A MEASUREMENT DATABASE OF US AND EUROPEAN CONCERT HALLS FOR REALISTIC AURALIZATION AND STUDY OF INDIVIDUAL PREFERENCE

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1 INTRODUCTION

Bridging the divide between the subjective and the objective lies at the core of concert hall acoustics. The objective quantification of rooms began with Sabine's reverberation time, quantifying the subjective impression of reverberance.[1] Rooms with acoustical purposes were built long before this development, but the basis of room design was rooted in practice and experience, rather than quantifiable metrics. After Sabine, new objective measurement techniques were applied to room acoustics, but the question of which measured quantities related back to perception was still unclear. In the 1950s and 60s, researchers and consultants began to question if reverberation alone could fully explain a listener's overall impression of a room.[2,3] Beranek's *Music, Acoustics, and Architecture* in 1962 provided 18 unique terms he proposed would be sufficient to describe the perceptual domain of concert hall acoustics.[4] Since Beranek, many attempts to quantify many different perceptions has occurred, but the central question for the individual still remains: "Which room do I like better?"

2 PREVIOUS CONCERT HALL MEASUREMENT TECHNIQUES

The main limitation of studying concert hall preference lies in modifying and controlling room properties, while still ensuring a realistic auralization. Some groups have performed live concert surveys,[5,6] and most notably Beranek conducted interviews of critiques and conductors about their experiences in each hall.[7] These methods are inherently realistic, but the variables of performer ability, music genre, visual aesthetic, and many other factors are uncontrolled. It is also difficult to compare subtle differences in perceptual judgements during two separate performances. One way to overcome these limitations is to use auralizations of hall environments. The first studies to incorporate these methods were conducted in the 1970s and used binaural techniques. By electroacoustically reproducing binaural recordings, hall differences alone could be compared side-by-side. Yamaguchi took recordings in Yamaha Music Hall with a single omnidirectional loudspeaker playing a passage of music, and recordings were made in ten different seats.[8] Recordings were made with a pair of omnidirectional microphones spaced 50 cm apart without a physical head model. The realism associated with this binaural measurement technique could call into question the validity of this study.

An improved measurement-based technique was implemented by three teams out of Göttingen, Tokyo, and Berlin. All three groups used binaural techniques with dummy heads and made recordings in a series of concert halls. Schroeder's group from Göttingen made recordings in 22 halls in Europe and reproduced them using cross-talk cancellation.[9] Kimura and Sekiguchi also conducted listening tests with dummy head recordings using an omnidirectional sound source on stage.[10] Their recordings were captured in 13 multipurpose auditoriums. These studies provided directly comparable auralizations, but the realism of the source(s) on stage were quite limited. Finally, work by Plenge et al.[11] and Wilkens[12] included recordings of the same live orchestra with dummy heads in six different halls. This method provided a realistic sound source but introduced non-controlled factors for differences in the orchestra's performance. For all of these studies, it is now known that using non-individualized or average HRTFs can cause localization errors, and this mismatch could also impact the realism of auralizations using a dummy head.[13]

In terms of large measurement efforts to create accurate auralizations of halls, a hiatus occurred from the core 1970s research until the work by Lokki's team out of Aalto University.[14,15] This team created an orchestra consisting of commercially available loudspeakers selected to have directional patterns similar to orchestral instruments.[14] Using this loudspeaker orchestra and a six-microphone intensity probe, room impulse responses (RIRs) were measured and processed using the Spatial Decomposition Method (SDM).[15] Since a six-microphone array was used, with a maximum capsule spacing of 100 mm, time delay of arrival (TDOA) methods identified the location of early reflections in the room. In each time segment, all the energy in the RIR was assigned to the auralization array's loudspeaker that was closest to the direction estimated by the TODA calculation. After a given cutoff time, energy in each time segment of the RIR was assigned randomly to one loudspeaker. This assumption created a diffuse reverberant tail, as individual reflections were not separable in time.

After a given time following the direct sound, room reflections overlap in time, and TDOA methods alone cannot separate one reflection from another. This inability to spatially resolve two reflections that overlap in time may result in two possible limitations. First, if two discrete early reflections overlap in time, a TDOA-based algorithm will place the energy from both reflections at a loudspeaker located between the two reflections. This limitation may result in a large perceptual error, since early reflections are important for spatial impression. Second, at some time following the direct sound, room reflections occur at an exponentially increasing rate and will eventually overlap in time. The spatial character of later arriving room energy in this method will suffer in spatial accuracy and could create an auralization not matching the hall's real acoustic experience. An additional limitation lies in the exclusion of any standard measurement loudspeakers. No RIRs were measured using an omnidirectional loudspeaker, so metric calculations can only be done using non-standardized methods, not in accordance with ISO 3382.[16] These effects can cause uncertainty in certain metrics.

3 ROOM MEASUREMENT AND PROCESSING METHODS

The goal of the present work was to create a comprehensive concert hall measurement database with two aims, both objective and subjective in nature. First, an objective measurement procedure was needed so that a wide variety of halls could be objectively compared using techniques that comply with and further advance existing standards specified by ISO 3382.[16] These measurements were taken using a custom-built three-part omnidirectional sound source and a 32-element spherical microphone array. Although repeatable and reproducible, these measurements are not realistic to a performance condition with an orchestra on stage. The second goal was to provide realistic, full-orchestral auralizations using a compact spherical loudspeaker array. This compact array is capable of recreating different directional radiation patterns, matching that of different orchestral instruments. Section 3.1 discusses the objectively-motivated measurements obtained with the omnidirectional source, section 3.2 describes the subjectively motivated measurements obtained with the compact spherical loudspeaker array, and section 3.3 overviews the database hall selection procedure.

