

THE AMBISONIC DECODER TOOLBOX: EXTENSIONS FOR PARTIAL COVERAGE LOUDSPEAKER ARRAYS

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What is Ambisonics?

- Extensible, hierarchical system for representing sound fields
 - Says how something should sound, rather than specific speaker signals.
- Capture or creation
 - Microphone arrays
 - 2-D or 3-D
 - Natural B-format, Tetrahedral, Spherical arrays
 - Ambisonic Panners
- Reproduction
 - 2-D, “horizontal” or 3-D “with height” loudspeaker arrays
 - “Any” size or shape array of loudspeakers

What is an Ambisonic Decoder?

- In Ambisonics, the program format is independent of the reproduction layout.
- The decoder's task is to create the best *perceptual impression* possible that the sound field is being reproduced accurately, given the resources available
 - Bandwidth, number of speakers, configuration of speakers ...
- We use the term “decoder” to mean the configuration for a decoding engine that does the actual signal processing

Goals for decoder design

- Mimic conditions of natural hearing
 - Constant amplitude gain for all source directions (P)
 - Constant energy gain for all source directions (E)
 - At low frequencies, correct reproduced wavefront direction and velocity (r_V)
 - At high frequencies, maximum concentration of energy in the source direction (r_E)
 - Matching high- and low-frequency perceived directions
- Getting r_E correct is the most difficult aspect
- Recent work shows that it is also the most important!

Designing Decoders

- Decoders for regular polygon and polyhedra loudspeaker arrays are easy to design
 - Build the speaker encoding matrix, K , by sampling the spherical harmonics at the speaker directions
 - Use pseudoinverse to find the basic decoding matrix M
 - rE guaranteed to point in same direction as rV
- However...
 - Room geometry or visual considerations often limit speaker placement
 - 3-D HOA requires placing more speakers above and *below* the listener

How you'd like to do it

AuraLab, San Francisco

A useful compromise



The Bubble, San Francisco

Tradeoffs

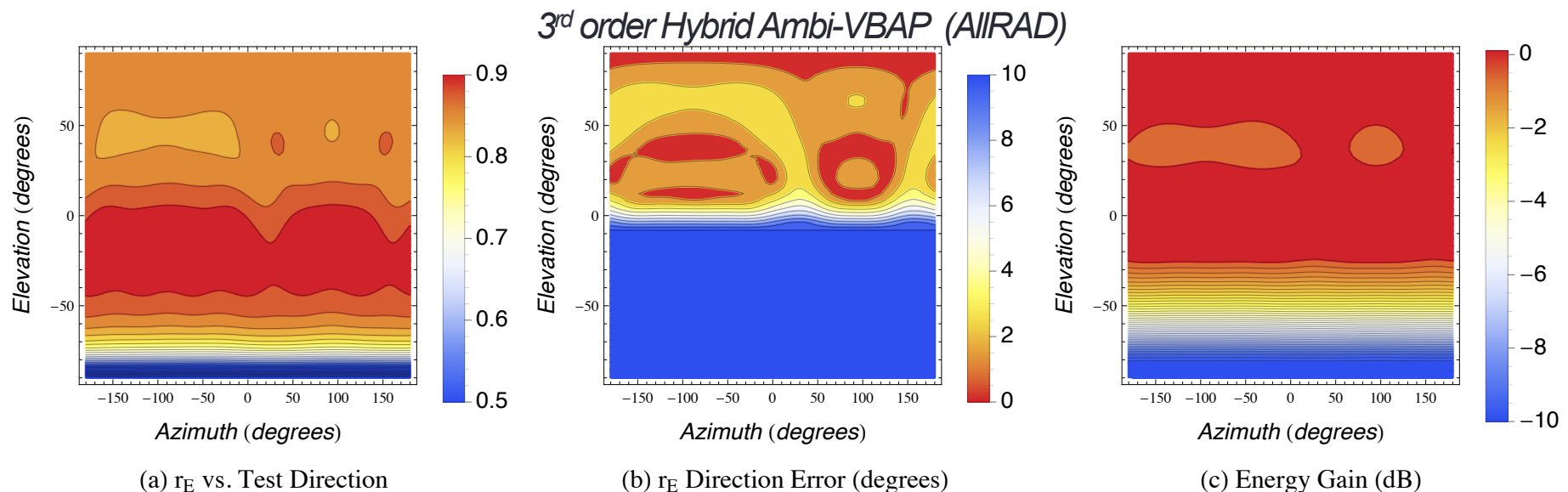
- Once we deviate from regular geometry
 - we must trade off localization accuracy for uniform loudness
 - Directions of r_E and r_V are not the same
- Localization degrades outside the area with a high density of loudspeakers
- Gerzon used nonlinear optimization for this
 - Many implementations: Wiggins, Moore & Wakefield, Tsang, BLaH
- Works well for small arrays (e.g., ITU 5.1)
- Convergence is slow for large HOA arrays (hrs)
- IDHOA (Scaini and Arteaga) looks promising
 - Better objective function and zero out small coefficients

New Strategies in Toolbox

- Use an inversion technique suited to ill-conditioned matrices
 - Constant energy decoder
 - Truncated SVD
 - Energy limited
- Invert a well-behaved full-sphere virtual speaker array, map to a real array
 - Hybrid Ambisonic-VBAP
 - AllRAD (Zotter and Frank)
- Derive a new set of basis functions for which inversion is well behaved
 - Spherical Slepian Functions
 - EPAD (Zotter, Pomberger, Noisternig)

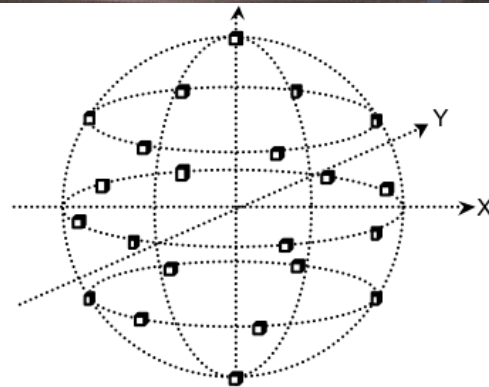
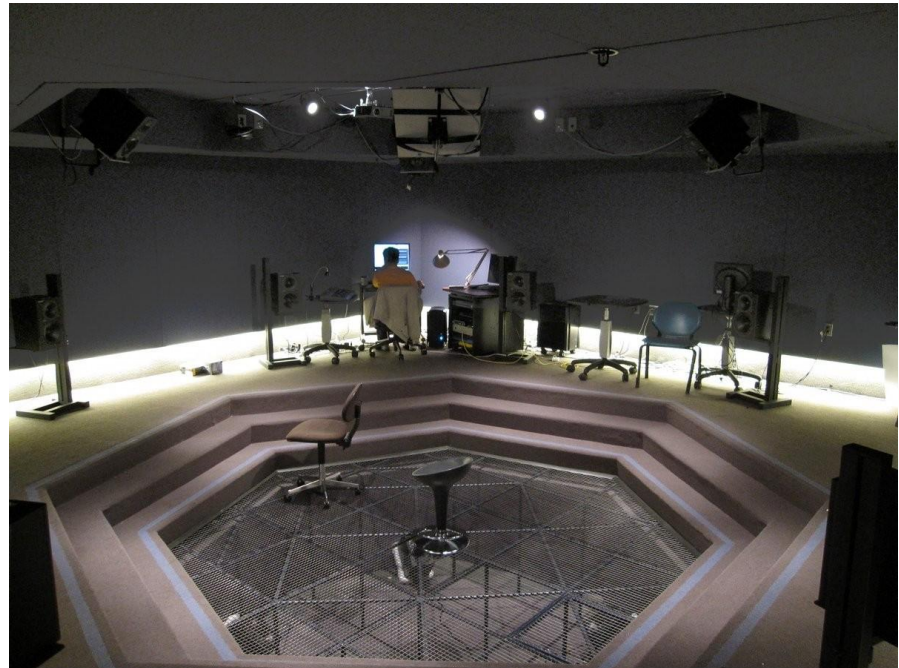
Are these decoders Ambisonic?

