CHARACTERIZING THE AUDIBILITY OF SOUND FIELD WITH DIFFUSION IN ARCHITECTURAL SPACES

by

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Dedication

To my family

Acknowledgements

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Abstract

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The significance of diffusion control in room acoustics is that it attempts to avoid echoes by dispersing reflections while removing less valuable sound energy. Some applications place emphasis on the enhancement of late reflections to promote a sense of envelopment, and on methods required to measure the performance of diffusers. What still remains unclear is the impact of diffusion on the audibility quality due to the geometric arrangement of architectural elements. The objective of this research is to characterize the audibility of the sound field with diffusion in architectural space.

In order to address this objective, an approach utilizing various methods and new techniques relevant to room acoustics standards was applied. An array of microphones based on beam forming (i.e., an Acoustic Camera) was utilized for field measurements in a recording studio, classrooms, auditoriums, concert halls and sport arenas. Given the ability to combine a visual image with acoustical data, the impulse responses measured were analyzed to identify the impact of diffusive surfaces on the early, late, and reverberant sound fields. The effects of the room geometry and the proportions of the diffusive surfaces were observed by utilizing computer simulations. The diffuseness in each space was measured by coherences from different measurement positions along with the acoustical conditions predicted by objective parameters such as T30, EDT, C80, and C50. Noticeable differences of the auditory experience were investigated by utilizing computer-based survey techniques, given the current software auralization capabilities. The results based on statistical analysis demonstrate the users' ability to localize the sound, and to distinguish the intensity, clarity, and reverberation created within the virtual environment.

The impact of architectural elements in diffusion control is evaluated by the design variable interaction, objectively and subjectively. The effectiveness of the diffusive surfaces is determined by the echo reduction and the sense of complete immersion in a given room acoustics volume. The application of such methodology at various stages of design provides the ability to create a better auditory experience by the users. The results based on the cases studied have contributed to the development of new acoustical treatment based on the diffusion characteristics.

Chapter 1
Introduction

Room acoustics design is governed by scientific principles that involve the sound source's properties, the sound propagation path, and human auditory perception. An architectural space is a sound propagation path. Often it is a design product that relies on the architect's intuition and desire where the acoustical condition is not well thought out prior to the design process. This assumption and the fact that subjective judgments rely on human hearing sensations has often categorized room acoustics more as a study of art than exact science (Kuttruff, 2009). Nevertheless, there is no doubt that the rapid development of methods and techniques in room acoustics have shown how amenable this field is to scientific solutions.

Room acoustics properties can now be measured with whatever degree of accuracy is required. The solutions lead to architectural spaces with adequate, purposeful audibility, which means spaces where one can simply understand speech, communicate easily, or distinguish musical tones in a scale.

The basic principles of room acoustics rely on the free-field condition. Inserting boundaries into the free-field through architectural manipulation will alter the acoustical condition depending on the frequency and wavelength of the occurring sound. As sound impinges on a surface, the energy can either be absorbed or reflected due to the acoustical properties of the surface. If a sound source is continuously generating energy in an enclosure, absorption by the air and surrounding surfaces (i.e., sound path) prevents the acoustic pressure amplitude from becoming infinitely large (Kinsler, 2000). Air absorption in small rooms can be neglected since air properties are nearly homogenous throughout the space. In a room with very high absorption, the propagating sound energy is reduced fast enough so that there is little or no reverberation and strong attenuation of high frequencies. The space is known as a “dead room.” A room with a small amount of absorption creates a highly reverberant room and is defined as a “live room.”

A diffusion control system has been suggested as a room acoustics solution for spaces with activities that cause conflicting acoustic demands. These activities encompass the need to eliminate excessive reverberation while maintaining a certain amount of the sound energy (Cox and D'Antonio, 2009). This has led to the current trend for acoustics treatment using a diffuser. In this chapter, a brief description of diffusers or diffusive surfaces (i.e., surfaces that have the ability to permit diffusion) is provided as well as the characteristics of a diffused sound field.

Some of the oldest halls with the best reputations were built with an architectural style that naturally contains highly diffuse surface finishes in forms of balconies, columns, alcoves and relief ornamentation (D'Antonio and Cox, 2000). The degree of diffuseness of the sound field within these halls, however, was not quantified. Meanwhile, a significant number of studies during the past years have placed an emphasis on methods to design, predict, and measure the performance of diffusers.

This study aims to characterize the audibility of a sound field with diffusion and to identify the geometrical arrangement and architectural elements of the space that significantly contribute diffusion within the sound field. Characterization is based on relationships among objective parameters and subjective attributes describing the auditory perception. A section on the human hearing system and fundamentals in psychoacoustics is provided to support information underlying the methodology.

Basic principles in room acoustics design is given in a section prior to the description of diffusion, diffusers, and diffused sound field. An historical review of objective parameters and subjective attributes profound in this field is also described. It provides insight into the research contribution, given the current method, techniques, and topics within this field. Methods and objectives applied are systematically described, which then leads to the outcomes and contributions. Research methodology includes the description of the objective and subjective parameters measured and methods for the data measurements.

1.1 Research in Room Acoustics

A brief description about the research method in room acoustics in general is shown in Figure 1.1.

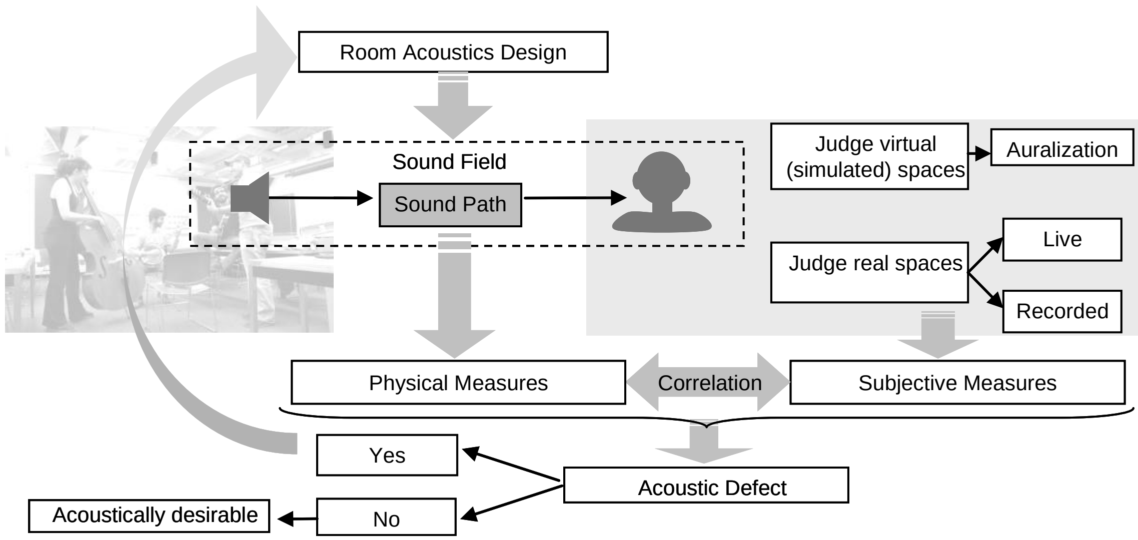


Figure 1.1. Research path in room acoustics design.

The three major research objectives are to identify design challenges that occur in real architectural spaces, to provide design improvements, and to evaluate the design outcome. Design challenges in an architectural space are identified through physical measurements, while the auditory quality of the space is evaluated by the occupants. A design improvement is then applied and the cycle continues with an evaluation.

While methods and techniques in room acoustics are quite sophisticated nowadays due to the advanced instruments that have been developed, very little information is available to help the architects during the early stages of design. Deficiencies in the guidelines have led to failure in delivering the appropriate design solution. Another factor that contributes to this failure is the complexity of the sound field in an enclosed space, since it is constructed with a large number of components. This is the main reason why no exact room acoustics treatment or a single design solution is available.

Evaluation and improvement of room acoustics is achieved through a thorough study of the sound-field characteristics. Figure 1.2 describes the classification of sound fields based on the frequency of the propagating sound waves. The first region of sound fields is where plane waves occur. At low frequencies, where the wavelength is greater than twice the length of the longest dimension of the room, only plane waves can be formed and the room behaves like a duct. This condition can occur in very small rooms, which in practice are rarely found.

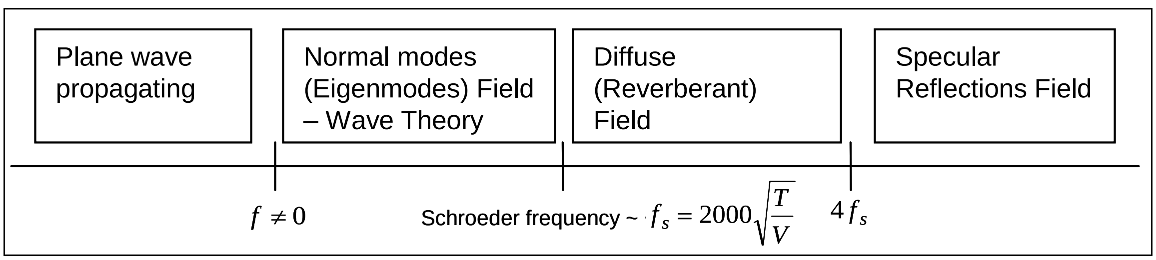


Figure 1.2. Regions of sound field in room acoustics based on the sound frequency.

Above the cutoff frequency of a room, normal modes are formed, which are manifested as standing waves having localized regions of high (antinodes) and low pressure (nodes). The wave theory method is used to characterize a sound field by its normal modes using eigenvalues and eigenfrequency for the wave equation solution. Closed path reflections, such as in a simple rectangular room with rigid boundaries, lead to standing wave resonances that create room modes (Blaszak, 2008).

In a large room and/or at higher frequencies, the density of modes is so great that there is a virtual continuum in each frequency range. It becomes more useful to model room behavior based on the energy density or other statistical considerations (Kuttruff, 2009). A crossover frequency that marks the transition from individual, well-separated resonances to many overlapping normal modes and diffused fields is known as the ”Schroeder frequency” (Schroeder, 1996). The most common room acoustics conditions fall into the range of diffused fields.

Parameters exist to measure and describe the properties of a sound field which are associated to the sound source of speech and/or music. A brief historical review[1] of research in developing important room acoustics parameters within the last decade is illustrated in Figure 1.3.

Important speech intelligibility predictors suggested by Bradley (Bradley, 1986) are speech transmission index (STI), clarity for speech (C50), articulation index, speech intelligibility index (SII), reverberation time, and noise criteria. As for musical comfort parameters, a study done by Cerdá (Cerdá *et al.*, 2009) categorized the parameters according to the energy, reverberation, and spatial parameters. Included as energy parameters are strength (G), clarity for speech (C50), clarity for music (C_{80}), and center time (Ts).

Parameters based on the reverberation are reverberation time (T_{30}), early decay time (EDT), bass ratio (BR), and treble ratio or brilliance (Br). Other important spatial parameters are the interaural cross correlation (IACC) and lateral fraction (LF). Related to these objective parameters are subjective impressions providing indicators of the acoustical quality of a room. The subjective impression is stated using index ratings from an “excellent” to a “poor” quality.

Several parameters describe the human audibility, such as loudness perception in the phon scale, which is a unit for the perceived loudness level (Fletcher and Munson, 1933). Based on this past work by Fletcher and Munson, the equal loudness contour of pure tone was clearly defined in 1957, which describes the threshold of hearing and the threshold of pain (Robinson and Dadson, 1957). Other subjective attributes to describe the human hearing sensation were further developed.

[1] The historical review is part of a literature review to observe past research on objective and subjective parameters and their attributes. It includes the study on the measurement method utilized and examples of implementation.

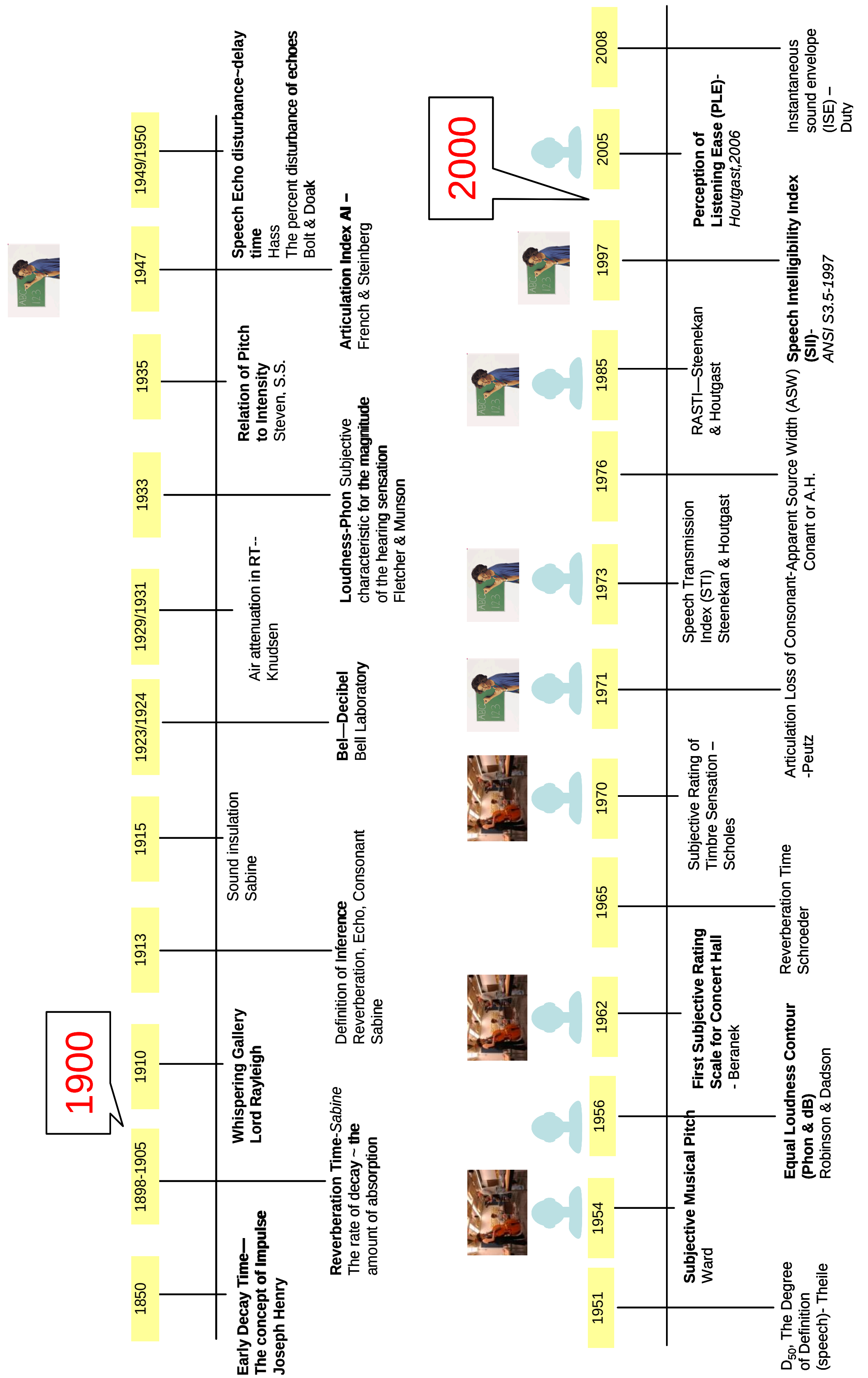


Figure 1.3.Historical review of room acoustics objective and subjective parameters.