3.1 Objectively-motivated Measurements for Sound Field Analysis

In order to objectively compare different rooms, a standard measurement procedure was implemented in each hall. The technique used is also repeatable and reproducible by different consultants or researchers. To connect perceptual hall attributes to objective calculations, care must be taken to ensure that measurements only contain differences due to the changing hall, and not changes in measurement setup. For this reason, a single measurement setup was created, and the same equipment was transported to each hall. A standard measurement grid, shown in Fig. 1, was created for comparable receiver locations. Receivers were placed first in four central hall seats closest to a source-receiver distance of 10, 15, 20, and 25 meters. The source was placed one meter from the front of the stage, at roughly the location of the orchestral conductor. Three additional mostly standard receivers were placed in the same seating row as receivers 2 through 4 but placed 5 meters from hall center. Finally, time permitting, up to nine additional receivers were placed throughout the hall to better sample unique locations pertaining to each individual hall's design.

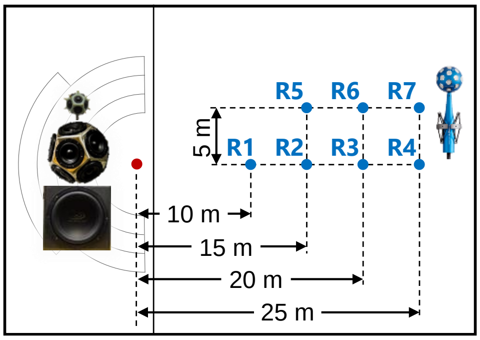

Figure 1: A standardized measurement grid for omnidirectional source measurements. Four central receivers, R1-R4, and three off-axis receivers, R5-R7, were included in the setup.

3.1.1 Loudspeaker, Microphone, and Measurement Setup

For the omnidirectional impulse responses, a three-part omnidirectional sound source was used to ensure good signal to noise ratio (SNR) and omnidirectional performance over a broadband range. The three loudspeakers for measuring low, mid, and high frequency RIRs are shown in Fig. 2. The low-frequency source was constructed using two 25-cm drivers designed to have a high linear excursion. The mid-frequency dodecahedron loudspeaker was constructed using twelve 10-cm drivers inside a welded steel enclosure. The high frequency dodecahedron loudspeaker was constructed with twelve 1.9-cm tweeters, designed to be as compact as possible in a 3D printed enclosure. Measurements were recorded over the entire audible range up to 20 kHz, and omnidirectional source radiation was achieved up to the 5 kHz one-third octave band.[17]

(a) (b) (c)

Figure 2: The three-part omnidirectional sound source using the objective database measurements. The source consists of a low-frequency loudspeaker (a) with two 25-cm (10-inch) drivers (30 cm x 30 cm x 30 cm), a mid-frequency dodecahedron (b) with 10-cm (4-inch) drivers (25 cm face-to-face diameter), and a high-frequency dodecahedron (c) with 1.9-cm (3/4-inch) tweeters (6 cm diameter).

Three separate measurements were taken at each seat location, one with each of the loudspeakers placed at the same position and height. This procedure reduced phase and scattering effects from non-collocated source positioning and self-radiation off of the other two loudspeaker enclosures when stacked. The three individual measurements were diffuse-field equalized, gain adjusted for sensitivity and amplifier gain differences, and combined using a set of three linear-phase crossover filters designed to sum flat with crossover frequencies at 100 and 1300 Hz. Measured RIRs exhibited omnidirectional source radiation up to 5 kHz and good SNR down to a lower limit around 40 Hz.

Measurement signals were generated and RIRs were processed in MATLAB, and measurement playback and recording was performed in a custom program developed in the Max 7 environment.[18] For the measurements, 5-second logarithmic sine sweeps were used with one pre-send signal and eight time-domain averages, increasing SNR. Logarithmic sine sweeps, as opposed to linear sweeps, were used to minimize harmonic distortion effects, and amplifier gains were limited to levels of very low and inaudible harmonic distortion. Measurements were made using a 32-element spherical microphone array, an Eigenmike em32, which is a commercially available product, shown in Fig. 4. At each receiver, 32 simultaneous RIRs were measured by each microphone element.

3.1.2 Spherical Array Beamforming Techniques

Using state-of-the-art spherical array processing techniques, high-resolution spatial analysis of a sound field is possible. In the same manor that a single-diaphragm microphone discretely samples the pressure of a sound field in time, a spherical microphone array discretely samples pressure in space. Using this spherical sampling of microphones, a sound field can be represented by estimating the set of spherical harmonic (SH) functions discretely in frequency, shown in Fig. 3. The benefit of representing a sound field using SH functions is found in the spherical Fourier series in Eqn. 1:[19]

$$p(\phi,\theta) = \sum_{n=0}^{\infty} \sum_{m=-n}^{n} P_{nm} Y_n^m(\phi,\theta). \qquad 1$$

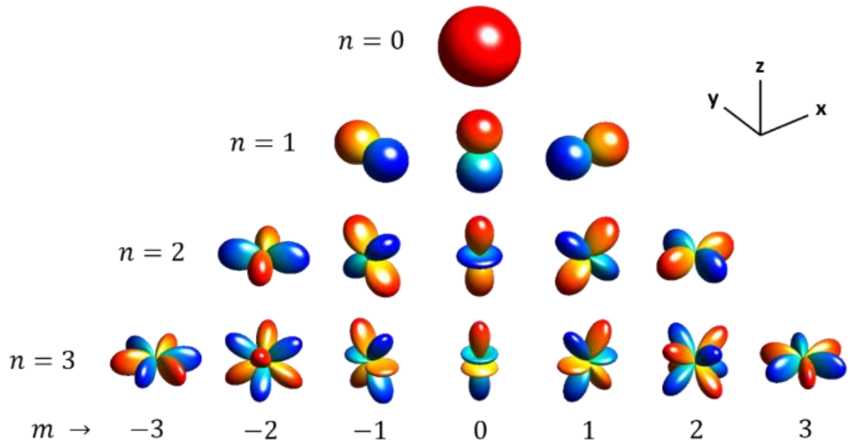

Figure 3: A plot of the real-valued spherical harmonics functions, defined by their order, n, and their degree, m. The set of functions shown is truncated at order $n = 3$.