- Ambisonic theory specifies performance goals, not how to design a decoder
- We use the same criteria for these decoders
- But...
 - Apply them only to source directions in the covered part of the sphere
 - Require them be “well behaved” in other directions

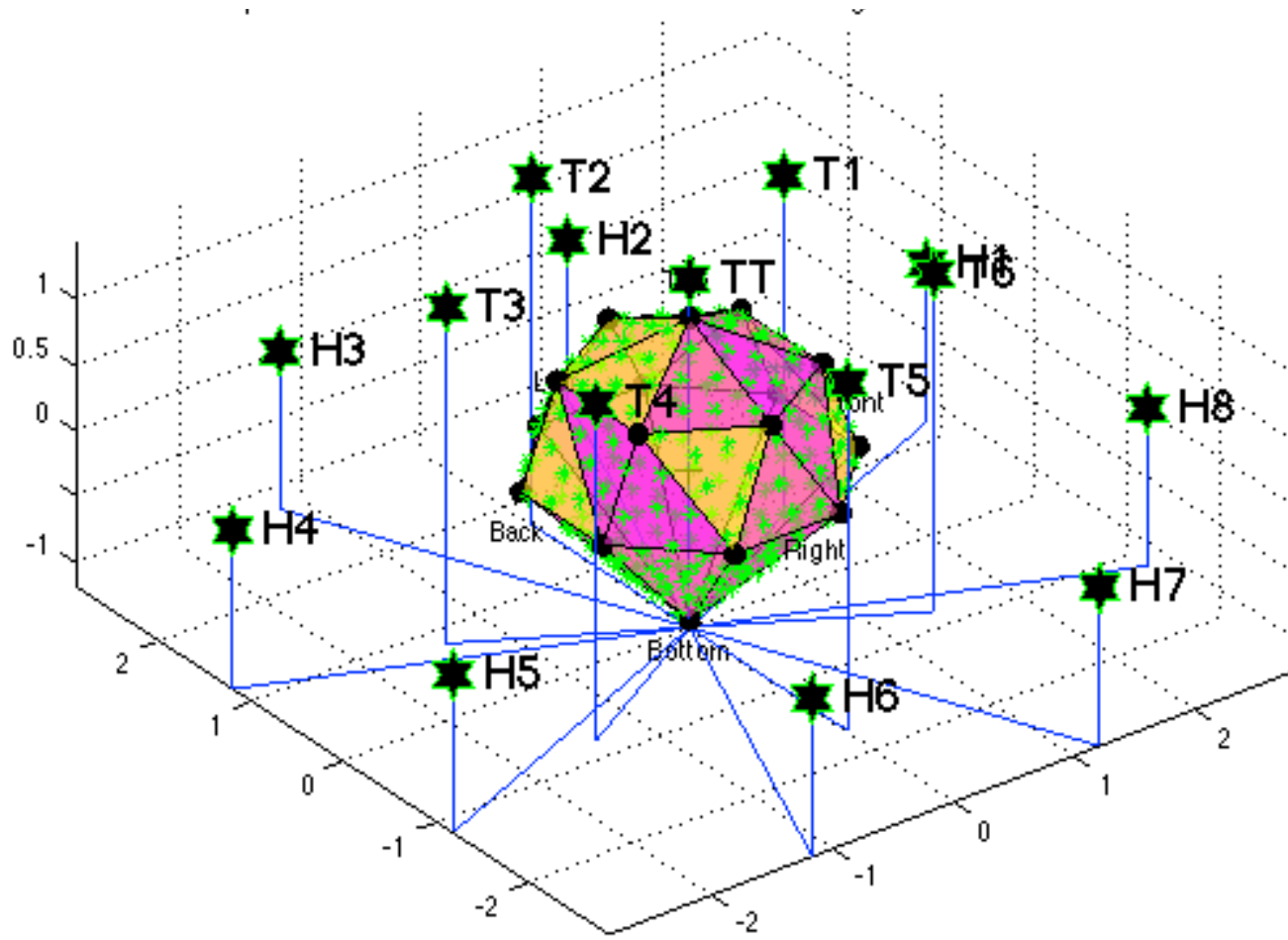


CCRMA Listening Room

- 22 identical loudspeakers in five rings
- Horizontal ring of 8 loudspeakers
- 2 rings of 6 loudspeakers, one 50° below horizontal and one 40° above
- 1 loudspeaker at each pole
- Array is almost regular
- Upper 15 used for hemispherical dome
- Full-sphere decoder described in our LAC2012 paper



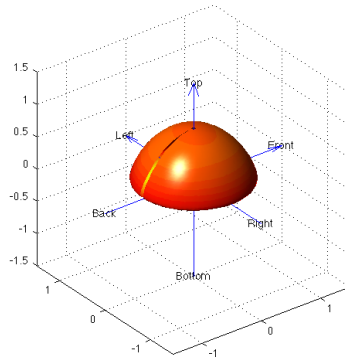
AIIRAD – Hybrid Ambi-VBAP



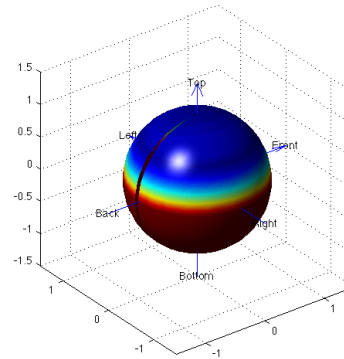
- 240 point spherical design for virtual speaker array
- Dome of upper 15 loudspeakers of CCRMA Listening Room, 8-6-1
- Imaginary speaker at bottom
- Design procedure detailed in paper

AIIRAD performance r_v

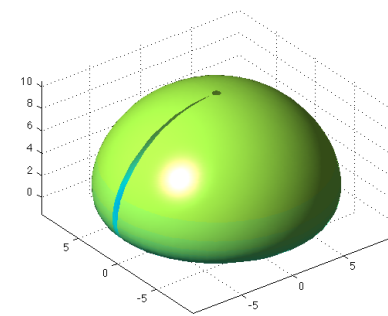
mag and dir of r_v



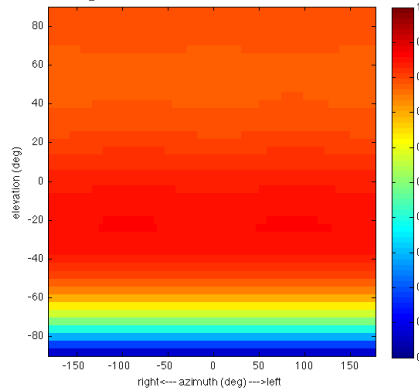
r_v angular error (degrees)



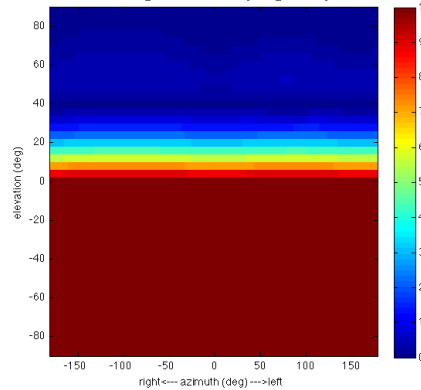
mag and dir of Pressure gain



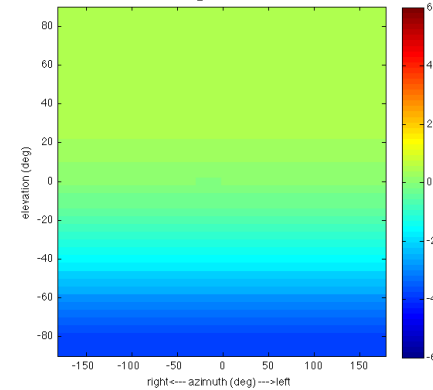
magnitude of r_v vs. test direction



r_v angular error (degrees)

