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# **Sound Field Analysis Toolbox**

## **Readme**

*Release 0.1dev*

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Contents:

## 1.1 Requirements

Obviously, you'll need [Python](#). The code has been developed and tested on Python 3.x only. [NumPy](#) and [SciPy](#) are needed for calculations. .. If you also want to plot the resulting sound fields, you'll need [matplotlib](#).

Instead of installing all of them separately, you should probably get a Python distribution that already includes everything, e.g. [Anaconda](#).

## 1.2 How to Get Started

Various jupyter notebooks are located in the root directory/

- **AE1\_IdealPlaneWave.ipynb**: Ideal unity plane wave simulation

## 1.3 Reference

Feel free to check out the full [Reference](#).

## GENERATORS

Generators explanation.

## PROCESSING

Processing explanation. Functions that act on the Spatial Fourier Coefficients

*fdt* Frequency to time transform

*itc* Fast Inverse Spatial Fourier Transform

*pdw* Plane Wave Decomposition

*rft* Radial filter Improvement

*stc* Fast Spatial Fourier Transform

*tdt* Time Domain Reconstruction

Not yet implemented:

*bsa* BEMA Spatial Anti-Aliasing

*sfe* Sound field extrapolation

*wdr* Wigner-D Rotation

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### Todo

Use more descriptive function names

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**PLOTTING**

Plotting explanation. Plotting functions Helps visualizing spherical microphone data.

Generally, you probably want to first extract the amplitude information in spherical coordinates: `>> plot.makeMTX(Pnm, dn, Nviz=3, krIndex=1, oversize=1)` And then visualize that: `>> plot3D(vizMTX, style='shape')`

Other valid styles are 'sphere' and 'flat'.

## REFERENCE

## 5.1 Generators

Module contains various generator functions:

**awgn** Generate additive White Gaussian noise

**gaussGrid** Gauss-Legendre quadrature grid and weights

**lebedev** Lebedev quadrature grid and weights

**mf** Modal Radial Filter

**swg** Sampled Wave Generator, emulating discrete sampling

**wgc** Wave Generator, returns spatial Fourier coefficients

`sound_field_analysis.gen.awgn(fftData, noiseLevel=80, printInfo=True)`

Adds White Gaussian Noise of approx. 16dB crest to a FFT block.

### Parameters

- **fftData** (*array of complex floats*) – Input fftData block (e.g. from F/D/T or S/W/G)
- **noiseLevel** (*int, optional*) – Average noise Level in dB [Default: -80dB]
- **printInfo** (*bool, optional*) – Toggle print statements [Default: True]

**Returns** **noisyData** – Output fftData block including white gaussian noise

**Return type** array of complex floats

`sound_field_analysis.gen.gaussGrid(AZnodes=10, ELnodes=5, plot=False)`

Compute Gauss-Legendre quadrature nodes and weights in the SOFiA/VariSphear data format.

### Parameters

- **ELnodes** (*AZnodes, )* – Number of azimuthal / elevation nodes [Default: 10 / 5]
- **plot** (*bool, optional*) – Show a globe plot of the selected grid [Default: False]

### Returns

- **gridData** (*matrix of floats*) – Gauss-Legendre quadrature positions and weights

<pre>[AZ_0, EL_0, W_0   ...   AZ_n, EL_n, W_n]</pre>
--

- **Npoints** (*int*) – Total number of nodes
- **Nmax** (*int*) – Highest stable grid order

`sound_field_analysis.gen.lebedev(degree, plot=False, printInfo=True)`

Compute Lebedev quadrature nodes and weights.



### Parameters

- **Degree** (*int*) – Lebedev Degree. Currently available: 6, 14, 26, 38, 50, 74, 86, 110, 146, 170, 194
- **plot** (*bool, optional*) – Plot selected Lebedev grid [Default: False]

### Returns

- **gridData** (*array\_like*) – Lebedev quadrature positions and weights: [AZ, EL, W]
- **Nmax** (*int*) – Highest stable grid order

`sound_field_analysis.gen.mf(N, kr, ac, amp_maxdB=0, plc=0, fadeover=0, printInfo=True)`  
Generate modal radial filters

### Parameters

- **N** (*int*) – Maximum Order
- **kr** (*array\_like*) – Vector or Matrix of kr values