Practically, the spherical Fourier series states that any spatially bounded functions, $p(\phi,\theta)$, can be represented as an infinite sum of the set of SH basis functions, $Y_n^m(\phi,\theta)$, using the proper weights, P_{nm}. These weights can be calculated using the spherical Fourier transform in Eqn. 2:[19]

$$P_{nm} = \int_0^{2\pi} \int_0^{\pi} p(\phi,\theta) [Y_n^m(\phi,\theta)]^* \sin\theta \, d\theta d\phi. \qquad 2$$

Much of the past RIR measurements in the field have been done using a single, diffuse-field microphone or a figure-of-eight microphone to isolate the lateral dipole response of a sound field. Some work has used a microphone to extract a B-format or first-order Ambisonic RIR, but this resolution only provides limited spatial analysis, sampling only the first-order SH components. With higher-order arrays providing greater detail in the SH domain, beamforming techniques can be used to create spatial maps of the energy in a RIR. This process can identify early reflections or generate maps of the average energy in specific time and frequency ranges using plane-wave decomposition (PWD), outlined in Fig. 4. The SOFiA toolbox in MATLAB was utilized for PWD analysis.[20]

First, the microphone RIRs are converted to a SH-RIR by the spherical Fourier transform. Next, radial filtering compensates for the physical effect of the array's size and the presence of a rigid sphere in the sound field. Then, PWD is used to create directional RIRs for the room, weighting and summing each SH component of the sound field so that the array has a beam-shaped or truncated SH-order plane wave directional response. This beam directivity pattern is steered to all points around a sphere, creating a set of directional RIRs oriented around the sphere. Finally, each directional RIR is time windowed and band-pass filtered to isolate specific energy regions, and the energy in each directional RIR is summed to create a 2D spatial grid. This type of high-resolution spatial analysis unlocks new potential in analysis of room sound fields in the time, frequency, and spatial dimensions.

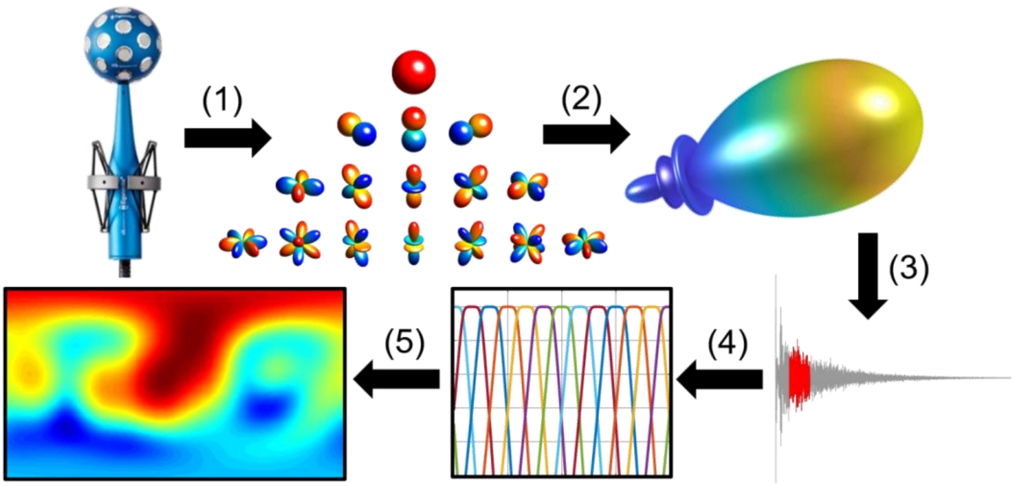

Figure 4: A process diagram showing how spatial energy maps are generated. Starting from a 32-channel microphone RIR, steps include: (1) a spherical Fourier transform and radial filtering, (2) applying plane wave SH weights & rotation, (3) RIR time windowing, (4) band-pass filtering, and (5) final energy summation into a spatial energy map.

3.2 Subjectively-motivated Measurements for Hall Auralization

The measurement procedure described in section 3.1 provides a high spatial resolution, repeatable technique that can produce omnidirectional RIRs over a broad frequency range, but these measurements are not realistic to an orchestral concert. An orchestra consists of a distributed array of unique sound sources, all with frequency-dependent radiation patterns. To achieve realistic source stage layout and directional radiation patterns, a compact spherical loudspeaker array was constructed, pictured in Fig. 5. With individual control of all 20 drivers in the source, it can be processed to mimic the directional radiation pattern of different orchestral instruments. The array contains twenty 40-mm drivers with a 200 Hz resonance and high linear excursion to maximize low-frequency SNR. This number of drivers enables up to third-order SH representation of instrument directivities. Both the compact size of the array, combined with the good low-frequency performance enables a directionally accurate response with good SNR over a broad frequency range.

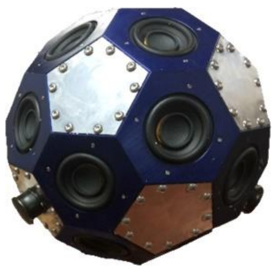

Figure 5: The compact spherical loudspeaker array for instrument directivity reconstruction, which consists of twenty 40-mm drivers and has an overall diameter of 15 cm (6 inches).

A set of 20 filters corresponding to each driver in the array was created for each orchestral instrument. A database of the spherical harmonic representation of 41 unique orchestral instruments was used as the basis for the filter design.[21] In practice, the array was positioned at a specific location on stage, and a logarithmic sine sweep measurement signal was processed with a bank of 20 instrument-specific driver filters, played back over the array, and recorded with the microphone array. This technique built-in the directivity of the instrument into the measured RIR. While taking individual RIR measurements of each driver in the compact array would allow for flexible post-processing of the source directivity, the measurement time would also increase by a factor of 20.