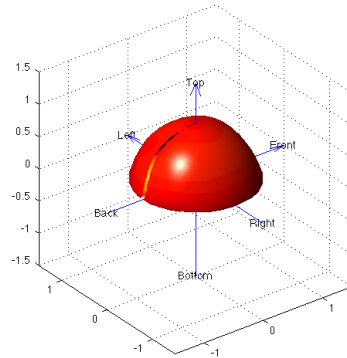


Pressure gain vs. test dir

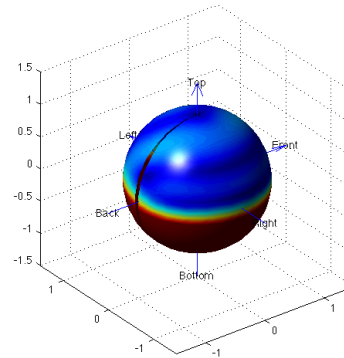


AIIRAD performance r_E

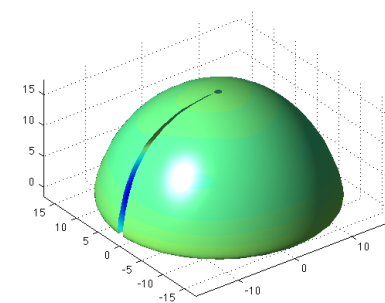
mag and dir of r_E



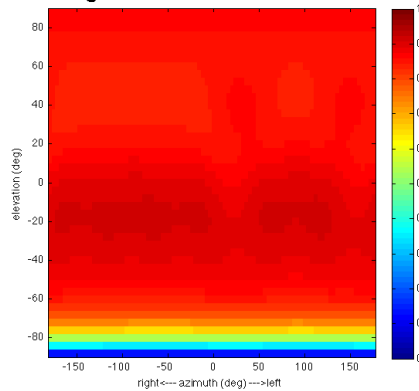
r_E angular error (degrees)



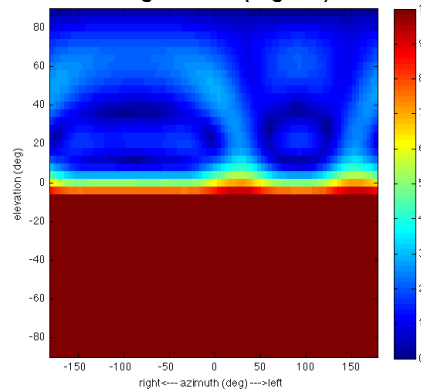
mag and dir of Energy gain



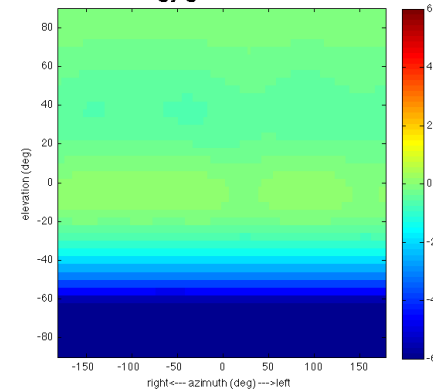
magnitude of r_E vs. test direction



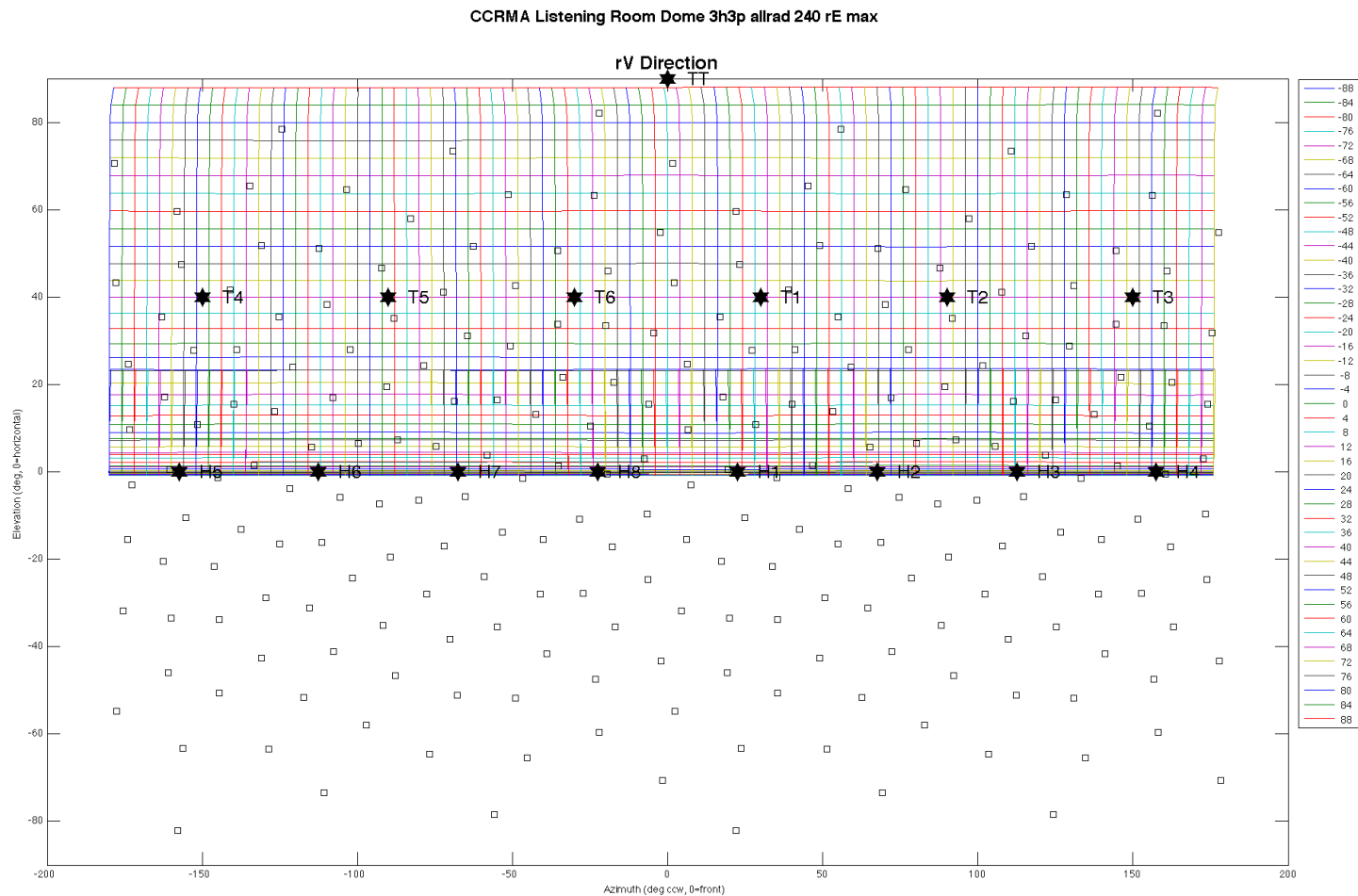
r_E angular error (degrees)



