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AMBISONIC ANALYSIS AND AURALIZATION FOR STAGE ACOUSTICS

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Abstract: *A spherical microphone array has been designed and constructed to measure 2nd order ambisonics, based on equations for scattering off a rigid sphere. The array was used to measure the stage acoustics of 9 performance halls around the state of New York. Acoustical data derived from a beamforming analysis of each stage is shown and compared to A.C. Gade's omnidirectional parameters (stage support, early ensemble level) and geometric parameters described by J. Dammerud, as validation of the microphone array and its potential use in the field of stage acoustics. Discussion of future auralizations is outlined.*

Spherical Microphone Array, Beamforming, Stage Acoustics, Ambisonics

1 INTRODUCTION

The field of stage acoustics has seen a great deal of maturation in the last 20 years. Starting with seminal work by A.C. Gade and continuing up to the present, with work by Dammerud, Ueno, and Jurkiewicz, among others, several parameters have been established as relevant for musicians onstage. However, with the exception of one parameter proposed by Dammerud, these parameters are omnidirectional. Listening tests used to determine these parameters are based on omnidirectional recordings or simulations, and reproduction systems such as binaural and VBAP (Vector-Based Amplitude Panning). The goal of this work has been to increase the accuracy of both the measurement and reproduction systems for stage acoustics measurements using real impulse responses measured with a spherical microphone array and reproduced over a sphere of loudspeakers using 2nd-order ambisonic decoding.

2 BACKGROUND AND THEORY

2.1. Stage Acoustics

A.C. Gade is best known for his contribution of the Stage Support (ST1) parameter, which calculates the ratio of direct sound to early energy onstage [1][2]. The development of this parameter came from several subjective tests utilizing both simulated and real impulse responses between 1981 and 1989. The results of his studies are shown in Table 1.

Subjective Parameter	**Objective Parameter**	**Preferred Values**
Reverberation (Soloist)	T_{20}, TA, C_{80}	Higher values preferred
Support (Soloist)	ST1	>-10dB preferred
Timbre (Soloist)	Early Reflection Spectra	High requencies preferred by violins, low frequencies preferred by cellists and flutists
Hearing Each Other (Ensemble)	EEL (Early Ensemble Level), EDT, C_{80}	High values of high-frequeny early energy and low values of reverberation preferred
Time Delay (Ensemble)	Direct sound delay	Delays of less than 20 ms preferred

Table 1: Correlation between subjective and objective parameters for stage acoustics, A.C. Gade

The definition of stage support is shown below:

$$\mathrm{ST1} = 10\log_{10}\left[\frac{\int_{20}^{100} p^2(t)dt}{\int_{0}^{20} p^2(t)dt}\right] \tag{1}$$

Another parameter, determined to be relevant for hearing each other in an ensemble situation, was developed through this work, Early Ensemble Level (EEL). This parameter is defined below:

$$\mathrm{EEL} = 10\log_{10}\left[\frac{\int_{0}^{80} p_{receiver}^2(t)dt}{\int_{0}^{20} p_{source}^2(t)dt}\right] \tag{2}$$

The relevance of ST1 was confirmed in work by Ueno, et al. [10] In 2005, Y. Jurkiewicz also validated this parameter, among others [5]. The results of his study are shown in Table 2.

Subjective Parameter	**Objective Parameter**	**Desired Range for Large Orchestra**
Reverberance	EDT_{10}	>1.5 s
Blending	C_{80} T_{30} G_{Late}	1 dB -- 4 dB > 1.9 s > 0 dB
Loudness	G	<6.5 dB
Support	ST1 ST3 G_{Early} G_{Late}	>-14 dB -15 dB – ST1 + 1 dB >3 dB 0 dB -- G_{Early}
Ease of Ensemble	ST1	>-14 dB

Table 2: Correlation between subjective and objective parameters for stage acoustics, Y. Jurkiewicz

Recently, work by J. Dammerud examined additional parameters, running reverberance (RR160) and strength (G, also examined by Jurkiewicz) [3]. These parameters are defined below:

$$\mathrm{RR160} = 10\log_{10}\left[\frac{\int_{160}^{320} p^2(t)dt}{\int_{0}^{160} p^2(t)dt}\right] \tag{3}$$

$$\mathrm{G}_i = 10\log_{10}\left[\frac{31200 \bullet T}{V} \bullet e^{-0.04r/T} \bullet e^{-1.11/T}\right] \tag{4}$$

His results led to the proposal of a new parameter, the ratio of stage enclosure height (measured from the brass section) to stage enclosure width (measured from the strings section). All of these omnidirectional parameters and the one directional height-width ratio have been calculated for the halls measured in this study.