To test the performance of this array, a complete set of directivity measurements were taken in an anechoic chamber. For the measurements, the directivity was measured for each driver element separately, so the results could be filtered to demonstrate the array's performance for each instrument. Fig. 6 shows the array's performance in reproducing radiation patterns for a trumpet at three discrete frequencies. In general, the array works well up to a spatial aliasing limit of 3 – 4 kHz.

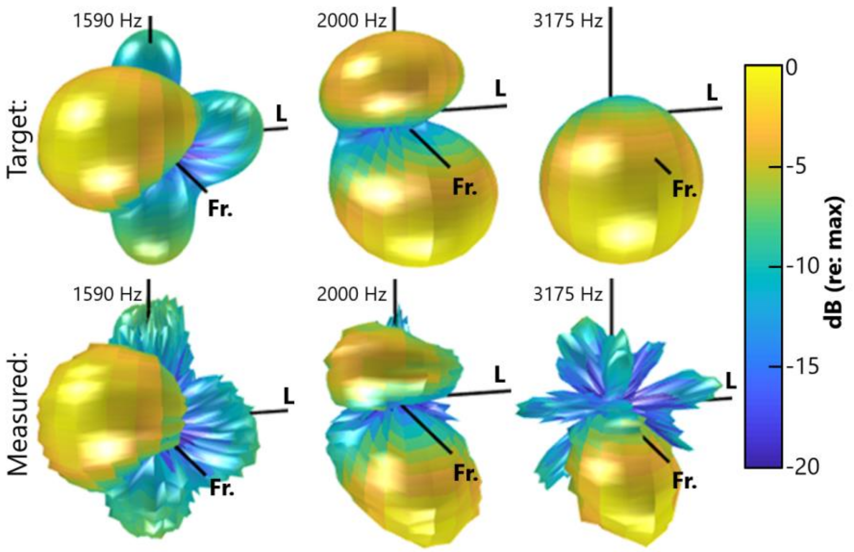

Figure 6: Example radiation pattern reconstructions for a trumpet at three discrete frequencies. The upper plots are the target responses and the lower plots are the measured reconstruction patterns using the compact spherical array. Patterns are well reconstructed up to around 3 – 4 kHz.

The source was positioned around the stage, and the measurement was repeated 20 times at different orchestra instrument locations. This technique was used in each hall, and a full set of orchestral RIRs was recorded at receiver R2, shown in Fig. 7a. The standard orchestral layout that was developed for source positioning is shown in Fig. 7b. In practice, the directional source could not achieve sufficient SNR below the resonance of the drivers, so an additional measurement with the omnidirectional source's subwoofer was taken at each location to extend the measurement range down to around 40 Hz. With 20 positions and two sources, the 40 measurements took a total of 1.5 to 2 hours to complete for a single receiver. Due to this time constraint, the full-orchestral set of RIRs was only taken at R2, and the omnidirectional source measurements sampled other seat locations. With the complete set of full-orchestral measurements, individual auralizations for each instrument can be virtually recreated in the AURAS facility, a 30-loudspeaker virtual acoustic facility at Penn State.[22] All 20 sources are superimposed to create a repeatable full-orchestral auralization.

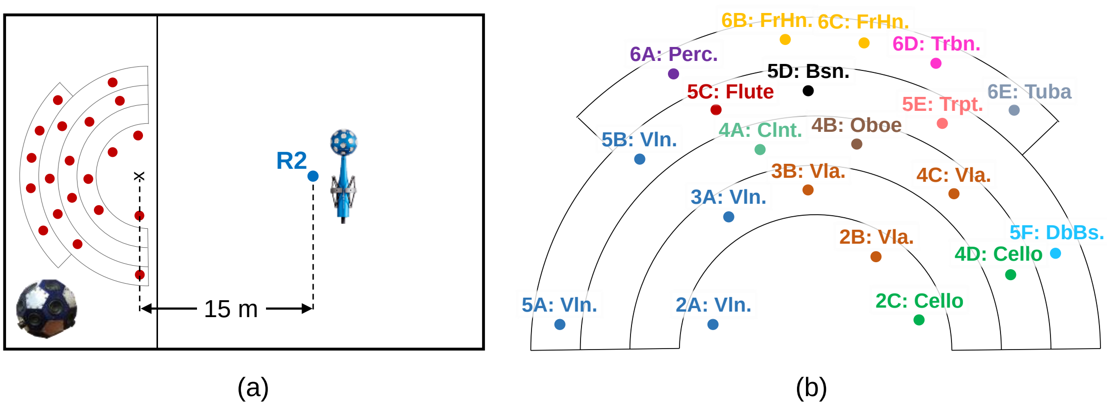

Figure 7: Measurement setup using the compact spherical loudspeaker array. This measurement was taken at (a) receiver R2, 15 meters from stage, for (b) 20 source locations.

3.3 Hall Selection Procedure

To draw broad conclusions about concert hall acoustics, it is highly important to include as wide a variety as possible in the database. For the current work, variety was desired in hall size, shape, RT, perceived reputation, and time period of construction. The hall selection began with an online survey that was distributed to consultants and researchers around the world, primarily located in the US and Europe. From the survey, over 70 responders provided 140 unique room suggestions. The list was initially reduced by only including halls with a rough minimum of 1200-1300 seats. The seat count was limited because many smaller halls do not have stages that are large enough to accommodate the orchestral setup shown in Fig. 7b, and the project was focused on rooms designed to accommodate full-orchestral symphonic performance. In addition, halls were limited to the US and Europe because these locations provided a large variety, while maintaining reasonable travel and budget considerations. These criteria reduced the list to a subset of 70 halls.

A final reduction resulted in a list of 40 halls, maintaining hall variety while considering geographic restraints for economical travel costs. Access was requested to around 30 halls, and measurements were taken in 21 different halls. Out of these halls, 16 were located in the US and 6 were located in Europe. The measurements took roughly six to eight hours in each hall. The current database includes 6 modern shoebox halls, 4 historic shoebox halls, 4 fan halls, 4 vineyard halls, and 3 halls of non-standard shapes. The seat counts range from 1200 to 2900. In two of the US halls (both modern shoebox halls), significant variable acoustic elements allowed for large adjustment in the halls' acoustic properties, and the complete hall measurements were taken for two settings. This increased the count to 23 unique room environments.