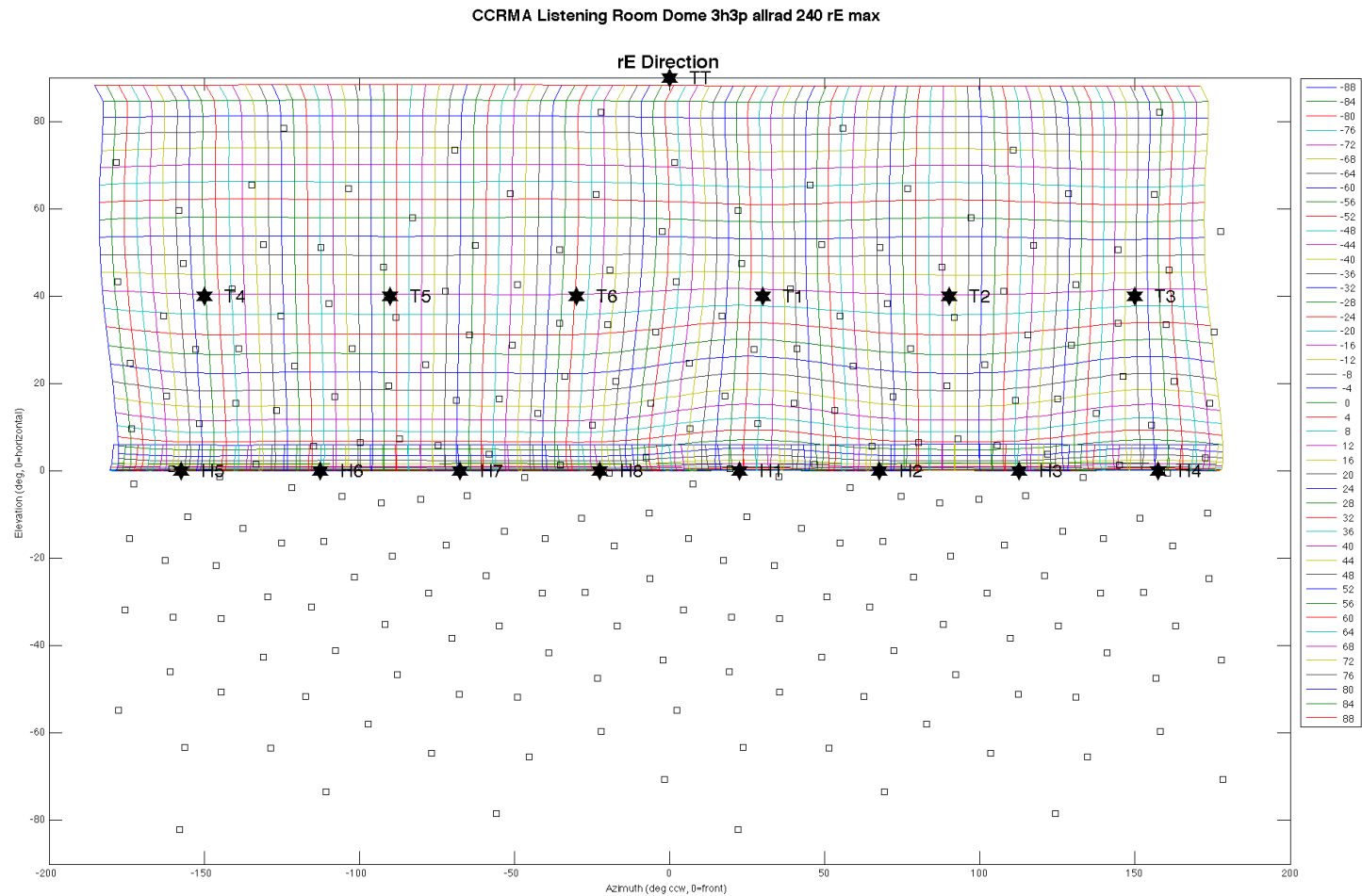
Energy gain vs. test dir



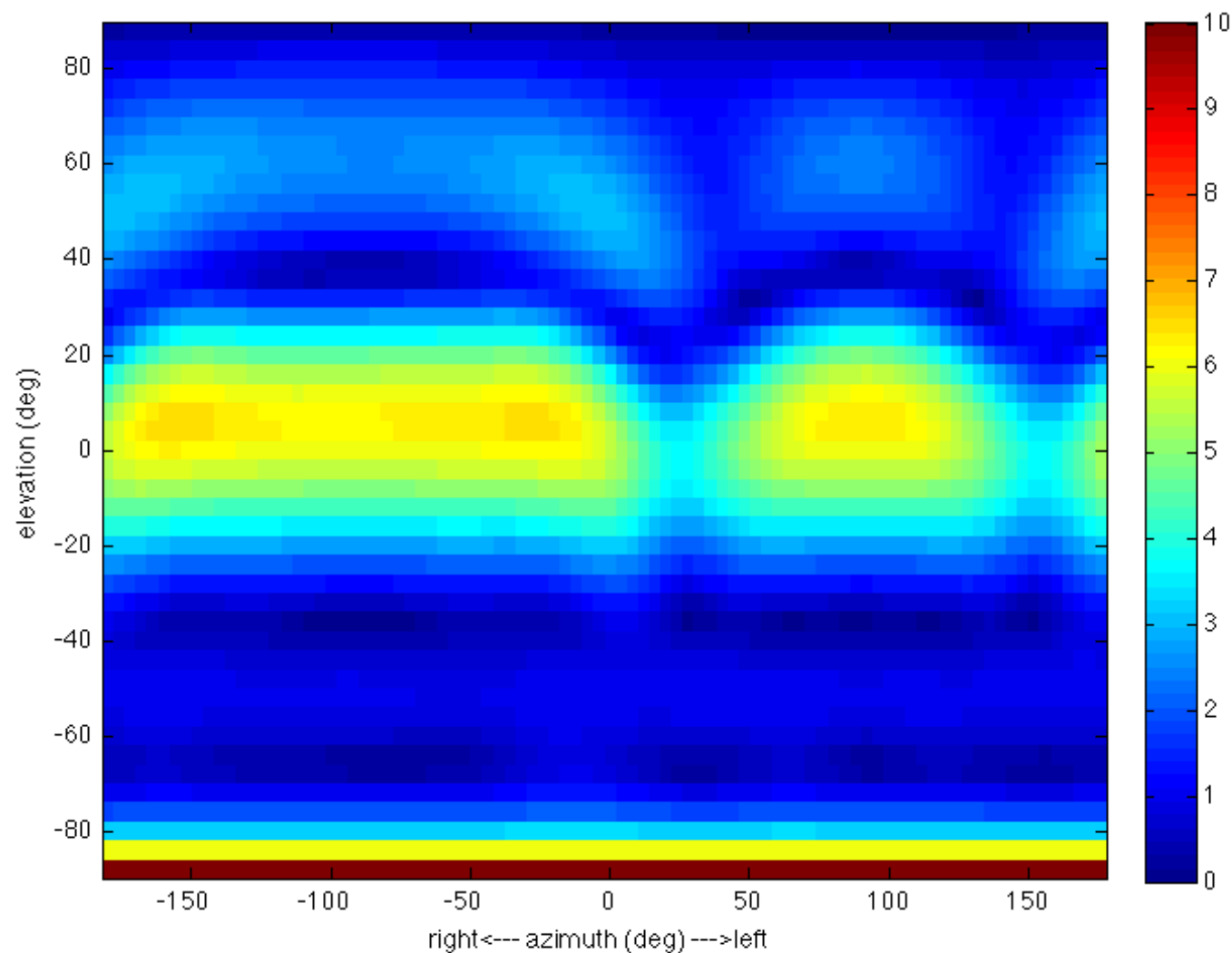
AIIRAD r_v direction grid



AIRAD r_E direction grid



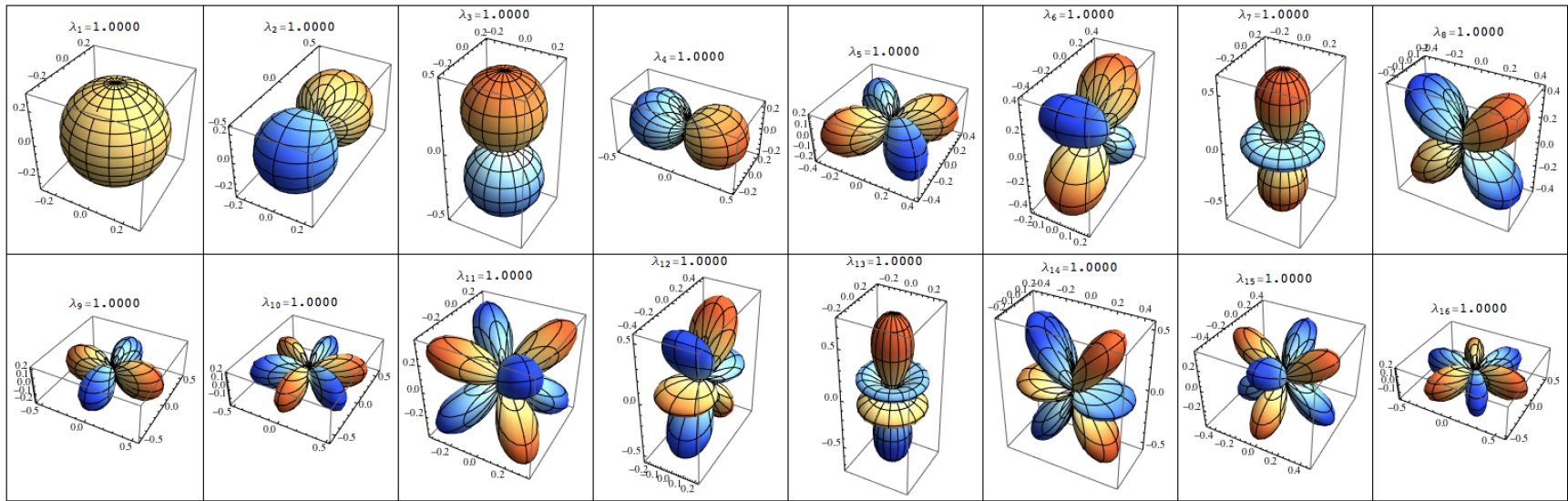
CCRMA Listening Room Dome 3h3p allrad 240 rE max rV rE Direction Difference