Understanding the parameters that are related to psychoacoustics is as important as the ability to select the suitable parameters to address a research objective. It is based on the assumption that the final interpretation of the acoustical quality relies on human perception, given the audibility characteristics. A standardized parameter to measure the diffuseness of a sound field is not yet available nor is the ability to characterize the audibility conditions within the sound field that is impacted by diffusion (D'Antonio and Cox, 2000).

Selected objective parameters and subjective attributes that are utilized within this study are described in detail in section 2.1. The selection is based on the information extracted from the historical review of acoustical parameters, standards measurement, and relevant applications of diffusion in room acoustics.

1.2 Introduction to Diffusion, Diffusers, and Diffuseness of Sound Field

Reflected sound may leave the surface as specular or non-specular reflections. A specular reflection is a condition where the angle of reflection equals the angle of incidence. Moreover, if the reflections are scattered to non-specular directions and uniformly dispersed (that is separation of the reflected sound into its frequency components), then diffusion will occur (see Figure 1.4). A surface that creates this diffusion phenomenon is known as a diffuser (D'Antonio and Cox, 2000).

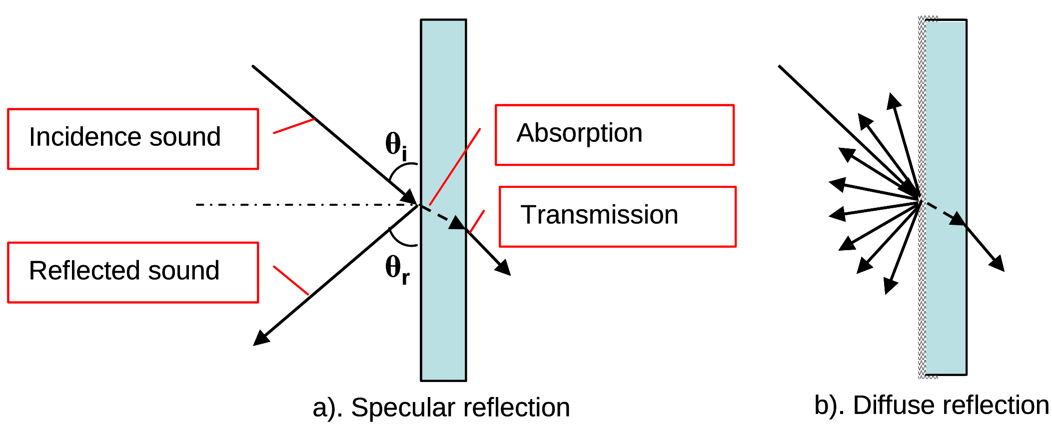


Figure 1.4. Illustration of a) specular reflection and b) diffuse reflection.

Diffusion can also be described as multiple scattering as shown in Figure 1.5. The performance of a diffuser is characterized by its diffusivity, which is the ability of the surface with its material properties to permit diffusion.

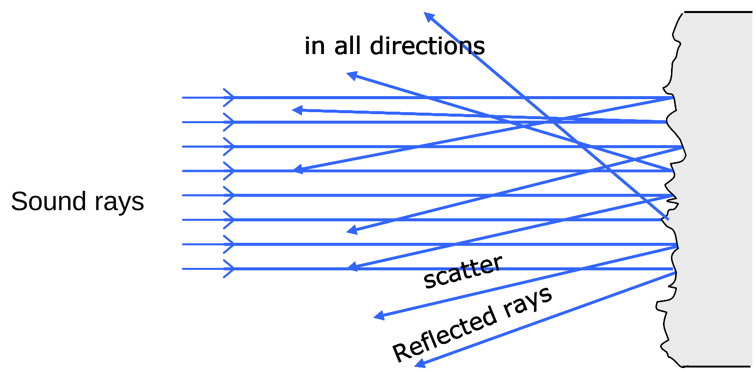


Figure 1.5. Sound waves scattering from a rough surface.

The roughness of a reflective surface causes the sound wave to reflect in all directions. On the edges of an object, diffraction becomes more dominant. Reflective surfaces with a certain ratio of surface roughness to surface size are intentionally designed as diffusers. They are known as numerical diffusers, which are comprised of wells of equal width with varying depth that is critical to a range of wavelengths (Schroeder, 1975). Illustration of this numerical diffuser is shown in Figure 1.6 (b).

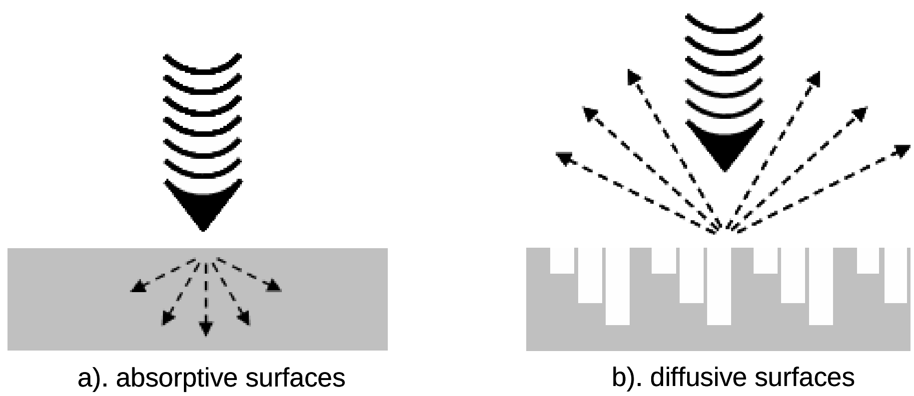
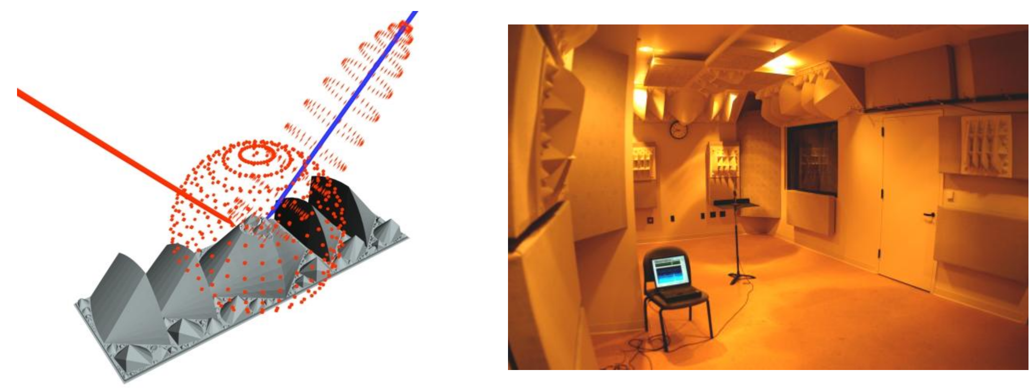


Figure 1.6. Sound waves impinging on a) absorptive and b) diffusive surfaces.

A diffuser was first introduced by Schroeder using the method of maximum length sequence (MLS), which results in the diffusion of a specific frequency band. This method relies on the creation of a series of reflection coefficients on a surface alternating between +1 and -1 in a predetermined periodic pattern (Schroeder, 1975). In 1979, Schroeder introduced a diffuser for a broader bandwidth known as the quadratic residue diffuser or Schroeder diffuser (Schroeder, 1979a).

In practice, a numerical diffuser might not be the critical element that creates the majority of diffusion within the sound field. Any non-planar surface with roughness, shape, and dimensions critical to a range of wavelengths has the potential to create diffusion. Diffusive surfaces are linked in their size to the frequencies over which they are intended to have an effect. On the basis of these mechanisms, diffusers in general can be defined as an obstacle in a sound path with surfaces that uniformly disperse (i.e., spatial and temporal dispersion) a significant portion of the reflected sound. Efficiency of performance depends on the effectiveness of sound diffusion in the frequency range and the direction where it is needed. Therefore, it is important to use the standardized method to measure the performance of the diffuser known as the diffuse coefficient.



a). Simulation of specular reflection, absorption and diffusion

b). Field measurement after renovation

Figure 1.7. Simulation of Golden Acoustic diffusers and field measurement of its application in a music practice chamber within the Detroit Orchestra Hall.

Most of the diffusers utilized in this research are the product of Golden Acoustics. The newly developed sonic panels by Golden Acoustics have silver-colored, studded surfaces with an ordered array of half- and full-size cones, which jut from every wall surface and the ceiling itself. The panels have shown positive performance by the ability to delay the low frequency attenuation within the space based on past research conducted within University of Michigan, Taubman College of Architecture and Urban Planning (TCAUP) acoustic simulation laboratory (unpublished). This was observed in the use of these diffusers in a music practice chamber in the Detroit Symphony Orchestra Hall (see Figure 1.7). Performance was mainly measured by on-site evaluation (before and after) and surveys of user satisfaction.

According to ISO 17497-1:2004(E), the methods to measure scattering or diffuse coefficients are classified as the free-field method and the reverberation chamber method. The free-field method was first introduced by Vorländer and Mommertz based on FFT post-processing of the measured impulse response for measuring scattering coefficients (Vorländer and Mommertz, 2000). Cox and D’Antonio (Cox and D'Antonio, 2009) introduced the polar distribution method based on the free-field method. It measures the diffuser’s performance by using the similarity between the scattered polar response and a uniform distribution. A different method based on wave field synthesis measures the total diffuse energy coming from a diffuser panel, and is defined as the surface diffusion coefficient (Farina, 2000). It takes out the amount of specular reflections, leaving only the diffused reflections. The last method discussed here is the subtraction method (Mommertz, 2000). It uses the impulse response measured in free-field to subtract the direct sound from the measured impulse response in the reflective room.

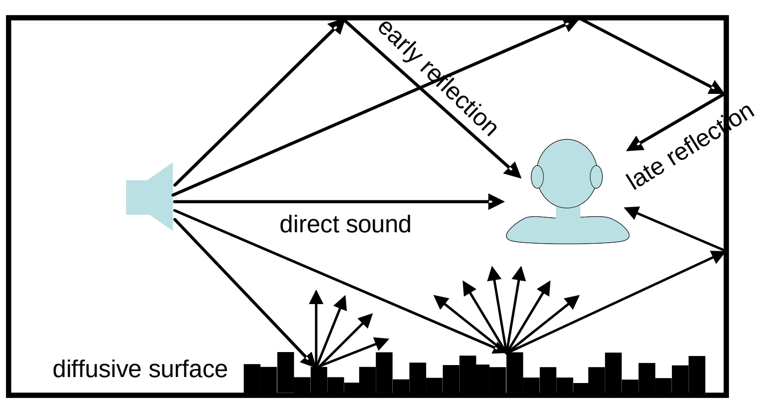


Figure 1.8. A sound path with diffusive surface and non-diffusive surface.

It should be noted that measuring the performance of a diffuser based on the diffusion coefficient is not the objective of this research. The goal is to evaluate the diffuseness of a sound field having a diffuser or diffusive surfaces in it (see Figure 1.8). Hypothetically, the diffuseness is not impacted by a diffuser alone, but rather by all the architectural elements. The method relies on observation of the impact of early and late reflections of the propagating sound, before and after the diffuser or other architectural elements are applied.

The first approach used to examine the contribution of a diffusive surface to the diffuseness of a sound field is by observing its ability to reduce comb-filtering as compared to a flat, reflective surface. This comb-filtering effect mostly occurs at the early reflections. Comb-filtering denotes an effect that creates the comb-like signal (see Figure 1.9). The series of constructive and destructive interferences will appear as periodic peaks and dips in a range of amplitude as a function of frequency response. It happens when two successive copies of a signal are summed together at the receiver. Comb-filtering can be created by multiple sources at different distances emitting the same signal, multiple-spaced microphones recording the same sound source, or a series of room reflections arriving at the microphone.