4 RESULTS AND DISCUSSION

4.1 Room Acoustic Parameter Calculations

Room acoustic parameters can be calculated from the omnidirectional microphone response of the spherical microphone array.[23] For the 21-hall database, Fig. 8 shows the wide variety in average mid-frequency unoccupied T30 of all 23 unique room environments (also averaged across receiver). T30 values range from 1.56 to 3.28 s, which covers a large, realistic range for most concert halls.[24] As well, variety is seen within each individual hall shape. In the entire database, 242 individual omnidirectional source RIRs were measured. This includes measurements at all combinations of concert hall, hall setting, and receiver location. For the parameters of EDT and C80, which can have large variation between seats in the same hall, a histogram of the mid-frequency average parameter values are shown in Fig. 9a and 9b for all 242 RIRs. An EDT range of 1.19 to 3.42 s and a C80 range of -6.6 to 2.8 dB was found.

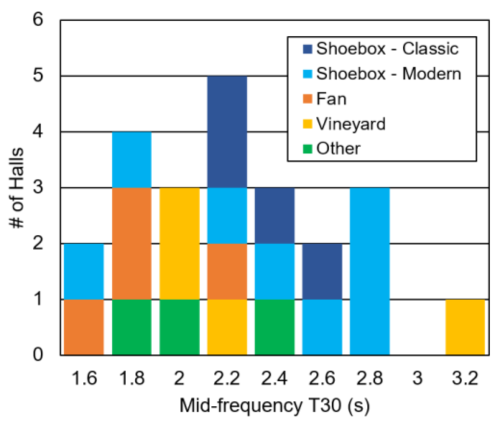

Figure 8: Reverberation time (T30) calculations shown across the 21 hall (23 room environments) database. For each room environment, one mid-frequency (500 – 1000 Hz) average unoccupied reverberation time of all measured receivers is shown.

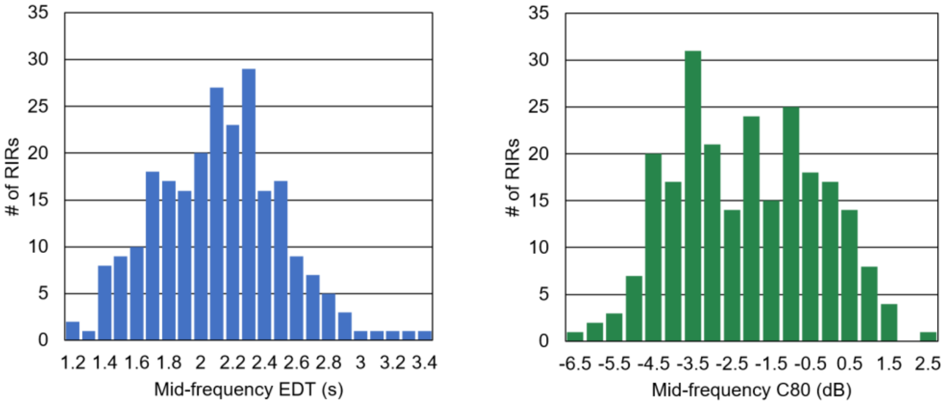

Figure 9: Histograms of the (a) mid-frequency EDT and (b) C80 averages for all 242 RIRs. For each parameter, a wide variety is found in the measurement database.

4.2 Spherical Array Beamforming Results

Using the PWD techniques described in Section 3.1.2, spatial energy maps were generated to show the energy distribution in fine-scale time sections of the RIR. With this level of detail, numerous different maps can be created, analyzing many combinations of frequency and time ranges. As well, these results could then be compared across all 23 different room environments. To demonstrate the potential of PWD in room acoustics, two different halls were selected for comparison. The first hall is fan-shaped, with a 2750 – 3000 seating capacity. The second hall is shoebox-shaped, with a 1750 – 2000 seating capacity. For both halls, PWD was performed on the RIRs in 1 ms time segments, for each octave band. For this study, the results shown are a summation of the energy in the 1000 - 4000 Hz octave bands. An example of a PWD spatial energy map is provided in Fig. 10 for visual interpretation of this style diagram.

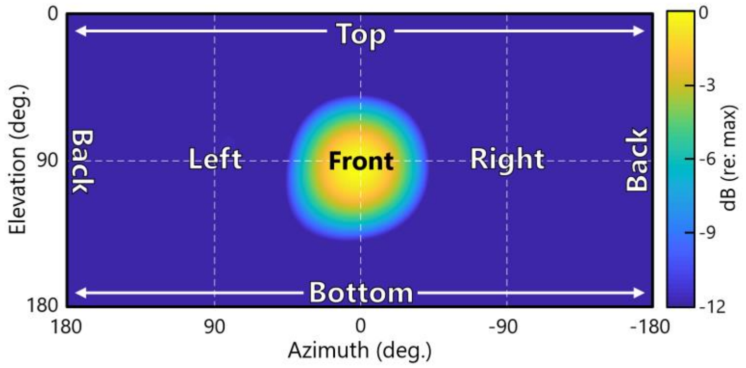

Figure 10: A sample spatial energy map resulting from the PWD of the direct sound. The sound field has been unwrapped over a rectangular grid. Azimuth is the clockwise-positive horizontal angle, and elevation is the vertical angle from the upward z axis.

For each room, the receiver R2 (at 15 m) was selected for comparison. The time-domain omnidirectional RIR was analyzed to identify locations of key early reflections in 1-2 ms time windows. Then, for each of these time windows, a spatial energy map was generated to identify the direction of arrival of the reflection, indicating a likely surface of origin. In both Fig. 11 and Fig. 12, the upper four plots show specific arrivals of the direct sound (a) and three early reflections (b)-(d). Each reflection is labeled with a suggested surface of origin. Finally, the fifth spatial energy map (e) shows a PWD of the reverberant energy, from 100 ms to 400 ms. This plot demonstrates the spatial character of the later sound field, which is thought to contribute to perceptual attributes such as listener envelopment.[24,25] Beamforming-type analyses are useful in analyzing individual reflections in the early sound field and larger spatial energy averages in the later sound field.