First Row (M=1) N: kr values microphone radius  
Second Row (M=2) N: kr values sphere/microphone radius  
[kr\_mic;kr\_sphere] for rigid/dual sphere configurations  
! If only one kr-vector is given using a rigid/dual sphere  
Configuration: kr\_sphere = kr\_mic

- **ac** (*int {0, 1, 2, 3, 4}*) –

#### Array configuration

- 0: Open Sphere with p Transducers (NO plc!)
- 1: Open Sphere with pGrad Transducers
- 2: Rigid Sphere with p Transducers
- 3: Rigid Sphere with pGrad Transducers
- 4: Dual Open Sphere with p Transducers

- **amp\_maxdB** (*int, optional*) – Maximum modal amplification limit in dB [Default: 0]
- **plc** (*int {0, 1, 2}, optional*) – OnAxis powerloss-compensation: - 0: Off [Default] - 1: Full kr-spectrum plc - 2: Low kr only -> set fadeover
- **fadeover** (*int, optional*) – Number of kr values to fade over +/- around min-distance gap of powerloss compensated filter and normal N0 filters. 0 is auto fadeover [Default]

### Returns

- **dn** (*array\_like*) – Vector of modal 0-N frequency domain filters
- **beam** (*array\_like*) – Expected free field on-axis kr-response

`sound_field_analysis.gen.swg(r=0.01, gridData=None, ac=0, FS=48000, NFFT=512, AZ=0, EL=1.5707963267948966, c=343, wavetype=0, ds=1, Nlim=120, printInfo=True)`

Sampled Wave Generator Wrapper

### Parameters

- **r** (*array\_like, optional*) – Microphone Radius [Default: 0.01]

Can also be a vector **for** rigid sphere configurations:  
[1,1] => rm Microphone Radius  
[2,1] => rs Sphere Radius (Scatterer)

- **gridData** (*array\_like*) – Quadrature grid [Default: 110 Lebedev grid]

Columns : Position Number 1...M
Rows : [AZ EL Weight]

- **ac** (*int {0, 1, 2, 3, 4}*) –

**Array Configuration:**

- 0: Open Sphere with p Transducers (NO plc!) [Default]
- 1: Open Sphere with pGrad Transducers
- 2: Rigid Sphere with p Transducers
- 3: Rigid Sphere with pGrad Transducers
- 4: Dual Open Sphere with p Transducers

- **FS** (*int, optional*) – Sampling frequency [Default: 48000 Hz]
  - **NFFT** (*int, optional*) – Order of FFT (number of bins), should be a power of 2. [Default: 512]
  - **AZ** (*float, optional*) – Azimuth angle in radians [0-2pi]. [Default: 0]
  - **EL** (*float, optional*) – Elevation angle in radians [0-pi]. [Default: pi / 2]
  - **c** (*float, optional*) – Speed of sound in [m/s] [Default: 343 m/s]
  - **wavetype** (*int {0, 1}, optional*) –
- Type of the wave:**
- 0: Plane wave [Default]
  - 1: Spherical wave
- **ds** (*float, optional*) – Distance of the source in [m] (For wavetype = 1 only)
  - **Nlim** (*int, optional*) – Internal generator transform order limit [Default: 120]

**Warning:** If NFFT is smaller than the time the wavefront needs to travel from the source to the array, the impulse response will be cyclically shifted (cyclic convolution).

**Returns**

- **fftData** (*array\_like*) – Complex sound pressures of size [(N+1)^2 x NFFT]
- **kr** (*array\_like*) – kr-vector

Can also be a matrix [krm; krs] <b>for</b> rigid sphere configurations: [1,:] => krm referring to the microphone radius [2,:] => krs referring to the sphere radius (scatterer)
---

**Note:** This file is a wrapper generating the complex pressures at the positions given in 'gridData' for a full spectrum 0-FS/2 Hz (NFFT Bins) wave impinging to an array. The wrapper involves the W/G/C wave generator core and the I/T/C spatial transform core.

S/W/G emulates discrete sampling. You can observe alias artifacts.

```
sound_field_analysis.gen.wgc(N, r, ac, fs, F_NFFT, az, el, t=0.0, c=343.0, wavetype=0,
                             ds=1.0, lowerSegLim=0, SegN=None, upperSegLim=None,
                             printInfo=True)
```

Wave Generator Core: Returns Spatial Fourier Coefficients *Pnm* and *kr* vector

## Parameters

- **N**(*int*) – Maximum transform order.
- **r**(*list of ints*) – Microphone radius

Can also be a vector for rigid/dual sphere configurations:  
`[1,1] => rm Microphone radius`  
`[2,1] => rs Sphere or microphone radius`  
 ! If only one radius (rm) is given using a rigid/dual sphere  
 Configuration: `rs = rm` and only one `kr`-vector is returned!

