Massive Audio Beamforming

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• *Introduction*

MIMO (Multiple Input Multiple Output) technology has gained an increasing interest in wireless communication due to its capability to increase the throughput of a communication system without an increased bandwidth or increased transmit power. The technology is based on the use of multiple antennas in both the transmitter and the receiver.

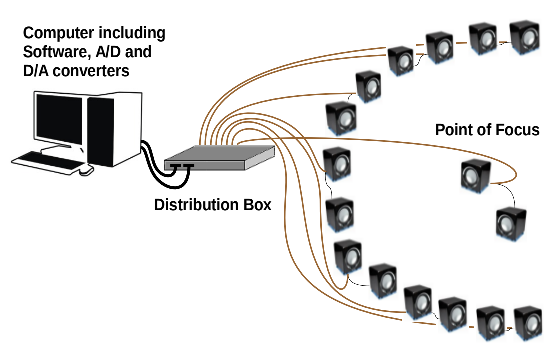
Beamforming is achieved by individually altering the phase of the transmit signals on the different antennas in order to create constructive interference at a certain point or in a certain direction.

The purpose of this project was to develop a system capable of demonstrating the effects of Massive MIMO, using audio signals instead of electromagnetic waves. It is carried out by substituting the antenna array by a set of loudspeakers, to perform audio beamforming.

• *The System*

The system consists of

- A computer equipped with A/D and D/A converters.
- A Distribution Box developed by the project team, with the purpose of channeling ingoing and outgoing signals, to and from the loudspeaker array.
- Eight pairs of loudspeakers, modified to be able to both transmit and record sound.



• *How the System Works*

1. One of the loudspeakers from the pair in Point of Focus transmits a pilot signal, while the loudspeakers of the array are recording.
2. The delay and phase difference are calculated for each loudspeaker of the array.
3. The individual phase difference is added to each signal phase and the resulting signals are transmitted back to the Point of Focus.
4. The sound level is measured in each of the loudspeakers in Point of Focus.

• *Phase Estimation*

Sent pilot *x[n]* with frequency *f*.

$$x[n]=\sin(2f\pi n)$$

Received signals $y_i[n]$ in array, where α_i is the attenuation of the channel and K is the number of speakers.

$$y_i[n]=\alpha_i x[n-k_i]=\alpha_i \sin(2f\pi n-\theta_i), \quad i=1,2,..,K$$

Estimation of delays k_i (in samples) and phases θ_i.

$$k_i=\underset{k}{\operatorname{argmax}}\, R_{xy}[k]=\underset{k}{\operatorname{argmax}} \sum_{n=-N}^{N} x[n-k]\, y_i[n], \quad i=1,2,..,K$$

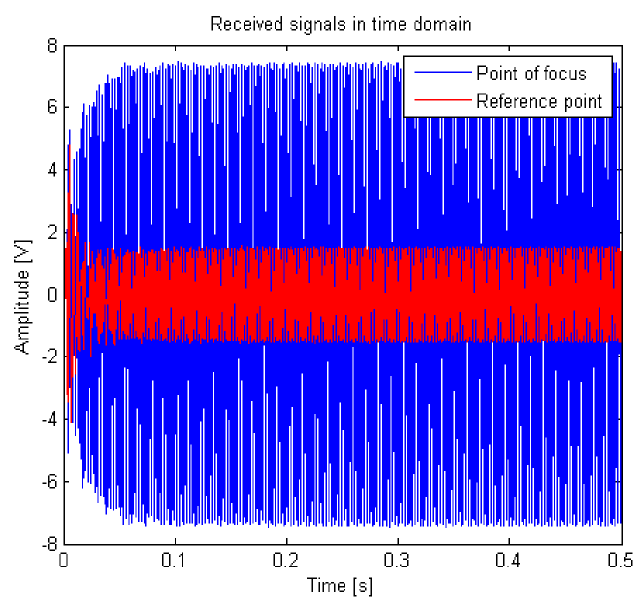
$$\theta_i=2\pi k_i \frac{f}{f_s} \mod 2\pi, \quad f_s - \text{sample frequency}$$

Theoretical received signal in point of focus

$$z[n]=\sum_{i\in K} y_i[n+k_i]=\sum_{i\in K}\alpha_i x[n]=KAx[n], \quad A - \text{average attenuation}$$

• *Results*

Testing has verified that the system is able to perform beamforming, to the extent that the difference between recorded sound energy in point of focus and a reference point about 30 cm away is in the region of 10-20 dB. In some cases, a maximum of 30 dB difference was achieved. A plot from a certain run is shown below. The plot shows the received electrical signal from the loudspeakers, measured in volts.



• *More Information*

More information about the project can be found on the project web page:

http://www.isy.liu.se/edu/projekt/tsks05/2014/projekt/