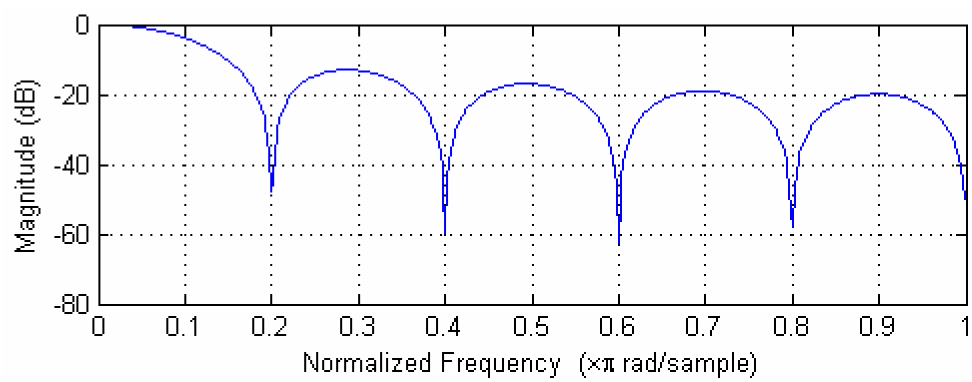


Figure 1.9. Example of comb-filtering computed with Matlab.

Less comb-filtering is depicted if there is a decrease in similarity between the direct sound and the reflection spectral content. An example is shown in Figure 1.10. The frequency responses[2] are obtained from impulse response

[2] Frequency responses in the logarithmic scale were plotted using SIA SmaartLive™, a sound system optimization and control software that can interpret information from the incoming signal at microphones.

measurements with an omni-directional microphone in room 2216-2219 in the Art & Architecture Building (AA21). Within this room, both absorber and diffuser panels are implemented.

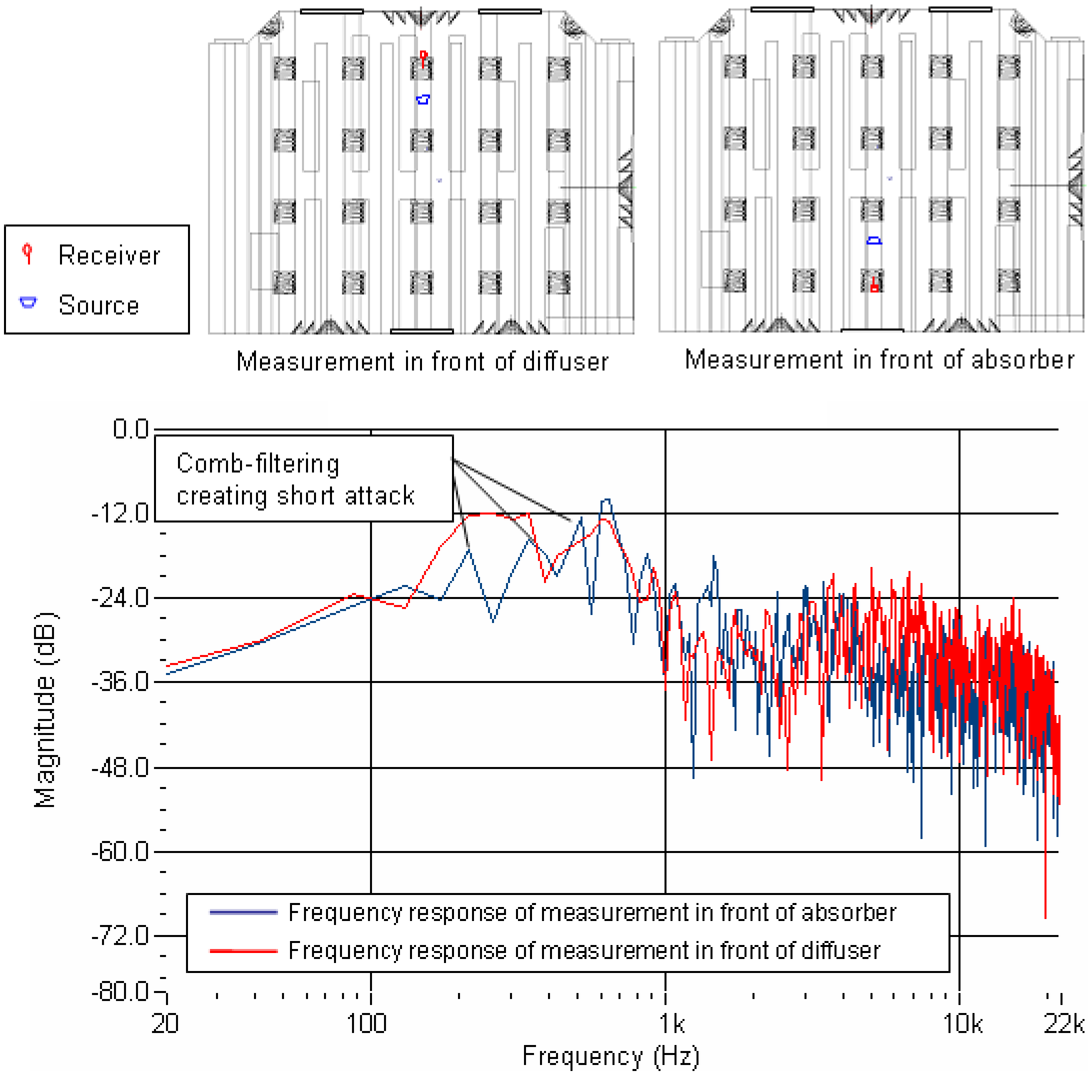


Figure 1.10. Example of the use of a diffuser to reduce comb-filtering.

Another application of diffusers is to achieve enhancement of late reflection to promote the sense of envelopment. This enhancement of the late reflections creates what is often known as the diffused sound field. A second approach to characterize the diffusivity of surfaces is by measuring the change of the degree of the diffuseness, before and after the diffuser is applied.

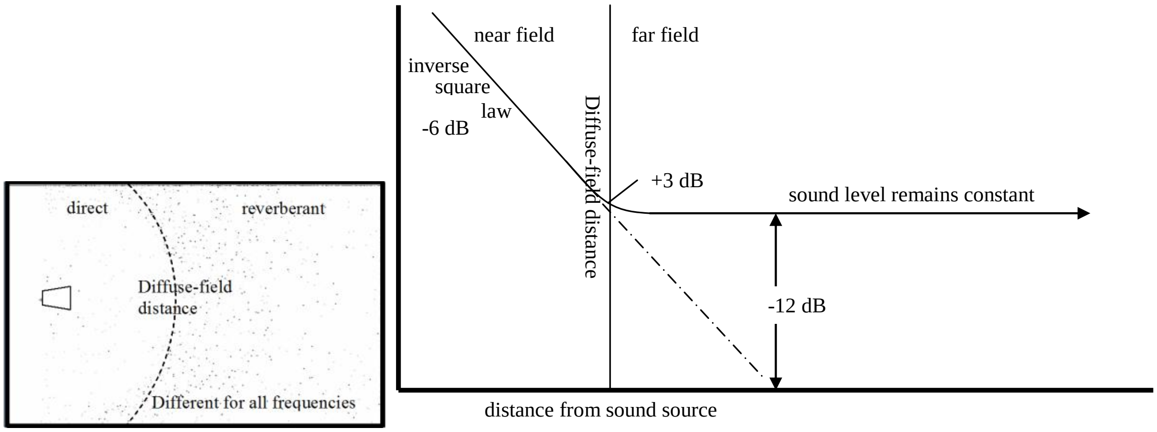


Figure 1.11. Critical distance or diffuse-field distance.

A sound field is considered diffuse under the following two conditions: the reflected sounds are coming from all directions with equal intensity, and the reverberant sound field is the same at every position in the room (Hodgson, 1994). The distance from the source that defines the “starting line” of the diffused sound field is called the critical distance, also known as diffuse-field distance (r_c). It is defined as the distance from the acoustic center of a sound source at which the mean-square sound pressure of the direct field in a specified direction is equal to the mean-square sound pressure of the reverberant sound in the room containing the source (ANSI S1 1-2004).

According to Schroeder, 1996, the diffuse-field distance can be determined from the reverberation time by

$$\varepsilon_d = \frac{P}{4\pi r^2 c} = \varepsilon_r = \frac{PT}{\log_e 10^6 V} \qquad (1\text{-}1)$$

where, ε_d is the direct energy density, ε_r is the reverberant energy density, P stands for power emitted by the source, T is the reverberation time of a room with certain volume, V. Based on equation (1-1), the critical distance is defined as,

$$r_c = \sqrt{\frac{\log_e 10^6 V}{4\pi c T}} \qquad (1\text{-}2)$$

Receiver placement within this distance will predominantly render direct-sound signals while placement outside will predominantly render diffuse-sound signals, which are perceived by a listener as spatial impression (Blauert *et al.*, 2008). Any input sounds measured beyond the critical distance in a diffused sound field should be similar. The coherence of two input sound measurements describe how well correlated (i.e., how similar) the sound waves are. The use of coherence to predict the degree of sound-field diffuseness is based on several references (Cook *et al.*, 1955; Yanagisawa and Takayama, 1983) and quantified by the autocorrelation and cross-correlation functions,

$$\rho_{11}(0) = \overline{e_1^2(t)} \tag{1-3}$$

$$\rho_{22}(0) = \overline{e_2^2(t)} \tag{1-4}$$

$$\rho_{12}(0) = \overline{e_1(t) \cdot e_2(t)} \tag{1-5}$$

where $e_1(t)$ and $e_2(t)$ are the microphone outputs, $\rho_{11}(0)$ and $\rho_{22}(0)$ show the respective autocorrelations, and $\rho_{12}(0)$ shows the cross-correlation. The bar over each function on the right hand-side represents the time-average. Using equation (1-3) to (1-5), the coherence can be obtained by,

$$C = \frac{\rho_{12}(0)}{\sqrt{\rho_{11}(0) \cdot \rho_{22}(0)}} \tag{1-6}$$

Receivers' distances describe the sound-field boundaries. An example of a comparison of the similarity between the energy decay at two measurement positions in Dennison Hall Room 170 (DH170) is illustrated in Figure 1.12.

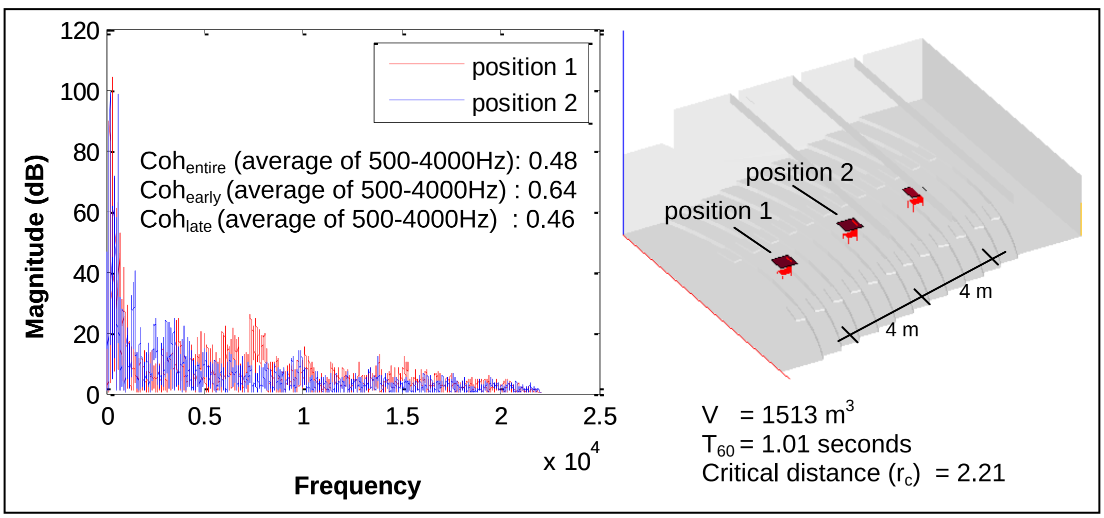


Figure 1.12. Energy decay of impulse responses at two measurement positions.

Utilizing this method, the coherence calculated value is 0.48, indicating the degree of diffuseness of the sound field within the boundary of position 1 and 2. Zero would be a condition of a non-diffused sound field, and a value of 1 would be completely diffused, assuring that the acoustical conditions are similar if one moves from one position to another within this sound field. Details of the signal processing and calculation of coherences is provided in section 2.1.

Another indicator of a diffused sound field is from a listener's point of view. Sound seems to be diffuse if the sound is perceived as coming from many directions. It is the sensation of being enveloped by the sound. This sensation can be measured by using the principle of coherence for input sounds at the left and right ear of a listener. Details of the parameter known as listener envelopment (LEV), which is calculated from the interaural cross correlation (IACC), are provided in section 2.1.4.

Beranek suggested two requirements to obtain sound with a good sound field using diffusion control: reverberation time must be fairly long, and irregularities of shape in ceilings and walls should be present (Beranek, 1962). A study has found the importance of taking into account the room shape, absorber location, and degree of sound diffusion into the reverberation-time calculation (Schroeder and Gerlach, 1974).

Figure 1.13. Architectural spaces with non-planar surfaces and with diffusers.

Diffuse sound fields in architectural spaces that were built 100 years ago were created by the ornamentation on interior surfaces and not necessarily in the form and appearance of diffusers (see Figure 1.13). An example shown here is the Detroit Orchestra Hall (DOH) with curved balconies and three-dimensional ornaments on the wall and ceiling surfaces. A different space to compare with the DOH is a lecture hall, Auditorium A in Angell Hall, University of Michigan, which recently was renovated for improvement of the acoustics. Diffusers applied here are Golden Acoustics panels.

To underline the discussion within this section, it is important to understand that diffusion created by diffusers can enhance early and late reflections. It is also important to identify the architectural elements (diffusers or other elements) that contribute to the room acoustics condition. Enhancement of the early reflections reduces comb-filtering, which reduces the chance of having a hollow sound and/or muddy and poorly defined sound due to short echoes. Meanwhile, diffusion for late reflections enhances the envelopment sensation within the diffused sound field. Both enhancements do not necessarily occur simultaneously in a sound field.