- **ac**(*int {0, 1, 2, 3, 4}*) –

### Array Configuration:

- 0: Open Sphere with p Transducers (NO plc!)
- 1: Open Sphere with pGrad Transducers
- 2: Rigid Sphere with p Transducers
- 3: Rigid Sphere with pGrad Transducers
- 4: Dual Open Sphere with p Transducers

- **FS**(*int*) – Sampling frequency
- **NFFT**(*int*) – Order of FFT (number of bins), should be a power of 2.
- **AZ**(*float*) – Azimuth angle in radians [0-2pi].
- **EL**(*float*) – Elevation angle in radians [0-pi].
- **t**(*float, optional*) – Time Delay in s.
- **c**(*float, optional*) – Speed of sound in [m/s] [Default: 343m/s]
- **wavetype**(*int {0, 1}, optional*) –

### Type of the Wave:

- 0: Plane Wave [Default]
- 1: Spherical Wave

- **ds**(*float, optional*) – Distance of the source in [m] (For wavetype = 1 only)
- **lSegLim**(*int, optional*) – (Lower Segment Limit) Used by the S/W/G wrapper
- **uSegLim**(*int, optional*) – (Upper Segment Limit) Used by the S/W/G wrapper
- **SegN**(*int, optional*) – (Segment Order) Used by the S/W/G wrapper
- **printInfo**(*bool, optional*) – Toggle print statements

**Warning:** If NFFT is smaller than the time the wavefront needs to travel from the source to the array, the impulse response will be cyclically shifted (cyclic convolution).

## Returns

- **Pnm**(*array of complex floats*) – Spatial Fourier Coefficients with nm coeffs in cols and FFT coeffs in rows
- **kr**(*array\_like*) – `kr`-vector

Can also be a matrix [`krm`; `krs`] **for** rigid sphere configurations:  
`[1,:]` => `krm` referring to the microphone radius  
`[2,:]` => `krs` referring to the sphere radius (scatterer)

## 5.2 Processing

Functions that act on the Spatial Fourier Coefficients

*fdt* Frequency to time transform

*itc* Fast Inverse Spatial Fourier Transform

*pdc* Plane Wave Decomposition

*rft* Radial filter Improvement

*stc* Fast Spatial Fourier Transform

*tdt* Time Domain Reconstruction

Not yet implemented:

*bsa* BEMA Spatial Anti-Aliasing

*sfe* Sound field extrapolation

*wdr* Wigner-D Rotation

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### Todo

Use more descriptive function names

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`sound_field_analysis.process.bsa` (*Pnm*, *ctSig*, *dn*, *transition*, *avgBandwidth*, *fade=True*)  
B/S/A BEMA Spatial Anti-Aliasing - NOT YET IMPLEMENTED

#### Parameters

- **Pnm** (*array\_like*) – Spatial Fourier coefficients
- **ctSig** (*array\_like*) – Signal of the center microphone
- **dn** (*array\_like*) – Radial filters for the current array configuration
- **transition** (*int*) – Highest stable bin, approx:  $\text{transition} = (\text{NFFT}/\text{FS} + 1) * (\text{N} * \text{c}) / (2 * \pi * \text{r})$
- **avgBandwidth** (*int*) – Averaging Bandwidth in oct
- **fade** (*bool*, *optional*) – Fade over if True, else hard cut {false} [Default: True]

**Returns** **Pnm** – Alias-free spatial Fourier coefficients

**Return type** *array\_like*

---

**Note:** This was presented at the 2012 AES convention, see <sup>1</sup>.

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### References

`sound_field_analysis.process.fdt` (*timeData*, *FFToversize=1*, *firstSample=0*, *lastSample=None*)

F/D/T frequency domain transform

#### Parameters

- **timeData** (*named tuple*) – timeData tuple with following fields

---

<sup>1</sup> B. Bernschütz, “Bandwidth Extension for Microphone Arrays”, AES Convention 2012, Convention Paper 8751, 2012. <http://www.aes.org/e-lib/browse.cfm?elib=16493>

```
.impulseResponses [Channels X Samples]
.FS
.radius           Array radius
.averageAirTemp   Temperature in [C]
(.centerIR        [1 x Samples] )
```

