Spatial Sound Field Concentration

Project Manager: Robert Höldrich (hoeldrich@iem.at)

Contributors: Franz Zotter

Markus Noisternig

Introduction

An essential part of the work in an air traffic control tower is the radio communication to airplane cockpits. As the usage of headphones over long time periods is demanding, thus loudspeakers are used which cause cross-talk between neighbouring working positions, so that it becomes even harder to understand the strongly band limited radio traffic. An approach to this problem is the usage of loudspeakers with a higher directivity, in order not to minimize the excitation of the diffuse sound field and the radiation towards neighbouring workspaces. In order to enable an appropriate freedom of movement the usage of beam-steering methods seems to be reasonable, thus a loudspeaker array is used to get better directivity. In this work the appropriate loudspeaker array is designed, beam-steering methods and a hardware implementation for this array are tested.

1. Methods

There are essentially 4 methods that are interesting for our beam-steering purpose:

- “stereo-pan” a simple 2 loudspeaker setup, the nearest loudspeaker is driven with more gain
- far-field beam-forming (delay & sum), based on delay stages with equal delay times, most common for beam-steering applications
- near-field beam-forming, using individual delay times
- (reduced) wave field synthesis, using individual gains and delays

The most interesting methods are near-field beam-forming and wave field synthesis combined each with a *harmonic nested* windowing technique. For very low cost implementation the “stereo-pan” approach is interesting, but doesn’t introduce any directivity except of the speaker’s own.

2. Loudspeaker Array

In order to get a most suitable design a MATLAB Simulation tool was developed to design the array's proportions. This tool calculates direct sound pressure level of the loudspeaker array driven by beam-steering methods and windowing techniques applied to harmonic input signals. The directivities of the simulated speakers are assumed to be omnidirectional. The simulation shows the magnitude of the sound pressure and directivity patterns.

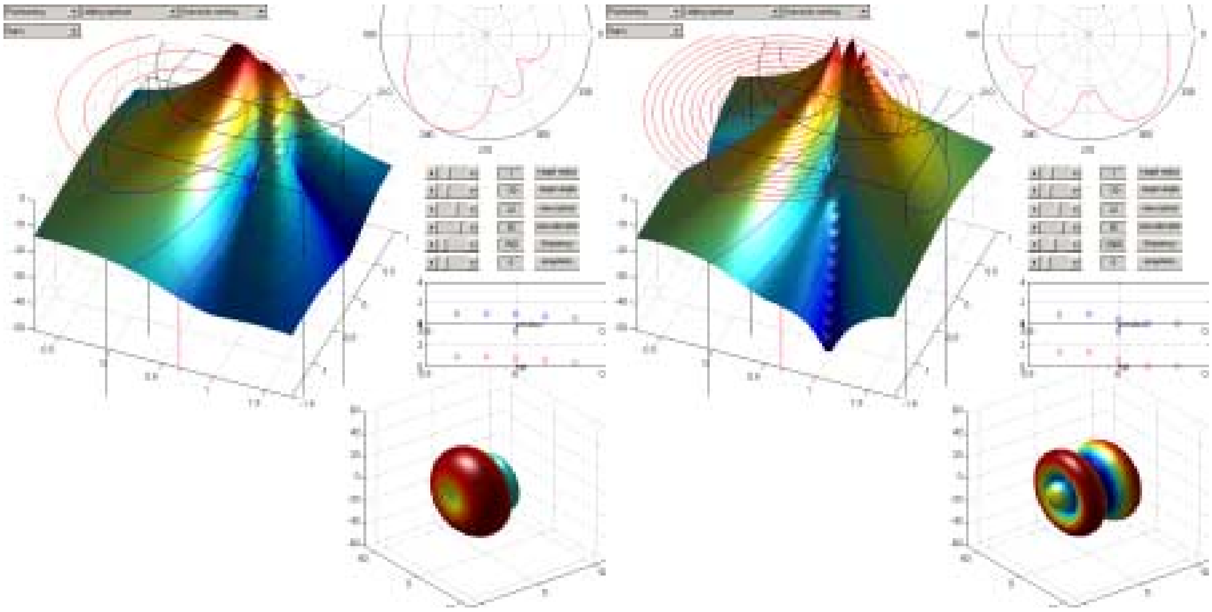

figure 1: near-field beam-forming with windowing technique, 5 speakers (700 Hz and 2000 Hz)