1.3 Introduction to Auditory Perception

Auditory perception is relevant to the sense of hearing, which involves three elements: 1) the physical nature of the signal, 2) the sensory detection by the nervous system, and 3) the final transformation into a perception. The portion of the sound energy that propagates within the environment and is detected by the hearing system is known as the auditory event. The auditory event is the stimulus that creates the auditory perception.

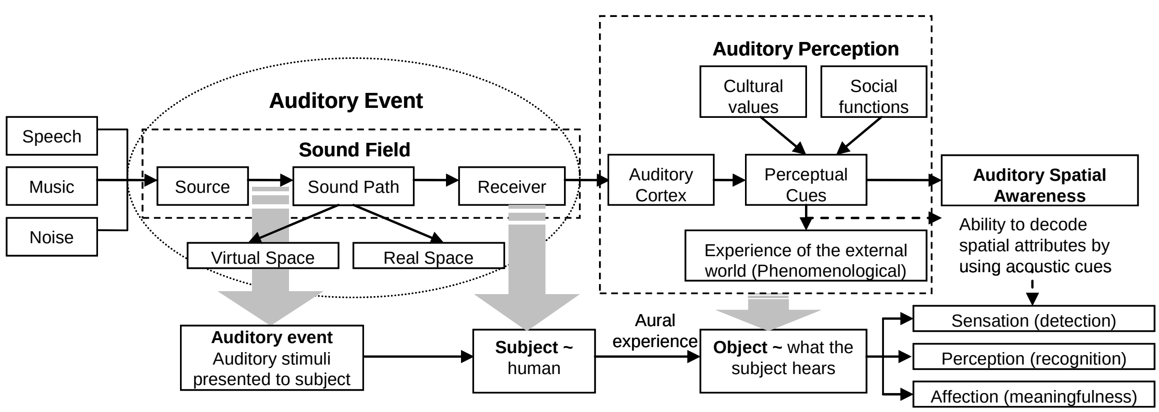


Figure 1.14. Aural architecture.

Aural experience for humans is categorized as detection of sound (sensation), recognition of sound (perception), and active reaction to the meaning and emotions (affection), the latter referring to the act of listening. Aural architecture studies the effect of space on auditory perception and spatial perception (Blesser and Salter, 2006).

Whenever sound as vibrational energy arrives at the ear, it is processed by a complex but distinct series of steps due to the anatomical division of the ear into the outer ear, middle ear, and inner ear. The ear pinnae, as part of the outer ear, serve to filter high-frequency sounds, to focus sound waves into the middle and inner portions of the ear, and also help to determine the direction from which a sound originates. The ear canal acts as an amplifier for sound frequencies between 3,000 and 4,000 Hz. The middle ear system has the ability to greatly amplify sound vibrations before they enter the inner ear. The last process in the inner ear is the conversion of vibrational energy into nerve impulses (electrical energy) that will travel to the brain. Auditory processing by the human brain allows sound to be perceived with a variety of pitch and loudness. Details of the hearing mechanism can be found in references within the field of psychoacoustics (Howard and Angus, 2006).

There are some aspects of the human auditory system that are important to consider in the study of room acoustics. The frequency analysis within the auditory system is the first aspect. The cochlea located in the inner ear, breaks down acoustic signals into frequency components. These components of signals are then carried by nerves, which behave as frequency channels that convey information about the energy and timing of the signal.

Interpretation of frequency ranging in auditory perception is described as pitch. Human ear sensitivity is within the frequency range of 20 Hz to 22 kHz. Absolute sensitivity varies with frequencies, and the bandwidth changes with the level of the input signal. The first research on this topic was conducted by Fletcher and Munson (Fletcher and Munson, 1933), who created the first equal-loudness curve (see Figure 1.15). Resonances within the ear canal create the down slope at above 4000Hz. It is, therefore, important to consider the sound event characteristic since it affects the auditory perception in a number of ways. For instance, speech is primarily conveyed by sound energy between 200 and 5000 Hz. To achieve the same sensitivity level, low frequencies require a higher sound pressure level given the normal hearing equal-loudness contour.

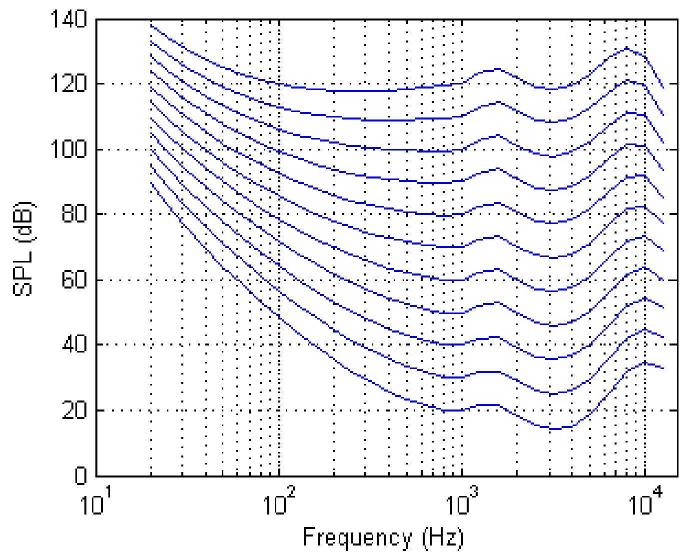


Figure 1.15. Normal equal-loudness contours for pure tones (ISO:226, 2003).

Auditory events occur in all directions from the person who perceives them. The totality of all possible positions of auditory events constitutes an auditory space (Blauert, 1997). The word "space" used in this expression is to be understood in the mathematical sense, as a set of points between which distances can be determined. The mode of the human auditory system function having two ears creates this spatial perception. This auditory system is defined as binaural hearing.

When multiple auditory events are occurring simultaneously, sounds interfere with one another in various ways. This creates another perceptual sensitivity known as a masking effect that often degrades the ability to discriminate and detect the sound tasks. Depending on the differences in arrival time of signals at each ear, the binaural hearing can reduce the masking effect. The phenomenon is known as binaural unmasking. With this ability, multiple sounds that are coming from different directions are actually detected more easily than if they are coming from only one direction.

Spatial perception is the ability to localize a sound source or dominant reflective surfaces given directionality and distance perception (Zwicker and Fastl, 1999; Shinn-Cunningham *et al.*, 2005; Vorlander, 2008). Localization cues are based on the comparison and interpretation of the time lag between the sounds reaching the right ear versus the left ear (Hartmann, 1999). The primary localization cues are interaural time differences (ITD) and interaural level differences (ILD), first proposed by Lord Rayleigh as the Duplex theory in 1900 describing human binaural sound localization (Zwicker and Fastl, 1999).

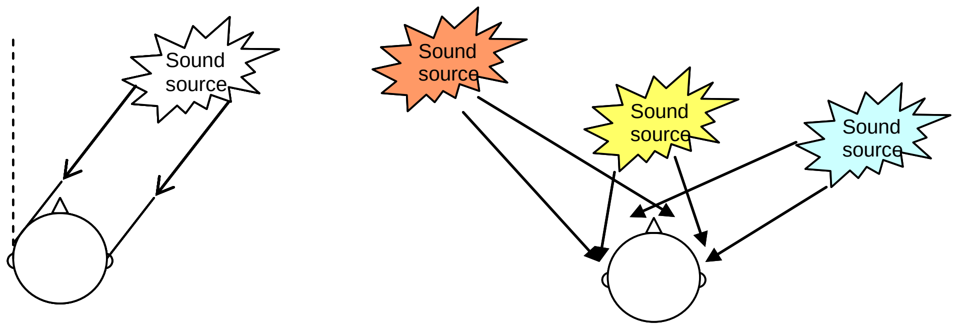


Figure 1.16. Illustration of human binaural sound localization.

The ITD is the difference in arrival times of waveform features at the two ears, and for pure tones, it is equivalent to a difference in phase. Observations of the listener's sound localization ability by using the ITD are only accurate at frequencies of 500 Hz or lower. The ILD is the standard comparison between intensities in the left and right ears. The effect becomes more noticeable in a large room where reflected sound dominates the direct sound, as compared to the ITD. The ILD is sensitive for all frequencies. However, at high frequencies the ILD is not only determined by the shape of the head, but is also greatly influenced by the shape of the pinna. In principle, localization is affected by resonances inside the ear and scattering of the head and upper torso. The loci of positions that cannot be resolved from binaural cues are described as a "cone of confusion" centered on the interaural axis.

Spectral cues are the main provider of elevation cues, which reduce the cone of confusion in source localization and depend on the relative position of the sound source and the listener's head (Batteau, 1967). Spectral cues arise as interactions of the outer ear (pinna) with the impinging sound wave. The delayed reflections create comb-filter interference effects on the received sound (Muller and Bovet, 1999).

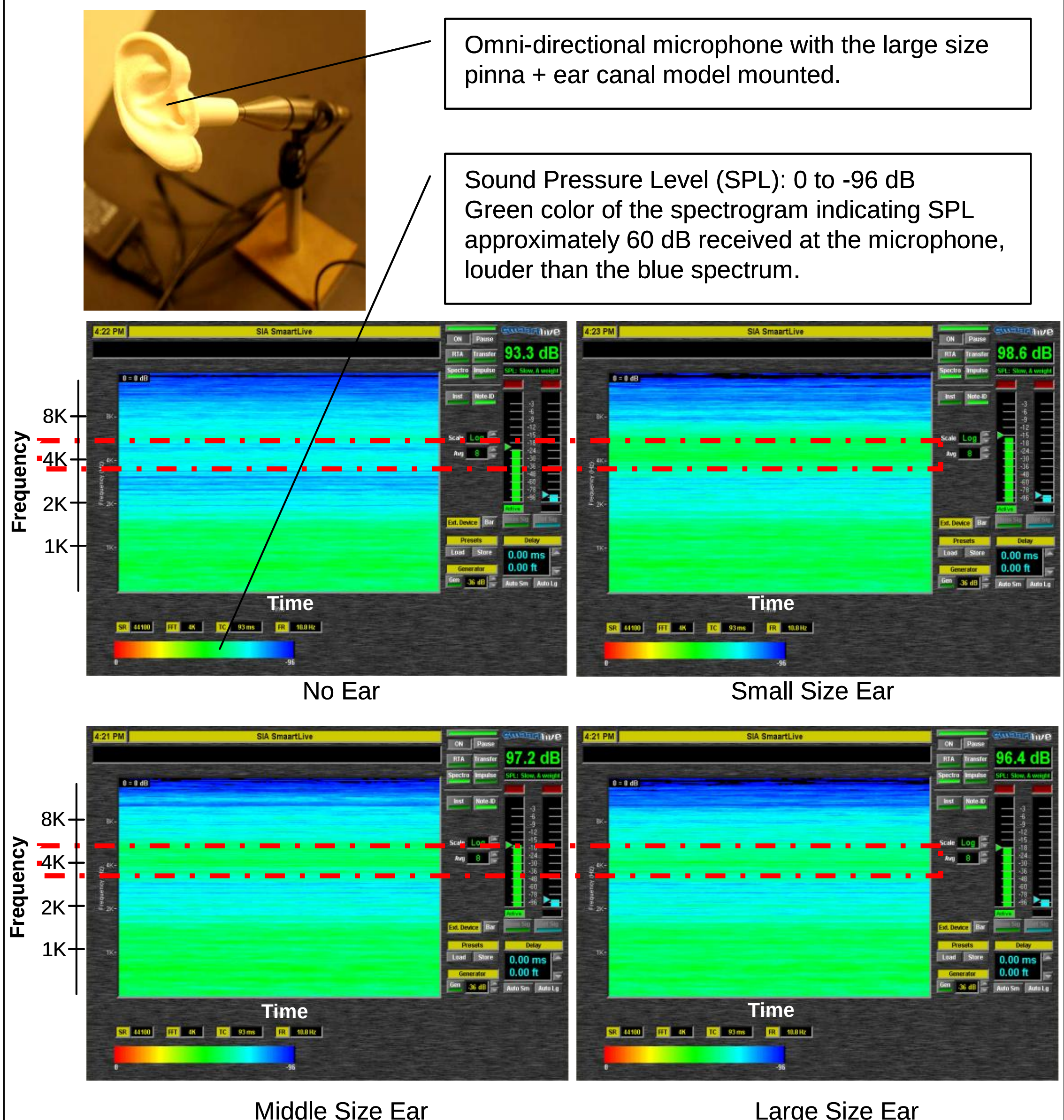


Figure 1.17.Observation on the SPL enhancement at 4 – 5 kHz by the ear pinna.[3]

[3] Ear pinna in this experiment is a rapid prototyping of a human ear using the 3D printing technique. The material used was a polymer resin. Three sizes of ear pinnae were used and categorized as large, normal, and small size of a scanned normal ear. Scaling up and down the normal ear is based on a study of anthropometric manikin (Burkhard, M. D., and Sachs, R. M. (**1975**). "Anthropometric manikin for acoustic research," The Journal of the Acoustical Society of America **58**, 214-222.

Individual differences of the pinna spectral filtering are large (Butler, 1975). To explore this phenomenon, the author recorded a broadband noise through a microphone without pinna and with three different sizes of pinna attached using SIA SmaartLive (see Figure 1.17). Enhancements on the incoming sound due to the ear pinna were detected at the frequency range between 4 kHz – 5 kHz shown by the area within the red dashed-line.

Changes in source position relative to the listener and head movements also affect the spatial cues. Listening tests utilizing headphones lose the effect of dynamic cues, and, therefore, it is important to correctly model the head and keep the source direction constant as the head moves. In fact, there are individual differences in the hearing system characteristics that create a uniquely different auditory experience. To serve the purpose for subjective assessment of the room acoustics condition, it is therefore important to use a consistent head and upper torso model during the binaural recording and sound reproduction process in the computer simulation.

1.4 Research Methodology

Methods used within this study address detailed research questions related to:

1. The diffuseness of a sound field (i.e., if there are any diffusion occurrences and the degree of diffuseness),
2. The acoustical impact of diffusers,
3. The acoustical impact of architectural elements that were not assigned as diffusers, and
4. The audibility characteristic of the sound field under consideration.