2.2. Spherical Microphone Array

In order to analyze the halls for spatial information, a spherical microphone array was designed to capture Spatial Impulse Responses (SIRs). The theory behind spherical arrays is based on the spatial sampling of a sphere and the decomposition of the pressure on this sphere into spherical harmonics [9]. The decomposition is achieved using the spherical Fourier transform, defined below:

$$f_{nm} = \int_{\Omega \in S^2} f(\Omega) Y_n^{m*}(\Omega) d\Omega \tag{5}$$

where Ω represents the domain of the unit sphere (θ,φ,r) and the spherical harmonics are the orthogonal, angular components, defined as follows:

$$Y_n^m(\theta,\varphi) = \sqrt{\frac{(2n+1)(n-m)!}{4\pi(n+m)!}} P_n^m(\cos\theta) e^{im\varphi} \tag{6}$$

where P are the Legendre functions for order n and degree m. In order to solve for the pressure on the sphere, the spherical harmonic expansion must be calculated for plane waves incident on a rigid sphere:

$$p_{0nm} = \mathrm{B_n}(kr) Y_n^{m*}(\Omega_0) \tag{7}$$

where $\mathrm{B_n}$ is the radial function defined below:

$$\mathrm{B_n}(kr) = 4\pi i^n \left(j_n(kr) - \frac{j_n'(kr_0)}{h_n'(kr_0)} h_n(kr) \right) \tag{8}$$

where k is the wave number r_0 is the radius of the sphere and r is the radius of the measurement point (in this case, $r_0 = r$), j_n are spherical Bessel functions and h_n are spherical Hankel functions.

Once the spherical harmonic expansion is achieved, the spherical harmonic components are weighted and summed to form a beam pattern facing a specific look direction [6][7]. This beam is swept around the soundfield to form a composite map of pressure arriving from all look directions. The linear system used to generate these beams is shown below:

$$\mathrm{AW} = c_N \mathrm{B}_N \qquad (9)$$
$$\mathrm{d}^T\mathrm{W} = 1$$

where A is the matrix of spherical harmonic expansions, c_N is the order coefficient, B_N is the harmonic weighting matrix, and W is the beamforming weights to be applied to the microphone array signals such that the product of W and a unit plane wave incident from the look direction (d^T) is equal to one. Because the system is over-determined (there are more microphone locations than spherical harmonics), the solution to this system is found using the Least-Squares method.

The spherical harmonics used in this microphone array are 0th, 1st, and 2nd-order harmonics. The number of harmonics in an order is given as $(N+1)^2$, which means that for 2nd-order, there are a total of 9 harmonics (one 0th-order monopole, three 1st-order dipoles, and five 2nd-order quadrupoles). These are shown in Figure 1:

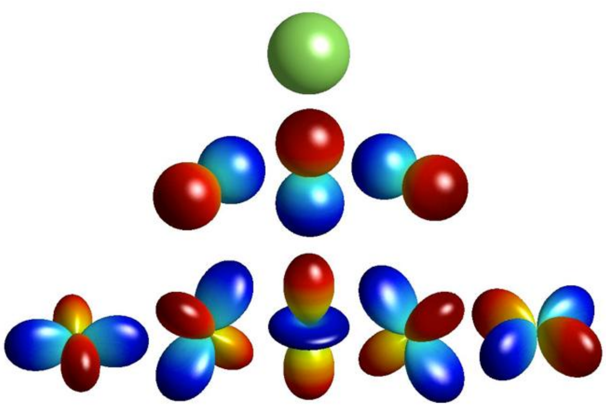

Figure 1: Spherical Harmonics for 2nd-Order Ambisonics

3 METHODOLOGY

The microphone was designed using J. Fliege's 16-node spacing (with appropriate weights for integration over the sphere) based on a facility dispersion method, providing a nearly-uniform weighting [4]. This method was chosen because of its efficiency: less than $2(N+1)^2$ capsules are required as opposed to $2(N+1)^2$ required by Gaussian spacing and $4(N+1)^2$ required by Equiangular spacing [9]. An image of the microphone is shown in Figure 2:

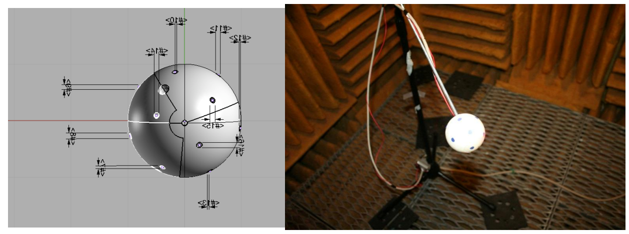

Figure 2: Microphone Design and Construction

The radius of the sphere was chosen to provide an aliasing upper limit frequency of 4.4 kHz, based on the equation $kr<N$, where k is the wavenumber of the upper limit frequency and r is the radius. for $N=2$, the sphere radius is 2.5 cm. In practice, the microphone actually provided up to 8 kHz of accurate beams. The sphere was then printed using a 3D Rapid-Prototype machine and 16 omnidirectional Panasonic capsules were soldered into place. The system was connected via National Instrument cable sampling at 62.5 kHz to a custom preamplifier/interface that connected to MaxSens, custom software designed by Dr. Ning Xiang.

The microphone capsules were first calibrated individually (outside of the sphere) to determine the differences in frequency response. Figure 3 shows the normalized response (average absolute gain differences were less than 2 dB).

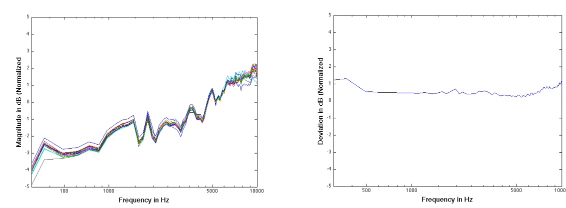

Figure 3: Normalized Frequency Response (Left) and Deviation (Right)

The microphone was then tested in its final construction. The source and receiver were set up 2 m apart in order to achieve far-field pressure down to 250 Hz. The nearest surface were also 2 m away and could therefore be windowed out, preserving the frequency content down to 250 Hz. The microphone was turned in 15-degree increments and impulse responses were captured using a directional Yamaha self-powered loudspeaker, and the resulting beampattern was swept around the azimuthal plane to generate a polar plot, shown in Figure 4. This plot shows the resulting patterns for each microphone direction increment, and it is clear from this graph that the average beam width is 70º (measured to -3dB at each side of the lobe). A first-order plot was also generated, and it was determined that the beam width was 100º. Both of these widths match theoretical data for this microphone array.

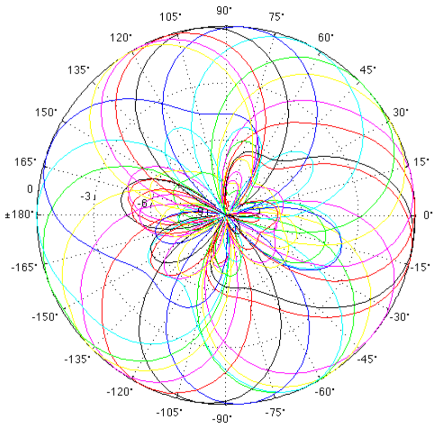

Figure 4: Beamforming analysis of microphone in the far-field (magnitude in dB)

This microphone was then taken to 10 halls and impulse responses were measured using a composite omnidirectional source consisting of one subwoofer, one dodecahedron and one miniature dodecahedron with crossovers between the three sources. Five onstage impulse responses were taken in different positions with both the spherical array and an omnidirectional Earthworks microphone. The halls measured are as follows:

1.Picotte Recital Hall, College of St. Rose, 400 Seats
2-3.SUNY Albany, Theater (500 seats) and Recital Hall (242 seats)
4.Richard B. Fisher Arts Center, Bard College (900 Seats)
5.Belle Skinner Hall, Vassar (325 Seats)
6-7.EMPAC Theater, RPI (400 Seats) and concert hall (1200 seats
8.Zankel Music Center, Skidmore College (600 Seats)
9.Lippes Concert Hall, SUNY Buffalo (670 Seats)
10.Kodak Hall, Eastman School of Music (2300 Seats)

4 RESULTS

Two halls are shown here in detail, Zankel Hall at Skidmore and Picotte Hall at The College of St. Rose. A CATT-Acoustic model was made of the stage enclosure in each case and individual reflections were compared to the magnitude mapping of a beam swept in 4.5º increments around azimuth and elevation. The mapping has been flattened out so the microphone is at 0º azimuth and 0º elevation and the look direction wraps around behind to both the left and right of the map. The abcissa of the map runs from floor at the bottom and ceiling at the top. The level is shown in dB with the dark red as the maximum level and the dark blue as the minimum level. Figures 5-9 show five reflections comparing measured and simulated data for St. Rose and Figures 10-14 show five reflection comparisons for Skidmore.