3. Hardware Implementation

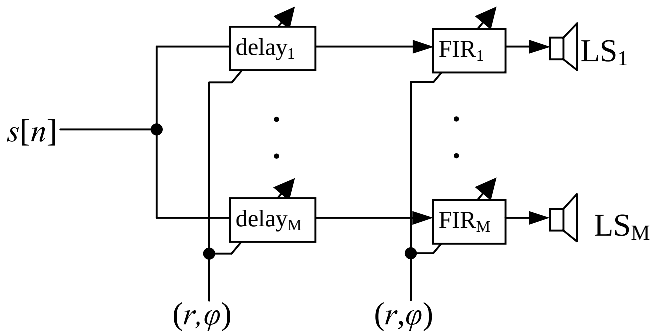

figure 2: signal flow chart for hardware implementation

The hardware implementation is based on "pd" (Pure Data, Miller S. Puckette). The signal processing consists simply of a delay and a FIR filter for each speaker channel. The AM radio channel used in airplane radio has a very small bandwidth of only 3.4 kHz, thus filtering can be done by low sampling rates. Our FIR filters are FFT based and use filter lengths of 64 samples at a sampling rate of 11025 Hz.

Filter coefficients are determined by the window function applied to the speaker array. Due to *harmonic nesting* and a window centre that slides along the array axis, window weights need to be

calculated for each steering position and frequency bin. The (high pass) frequency response compensation is multiplied to the filter responses, so we need only one filter per loud speaker. The windows are determined by their off-center shift and the desired beam width (*harmonic nesting*). The maximum number of speakers in our implementation is restricted to 8.
8 JBL Control 1 Speakers and two digital 4 channel Amplifiers are used, the relative sound pressure level of the loudspeakers have been calibrated by spl measurements. A 800 MHz Pentium III processor was quite sufficient for our real time signal processing and calculation purposes.

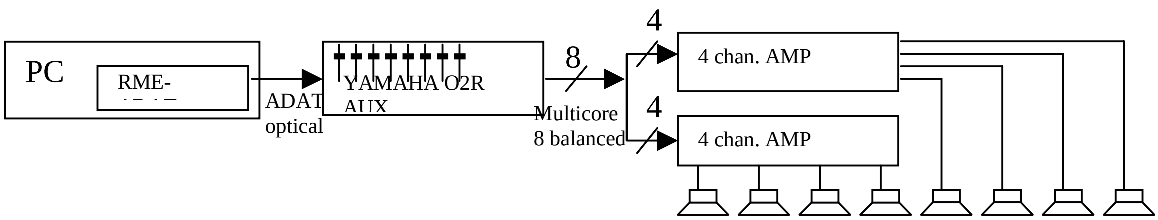

figure 3: cabling and devices

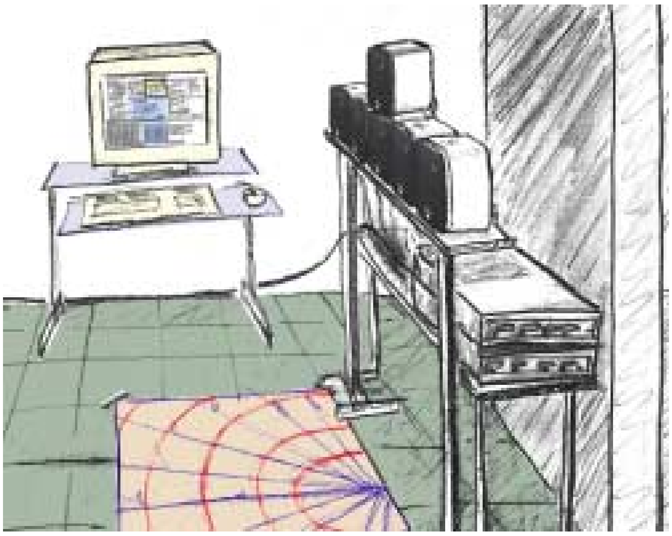

figure 4: sketch of the experimental setup

As expected near-field beam-forming or wave field synthesis combined with windowing techniques (sliding window center, harmonic nesting) has the most desirable results. Our minimum spacing between the speakers was restricted to 15.5 cm. In a system implementation it is recommended to reduce this spacing further to suppress the effects of spatial aliasing. The use of 5 or even more speakers is recommended.
For low computation effort it is also recommended to use only a discrete number of listener positions in calculation, so that only a small set of filter configurations and delay times remain (e.g. for -60, -45, -30, -15, 0° and mirrored.).