A solution for an adequate method to characterize the diffuseness of a sound field in architectural spaces is the use of an integrated method of objective measurements and subjective assessments shown in Figure 1.18.

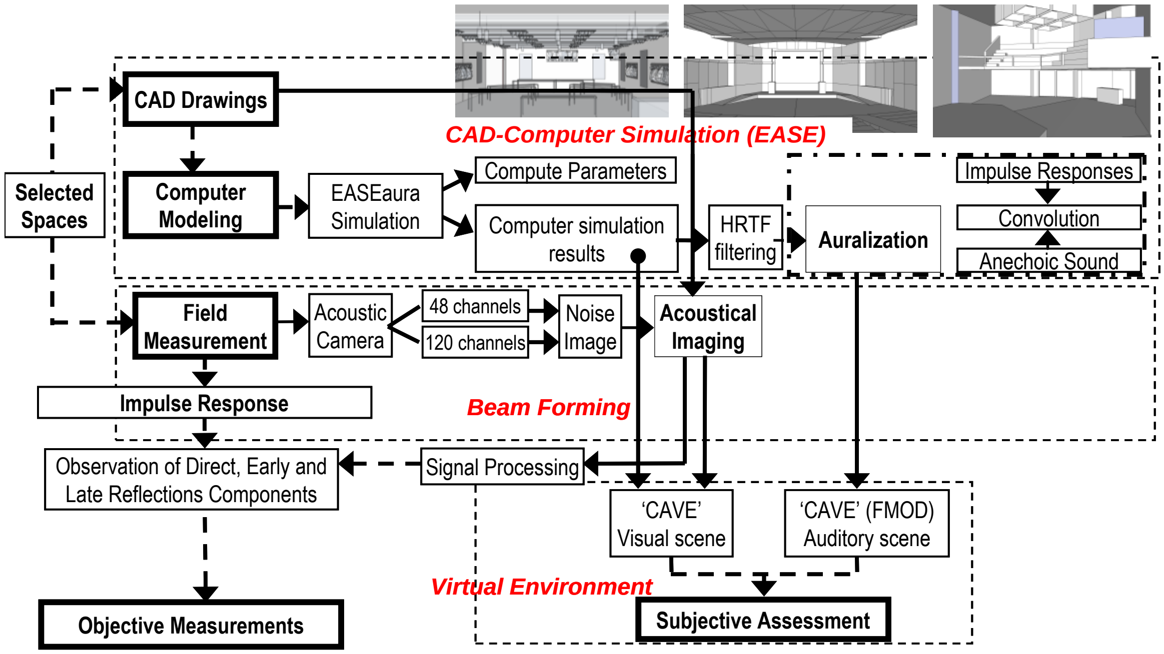


Figure 1.18. The integrated method applied on selected cases studied.

Investigations of the impact of architectural elements require the capability of computer simulation to model a variety of geometrical configurations for the space of interest. Objective parameters are computed within the software while simulated impulse response can be obtained as raw data to predict the sound field diffuseness. Field measurement enables the observation of existing conditions and provides complementary data to insure the accuracy of the computer modeling. Characterizing the audibility of the sound field with diffusion requires a subjective assessment of noticeable differences in the loudness, clarity, and reverberation perception. Details of the techniques applied within the integrated method are provided in Chapter 2.

1.4.1 Measurement of Impulse Response

A given room being measured for its acoustical condition is representing a linear time-invariant system, which is a response to an arbitrary input signal. Linearity means that the relationship between the input and the output of the system is mathematically linear. If input $x_1(t)$ produces response $y_1(t)$, and input $x_2(t)$ produces response $y_2(t)$, then one will obtain,

$$a_1 x_1(t) + a_2 x_2(t) \rightarrow a_1 y_1(t) + a_2 y_2(t) \quad (1\text{-}7)$$

Time invariance means that when the system is triggered with an input at *t=0* (i.e. an impulse), the output will be identical except with a time delay of *t* seconds.

$$x(t) \rightarrow y(t) \quad (1\text{-}8)$$

$$x(t-\tau) \rightarrow y(t-\tau) \quad (1\text{-}9)$$

The fundamental result in LTI system theory is that any LTI system can be characterized by a single function. This single function is called the system's impulse response.

Being excited by an impulse, a room will have an impulse response, which is the time response created by the totality of sound waves that travel from a source to a receiver along a multitude of propagation paths. At a receiver (i.e., microphone), the impulse response consists of direct sound with a series of delayed reflections. Each of the reflections is specified by its time delay and intensity level with respect to the direct sound. The frequency response function may be obtained from the impulse response by employing the Fourier transform (ISO, 2006). An example of the graphical representation of a room impulse response is shown in Figure 1.19, also known as a "reflectogram."

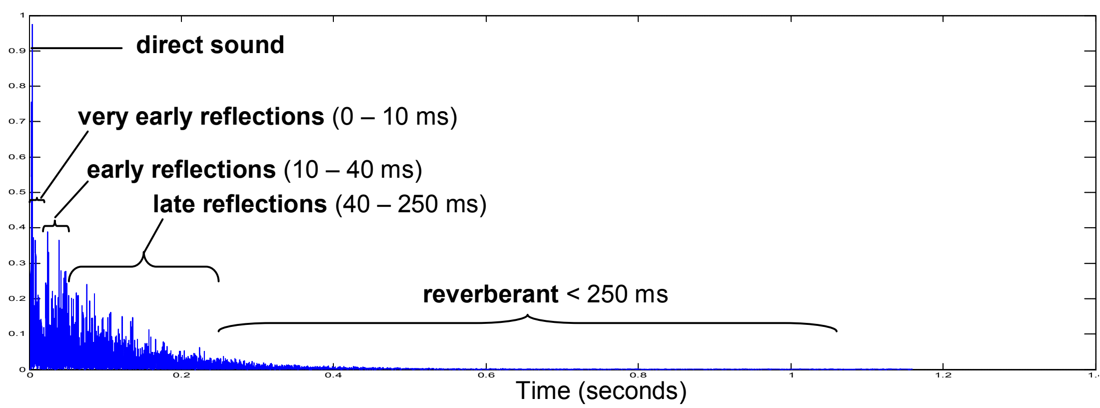


Figure 1.19 An example of a reflectogram of a room impulse response in a time domain.

The direct sound is measured as the first pulse, with its characteristics being highly related to the distances between source and receiver. Early reflections following the direct sound as secondary components are delayed due to the longer path of propagation or the presence of a room element blocking the direct sound. Early reflections enhance the loudness and support some subjective hearing impressions related to clarity, intelligibility, and spaciousness (Beranek, 2004). Some portion of the excited sound will propagate longer and arrive at the receiver as late reflections. The impulse response characteristic is significantly affected by the architectural elements of the room. The impulse responses within this research are obtained from field measurement and computer simulation.

A single impulse response does not include the information of the direction of the incoming sound waves. Meanwhile, a sound field can only be characterized as diffused if the reflected sounds are coming from all directions with equal intensity. Basically, impulse responses measured within a diffused sound field are expected to have similar acoustical properties. At least two impulse responses measured simultaneously at two receiver positions in a space are needed to measure this similarity. The sphere-microphone array, a product of Acoustic Camera, and the computer simulation provide this ability to measure simultaneously impulse responses that arrive from many directions. Details of these methods are discussed in the following sections.

1.4.2 Objective and Subjective Parameters

Haan and Fricke linked the diffusivity of a surface to the acoustical quality (Haan and Fricke, 1992). They showed that the surface diffusivity index, which is a quantification of surface diffusion from a visual inspection, correlates very highly with acoustic quality. The relationship between objective indicators with subjective attributes, however, is barely discussed.

Torres *et al.* found that changes in the amount of diffuse reflections in a computer model were audible, but these changes only approximately modeled the effects of scattering; further conclusions for real spaces were questionable (Torres *et al.*, 2000).

The parameters and indicators that are used in this research are listed in Table 1.1. Selecting the most sensitive indicators to measure the sound-field diffuseness and to characterize its audibility is one of the important achievements of this study. The process included literature review of parameters that have been developed since 1900 through the present (see Figure 1.3), review of standard measurements in room acoustics, and preliminary research on subjective assessment (see section 2.4.2). In section 2.1, further discussion on the logic of using these objective parameters and subjective attributes is provided.

Table 1.1. Objective parameters and subjective attributes for analysis on diffusion.

Objective Parameters	**Unit**	**Objective Attributes measured/observed**	**Subjective attributes measured**
Total Sound Pressure Level (Total SPL)	dB	Intensity Level	Loudness perception
Reverberation Time (T_{30})	sec	Diffusion and Total absorption (Sabin)	Liveliness perception
Early Decay Time (EDT)	sec	Diffusion on early reflections	Liveliness perception
Clarity of Speech (C_{50})	-	Diffusion on early reflections	Clarity
Clarity of Music (C_{80})	-	Diffusion on early reflections	Clarity
Interaural Cross Correlation of the late energy ($IACC_{late,mid}$)	-	Listener Envelopment (LEVcalc) Diffusion on early and late at the listener's ear	Listener Envelopment (LEV), the perception as if sound is coming from all direction
Source Strength factor ($G_{late,mid}$)	dB		

One of the most recent parameters developed by other researchers is the calculated listener envelopment (LEV) proposed by Beranek (Beranek, 2010). It is the sensation of being surrounded by the music or sound source due to diffusion occurring at the human ears. It utilizes the parameters of clarity index for music (C_{80}), strength factor of late energy (G_L), and interaural cross correlation of the late energy ($IACC_L$). This research explored the possibility to use LEV as an indicator to predict the sound-field diffuseness.

1.4.3 Field Measurement

The main instrument used for field measurement is the Acoustic Camera. The base configuration consists of a microphone array with an implemented camera, a data-recording device, a notebook computer, and the Noise-Image software as illustrated in Figure 1.20 (detailed specifications can be found at http:// www.acoustic-camera.com). There are many types of microphone arrays and the one used in this study is the sphere-microphone array with 48 and 120 channels of microphones. The system is based on a common principle known as the "delay-and-sum" beamforming method. Details of this method are described in Figure 1.21.

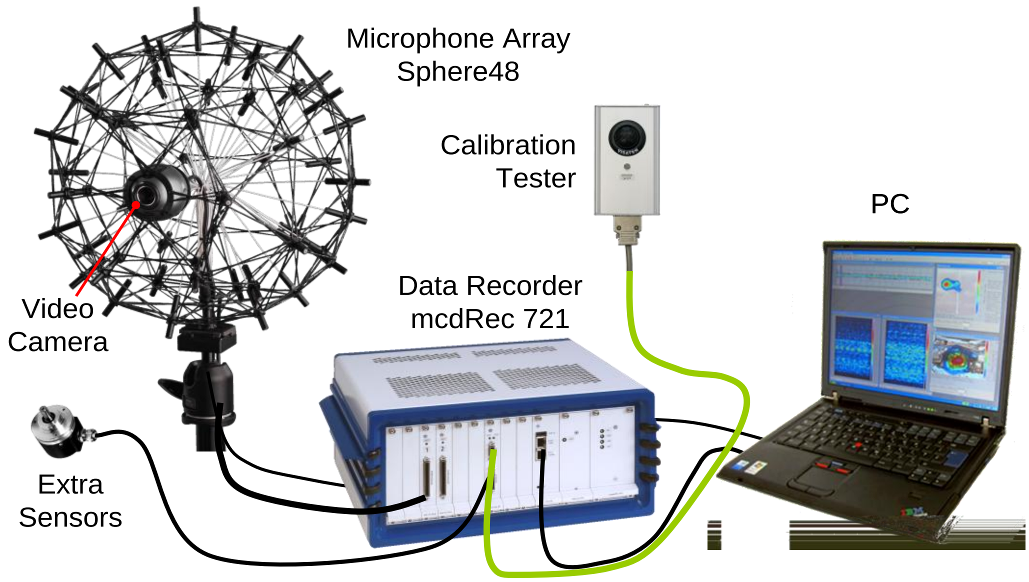


Figure 1.20. The base configuration of the Acoustic Camera system.

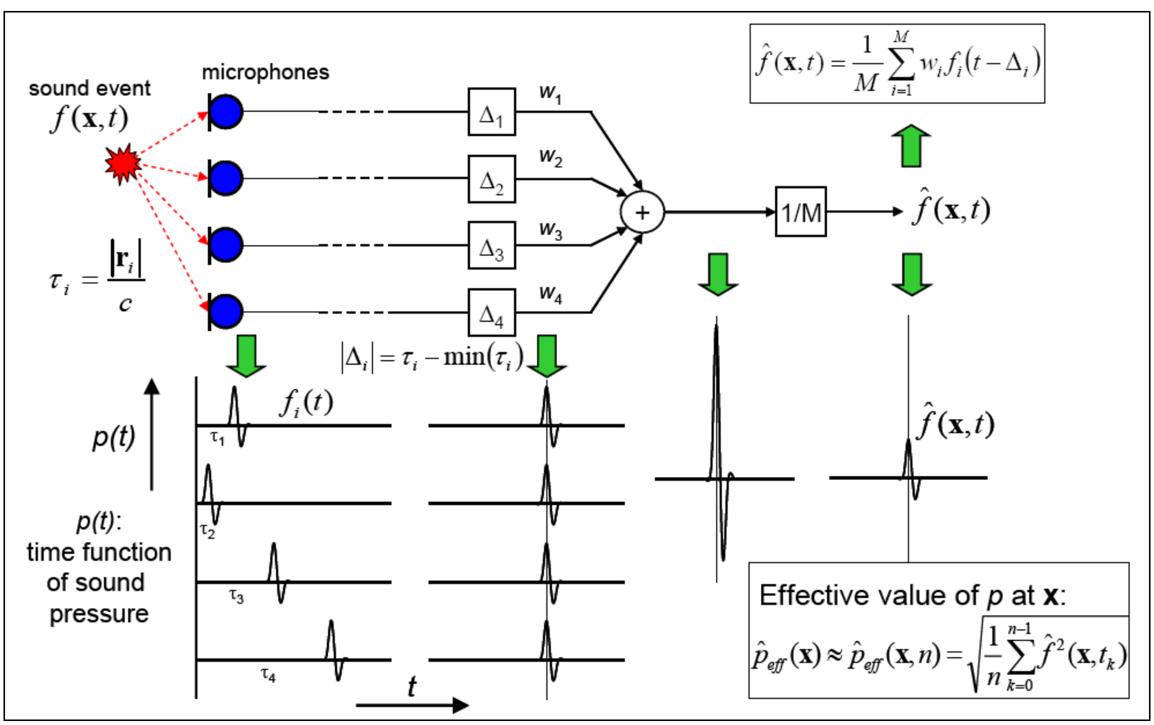


Figure 1.21. "Delay-and-sum" beamforming method (Jaeckel, 2006).