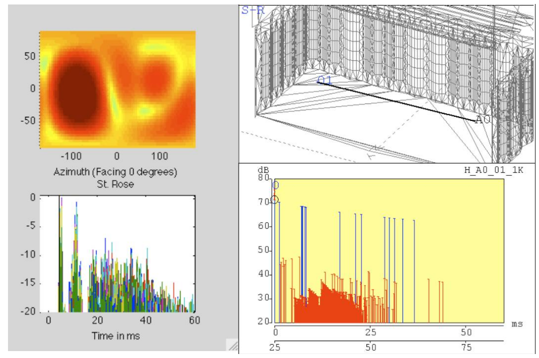

Figure 5: St. Rose direct sound

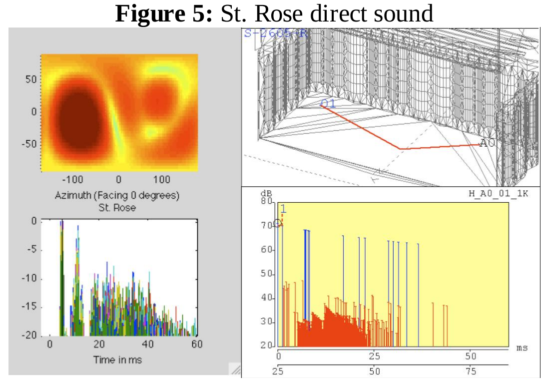

Figure 6: St. Rose floor reflection

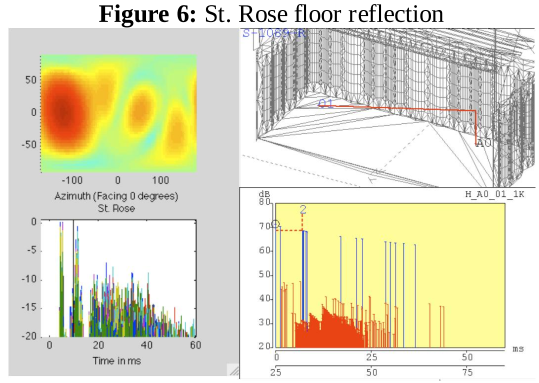

Figure 7: St. Rose rear-wall reflection

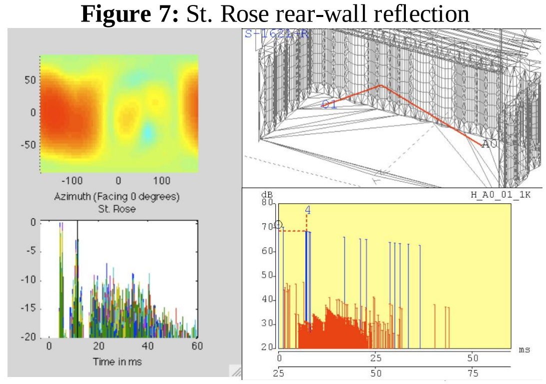

Figure 8: St. Rose rear-wall reflection

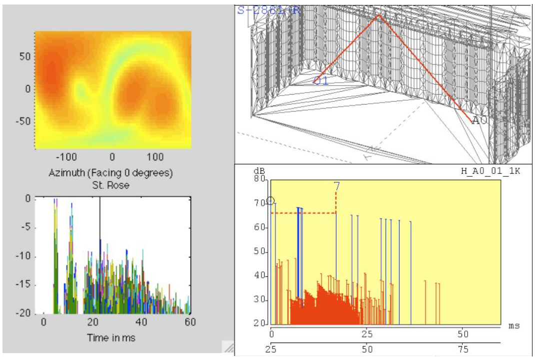

Figure 9: St. Rose ceiling reflection

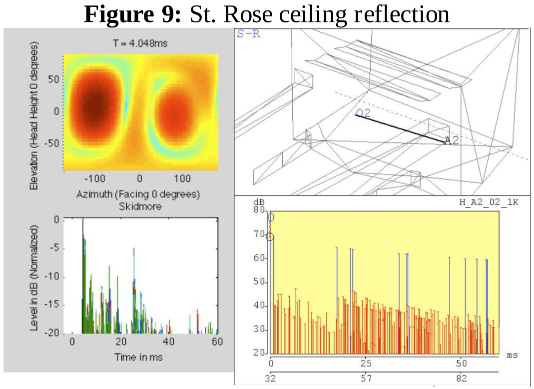

Figure 10: Skidmore direct sound

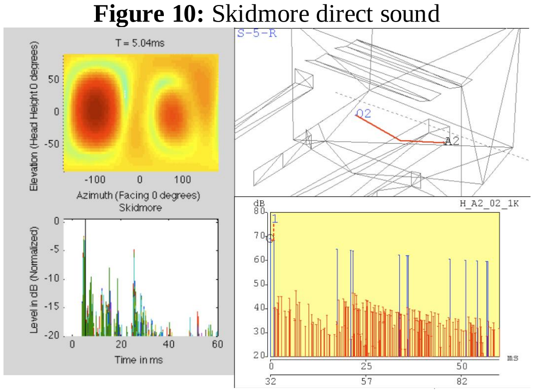

Figure 11: Skidmore floor reflection

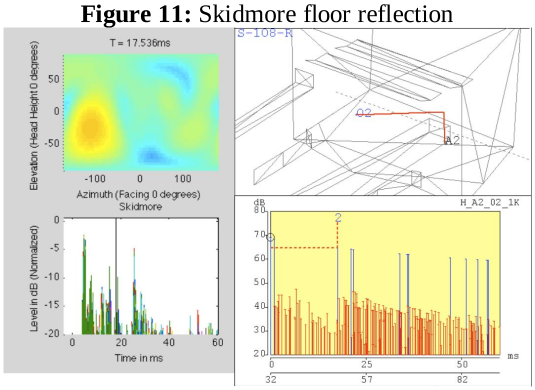

Figure 12: Skidmore rear-wall reflection

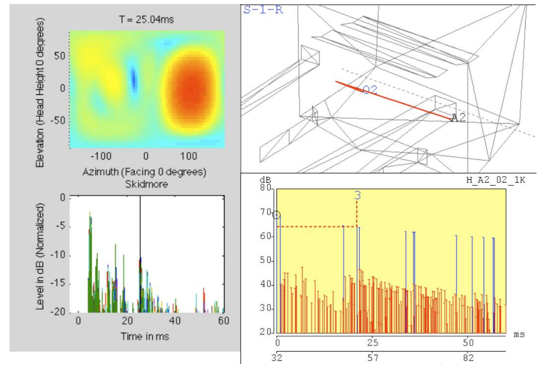

Figure 13: Skidmore side-wall reflection

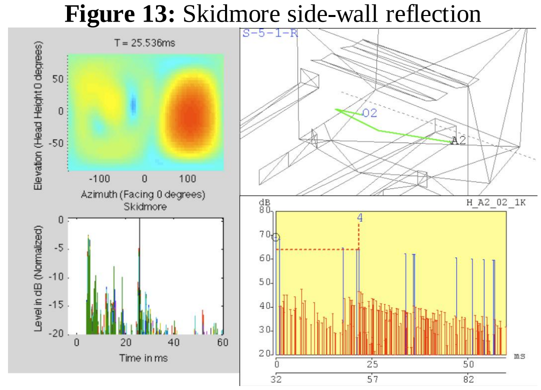

Figure 14: Skidmore side-wall reflection

5 DISCUSSION AND FUTURE WORK

The omnidirectional parameters and the one spatial parameter are shown in Table 3 for the two halls.