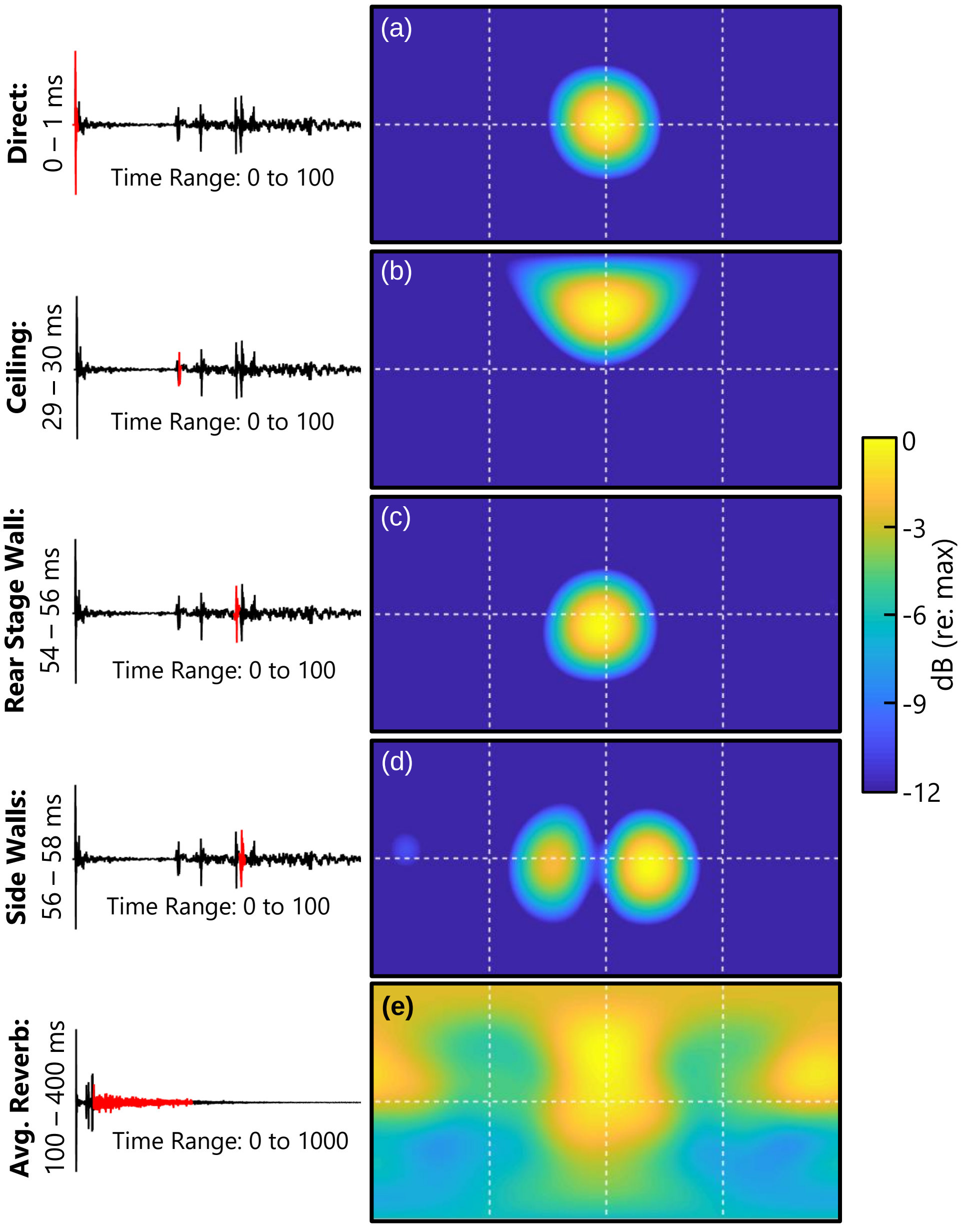

Figure 11: These plots show PWD spatial energy maps for five different time regions in a fan-shaped concert hall with 2750 – 3000 seats. The time-domain omnidirectional RIR is shown on the left side, highlighting the analysis time range in red, and the spatial energy map is shown on the right. The direct sound is shown in (a), three significant 1-2 ms regions containing early reflections in (b)-(d), and the average reverberant energy from 100 to 400 ms is shown in (e). Due to its fan shape, the side wall reflections come from a more frontal angle (~ 33º), and the reverberant energy does not include as much energy arriving from the lateral directions.

***Note:** The ceiling reflection's spread is due to the rectangular unwrapping of a spherical function.*

Highly interpretable results, directly linked to the two room geometries, emerge from comparing the early reflections and reverberation in the fan- and shoebox-shaped halls in Fig. 11 and 12, respectively. Both receivers are located 15 meters from the source position, but the ordering and timing of the reflections is unique to each room. In the fan-shaped hall, which is larger and much wider than the shoebox hall, the first arriving reflections come from the ceiling, followed by a reflection from the rear stage wall, and then the two sidewall reflections (the second ceiling reflection is not shown). In the rectangular hall, which is smaller and narrower with a high ceiling, this order switches. The first