The basic principle of the delay-and-sum beamforming method can be described as follows: the measurement using the beamforming is done by making use of a set of spatially separated microphones. A microphone array will be successively focused to many points lying on a measurement plane or on an object's surface. For each focus point, the audible excitation arrives at a different time and, therefore, compensation is accounted for by the relative runtime delays between the microphone channels. Dividing by the channel number gives an estimated time function, which is comparable in its power content to the original time signal at this focus location (Jaeckel, 2006). In theory, this focus distance can even be considered to be infinitely long, which is equivalent to the model assumption of plane waves passing through the sensor array. In this case the estimated time function is generated by selecting a direction from which to accept signals, while rejecting signals from other directions.

Using the Noise-Image® software capability, the Acoustic Camera measurement can be visualized in a format of sound pressure level mapping on a virtual 3D surface. The SPL is mapped for each focus point at the 3D-model surface, which has already considered the compensated runtime delays between the microphone channels. A 3D-model of the measurement object is therefore needed, preferably available in a standard CAD file format. The polygon model has to be reduced in resolution of its model planes (i.e., triangles) before the actual acoustic mapping takes place. These triangles are intentionally oriented in space and are modeling the actual surface of the measured room. A recent development in this software has emphasized room and building acoustics applications (Acoustic-Camera, 2009).

In one particular source and receiver position, there are multiple signals recorded relevant to the number of microphones on the Acoustic Camera microphone array (i.e., 48 or 120 channels). Therefore, for an impulse excited within a space, there can be N number of impulse responses, where N = number of microphones x number of positions. Data for each probe microphone has the properties of sampling frequency 96 kHz and 16 bit, in *.wav* format. Measurement results obtained from the Acoustic Camera can be analyzed through further computation given the impulse response in the time and frequency domain.

The type of sound source in a field measurement will determine the quality of the measurement. Several criteria for a desired impulse sound source for acoustic measurements can be described as follows: the source is preferred to be as omni-directional as possible in order to obtain reflections from all surfaces. The sound produced should have a sufficient SPL to provide decay curves with the required minimum dynamic range without contamination of background noise. The last criteria would be a source that provides a spectrum with sufficient acoustic output at the frequencies desired, and with reproducibility from shot to shot (Galloway *et al.*, 1955). Several studies have discussed comparison of sources that are available in room acoustics measurement. Watters (Watters, B.G., 1963) work is mostly used for the comparison of balloon to the other impulsive sources noted above. A study was recently done to evaluate the effect of sound source directionality to the results obtained in room acoustics computer simulation including the auralization output (Wang and Vigeant, 2008).

Based on past studies done by others and experiences during many experimental setups, there are several advantages and disadvantages to the use of a variety of sound sources for a room acoustics measurement. A clapper, hand-gun, balloon, yacht cannon, and dodecahedron[4] speaker are among the available sound sources that were compared in Table 1.2. Comparison is used to determine the source for field measurement in a variety of cases studied within this research and to comply with standard measurement in room acoustics. Based on this comparison, balloon burst is the most reliable source for the research with supporting evidence of this claim provided in Figure 3.20.

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[4] Dodecahedron speaker is a speaker with 12 sides created by 12 loudspeakers mounted together in a dodecahedron shape. The sound output is expected to be omni-directional source.

Table 1.2. Comparison of sound sources' capabilities for room acoustics field measurement.

Advantages	Clapper	Hand-gun	Balloon	Yacht cannon	Dodeca-hedron
An inexpensive source	X	X	X		
Easy to carry around	X		X		
Practical	X		X		
High reproducibility		X			X
Safety	X		X		X
A sufficient sound pressure level for the entire frequency band				X	X
A sufficient sound pressure level for mid frequency band		X	X	X	X
A sufficient sound pressure level for low frequency band		X	X	X	X
Suitable for structures with high attenuation and large physical dimensions				X	
No chance of clipping during measurement	X		X		X

1.4.4 Computer Simulation

The most common computational methods for simulating the propagation of sound through an environment are based on geometrical acoustic modeling (e.g., image source methods, ray tracing, and beam tracing). The source emission patterns, atmospheric scattering, surface reflectance (i.e., geometry, absorption coefficients, and diffusion coefficient), edge diffraction, and receiver sensitivity must be defined as mathematical objects of the input data required for computer modeling.

The diffuse reflected energy is modeled as radiating from a surface with a particular spatial distribution. In the majority of current geometrical room acoustics models, Lambert's law is used to determine this distribution of the diffuse energy (Cremer and Muller, 1982). Problems would occur if only part of the surface is illuminated, objects cast shadows on surfaces, or in the case of directional sources. A valid approximation for room acoustics simulation occurs by treating the sound sources as omni-directional sources since this provides the opportunity to obtain acoustical impacts from all surfaces.

The sound energy profile of a computer simulated impulse response can be quite different than an impulse response obtained from field measurement (Astolfi, 2005; Astolfi *et al.*, 2008; Astolfi and Pellerey, 2008). Several attempts however, have been done to eliminate these differences by improvements in the techniques and simulation algorithm (Wang and Vigeant, 2008). These past studies help with decision making on a variety of setups for simulation using current room acoustics software. Assurance concerning the algorithm used to compute parameters obtained from the simulation, such as the reverberation properties, is no less important.

Measurement sensitivity depends on the geometry of the case studied, and particularly on the number of microphones that fall inside the specular zone. Although, simplification of the modeling reduces the computational time, it has a downside to the accuracy of predicted room acoustics (Zeng *et al.*, 2006; Vorlander, 2008). Complexity of the acoustical model is relevant to the need of a sufficient number of sound rays in order to obtain a reliable simulation result.

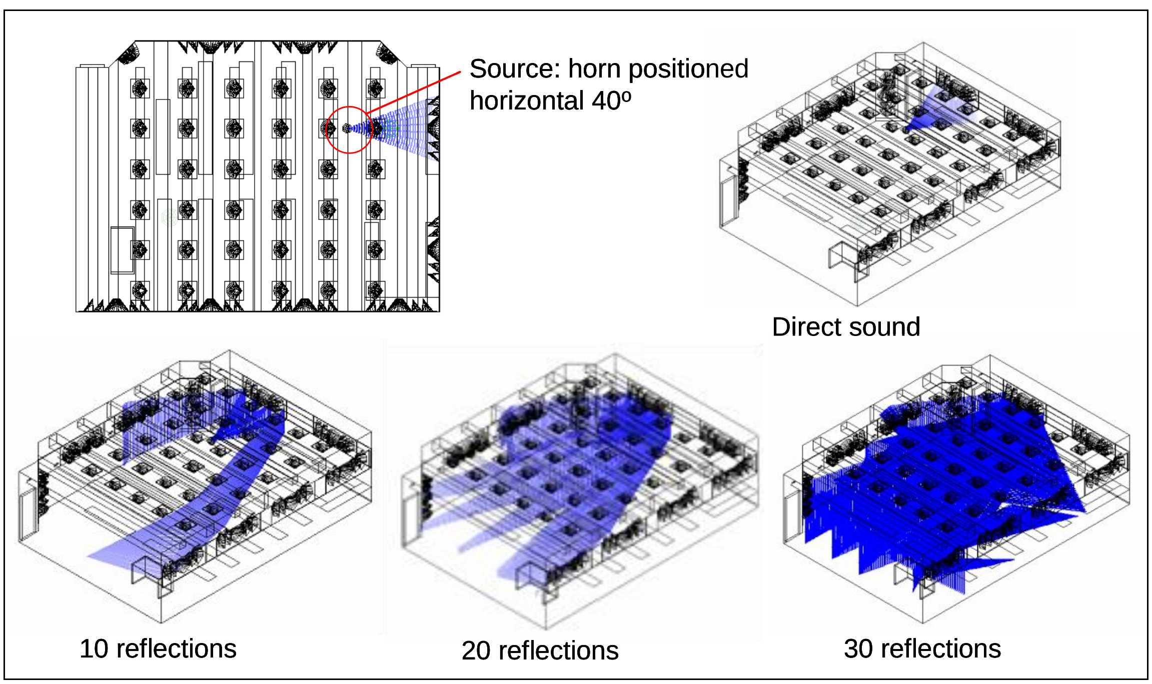

Figure 1.22. Different number of reflections in AC3 simulation.

Determination of the sufficient number of rays required to achieve a reliable simulation result depends on several factors, including the volume of the modeled space and the number of reflections observed. An example is shown in Figure 1.22. In order to observe reflections from diffusers on the side walls, more than 30 reflections were required for the ray tracing. This example is one of the preliminary results used to check the validity of the software and to determine the computer simulation setup.

Room acoustics computer software utilized within this research are EASE 4.3 and Ecotect. Many algorithms designed to obtain better prediction rates in modeling the diffuse reflections in computer simulation are based on geometrical room acoustics as demonstrated in a number of studies (Miles, 1984; Lam, 1996; Howarth and Lam, 2000).

1.4.5 Subjective Assessment (Listening Test)

Any objective parameters derived from impulse response measurement can represent an average impression of the room acoustics. However, the true auditory event is only covered through a full auditory experience of the space (Vorlander, 2008). Listeners might be seated in real space to evaluate the auditory event. They might also listen to audible numerical (i.e., simulated, measured, or synthesized) data that represent the actual acoustic conditions without being seated in the real space through a process known as auralization. The methods and techniques for the subjective assessment should consider all the principles in binaural hearing as described earlier in section 1.3.

1.4.5.1 Synthesizing the Auditory Stimuli

Basic methods and techniques to construct an auditory representation in a virtual environment are provided within this section. Discussion focuses on production of the binaural room-simulation, a system that creates a listening situation of the virtual environment given the modeled sound-field characteristics. The process can be conceived as being composed of two main operational parts, which are the sound-field modeling and auralization (Lehnert and Blauert, 1992).

In the binaural room-simulation, the auralization uses a simulated impulse response obtained from computer modeling and a sound recorded in an anechoic space to generate the auditory representation of a virtual space through a signal processing technique known as convolution (Vorlander, 1989). Details of the steps in the binaural room-simulation are shown in Figure 1.23.

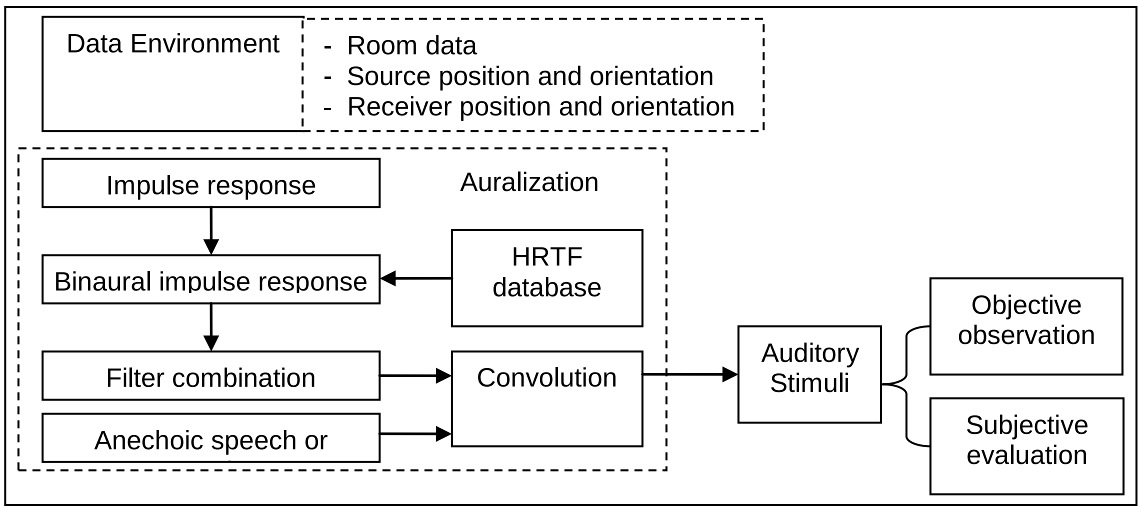

Figure 1.23. Diagram path of the auralization process within the computer simulation.

The advantage of auralization is that it permits easy and rapid variations of sound-field parameters and the immediate comparison of different room configurations. This is in spite of several potential response biases listed by Cremer for on-site subjective assessment within real spaces (Cremer and Muller, 1982). There are numerous uncontrolled variables in an on-site subjective assessment that may create response bias, such as the inconsistency of the stimuli or sound output (i.e., live performance of music or speech) from one experiment to another. The attempt to avoid response biases is associated with the selection of the data collection technique, which is discussed in the following section.

