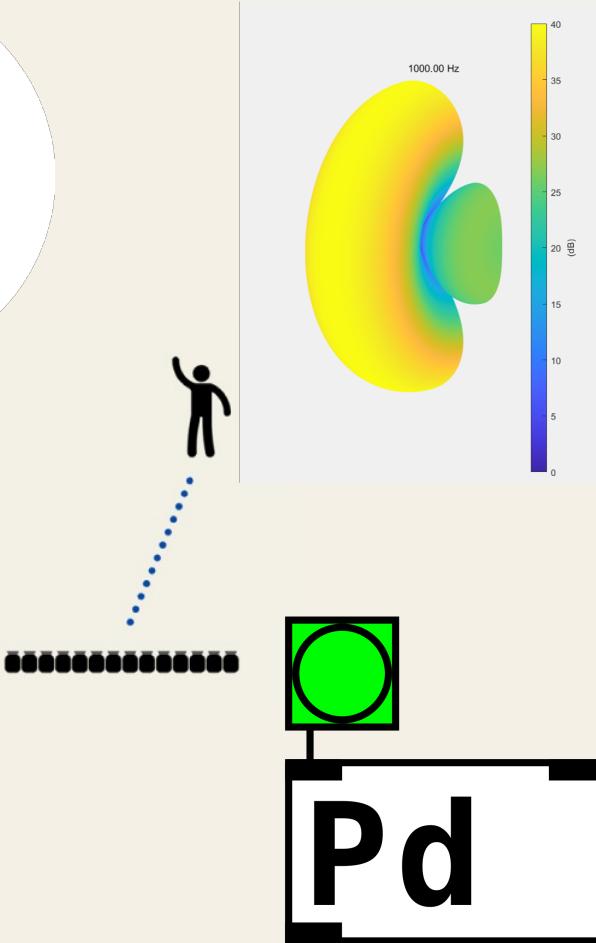
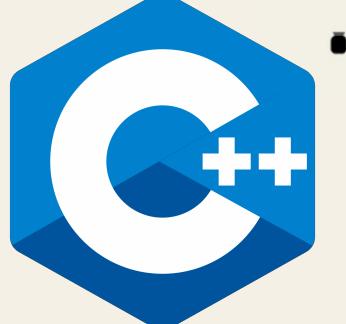
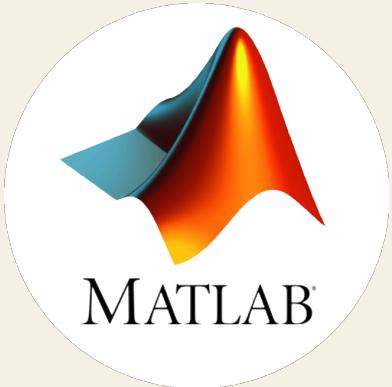


Adaptive Beamforming Algorithm

for 14-channels speakers

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Overview

Started With:

- MATLAB
- Max
- Fixed Beamforming

Goal:

- C++
- Pure Data
- Real Time Adaptive Beamforming

Beamforming Class

Class setup

- Created a class that encloses beamforming algorithm.
- Class consists of 3 major components - constructor, assignment of parameters of the system, filter calculator
- By using Eigen library, we could streamline translation from Matlab

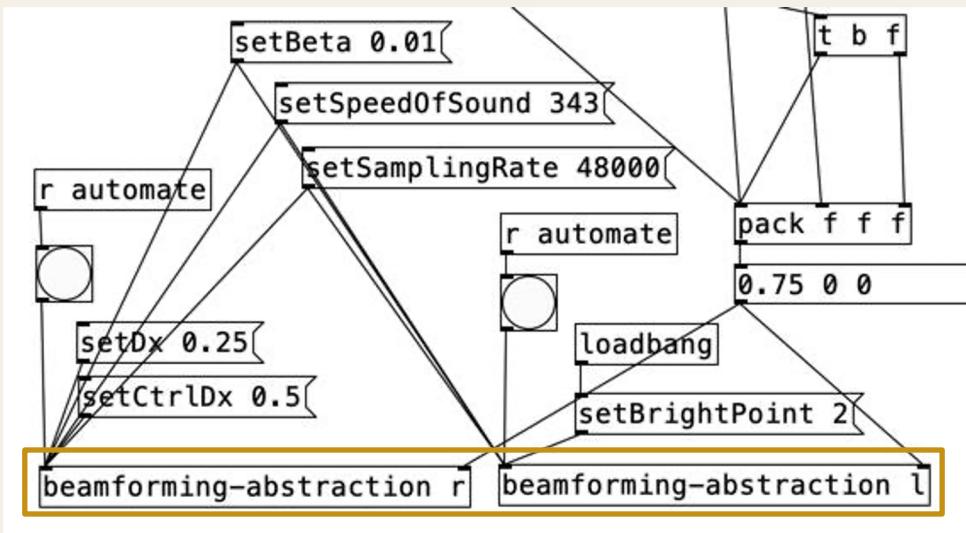
```
public:  
    // constructor  
    Beamforming(int nSpeakers=14, int nCtrlPts=1);  
    // destructor  
    ~Beamforming();  
  
    bool control_point_geometry(); //set up control point geometry TODO: maybe remove  
    MatrixComplex solve_filters(); //main entry point for solving filters  
  
    // Getters  
    vector<t_float> get_pf();  
    t_float get_fs();  
    t_float get_c();  
    unsigned int get_nfft();  
    unsigned int get_nf();  
    VectorReal get_ff();  
    t_float get_beta();  
    int get_n();  
    t_float get_dx();  
    bool get_flatten_filters();  
    bool get_normalize_filters();  
    bool get_crop_filters();
```