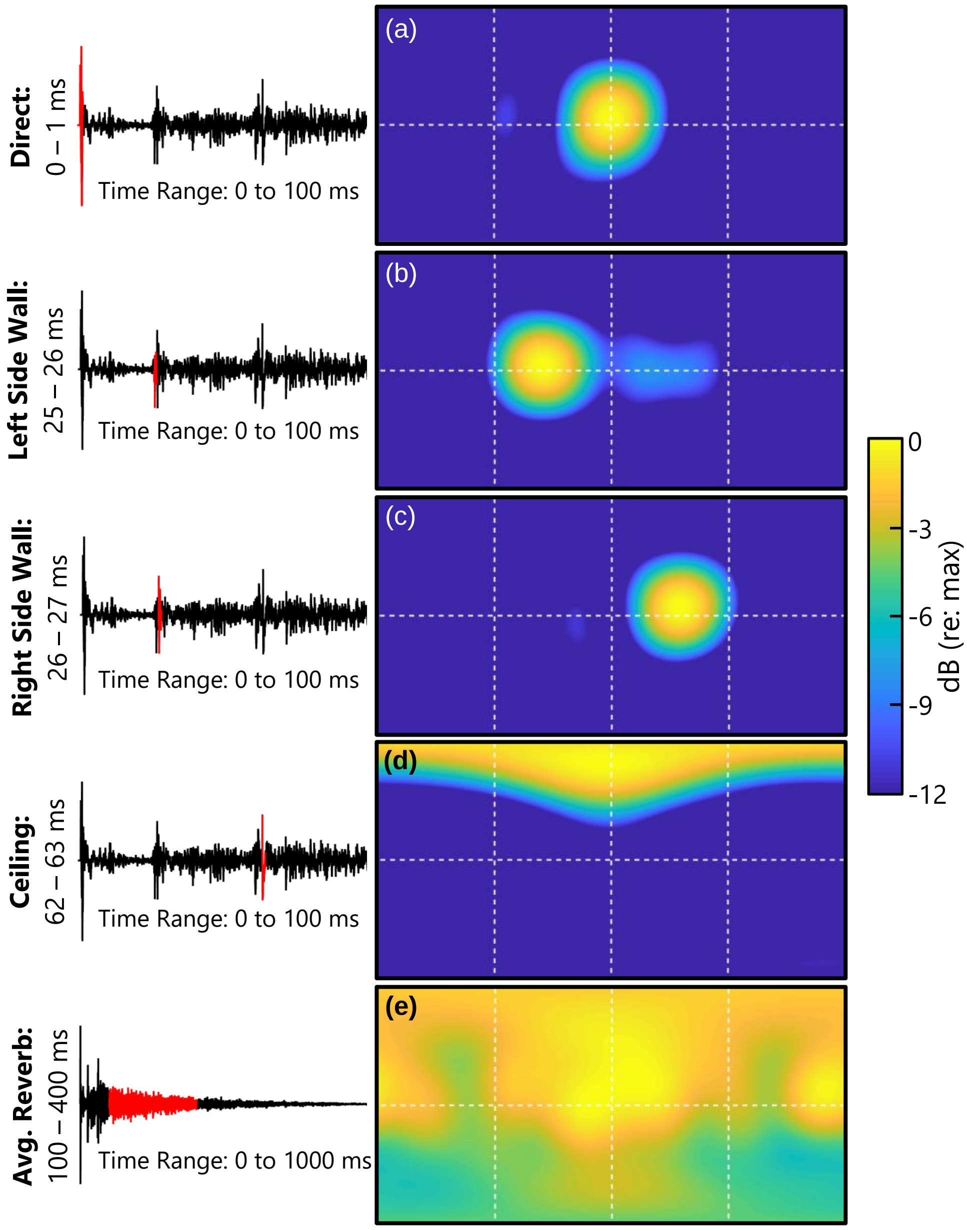

Figure 12: This figure is the same analysis set, technique, and organization from Figure 11, but now the analysis is for a shoebox hall. For this hall, the lateral reflections in (b) and (c) arrive much more quickly than those from the fan-shaped hall, and they arrive at a more lateral direction (~ 54º). The ceiling reflection in (d) arrives at a very steep angle, as the ceiling in this hall is higher and flat, compared to the ceiling in the fan-shaped hall which is angled towards the audience. Finally, the reverberant energy in (e) from 100 to 400 ms contains a much more significant and uniform spatial distribution of energy, adding lateral energy which is important for promoting listener envelopment.

***Note:** The ceiling reflection's spread is due to the rectangular unwrapping of a spherical function.*

reflections to arrive are the strong lateral reflections, which occur at a more lateral angle of 54º, followed by next most prominent reflections coming from the rear stage wall (~47 ms, not pictured) and the ceiling reflection. In the shoebox hall, the ceiling is flat, and higher, to accommodate adequate volume. In the fan-shaped hall, the ceiling is sloped, coming to an angle towards the stage, providing more frontal reflections earlier in time. The later and steeper arrival of the shoebox ceiling reflection (Fig. 12) could also indicate that it is a second order reflection, first reflecting off of the stage floor.

Another important note can be observed regarding the analysis of the side wall reflections from the fan-shaped hall (Fig. 11). By observing the RIR in time, these two reflections were indistinguishable, occurring even within the same window of 1 ms. Traditionally, it would be difficult to know that two reflections were occurring, but with the added spatial analysis using high-resolution spherical array techniques, much more understanding of the existing sound field in the room is possible. In further analysis, these two reflections were found to be still indistinguishable using both first- and second-order PWD, as compared to the plots here shown using third-order PWD. It would be very difficult or even impossible to separate these two reflections with an omnidirectional microphone, TDOA techniques, or lower first-order microphone arrays, which are more common in room acoustics.

Finally, an important overall difference between the two halls is in the spatial energy distribution of the later reverberant energy. For the fan-shaped hall in Fig. 11, the later reverberant energy lacks evenness, especially in the lateral directions. In the shoebox hall, this energy is not spatially uniform or diffuse, but it does include much more energy from the lateral directions, contributing to a higher sense of listener envelopment.[25] Although lateral energy has traditionally been measured using a figure-of-eight microphone, PWD can provide a more complete spatial map of a room's reverberation.