- **FFTOversize** (*int*, *optional*) – FFToversize > 1 increase the FFT Blocksize. [Default: 1]
- **firstSample** (*int*, *optional*) – First time domain sample to be included. [Default: 0]
- **lastSample** (*int*, *optional*) – Last time domain sample to be included. [Default: -1]

#### Returns

- **fftData** (*array\_like*) – Frequency domain data ready for the Spatial Fourier Transform (stc)
- **kr** (*array\_like*) – kr-Values of the delivered data
- **f** (*array\_like*) – Absolute frequency scale
- **ctSig** (*array\_like*) – Center signal, if available

---

**Note:** A FFT of the blocksize (FFTOversize\*NFFT) is applied to the time domain data, where NFFT is determined as the next power of two of the signalSize which is signalSize = (lastSample-firstSample). The function will pick a window of (lastSample-firstSample) for the FFT.

Call this function with a running window (firstSample+td->lastSample+td) iteration increasing td to obtain time slices. This way you resolve the temporal information within the captured sound field.

---

sound\_field\_analysis.process.**itc** (*Pnm*, *angles*, *N=None*, *printInfo=True*)  
I/T/C Fast Inverse spatial Fourier Transform Core

#### Parameters

- **Pnm** (*array\_like*) – Spatial Fourier coefficients with FFT bins as cols and nm coeffs as rows (e.g. from SOFiA S/T/C)
- **angles** (*array\_like*) – Target angles of shape

```
[AZ1, EL1;
 AZ2, EL2;
 ...
 AZn, ELn]
```

- **[N]** (*int*, *optional*) – Maximum transform order [Default: highest available order]

**Returns** **p** – Sound pressures with FFT bins in cols and specified angles in rows

**Return type** array of complex floats

---

**Note:** This is a pure ISFT core that does not involve extrapolation. (=The pressures are referred to the original radius)

---

sound\_field\_analysis.process.**pdC** (*N*, *OmegaL*, *Pnm*, *dn*, *cn=None*, *printInfo=True*)  
P/D/C - Plane Wave Decomposition

#### Parameters

- **N** (*int*) – Decomposition order

- **OmegaL** (*array\_like*) – Look directions of shape

```
[AZ1, EL1;
 AZ2, EL2;
 ...
 AZn, ELn]
```

- **Pnm** (*matrix of complex floats*) – Spatial Fourier Coefficients (e.g. from SOFiA S/T/C)
- **dn** (*matrix of complex floats*) – Modal array filters (e.g. from SOFiA M/F)
- **cn** (*array\_like, optional*) – Weighting Function. Either frequency invariant weights as 1xN array or with kr bins in rows over N cols. [Default: None]

**Returns** **Y** – MxN Matrix of the decomposed wavefield with kr bins in rows

**Return type** matrix of floats

`sound_field_analysis.process.rfi(dn, kernelDownScale=2, highPass=0.0)`

R/F/I Radial Filter Improvement [NOT YET IMPLEMENTED!]

#### Parameters

- **dn** (*array\_like*) – Analytical frequency domain radial filters (e.g. SOFiA M/F)
- **kernelDownScale** (*int, optional*) – Downscale factor for the filter kernel [Default: 2]
- **highPass** (*float, optional*) – Highpass Filter from 0.0 (off) to 1.0 (maximum kr) [Default: 0.0]

#### Returns

- **dn** (*array\_like*) – Improved radial filters
- **kernelSize** (*int*) – Filter kernel size (total)
- **latency** (*float*) – Approximate signal latency due to the filters

---

**Note:** This function improves the FIR radial filters from SOFiA M/F. The filters are made causal and are windowed in time domain. The DC components are estimated. The R/F/I module should always be inserted to the filter path when treating measured data even if no use is made of the included kernel downscaling or highpass filters.

Do NOT use R/F/I for single open sphere filters (e.g.simulations).

**IMPORTANT** Remember to choose a fft-oversize factor (F/D/T) being large enough to cover all filter latencies and reponse slopes. Otherwise undesired cyclic convolution artifacts may appear in the output signal.