Optimization of the detailed procedure in binaural room-simulation is an important key factor in obtaining a “real” human perception in the virtual environment. Careful consideration of the physical properties of a modeled space is important since information of the sound field will be embedded within the simulated impulse response. The properties include the geometric and acoustic data of the surfaces in modeling and the position, orientation, and directional characteristics of the source and receiver. At the receiver point of view, application of a filter known as the head relative transfer function (HRTF) is required to transform a room impulse response (RIR) into a binaural room impulse response (BRIR). Here, the delay time of the sound arriving at the left and right ear and sound scattering due to the head, ear pinna, and upper torso are considered. This process creates a realistic condition as if the impulse response were recorded at the human ears.

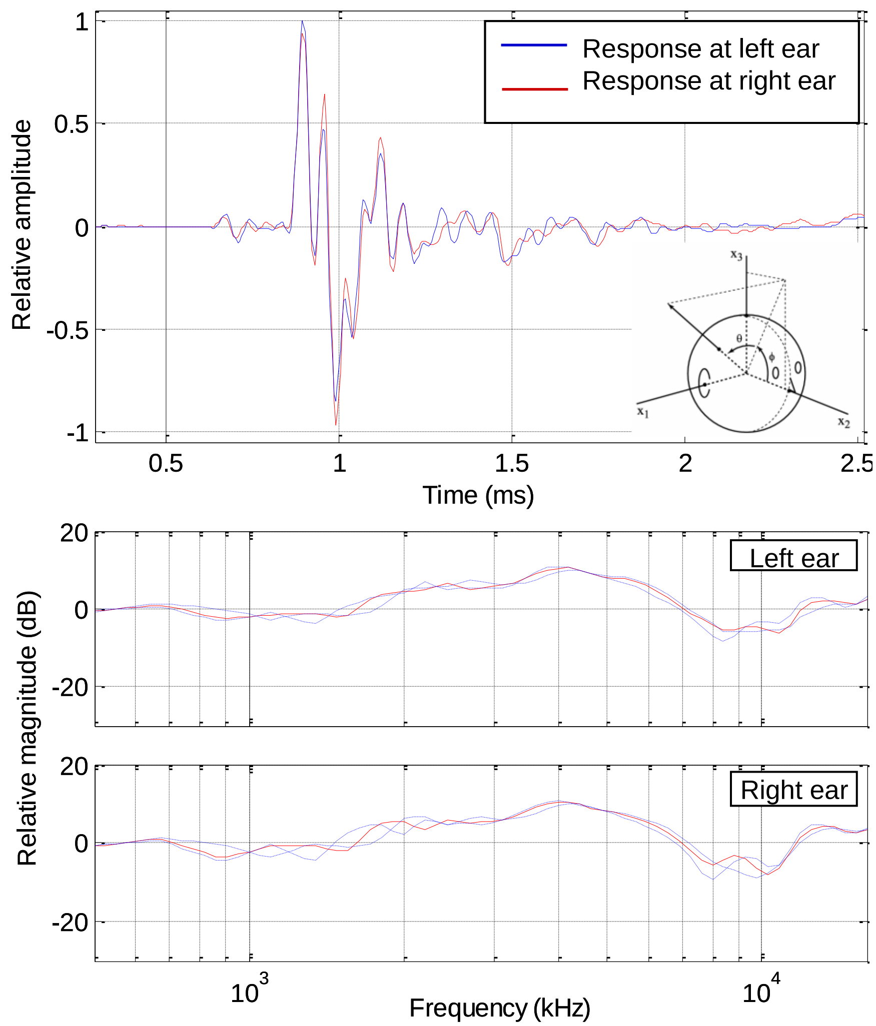

Figure 1.24. Example of a HRIR and HRTF data from CIPIC database.[5]

[5] The plotted graphs were generated from Matlab, using CIPIC database of HRTF and Matlab script to read the data, provided in the web link: http://interface.cipic.ucdavis.edu/sound/hrtf.html.

The HRTF or ATF (Anatomical Transfer Function) describes the amount of scattering by the listener's outer ears, head, and upper torso. It is the Fourier transform (i.e., frequency domain) of a head-related impulse response (HRIRs), an impulse response measured at the listener's ears from sources presented in an anechoic space. The scattering causes selective amplification or attenuation at certain frequencies, depending on source location. It is actually a filtering process of incoming signals at the left and right ears. It is defined by the sound pressure measured at the eardrum or at the ear canal entrance divided by the sound pressure measured with the microphone at the center of the head but with the head absent. The HRTF utilized within the computer simulation is using HRTF of the Knowles Electronics Mannequin for Acoustic Research (KEMAR) provided in EASE. The CIPIC, UC/Davis HRTF database (Algazi *et al.*, 2002) was used as a reference to observe the possibility of converting an RIR from the Acoustic Camera measurement into a BRIR (see Figure 1.24).

The ability to compare and interpret the time lag between the sounds reaching the right ear versus the left ear provides the localization cues (Zwicker and Fastl, 1999). More advanced techniques that provide the ability for real-time auralization have been developed by others (Funkhouser, Carlbom *et al.*, 1999; Lentz *et al.*, 2007). Some have studied the selections of system and technology based on physical design criteria for various applications such as navigation aids, virtual control rooms, integrated multi-modal virtual environment generators, and psychophysical research (Sahrhage, 1999; Lokki, 2000).

The final stage of auralization is the reproduction of a three-dimensional (3D) sound field for the listener. The sound reproduction utilizes 3D auditory display techniques that can be classified as: 1) binaural and transaural techniques, focused on recreating the sound field at both ears of the listener using headphones (binaural) or loudspeakers (transaural), and 2) multi-channel auditory displays, to construct a 3D sound field using an array of loudspeakers.

1.4.5.2 Design of the Data Collection Technique

The most common data collection techniques for psychoacoustics tasks in room acoustics are (Blauert, 2005): 1) ranking methods - stimuli are ranked in an order preferred from best to worst with respect to their acoustical quality; 2) the semantic differential - provides hints about what sounds are suitable to convey an intended message using certain adjective scales; 3) category scaling, where the most commonly used are 5- to 7-step scaling; and 4) magnitude estimation, where the subject indicates the discrepancy of a sound quality by using a reference sound assigned with a given magnitude.

Category rating judgment is frequently used in room acoustics studies. It relies on the relationship of a stimulus to a range of contextual values and also habits of biases governing the frequency with which different categories of the rating scale are used (Parducci and Perrett, 1971). Four potential causes of bias in the category rating are order effect, response range bias, anchoring the response bias, and grouping bias (Poulton, 1989).

The number of categories in a response scale must reflect the ability of subjects to use categories as well as the accuracy with which the recorded data represents the subjects' intended response. Heise used a seven-point range to demonstrate typical semantic differential scaling (Heise, 1969). More than seven categories tends to create confusion (Miller, 1994). Anchoring helps in any experiment with untrained subjects to develop internal criteria of the upper and lower ends of the response scale.

Different methods were explored and applied during the preparation stage of the subjective assessment. Methods ranged from on-site listening tests, computer-interface listening tests, Web-surveys, to the possibility of using an immersive virtual environment.

Avoidance of potential sources for data collection or survey errors is important (Schonlau *et al.*, 2002). Types of survey errors include coverage errors, non-response errors, and measurement errors. Given that the subjective assessment is a laboratory experiment set up with a sample of listeners recruited selectively, the data cannot be generalized to a larger population. The relevant potential data collection error is, therefore, the measurement error, which is an error that accounts for how far off from true values the respondent's answers are (Conrad *et al.*, 2008).

Sources of measurement errors in general arise from the effect interviewers have on respondents' answers, respondent-related error (e.g., problems in respondents' comprehension, memory, or judgment), the mode of data collection, and weakness in the wording of survey questionnaires (Groves, 2004). The latter is the most common source of measurement errors. Acousticians struggle to demonstrate how individual perceivers communicate internal experience to external observers (Blesser and Salter, 2006). Descriptive analysis is the most basic method, but it is still found to lack an intellectual foundation. Prominent labels of subjective attributes evaluating a typical acoustic space are then developed. Beranek showed that musical experts, at least within a shared time period, evaluated the quality of concert halls consistently (Beranek, 2004). A preliminary study was conducted to explore the effectiveness of several of the prominent labels in providing numerical answers associated with the acoustical condition measured (see section 2.4.2).

Samples were selected by randomization, and the potential for coverage errors was noticed. The result of a failure to include all units of the target population on the survey frame represents the coverage errors. It might create missing data reported as errors of non-observation. Important terms to understand concerning coverage error include "target population" and "frame population," described in Figure 1.25.

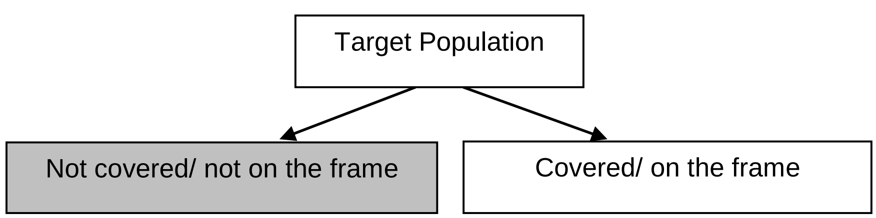

Figure 1.25. Types of coverage errors.

Target population is the set of units about which data are sought and inferences are to be made. Frame population is the set of units from which the survey sample is actually selected. Non-response error is the failure to obtain complete measurements on the survey sample. It is indicated by the non-response rate. Non-response rate is the proportion of non-response of the total sample. Having the response rate, one can measure the non-response rate and use it as an indicator to measure non-response error.

1.4.5.3 Sound Source Characteristics

Auditory perception is affected by the source characteristics in a number of ways. Careful attention to the characteristics of the sound source (i.e., the dry-signal) used in the reproduction of the auditory stimuli for subjective assessment is important. An example is related to the spectral filtering of the hearing system. The filtering system naturally selects the frequency bandwidth of the stimulus that can be heard and creates an auditory perception with a different level of sensitivity.

Most experiments of interaural time differences (ITD) are accurate at frequencies of 500 Hz or lower, while an interaural level differences (ILD) effect becomes more noticeable at high frequencies. There is a chance of exclusion of certain frequency ranges from hearing sensitivity, which might create ambiguity in sound localization. This phenomenon is due to diffracted waves at certain wavelengths or frequencies (i.e., an acoustical “bright spot”) caused by the presence of an object. This acoustical “bright spot” is also influenced by the dominance of the center frequency during the frequency channeling of the nerve system and the existence of a particular frequency range that creates the diffractive deception. As a consequence, careful attention to the source frequency components between the low-frequency region and the high-frequency region is required during spatial perception testing (Macaulay *et al.*, 2010).

The room reflections are important in creating a large ILD even at low frequencies. This is true especially when the reflections are coming from the same direction as the direct sound causing an acoustical phenomenon called the precedence effect (Litovsky *et al.*, 1999) or also known as the Haas effect. This acoustical phenomenon can be described by two conditions. First, human ears localize sound that is arriving first as a single auditory event despite the presence of another single reflection from a different direction. Second, other reflections arriving before 30 milliseconds (ms) are fused into the perception of the first arrival. Reflections that arrive after 30 ms will be perceived as echoes. This condition can occur, for instance, when reflective surfaces are applied at the backstage walls. Since early lateral reflections tend to delocalize sound, room geometry becomes an important factor. When reverberation and echoes are present, the onset and offset of a signal increases the accuracy of sound localization (Rakerd and Hartmann, 1986).

When multiple sound sources are presented, the loudness perception of an intensity level can be a good indicator to predict distance. The effectiveness of distance cues using source recognition also depends on familiarity with the sounds. In an enclosed space with sound reflections, the ratio and the time delay between direct and reflected sound provide cues to distance (Moore, 2003). The type of sound source and its angular position are also the other factors influencing distance cues. It includes the fact that acoustical characteristics of surfaces create reflections with various frequency components while a sound source also varies by its directivity properties. Combining information from all factors is suggested to obtain accurate judgments of distance (Zahorik, 2002).

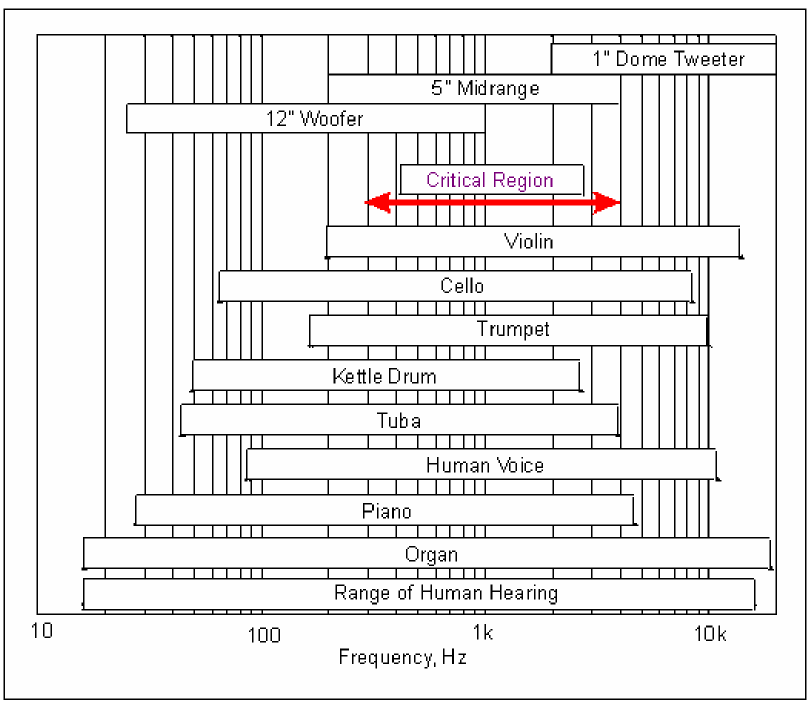

Figure 1.26. Frequency ranges of music and speech sources.

Frequency ranges of musical instruments relative to the range of human hearing and the critical region of hearing sensitivity is shown in Figure 1.26.

