Questions A

A1. Difference equation (slide 5)

- Explain the meaning/function of both summation signs. What entities do $y_m(n-i)$ and $g_{m,i}(n)$ represent?
- Which index can be dropped in the case of a fixed beamformer?

A2. Receiving characteristic

- What does a cardioid characteristic look like?
- Have a look on slide 10. Find out in which parameter the dependency on the angle of incidence is hidden.

A3. Delay & Sum

- Describe the delay-and-sum beamformer in your own words.
- Why isn't the time-domain solution simply a delay of τ_m samples?

A4. Interference compensation

- Explain the structure with advantages/drawbacks of one of the presented schemes to block the desired signal.
- Based on the interference compensation, how can the interference signal power be estimated?

Questions B

B1. Have a look at the time delay equation on the bottom of slide 7.

- Which units do the variables have?
- Show that the units in the equation are consistent.

B2. Beampattern

- Which conditions have to be fulfilled in order to be able to separate the influences of the microphones and of the signal processing?
- Which are the extreme cases of the array gain? I.e., what are the highest and lowest possible gains?

B3. Filter & Sum

- What does superdirectivity mean?
- What is the optimization criterion of the filter-and-sum beamformer?
- Why do we introduce a constraint?
- On slide 25, what is the origin of the sidelobes?

Answers B

B1. Have a look at the time delay equation on the bottom of slide 7.

- $f_s[1/\mathrm{s}],\ c[\mathrm{m/s}],\ \boldsymbol{r}_m[\mathrm{m}]$. ***r*** is dimensionless because of the normalization to 1.
- Consistency check: $$[\mathrm{samples}] = \frac{[\mathrm{samples/s}]}{[\mathrm{m/s}]}[\mathrm{m}].$$

B2. Beampattern

- Condition: The microphones must have the same characteristic (including same direction).
- a) lowest: The beamformer has omnidirectional characteristic, $Q = 1,\ Q_{\log} = 0$.
  b) highest: The beamformer has an extremely thin lobe: Q goes to infinity.

B3. Filter & Sum

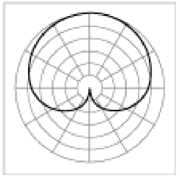
- A characteristic that is better than delay and sum is called superdirective.
- See slide 20.
- Minimizing $\tilde{u}(n)$ without constraints would lead to all filter coefficients converging to 0.
- The sidelobes can be explained by the microphone distances which are getting larger than half the wave length.

Answers A

A1. Difference equation (slide 5)

- Summation signs: 1) Sum over all microphones; 2) Convolution sum.
- $y_m(n-i)$: microphone m, sample index $n-i$.
  $g_{m,i}(n)$: Filter of microphone m at time index n and filter coefficient i .
- Fixed beamformer: $g_{m,i}$ -> time index n can be dropped

A2. Receiving characteristic

- See equation on slide 9 and following figure: 
- Angle of incidence of the sound is hidden in the parameter $\tau_m = \tau_m(\boldsymbol{r})$.

A3. Delay & Sum

- See slides 14, 15.
- τ_m is generally not integer, so the solution is a fractional delay filter (see slides 16, 17).

A4. Interference compensation

- See slides 27-35.
- The output of the interference compensation is an estimate of the interference. Its squared absolute value can be smoothed and used as interference signal power estimate.