**HIGHPASS** If HPF is on (highPass>0) the radial filter kernel is downscaled by a factor of two. Radial Filters and HPF share the available taps and the latency keeps constant. Be careful using very small signal blocks because there may remain too few taps. Observe the filters by plotting their spectra and impulse responses. > Be very carefull if  $NFFT/\max(kr) < 25$  > Do not use R/F/I if  $NFFT/\max(kr) < 15$

---

`sound_field_analysis.process.sfe(Pnm_kra, kra, krb, problem='interior')`

S/F/E Sound Field Extrapolation. CURRENTLY WIP

#### Parameters

- **Pnm\_kra** (*array\_like*) – Spatial Fourier Coefficients (e.g. from SOFiA S/T/C)
- **kra, krb** (*array\_like*) –  $k * r_a/r_b$  vector
- **problem** (*string{'interior', 'exterior'}*) – Select between interior and exterior problem [Default: interior]

`sound_field_analysis.process.stc` (*N*, *fftData*, *grid*)

S/T/C Fast Spatial Fourier Transform

#### Parameters

- **N** (*int*) – Maximum transform order
- **fftData** (*array\_like*) – Frequency domain sounfield data, (e.g. from SOFiA FDT) with spatial sampling positions in cols and FFT bins in rows
- **grid** (*array\_like*) – Grid configuration of AZ [0 ... 2pi], EL [0...pi] and W of shape

```
[AZ1, EL1, W1;
 AZ2, EL2, W2;
 ...
 AZn, ELn, Wn]
```

**Returns** **Pnm** – Spatial Fourier Coefficients with nm coeffs in cols and FFT bins in rows

**Return type** `array_like`

`sound_field_analysis.process.tdt` (*Y*, *win=0*, *minPhase=False*, *resampleFactor=1*, *printInfo=True*)

T/D/T - Time Domain Transform

#### Parameters

- **Y** (*array\_like*) – Frequency domain data over multiple channels (cols) with FFT data in rows
- **float, optional** (*win*) – Window Signal tail [0...1] with a HANN window [Default: 0] - NOT YET IMPLEMENTED
- **int, optional** (*resampleFactor*) – Resampling factor (FS\_target/FS\_source)
- **bool, optional** (*minPhase*) –
- **minimum phase reduction - NOT YET IMPLEMENTED [Default (Ensure)]** –

**Returns** **y** – Reconstructed time-domain signal of channels in cols and impulse responses in rows

**Return type** `array_like`

---

**Note:** This function recombines time domain signals for multiple channels from frequency domain data. It is made to work with half-sided spectrum FFT data. The impulse responses can be windowed. The IFFT blocklength is determined by the Y data itself:

Y should have a size [NumberOfChannels x ((2<sup>n</sup>)/2)+1] with n=[1,2,3,...] and the function returns [NumberOfChannels x resampleFactor\*2<sup>n</sup>] samples.

---

`sound_field_analysis.process.wdr` (*Pnm*, *xAngle*, *yAngle*, *zAngle*)

W/D/R Wigner-D Rotation - NOT YET IMPLEMENTED

#### Parameters

- **Pnm** (*array\_like*) – Spatial Fourier coefficients
- **yAngle, zAngle** (*xAngle*,) – Rotation angle around the x/y/z-Axis

**Returns** **PnmRot** – Rotated spatial Fourier coefficients

**Return type** `array_like`

## 5.3 Plotting

Plotting functions Helps visualizing spherical microphone data.

Generally, you probably want to first extract the amplitude information in spherical coordinates: `>> plot.makeMTX(Pnm, dn, Nviz=3, krIndex=1, oversize=1)` And then visualize that: `>> plot3D(vizMTX, style='shape')`

Other valid styles are 'sphere' and 'flat'.

`sound_field_analysis.plot.genFlat(vizMTX)`

Returns trace of flat surface with intensity as surface elevation and color

**Parameters** `vizMTX` (*array\_like*) – Matrix holding spherical data for visualization

**Returns** `T` – Trace of desired surface

**Return type** `plotly_trace`

---

### Todo

Fix orientation and axis limits

---

`sound_field_analysis.plot.genShape(vizMTX)`

Returns trace of shape with intensity as radial extension

**Parameters** `vizMTX` (*array\_like*) – Matrix holding spherical data for visualization

**Returns** `T` – Trace of desired shape

**Return type** `plotly_trace`

---

### Todo

Fix camera position

---

`sound_field_analysis.plot.genSphCoords()`

Generates cartesian (x,y,z) and spherical (theta, phi) coordinates of a sphere :returns: **coords** – holds cartesian (x,y,z) and spherical (theta, phi) coordinates :rtype: named tuple