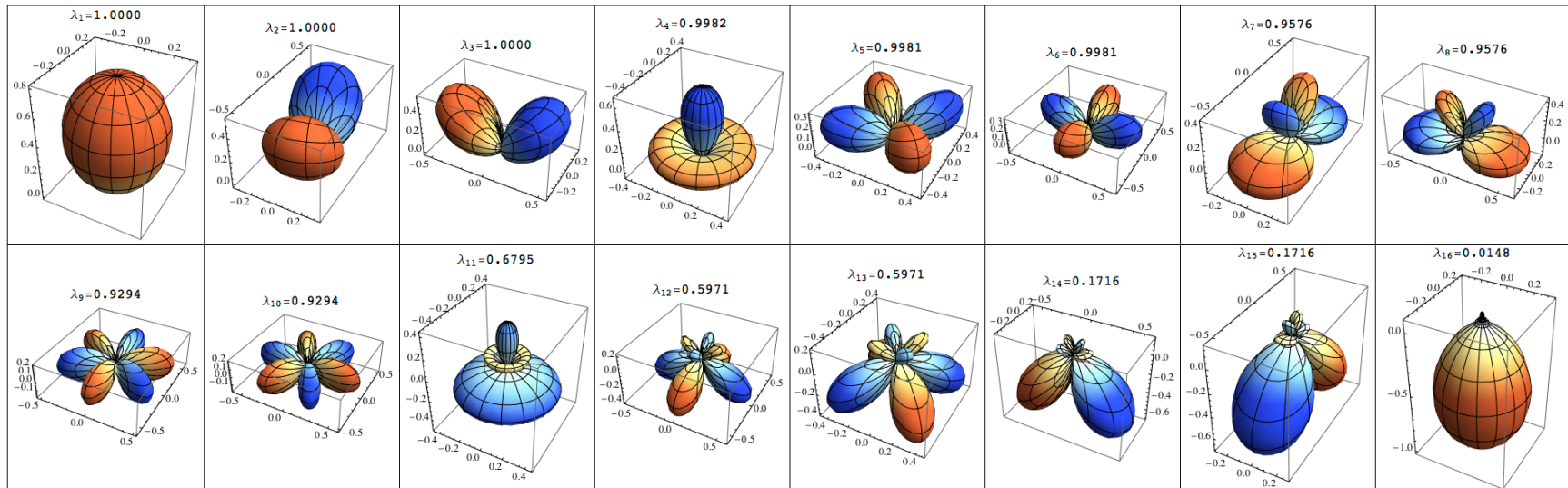
Spherical Slepian Functions

- Linear combinations of spherical harmonics
- Produce a new set of basis functions that are zero outside the region of interest on the sphere
- Remain orthogonal within the region
- Used in satellite geodesy to model earth's gravitational and magnetic fields from incomplete data
- In Ambisonic decoding, we can specify a region of the sphere, a dome or a ring, and derive a well behaved set of basis functions for that region.
- Design procedure detailed in paper

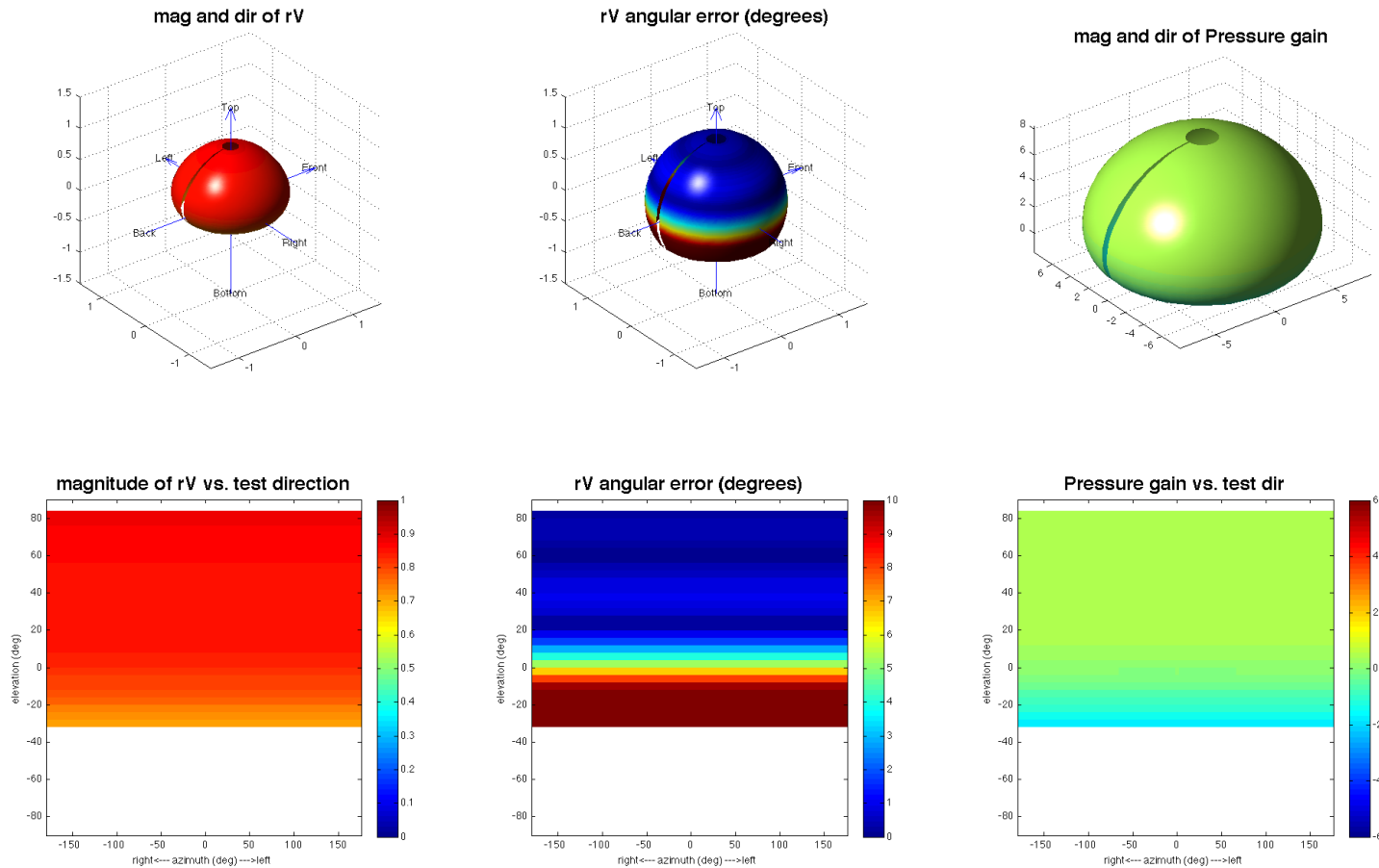
3rd order spherical harmonics (blue = inverted polarity)