5 CONCLUSIONS AND FUTURE WORK

A concert hall RIR database has been created that includes both objectively-motivated measurements for sound field analysis and subjectively-motivated measurements for realistic auralization. Objective measurements exhibit omnidirectional source radiation over a wide frequency range, and high resolution spatial analysis has been enabled using spherical microphone array processing techniques. As well, measurements made using a compact spherical loudspeaker array enable full-orchestral auralizations that have accurate instrumental source radiation patterns. The compact spherical array can be processed to mimic different orchestral instruments flexibly without requiring many different sound sources. This source was shown to produce directionally accurate radiation patterns up to a spatial aliasing frequency of 3 – 4 kHz.

The results from both standard metric calculations and PWD spatial energy maps show the variety included in the database and the capabilities of spherical array processing techniques. In the database, all 23 room environments have mid-frequency hall average T30s that range from 1.6 to 3.3 s. Including all of the receivers measured in all hall conditions, 242 omnidirectional RIRs were captured. The variety is further shown in the database by looking at the variation of room acoustics parameters. The decay parameter of mid-frequency EDT varies from 1.19 to 3.42 s, and the energy ratio of C80 shows a variation in its mid-frequency average of -6.55 to 2.75 dB. Further, the PWD analyses show intuitive and highly detailed results. This method yields fine time and spatial resolution, 1 ms and 3 degrees, respectively, which can be used to compare different rooms in a perceptually intuitive way. The results from this approach indicate the benefits of spherical array processing in concert hall acoustics. For future work, perceptual rating data will be collected using realistic auralizations from the measurement database. Perceptual ratings will then be correlated to the highly detailed objective analysis in the time, frequency, and spatial domains. These techniques can provide a better understanding of the many subjective impressions in concert hall acoustics, how they inform preference, and how preference varies between individuals.

6 ACKNOWLEDGEMENTS

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7 REFERENCES

1. W. Sabine, “Collected Papers on Acoustics, Caimbridge”, Harvard University Press (1922).
2. T. Somerville, “An Empirical Acoustic Criterion,” Acustica, 3 (6), 365-369 (1953).
3. A. Marshall, “A Note on the Importance of Room Cross-Section in Concert Halls,” J. Sound & Vib., 5 (1), 100-112 (1967).
4. L. Beranek, “Music, Acoustics, and Architecture,” Wiley & Sons (1962).
5. L. Beranek, “Subjective Rank-Orderings of Acoustical Measurements for Fifty-Eight Concert Halls,” Acta Acust. united Ac., 89, 494-508 (2003).
6. R.J. Hawkes and H. Douglas, “Subjective Acoustic Experience in Concert Auditoria,” Acustica, 2, 235-250 (1971).
7. M. Barron, “Subjective Study of British Symphony Concert Halls,” Acustica, 66 (1), 1-14 (1988).
8. K. Yamaguchi, “Multivariate Analysis of Subjective and Physical Measures of Hall Acoustics,” J. Acoust. Soc. Am., 52 (5A), 1271-1279 (1972).
9. M.R. Schroeder, D. Gottlob, and K.F. Siebrasse, “Comparative study of European concert halls: correlation of subjective preference with geometric and acoustic parameters,” J. Acoust. Soc. Am., 56 (4), 1195-1201 (1974).
10. S. Kimura and K. Sekiguchi, “Study on Criteria for Acoustical Design of Rooms by Subjective Evaluation of Room Acoustics,” J. Acoust. Soc. Jpn., 32 (10), 606-614 (1976).
11. G. Plenge, R. Lehmann, R. Wettschureck, and H. Wilkens, “New methods in architectural investigations to evaluate the acoustic qualities of concert halls,” J. Acoust. Soc. Am., 57 (6), 1292-1299 (1975).
12. H. Wilkens, “A Multidimensional Description of Subjective Judgement of Concert-Hall Acoustics,” Acustica, 38, 10-23 (1977).
13. S. Kim and W. Choi, “On the externalization of virtual sound images in headphone reproduction: a Wiener filter approach,” J. Acoust. Soc. Am., 117 (6), 3657-3665 (2005).
14. J. Pätynen, “A virtual symphony orchestra for studies on concert hall acoustics,” Doctoral Dissertation, Aalto University (2011).
15. S. Tervo, J. Patynen, A Kuusinen, and T. Lokki, “Spatial Decomposition Method for Room Impulse Responses,” J. Audio Eng. Soc., 61 (1/2), 17-28 (2013).
16. ISO 3382: 2009, “Acoustics – Measurements of room acoustics parameters – Part 1: Performance Spaces.”
17. D. Dick, “A new metric to predict listener envelopment based on spherical microphone array measurements and higher order Ambisonic reproductions,” Doctoral Dissertation, The Pennsylvania State University (2017).
18. Max 7, Cycling 74 [Online], https://cycling74.com/products/max. [Accessed 30 August, 2018].
19. E. Williams, “Fourier Acoustics: Sound Radiation and Nearfield Acoustical Holography,” Academic Press (1999).
20. B. Bernschutz, C. Porschmann, S. Spors, and S. Weinzierl, “SOFiA Sound Field Analysis Toolbox,” Proc. Int. Conf. on Spatial Audio (ICSA), Detmold, Germany (2011).
21. N. Shabtai, G. Behler, M. Vorländer, and S. Weinzierl, “Generation and analysis of an acoustic radiation pattern database for forty-one musical instruments,” J. Acoust. Soc. Am., 141 (2), 1246-1256 (2017).
22. M. Neal, “Investigating the sense of listener envelopment in concert halls using third-order Ambisonic reproduction over a loudspeaker array and a hybrid room acoustics simulation method,” MS Thesis, The Pennslyvania State University (2015).
23. D. Dick and M. Vigeant, “A comparison of measured room acoustics metrics using a spherical microphone array and conventional methods,” Appl. Acoust., 107, 34-45 (2016).
24. L. Beranek, “Concert Halls and Opera Houses: Music, Acoustics, and Architecture,” Springer (2004).
25. J. Bradley and G. Souludre, “Objective measures of listener envelopment,” J. Acoust. Soc. Am., 98 (5), 2590-2597 (1995).