ALaRI: Digital Signal Processing

Exercise Session 2: Acoustic channel measurement & modeling

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May 2013

Goal: to create m-files for estimating the acoustic channel impulse response and the transmitter and receiver sampling frequency mismatch.

Requirements: Matlab/Simulink in Windows, a sound card with line in/out, a loudspeaker and a microphone (possibly in a headset), 1 m-file (initparams.m), and 1 Simulink model (audio_io.mdl).

Outcome: 2 m-files (fs_mismatch.m, channelident_time.m).

Exercise 2-1: The synchronization problem – a first glance

Unfortunately, in many sound cards the signal at the line out (i.e., the transmitted signal) and the signal at the line in (i.e., the received signal) are sampled with a different clock. Therefore, not only sampling instants will differ, but also the sampling frequencies will be (slightly?) different, i.e., $f_s^{\text{receiver}} = f_s^{\text{transmitter}} + \triangle f_s$.

1. Try to define an experiment to roughly estimate the sampling frequency mismatch $\triangle f_s$ between the transmitter and the receiver, reusing the m-file initparams.m and the Simulink model audio_io.mdl created in the previous session.

2. Create an m-file "fs_mismatch.m" with which the sampling frequency mismatch $\triangle f_s$ can be estimated for a user-specified sampling frequency, and evaluate this m-file for $f_s^{\text{transmitter}} = 8000, 8100, 16000, 32000$ Hz.

3. Can you now answer the last question of Exercise 1-2?

Exercise 2-2: Channel characteristics – a first attempt to find the impulse response

1. What is an impulse response (IR)?

2. Set up a simple experiment and create an m-file "channelident_time.m" to estimate the impulse response of the channel between transmitter and

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receiver, by literally applying the definition of the IR and reusing the m-file initparams.m and the Simulink model audio_io.mdl created in the previous session. What do you observe? What is the length of the impulse response?

3. Is the acoustical environment the only factor that determines the IR? What else will have an influence?

4. *Optional:* Perform the white noise experiment from Exercise 1-2, but now use the estimated IR to compute the channel output signal. Does the spectrogram look more or less the same?

(Optional) Exercise 2-3: Channel characteristics – a robust estimate of the IR and frequency response

Our aim is now to obtain a better estimate of the IR of the channel.

1. How do the transmitted signal u, the (unknown) channel IR h, and the recorded signal y relate (assuming that noise can be neglected)? Provide a matrix description of this relation (in the time domain). In this way, a linear system of equations can be obtained, where the vector h contains the unknown variables. Transmit a well-chosen signal and record it. Now estimate h, based on this system of equations.

2. We did not incorporate noise in this model. Is this a big problem?

3. Does the $\triangle f_s$-problem limit this experiment? How can you check?

4. Plot the channel frequency response from the estimated IR. Does this resemble the recorded white noise spectrogram from Exercise 1-2?

5. Predict what will happen to the frequency response of the channel if you would put your hand against the loudspeaker. How will this change the shape of the frequency response of the channel, besides the obvious higher attenuation? Now do the experiment and check if you were right.