`sound_field_analysis.plot.genSphere(vizMTX)`

Returns trace of sphere with intensity as surface color

**Parameters** `vizMTX` (*array\_like*) – Matrix holding spherical data for visualization

**Returns** `T` – Trace of desired sphere

**Return type** `plotly_trace`

`sound_field_analysis.plot.genVisual(vizMTX, style='shape', normalize=True)`

Returns desired trace after cleaning the data

**Parameters**

- **vizMTX** (*array\_like*) – Matrix holding spherical data for visualization
- **style** (*string*{'shape', 'sphere', 'flat'}, *optional*) – Style of visualization. [Default: 'Shape']
- **normalize** (*Bool*, *optional*) – Toggle normalization of data to [-1 ... 1] [Default: True]

**Returns** `T` – Trace of desired visualization

**Return type** `plotly_trace`



`sound_field_analysis.plot.makeFullMTX (Pnm, dn, kr, Nviz=3)`

Generates visualization matrix for a set of spatial fourier coefficients over all kr :param Pnm: Spatial Fourier Coefficients (e.g. from S/T/C) :type Pnm: array\_like :param dn: Modal Radial Filters (e.g. from M/F) :type dn: array\_like :param kr: kr-vector

:: Can also be a matrix [krm; krs] for rigid sphere configurations: [1,:] => krm referring to the microphone radius [2,:] => krs referring to the sphere radius (scatterer)

**Parameters** `Nviz (int, optional)` – Order of the spatial fourier transform [Default: 3]

**Returns** `vizMtx` – Computed visualization matrix over all kr

**Return type** array\_like

`sound_field_analysis.plot.makeMTX (Pnm, dn, krIndex=1, Nviz=3, oversize=1)`

`mtxData = makeMTX(Nviz=3, Pnm, dn, krIndex)`

**Parameters**

- **Pnm** (*array\_like*) – Spatial Fourier Coefficients (e.g. from S/T/C)
- **dn** (*array\_like*) – Modal Radial Filters (e.g. from M/F)
- **krIndex** (*int*) – Index of kr to be computed [Default: 1]
- **Nviz** (*int, optional*) – Order of the spatial fourier transform [Default: 3]
- **oversize** (*int, optional*) – Integer Factor to increase the resolution. [Default: 1]

**Returns** `mtxData` – 3D-matrix-data in 1[deg] steps

**Return type** array\_like

---

**Note:** The file generates a SOFiA `mtxData` Matrix of 181x360 pixels for the visualisation with `visualize3D()` in 1[deg] Steps (65160 plane waves). The HD version generally allows to raise the resolution (`oversize > 1`). (`visual3D()`, `map3D()` admit 1[deg] data only, `oversize = 1`)

---

`sound_field_analysis.plot.normalizeMTX (MTX)`

Normalizes a matrix to [-1 ... 1]

**Parameters** `MTX (array_like)` – Matrix to be normalized

**Returns** `MTX` – Normalized Matrix

**Return type** array\_liked

`sound_field_analysis.plot.plot2D (data, type=None, fs=44100)`

Visualize 2D data using plotly.

**Parameters**

- **data** (*array\_like*) – Data to be plotted, separated along the first dimension (rows).
- **type** (*string{None, 'time'}*) – Type of data to be displayed. [Default: None]
- **fs** (*int*) – Sampling rate in Hz used to display time signals. [Default: 44100]

`sound_field_analysis.plot.plot3D (vizMTX, style='shape', layout=None, colorize=True)`

Visualize matrix data, such as from `makeMTX(Pnm, dn)`

**Parameters**

- **vizMTX** (*array\_like*) – Matrix holding spherical data for visualization
- **style** (*string{'shape', 'sphere', 'flat'}, optional*) – Style of visualization. [Default: 'shape']

- **normalize** (*Bool, optional*) – Toggle normalization of data to [-1 ... 1] [Default: True]

---

**Todo**

Colorization, contour plot

---

`sound_field_analysis.plot.showTrace` (*trace, layout=None, colorize=True*)

Wrapper around plotly's offline `.plot()` function

**Parameters**

- **trace** (*plotly\_trace*) – Plotly generated trace to be displayed offline
- **colorize** (*Bool, optional*) – Toggles bw / colored plot [Default: True]

