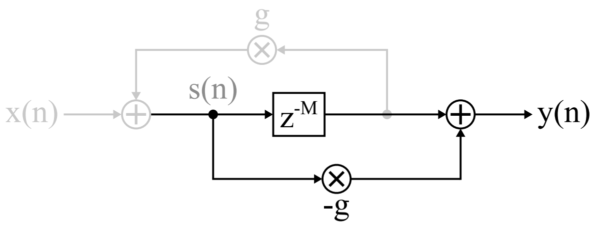
   Solution:

   - With an allpass filter we can avoid spectral shaping observed previously and get a frequency response which is flat enough.
   - see Matlab script 'dasp_ex7_3_1'

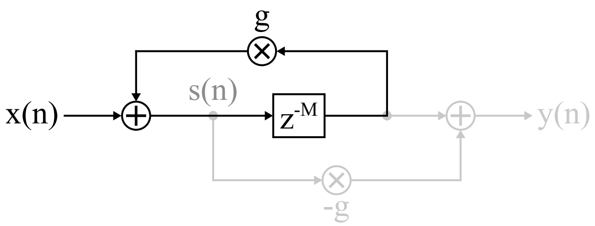
   (b) Derive the transfer function and show the pole/zero locations, and the impulse response, the magnitude, and phase responses.

   Solution:

   - *Decomposition of the system:*



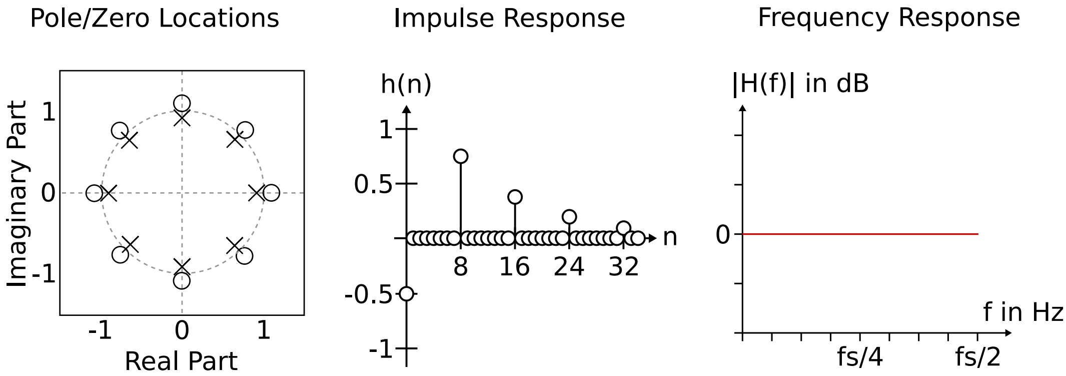
$$y(n) = -g \cdot s(n) + s(n - M)$$
$$\Rightarrow H_{ys}(z) = \frac{Y(z)}{S(z)} = -g + z^{-M}$$



$$s(n) = x(n) + g \cdot s(n - M)$$
$$\Rightarrow H_{sx}(z) = \frac{S(z)}{X(z)} = \frac{1}{1 - g \cdot z^{-M}}$$

$$H(z) = \frac{Y(z)}{X(z)} = \frac{Y(z)}{S(z)} \cdot \frac{S(z)}{X(z)} = H_{ys}(z) \cdot H_{sx}(z) = \frac{-g + z^{-M}}{1 - g \cdot z^{-M}}$$

   - *Example:* $M = 8$, $g = 0.5$



(c) Perform the filtering of an audio signal with the two filters and estimate the delay length M, which leads to a perception of a room impression.

Solution:

- see Matlab script 'dasp_ex7_3_2'

4 Feedback Delay Networks

Write a Matlab program which realizes an FDN system.

Solution:

- see Matlab script 'dasp_ex7_4'

1. What is the reason for a unitary feedback matrix?

   Solution:

   - Unitary feedback matrices ensure the system's stability, they preserve the energy of the input signal by endlessly circulating it through the network and makes the FDN prototype lossless.

2. What is the advantage of using a unitary circulant feedback matrix?

   Solution:

   - There are two main advantages in the use of unitary feedback matrices:
     - Specific eigenvalues can be specified during the design of the system.
     - The matrix multiplication only takes about $N \log(N)$ operations to compute since the FFT can be used.

3. How do you control the reverberation time?

   Solution:

   - The decay time of Unitary FDN can be precisely controlled by inserting an attenuation g_i with each delay line. The decay time can be made frequency dependent by inserting with each delay line a filter $G_i(z) : 20\log_{10}|G_i(e^{j\omega T})| = -60\frac{\tau_i}{T_{60}(\omega)}$.