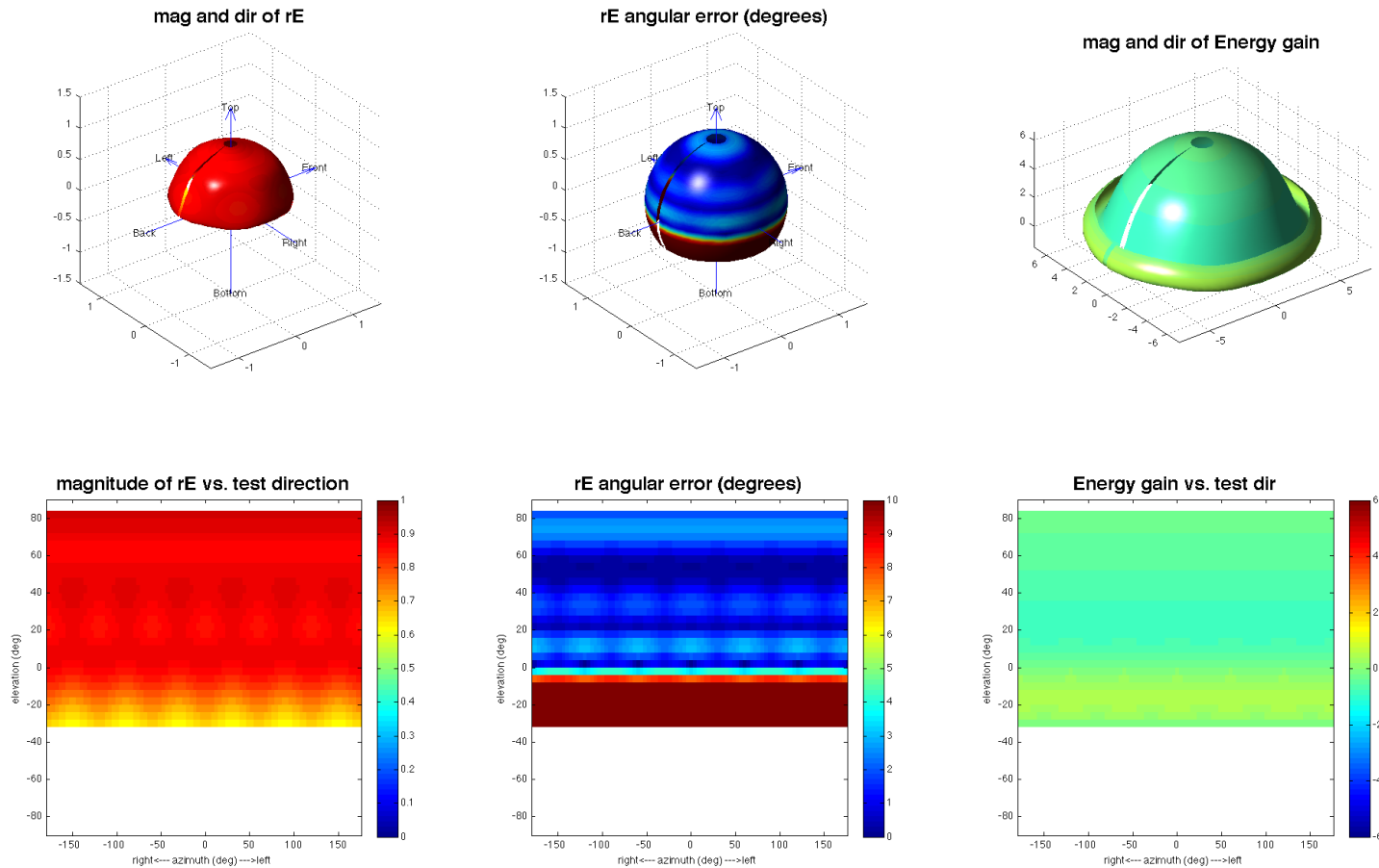
3rd order spherical Slepian functions for +90° to -30° dome (first 13 used for decoder)