**Returns** **fig** – JSON representation of generated figure

**Return type** `plotly_fig_handle`

`sound_field_analysis.plot.sph2cartMTX` (*vizMTX*)

Converts the spherical vizMTX data to named tuple containing `.xs/.ys/.zs`

**Parameters** **vizMTX** (*array\_like*) – [180 x 360] matrix that hold amplitude information over phi and theta

**Returns** **V** – Contains `.xs, .ys, .zs` cartesian coordinates

**Return type** `named_tuple`

## 5.4 Sphericals

Collection of spherical helper functions:

**sph\_harm** More robust spherical harmonic coefficients

**spbessel / dspbessel** Spherical Bessel and derivative

**spneumann / dspneumann** Spherical Neumann (Bessel 2nd kind) and derivative

**sphankel / dsphankel** Spherical Hankel and derivative

**cart2sph / sph2cart** Convert cartesian to spherical coordinates and vice versa

`sound_field_analysis.sph.besselh` (*n, z*)

Bessel function of third kind (Hankel function). Wraps `scipy.special.hankel1(n, z)` :param *n*: Order (float)  
:type *n*: array\_like :param *z*: Argument (float or complex) :type *z*: array\_like

**Returns** **H** – Values of Hankel function of order *n* at position *z*

**Return type** `array_like`

`sound_field_analysis.sph.besselj` (*n, z*)

Bessel function of first kind. Wraps `scipy.special.jn(n, z)` :param *n*: Order (float) :type *n*: array\_like :param *z*: Argument (float or complex) :type *z*: array\_like

**Returns** **J** – Values of Bessel function of order *n* at position *z*

**Return type** `array_like`

`sound_field_analysis.sph.bn` (*n, krm, krs, ac*)

`sound_field_analysis.sph.bn_dualOpenP` (*n, kr1, kr2*)

`sound_field_analysis.sph.bn_npf` (*n, krm, krs, ac*)

Microphone scaling

**Parameters**

- **n** (*int*) – Order
- **krm** (*array of floats*) – Microphone radius
- **krs** (*array of floats*) – Sphere radius
- **ac** (*int {0, 1, 2, 3, 4}*) –

**Array Configuration:**

- 0: Open Sphere with p Transducers (NO plc!) [Default]
- 1: Open Sphere with pGrad Transducers
- 2: Rigid Sphere with p Transducers
- 3: Rigid Sphere with pGrad Transducers
- 4: Dual Open Sphere with p Transducers

**Returns b****Return type** array of floats`sound_field_analysis.sph.bn_openP (n, krm)``sound_field_analysis.sph.bn_openPG (n, krm)``sound_field_analysis.sph.bn_rigidP (n, krm, krs)``sound_field_analysis.sph.bn_rigidPG (n, krm, krs)``sound_field_analysis.sph.cart2sph (x, y, z)`

Converts cartesian coordinates x, y, z to spherical coordinates az, el, r.

`sound_field_analysis.sph.dspbessel (n, kr)`

Derivative of spherical Bessel

**Parameters**

- **n** (*int*) – Order
- **kr** (*int*) – Degree

**Returns J'** – Derivative of spherical Bessel**Return type** complex float`sound_field_analysis.sph.dsphankel (n, kr)`

Derivative spherical Hankel function hn'

**Parameters**

- **n** (*int*) – Order
- **kr** (*int*) – Degree

**Returns hn'** – Derivative of spherical Hankel function hn'**Return type** complex float`sound_field_analysis.sph.dspneumann (n, kr)`

Derivative spherical Neumann (Bessel second kind) of order n

**Parameters**

- **n** (*int*) – Order
- **kr** (*int*) – Degree

**Returns Yv'** – Derivative of spherical Neumann (Bessel second kind)**Return type** complex float`sound_field_analysis.sph.spbessel (n, kr)`

Spherical Bessel function

**Parameters**

- **n** (*int*) – Order
- **kr** (*int*) – Degree

**Returns** **J** – Spherical Bessel**Return type** complex float`sound_field_analysis.sph.sph2cart (az, el, r)`

Converts spherical coordinates az, el, r to cartesian coordinates x, y, z.