Parameter	Skidmore	St. Rose
RT in s	1.96	1.68
V in m3	7000	1140
C80 in dB	0.54	0.96
ST1 in dB	-12.5	-9.5
G in dB	8.9	15.9
EDT in s	2.13	1.68
H/W	0.50	0.52

Table 3: Comparison of Stage Acoustic Parameters

As shown in Table 3, the stages are quite different when examined from an omnidirectional standpoint. Based on these parameters, St. Rose would seem to be better for musicians hearing eachother but worse for blending and loudness [1][2][5]. However, an examination of the H/W parameter shows that the stages are nearly the same, indicating that hearing each other will be the same in both conditions [3]. The spatial analysis of each hall shows that there is a much wider spatial distribution of early energy in the St. Rose enclosure. Additional comparisons of Skidmore with and without an orchestra shell show much greater early energy from above and below the musician in the case of the orchestra shell. This may be beneficial or detrimental to the musicians, depending on the musical cue. Subjective preference tests with musicians could help confirm the value of this information.

Future work in this research would include real-time auralizations with performing musicians in the Arup SoundLab, a sphere of 12 loudspeakers designed to decode 2^{nd}-order ambisonics. The spherical harmonic encoding in this case would utilize real harmonics with Furse-Malham weightings [8]. Multi-dimensional scaling with preference tests would yield the relevance of spatial information for musicians performing onstage with and without other players.

6 CONCLUSIONS

A spherical microphone array has been designed, constructed, and calibrated to measure and auralize 2^{nd}-order ambisonics. This array has been validated in real stage enclosures in comparison to CATT-Acoustic image source models and is shown to provide not only the same but also additional spatial information about the reflections in the stage enclosure. The goal of this work is to determine spatial parameters relevant to the ease and quality of performance for musicians playing alone and together onstage.

REFERENCES

[1] A. C. Gade. “Investigations of musicians room acoustic conditions in concert halls .1. methods and laboratory experiments.” *Acustica*, 69(5):193–203, 1989.

[2] A. C. Gade. “Investigations of musicians room acoustic conditions in concert halls .2. field experiments and synthesis of results.” *Acustica*, 69(6):249–262, 1989.

[3] J. Dammerud and M. Barron. *Stage Acoustics for Symphony Orchestras in Concert Halls*. Doctor of philosophy, University of Bath, September 2009.

[4] J. Fliege. *A Two-Stage Approach for Computing Cubature Formulae for the Sphere*. Report, Fachbereich Mathematik, Dortmund.

[5] Y. Jurkiewicz. *Investigation of Musicians' Conditions and Acoustical Environments on Orchestra Platforms*. Master's Thesis in collaboration with Arup NY, École Centrale de Lyon, 2005.

[6] Z. Li and R.Duraiswami. “Flexible and Optimal Design of Spherical Microphone Arrays for Beamforming.” *Audio, Speech, and Language Processing, IEEE Transactions on*, 15(2):702-714, 2007.

[7] J. Meyer, et al. “Spherical microphone array for spatial sound recording.” *115th Convention of the Audio Engineering Society*, New York, 2003.

[8] S. Moreau, et al. “3d sound field recording with higher order ambisonics ob jective measurements and validation of a 4th order spherical microphone.” *120th Convention of the Audio Engineering Society*, Paris, 2006.

[9] B. Rafaely. “Analysis and design of spherical microphone arrays.” *Speech and Audio Processing, IEEE Transactions on*, 13(1):135–143, 2005.

[10] K. Ueno, et al. “Experimental study on stage acoustics for ensemble performance in chamber music.” *Acoustical Science and Technology*, 26(4):8, 2005.