Beamforming Class

```
public:  
// constructor  
Beamforming(int nSpeakers=14, int nCtrlPts=1);  
// destructor  
~Beamforming();  
  
bool control_point_geometry(); //set up control point geometry. TODO: maybe remove  
MatrixComplex solve_filters(); // MatrixComplex Beamforming::solve_filters() {  
    // configControlPoints();  
    return a.reshaped<Eigen::RowMajor>().eval();  
}  
  
// Getters  
vector<t_float> get_pf();  
t_float get_fs();  
t_float get_c();  
unsigned int get_nfft();  
unsigned int get_nf();  
VectorReal get_ff();  
t_float get_beta();  
int get_n();  
t_float get_dx();  
bool get_flatten_filters();  
bool get_normalize_filters();  
bool get_crop_filters();  
  
// Sampling rate  
fs = 48000;  
// speed of sound  
c = 343;  
// fft size (samples)  
nfft = 512;  
// number of frequencies  
nf = nfft/2+1; //Not used else except below  
// frequency vector  
ff = linspace(0, fs/2, nf);  
// Regularization Parameter  
beta = 0.01; //Limit: 0 to infinity.  
  
// Speaker setup  
n = nSpeakers; //num speakers  
dx = 0.03874; //speaker_spacing
```

- Created a class that encloses beamforming algorithm.
- By using Eigen library, we could streamline translation from MATLAB
- Class consists of 3 major components
 - Constructor with default parameters
 - Update of parameters
 - Filter calculator