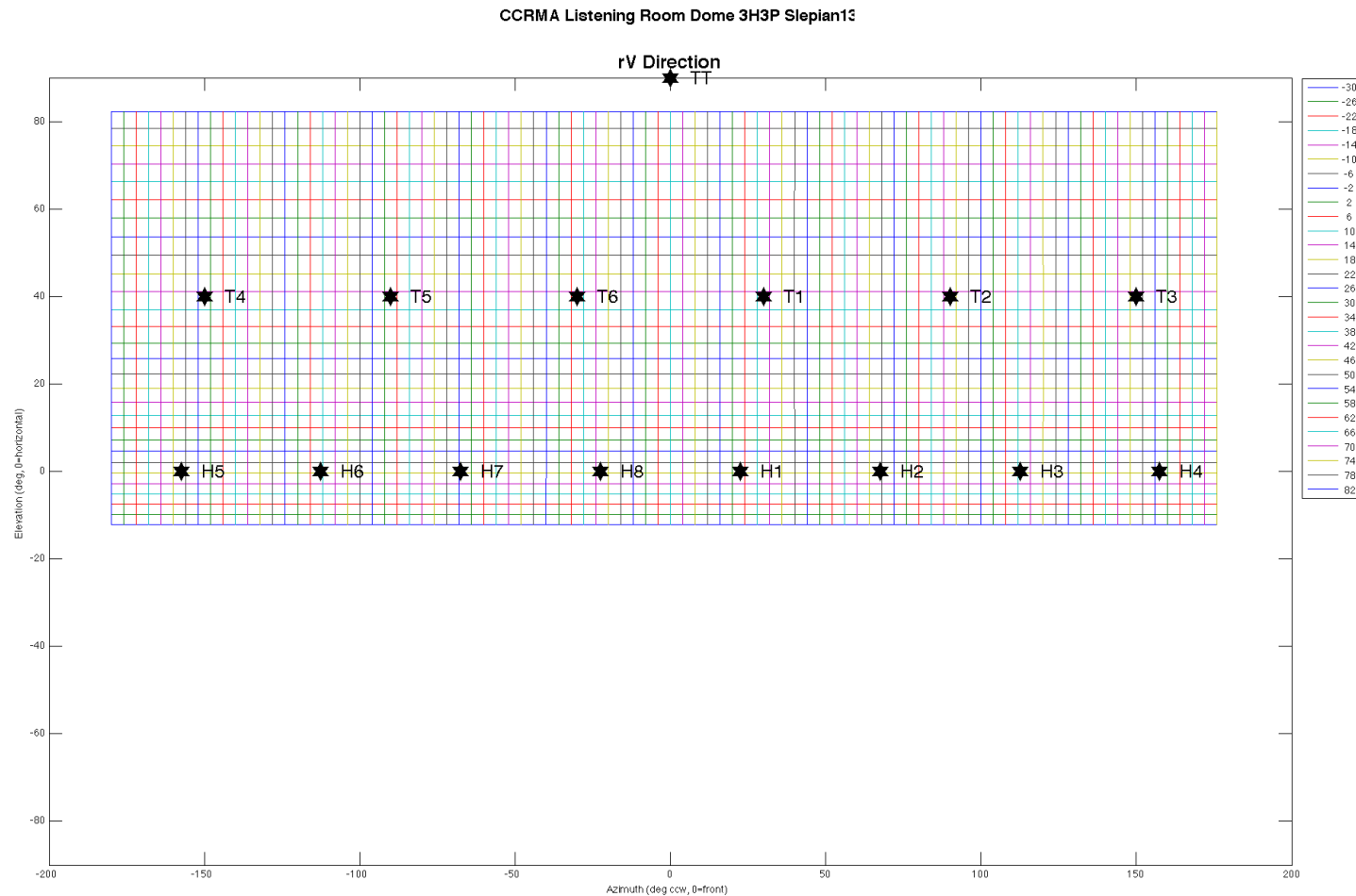
Spherical Slepian performance r_v



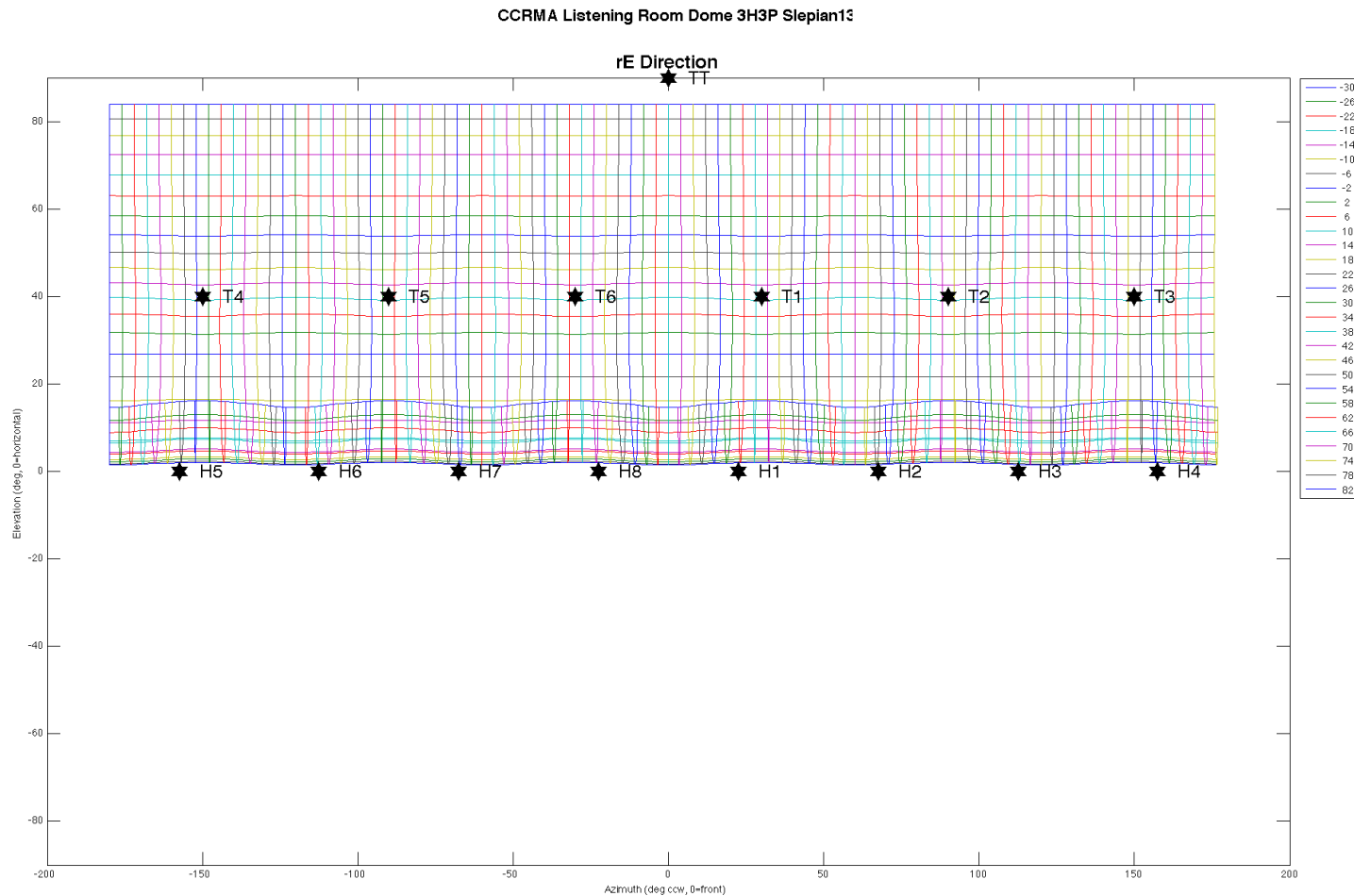
Spherical Slepian performance r_E



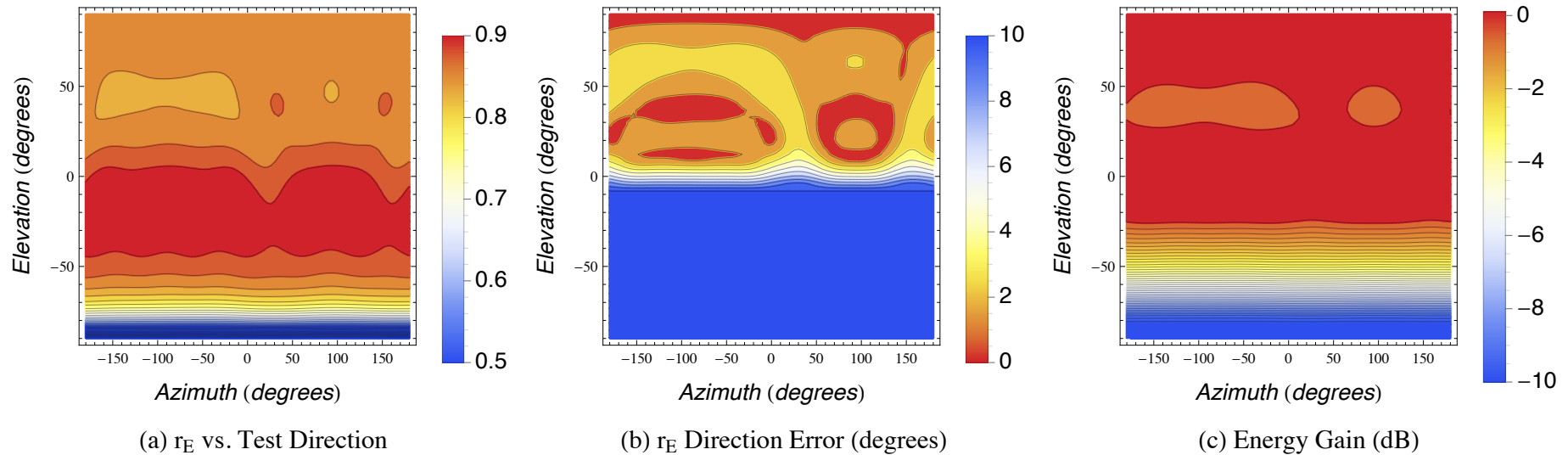
Spherical Slepian r_v direction grid



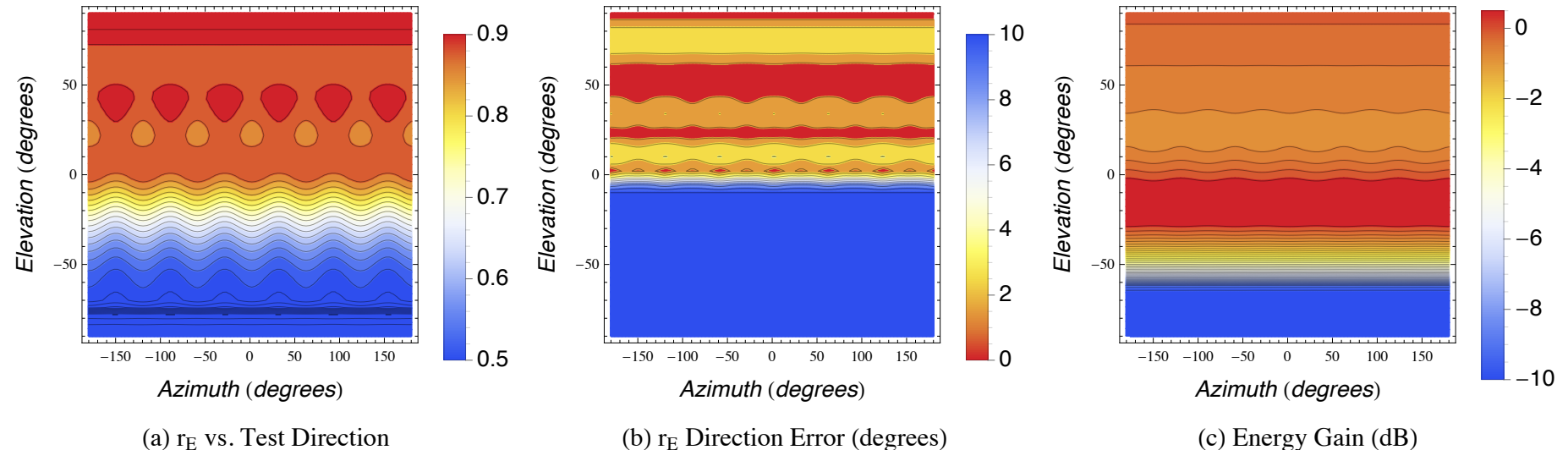
Spherical Slepian r_E direction grid



3rd order Hybrid Ambi-VBAP (AIRAD)