`sound_field_analysis.sph.sph_harm (m, n, az, el)`

Compute spherical harmonics

**Parameters**

- **m** (*int*) – Order of the spherical harmonic.  $\text{abs}(m) \leq n$
- **n** (*int*) – Degree of the harmonic, sometimes called  $l$ .  $n \geq 0$
- **az** (*float*) – Azimuthal (longitudinal) coordinate  $[0, 2\pi]$ , also called Theta.
- **el** (*float*) – Elevation (colatitudinal) coordinate  $[0, \pi]$ , also called Phi.

**Returns**

- **y\_mn** (*complex float*) – Complex spherical harmonic of order  $m$  and degree  $n$ , sampled at  $\text{theta} = \text{az}$ ,  $\text{phi} = \text{el}$
- $Y_{n,m}(\text{theta}, \text{phi}) = ((n - m)! * (2l + 1)) / (4\pi * (l + m)!^{0.5} * \exp(i * m * \text{phi}) * P_n^m(\cos(\text{theta}))$
- as per <http://dlmf.nist.gov/14.30>
- $P_m(z)$  is the associated Legendre function of the first kind, like `scipy.special.lpmv`
- `scipy.special.lpmn` calculates  $P(0..m, 0..n)$  and its derivative but won't return  $+\infty$  at high orders

`sound_field_analysis.sph.sphankel (n, kr)`Spherical Hankel  $h_n$ **Parameters**

- **n** (*int*) – Order
- **kr** (*int*) – Degree

**Returns** **hn** – Spherical Hankel function  $h_n$ **Return type** complex float`sound_field_analysis.sph.spneumann (n, kr)`

Spherical Neumann (Bessel second kind)

**Parameters**

- **n** (*int*) – Order
- **kr** (*int*) – Degree

**Returns** **Yv** – Spherical Neumann (Bessel second kind)**Return type** complex float

## 5.5 I/O

Input-Output functions

`sound_field_analysis.io.readMiroStruct (matFile)`

Reads miro matlab files.

**Parameters** `matFile` (*filepath*) – .miro file that has been exported as a struct like so

```
load SOFiA_A1;
SOFiA_A1_struct = struct(SOFiA_A1);
save('SOFiA_A1_struct.mat', 'SOFiA_A1_struct');
```

**Returns**

- **timeData** (*named tuple*)
- *timeData* tuple with following fields
- : – .impulseResponses [Channels X Samples] .FS .radius Array radius .averageAirTemp Temperature in [C] (.centerIR [1 x Samples] )

## 5.6 lebedev

Generate Lebedev grid and coefficients This module only exposes the function `lebGrid = lebedev.genGrid(degree)`.

`lebGrid` is a named tuple containing the coordinates `.x`, `.y`, `.z` and the weights `.w` Possible degrees: 6, 14, 26, 38, 50, 74, 86, 110, 146, 170, 194

Adapted from Richard P. Mullers Python version, [https://github.com/gabrielelanaro/pyquante/blob/master/Data/lebedev\\_write.py](https://github.com/gabrielelanaro/pyquante/blob/master/Data/lebedev_write.py) C version: Dmitri Laikov F77 version: Christoph van Wuelen, <http://www.ccl.net>

Users of this code are asked to include reference [1] in their publications, and in the user- and programmers-manuals describing their codes.

[1] V.I. Lebedev, and D.N. Laikov ‘A quadrature formula for the sphere of the 131st algebraic order of accuracy’ Doklady Mathematics, Vol. 59, No. 3, 1999, pp. 477-481.

`sound_field_analysis.lebedev.genGrid (n)`

Returns Lebedev coefficients of n'th degree

**Parameters** `n` (*int*{6, 14, 26, 38, 50, 74, 86, 110, 146, 170, 194}) – Lebedev degree

**Returns** `lebGrid` – `lebGrid` is a named tuple containing `.x`, `.y`, `.z` and `.w`

**Return type** named tuple

## 5.7 Utilities

Miscellaneous utility functions

`sound_field_analysis.utils.env_info()`

Guess environment based on sys.modules.

**Returns** `env` – Guesed environment

**Return type** string{‘jupyter\_notebook’, ‘ipython\_terminal’, ‘terminal’}

`sound_field_analysis.utils.progress_bar (curIDX, maxIDX=None, description='Progress')`

Display a spinner or a progress bar

**Parameters**

- **curIDX** (*int*) – Current position in the loop
- **maxIDX** (*int*, *optional*) – Number of iterations. Will force a spinner if set to None. [Default: None]

- **description** (*string*, *optional*) – Clarify what's taking time

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