Beamforming External

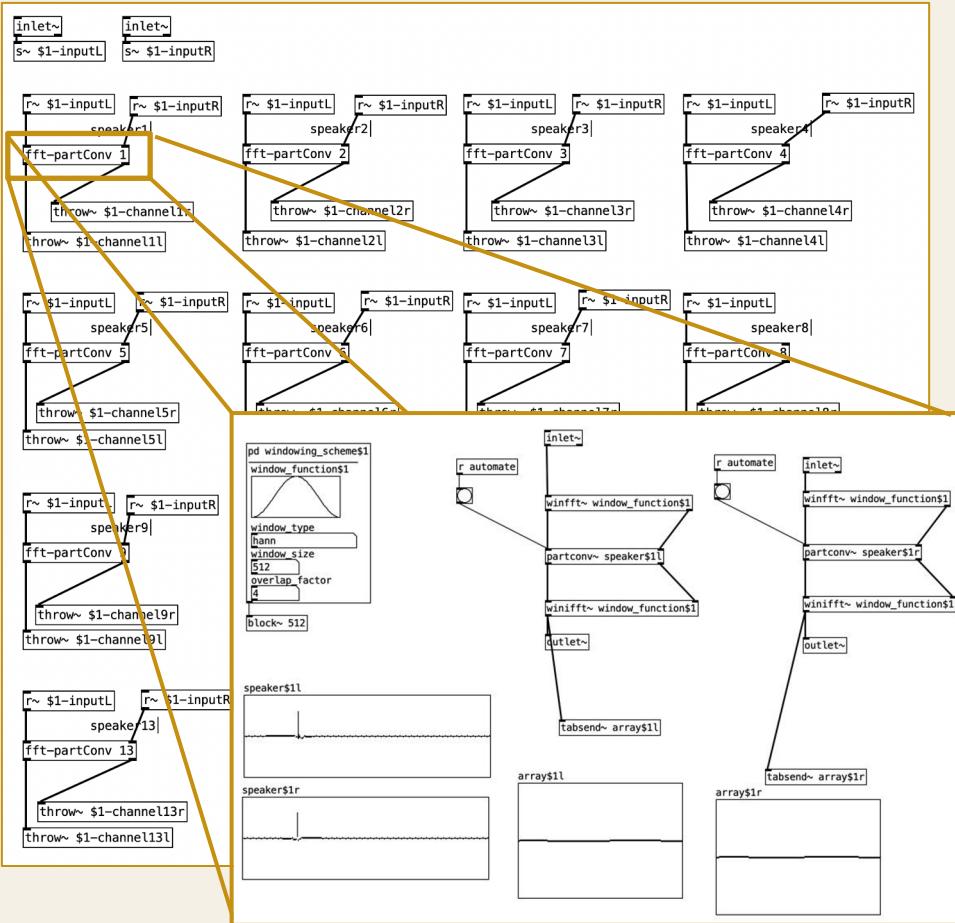


- Pd external in C++ that interfaces the Beamforming class
- Pdlibbuilder
- Customization and parameter setting
- Positional data is passed in
- Filters are outputted for use in convolution

Real-time convolution

partconv~ object and Pd patches

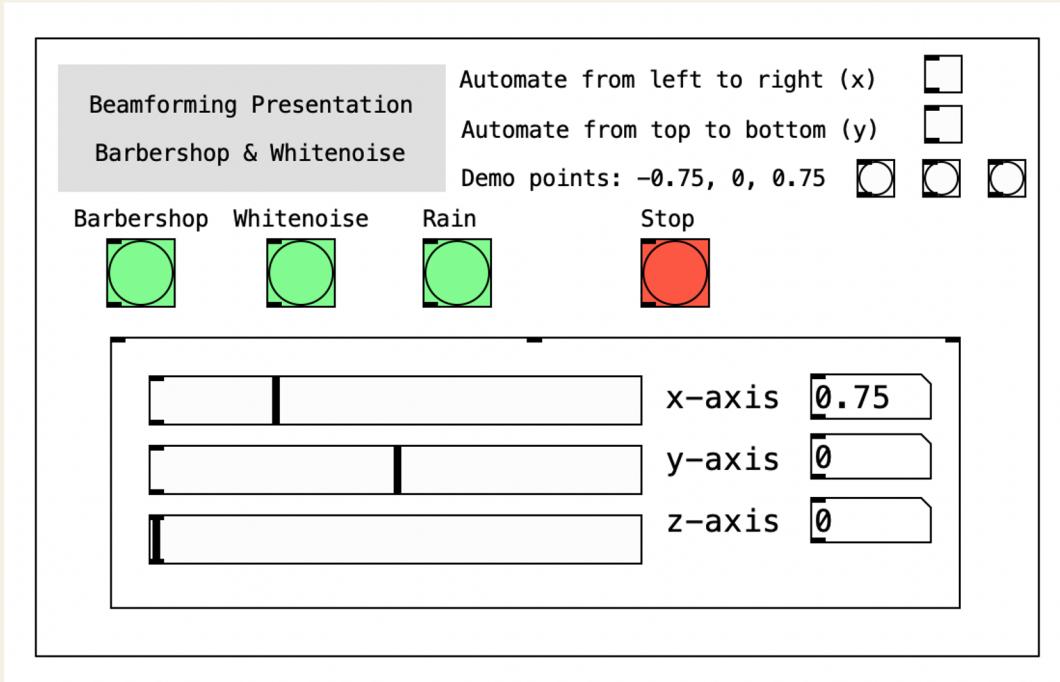
- Pd Spectral Library and modification
- Implementation of 14-channel streaming in Pure Data
- Pure Data Abstraction – ready for any number of speakers



```

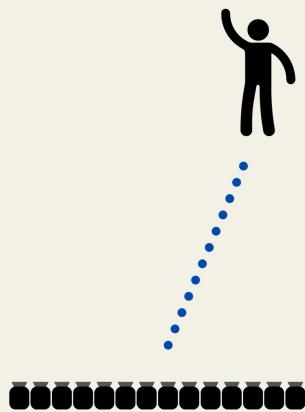
299 static void partconv_analyze_impulse( t_partconv* object )
300 {
301     // store parts and spectra_mem_size for use in dsp loop
302     object->parts          = parts;
303     object->spectra_mem_size = spectra_mem_size;
304
305     // report analysis info
306     // post( "partconv~: analyzed %s array", object->impulse_name->s_name );
307
308     object->analyzed_flag = TRUE;
309 }

```



- Graphical User Interface (GUI)
- Provides Barbershop binaural sample audio and white noise for a diverse experience with adaptive beamforming
- Beam automation allowing users to feel and hear the beam moving

Final Product



Real Time Adaptive Beamforming

Visualization