3rd order spherical Slepian function (EPAD)



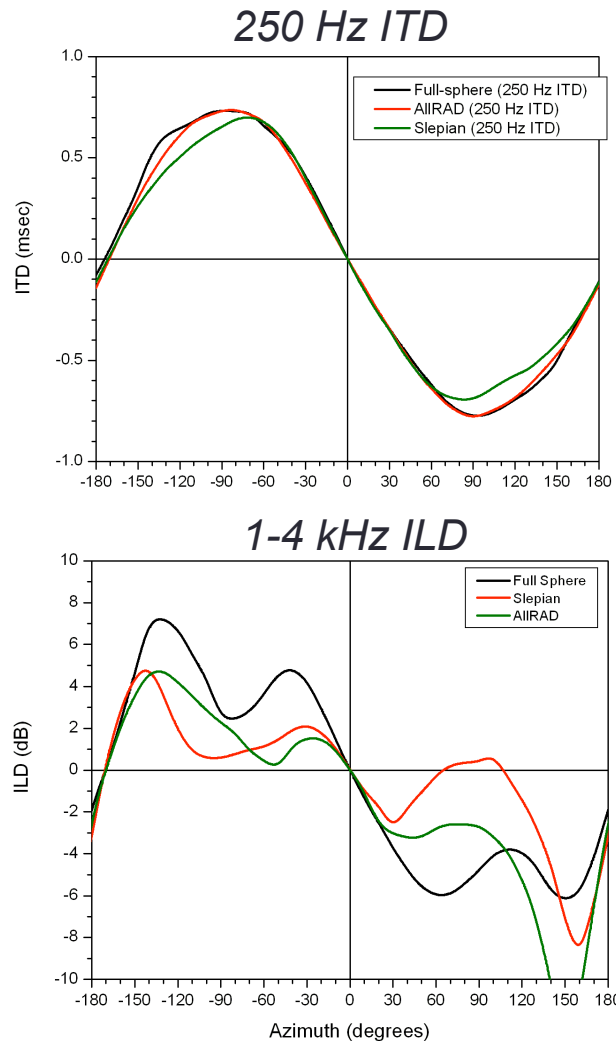
In situ performance measurements



- Dummy head and reference omni
- Dome array using upper 15 speakers in CCRMA's listening room (8-6-1)

- Tested
 - AllRAD Dome
 - Spherical Slepian Dome
 - Full-sphere (from LAC2012)
- Collected
 - individual speaker IRs
 - Ambisonically panned IRs at 10° azimuth, 30° elevation intervals for each decoder
- Analyzed horizontal data
 - 250 Hz ITD (r_V)
 - 1 to 4 kHz ILD (r_E)

ITD and ILD measurements



Observations

- The measured ITDs were similar with the three decoders but ILDs were very different
- This supports the subjective observations that the three decoders sound different
- Detailed analysis is pending

Informal listening tests

- 3rd-order test programs
 - Full-sphere mix of “Babel” by Allette Brooks (Jay Kadis)
 - *Chroma XII* by Rebecca Sanders (Jörn Nettingsmeier)
- Both dome decoders sounded good subjectively (but different!)
 - Compact and directionally accurate localization down to horizon
 - Faded below horizon
 - SSF decoder sounded brighter and more detailed than AllRAD
- Neither decoder sounded as good as the full-sphere reference decoder
- 1st-order orchestral recording not reproduced well
 - Most of orchestra is below the horizon

Decoding Engine

- New decoding engine written in FAUST
 - No inherent limit on order
 - Dual band, NFC filters, distance compensation, ...
- Toolbox writes out configuration section, appends implementation
- Compiles to LADSPA, LV2, Pd, Supercollider, VST, AU ...
- Can be used independently of toolbox
- Drawback: Configuration “baked into” plugin
- Toolbox also writes out configuration files for
 - Kronlachner’s ambiX plugin suite
 - Adriaensen’s Ambdec

Implementation

- Toolbox runs in MATLAB and GNU Octave
 - Implements all known channel ordering and normalization conventions; both mixed-order conventions (HP and HV)
 - No inherent limit on Ambisonic order
 - Actively in use by a few beta testers
 - Mixed results for graphics output in Octave
 - Moving graphics output code to Python with MayaVi
 - Interface to IDHOA optimizer
- GNU Affero General Public License
- Faust decoder engine BSD 3-Clause License
- Git repo at <https://bitbucket.org/ambidecodertoolbox/adt>

Summary and Conclusions

- Extensions to Ambisonic Decoder Toolbox to handle speaker configurations that do not cover full sphere
- New decoder engine is written in Faust
- Ability to generate decoders quickly has proven valuable in performance settings
- Plans
 - Dual-band AllRAD and Slepian decoders
 - Optimizer to refine decoders
- Open question:
 - What to do when sources move into areas of poor coverage.
 - Current implementation fades them out.
 - Decorrelate and mix into other speakers?
 - Should transmission standards include “rendering hints”?

Thanks!

- Fernando Lopez-Lezcano for helping with the listening tests and in-situ measurements, and overall feedback and encouragement.
- Andrew Kimpel, Marc Lavallée, and Paul Power who are active users.
- Richard Lee, Jörn Nettingsmeier, and Bob Oldendorf who read early drafts and provided feedback.
- LAC 2014 reviewers and organizers

Human Auditory Localization

- At low frequencies (up to about 800 Hz) works by Interaural *Time* Differences (ITDs)
- At middle frequencies (800 Hz to 5 kHz) works by Interaural *Level* Differences (ILDs)
 - Transition is fairly sharp
 - due to the ITDs becoming ambiguous once the wavelength become smaller than ear spacing.
- 2-channel stereo doesn't get it right
 - ILD cues are such that the images tend to stick to nearest speaker
- Ambisonics was designed from the beginning to get this correct with modest resources.
 - Small number of program channels and loudspeakers

Gerzon's Theory of Auditory Localization

- Early workers in stereo did theoretical analysis showing how stereo did (or didn't) provide proper localization cues
- Gerzon's contribution was to integrate those theories and came up with a theory that defined
 - \mathbf{r}_V , the vector sum of the signals from the loudspeakers
 - \mathbf{r}_E , the vector sum of the squares of the signals from the loudspeakers.
- By providing a simple mathematical encapsulation, we can use these to
 - design decoders
 - prove theorems, e.g., polygonal decoder theorem
 - help understand what various spatial sound reproduction systems can and cannot do

Localization Vector Theory

- \mathbf{r}_V predicts low-frequency localization almost perfectly.
 - If $\mathbf{r}_V=1$, then low-frequency sounds will be precisely located.
- \mathbf{r}_E predicts mid-frequency localization moderately well.
 - If $\mathbf{r}_E=1$, then mid-frequency localization will be good
 - BUT... \mathbf{r}_E is always less than 1, unless the sound is coming from a single point source.
 - At best $\mathbf{r}_E = \cos(\theta/2)$, where θ is the angle between the loudspeakers, so for a square array $\mathbf{r}_E \leq 0.707$.
 - In general, \mathbf{r}_E is low in directions with few loudspeakers
 - Best we can do is have it change smoothly in performance from dense areas to sparse areas.

Energy Localization Vector

- Maximizing \mathbf{r}_E *and* getting it to point in the right direction is the crux of the decoder design problem.
 - Easy with regular arrays
 - Irregular arrays always involve tradeoffs
 - Virtually all real world arrays are irregular!
 - Arrays need to fit in real rooms
 - ITU 5.1 is the dominant domestic standard, rear speakers 120° apart.
- Because it is a non-linear function of speaker position, we currently need to use numerical optimization methods.