Some musical tone variables, such as tone onset, tone duration, and tone offset, affect sound localization. An abrupt offset results in small but measurable improvement in localization for long tone duration. As for tone onset, the onset rate has a greater influence than onset duration and becomes more significant when it occurs instantaneously (Boder and Goldman, 1942).

Speech intelligibility test materials are typically sentences, one syllable words, and random syllables that do not form words, with each type being increasingly more difficult to understand in the presence of noise. The most direct method of measuring intelligibility is to use sentences containing individual words. These can be presented at various levels in the presence of background noise or reverberation.

1.5 Research Objectives

As discussed earlier, acoustical treatments in current use rely on both absorptive and diffusive surfaces. Many variables however, are still in need of investigation to eliminate unwanted acoustical conditions without the use of electrical sound systems. Although methods and techniques for measuring the performance of diffusers are currently available, an adequate method to characterize the sound field with diffusion is still needed. The characterization is based on the fact that diffusion creates a variety of impacts to the early and late reflections. Furthermore, diffusion can be created by diffusive surfaces in the form of diffusers or other architectural elements within the enclosure space.

The solution for an adequate method to characterize the diffuseness of sound field in architectural spaces requires the use of an integrated method of objective measurements and subjective assessment. This study also aims to investigate architectural design variables that significantly impact the sound field with diffusion. Selection of the most appropriate indicators among currently available room acoustics parameters is as important as the understanding of the human audibility perception. The description of the human hearing system and its perception that specifically relates to the auditory perception in architectural spaces was discussed earlier, and entails some of the fundamentals in psychoacoustics.

Details of techniques used within the integrated method are presented in Chapter 2. A variety of architectural spaces have been selected as case studies relevant to acoustical function, the diffuser application, the room size, and the room shape and other architectural elements that are not considered as diffusers. Results are presented in Chapter 3 within sections categorized by these architectural design variables. Chapter 4 describes the analysis characterizing the acoustical conditions in order to determine the actual auditory impact of diffusers and other geometrical arrangements of architectural elements. The analysis focuses on the audibility conditions, and is based on the relationship among objective parameters and subjective attributes. Indicators are the degree of diffuseness, the energy based parameters, and the associated auditory perception. In Chapter 5, important key findings are listed and implemented into the draft of the guideline for diffusion control presented along with examples of architectural design applications. Chapter 6 describes several conclusions and ideas for future work.

1.6 Outcomes and Contribution

Upon completing the research objectives, the research outcomes provide evidence for architects and acousticians about benefits, impacts, and results of acoustic treatments utilizing diffusion control with different geometrical sizes and shapes of rooms.

Examples of the architectural spaces investigated in this study are chosen to provide the framework of the capabilities and procedures of the integrated method applied in characterizing the sound field with diffusion of architectural spaces. The different cases studied, with a variety of design alternatives, can be analyzed using the field measurement with a spherical-microphone array system, computer simulation, and subjective assessment. This provides a large variety of information and analysis capabilities that can assist and accelerate the design decision-making process for an appropriate diffusion control system. Guidelines in architectural design can be developed as a future work to help architects create a better auditory space.

The integrated method capabilities allow designers to employ a rich array of data analysis techniques to observe the diffuseness of a sound field, the acoustical and audibility quality within it, and to identify architectural elements including diffusers that most effectively impact the room acoustics characteristic.

Research in spatial perception benefits from the use of this integrated method, given the ability to stimulate audio and visual conditions simultaneously in the virtual environment. Other advantages of this integrated method, for example in psychoacoustics studies, include the ability to control environmental variables, repeatability, and the prevention of hazardous conditions from uncontrolled stimuli within real spaces.

Chapter 2

Technique Details of the Integrated Method

The audibility characteristics of the sound field are derived from the relationship between objective parameters and subjective attributes. Deriving the relationship requires subjective assessment of noticeable differences in the loudness, clarity, reverberation perception, and the ability of sound localization, in addition to the field measurement and computer simulation. A detailed algorithm of the objective parameters and steps to process the impulse responses measured is provided within this chapter.

Observations from the selected case studies are described along with the experimental setup of both field measurement and computer simulation. Computer simulation emphasizes the objective to explore a variety of geometric and acoustic configurations with the diffusers applied. Owing to the limitation on the accessibility of certain spaces and some technical issues, the field measurement using the Acoustic Camera was not conducted in all cases studied. Validity and accuracy of the integrated method were examined through several preliminary studies that will be discussed in the following sections.

2.1 Data Processing of Impulse Response

The techniques and principles utilized in this research are based on the standard measurement of room acoustics (ISO, 1997; 2008; 2009). The multi-microphone array of the Acoustic Camera and the computer simulation provide the ability to simultaneously measure the impulse responses that arrive from many directions, a condition expected in sound fields with diffusion. This section conveys the detailed calculation of the objective parameters. It also demonstrates steps of the raw data (i.e., impulse response) processing for analysis of the sound-field diffuseness. From this point, the field measurement result is referred to as the measured impulse response while the computer simulation result is the simulated impulse response.

It is difficult to interpret the sound-field diffuseness by using the coherence of measured impulse responses alone. One theoretical assumption states that an indicator of a diffuse sound field is if the reverberant fields are the same at any position; for this reason, coherence of the late reflections (Coh_{late}) of impulse responses was employed. This is based on the principle that coherence is a measure of similarity of the properties of two signals. The boundary of the sound field is defined by the spacing between microphones. The coherences calculated from the measured impulse response are obtained from two microphones with opposite positions on the Acoustic Camera. The radius of the 48-channel sphere-microphone array is 35 cm, while the 120-channel radius is 60 cm. Therefore, the sound field observed is also within the system's radius. The microphones are labeled with numbers that can be identified during the post processing of data.[6] Coherences calculated from three pairs of microphones were used for the analysis: first is a pair parallel to the length of the space identified as the front and rear microphones; second is a pair of microphones measuring incoming sound from left and right walls; and third is a pair facing up (top) and down (bottom).

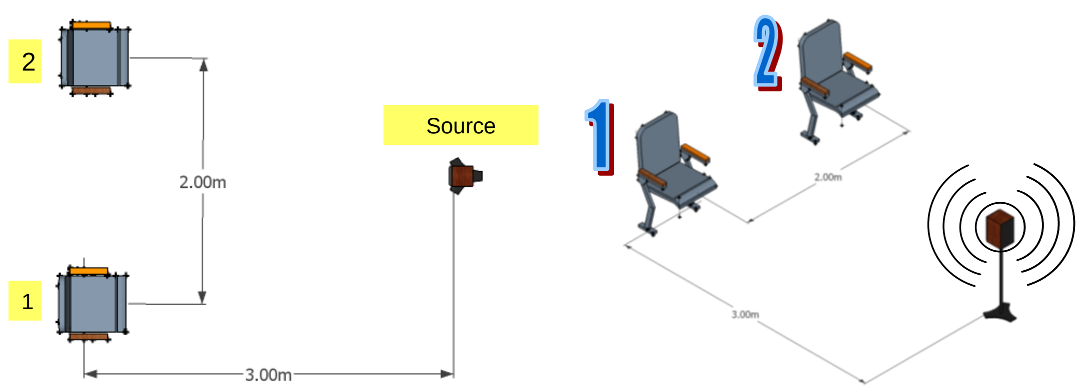

Figure 2.1. The main distances between source and receiver and between receivers in the room data of the computer modeling for all cases studied.

[6] Data being processed from the sphere-microphone array system are the measured impulse responses. They can be obtained from the measurement system through the Noise Image software and exported as separate *.wav* format audio files for each microphone.

Meanwhile the sound field boundaries in the computer models are determined by the spacing between receivers (seats). At least two receivers are assigned to each of the spaces for calculating the coherences (see Figure 2.1). The receiver-to-source distance did consider the critical distance.[7]

In section 1.2 it is mentioned that the coherences are calculated using the entire (Coh_{entire}), early (Coh_{early}), and late (Coh_{late}) portions of impulse responses in octave frequency bands. A detailed calculation of the coherence using the impulse response can be described with the following equations:

$$Coh_{entire}(t) = \frac{1}{T_D \sqrt{P_{up} P_{down}}} \sum_{t=0}^{T_D} E_{up}(t) E_{down}(t) \tag{2-1}$$

$$P_{up} \rightarrow P_{E,up} = \frac{1}{T_D} \sum_{t=0}^{T_D} E_{up}(t)^2 \; ; P_{down} \rightarrow P_{E,down} = \frac{1}{T_D} \sum_{t=0}^{T_D} E_{down}(t)^2 \tag{2-2}$$

The degree of diffuseness is observed in a frequency-dependent manner. Instead of using a time response, the correlation function uses a frequency response since it is easier to measure and provides a better predictor for characterizing the diffusion (Schroeder, 1962). The coherences are calculated using the entire, early, and late portions of the impulse responses for octave bands of 63 Hz to 8 kHz. Using the frequency response, the envelope energy in equation (2-2) is replaced by,

$$F_{up}(f) = fft(E_{up}(t)) \; ; F_{down}(f) = fft(E_{down}(t)) \tag{2-3}$$

where, F_{up} (f) is the frequency response obtained from fast Fourier transform of the time response. The coherence of the entire impulse response (Coh_{entire}) at octave band *f* Hz, is the average Coh_{entire} (f) for that particular octave band.

An example of an impulse response plot of two microphone outputs measured at a pair of microphones in Dennison Hall room 170 (DH170) is shown in Figure 2.2. The coherence for the entire impulse response at an octave band of 1000 Hz is 0.82, indicating a high similarity between the signals measured at

[7] See Figure 1.11 and the detailed discussion in section 1. 2.

the microphone facing up and facing down, given 0 as no correlation and 1 as the maximum correlation value. Fourier transform of the impulse response was done in Matlab. The microphones in this sphere-microphone array were 60 cm apart, which defined the sound-field boundaries.

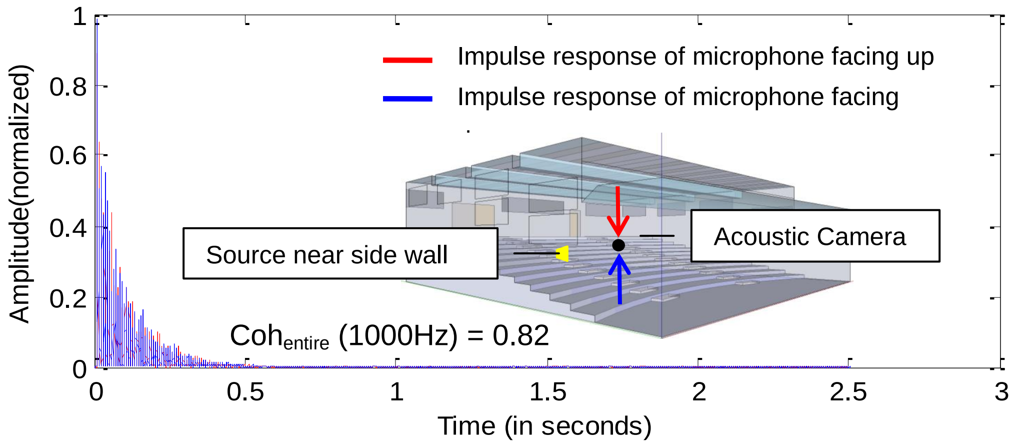

Figure 2.2. Coherence for the entire impulse response (Coh$_{entire}$) at octave band 1000 Hz in Dennison Hall room 170 (DH170), using the Acoustic Camera top and bottom microphones' output.

Hidaka *et al.* have summarized several past studies regarding the boundary point of early and late reflections based on subjective attributes (Hidaka *et al.*, 2007). Based on these references, 80 ms is chosen as the boundary point for the early response.

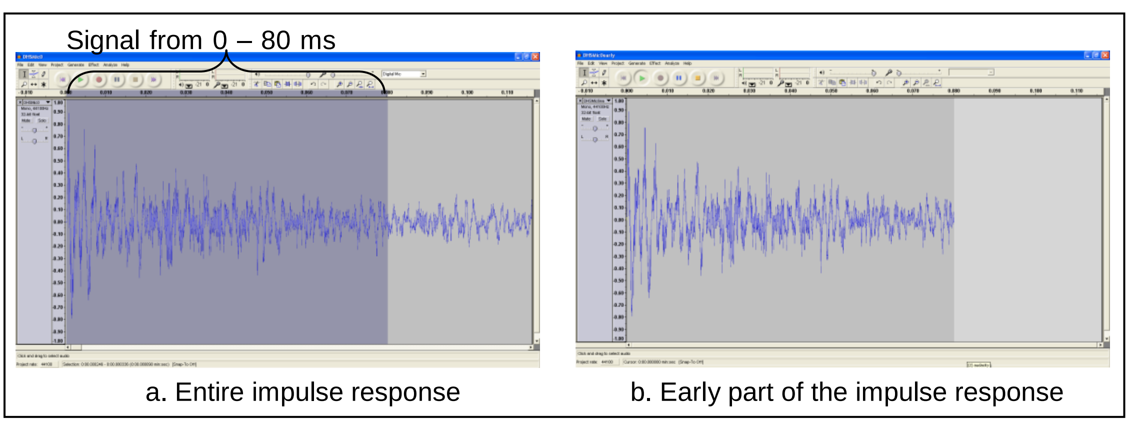

Figure 2.3. Processing the signal to define boundary point of the early reflections (80 ms).

Coherences of early reflections are utilizing the 0 to 80 ms portion of impulse responses. Late reflections are the portion of impulse response that reaches the listener 80 ms after the process of decay has begun. The early and late energy were separated by tracking the time length of the impulse response using Audacity[8] (see Figure 2.3) and Matlab.[9] Using the same impulse responses in Figure 2.2, the coherence of early reflections (Coh_{early}) value was 0.84 for octave band 1000 Hz, which indicated a high similarity for the early reflections. The late reflections of the two impulse responses were less similar with a coherence of late reflections (Coh_{late}) value of 0.77; the impulse responses and coherence plot are illustrated in Figure 2.4.

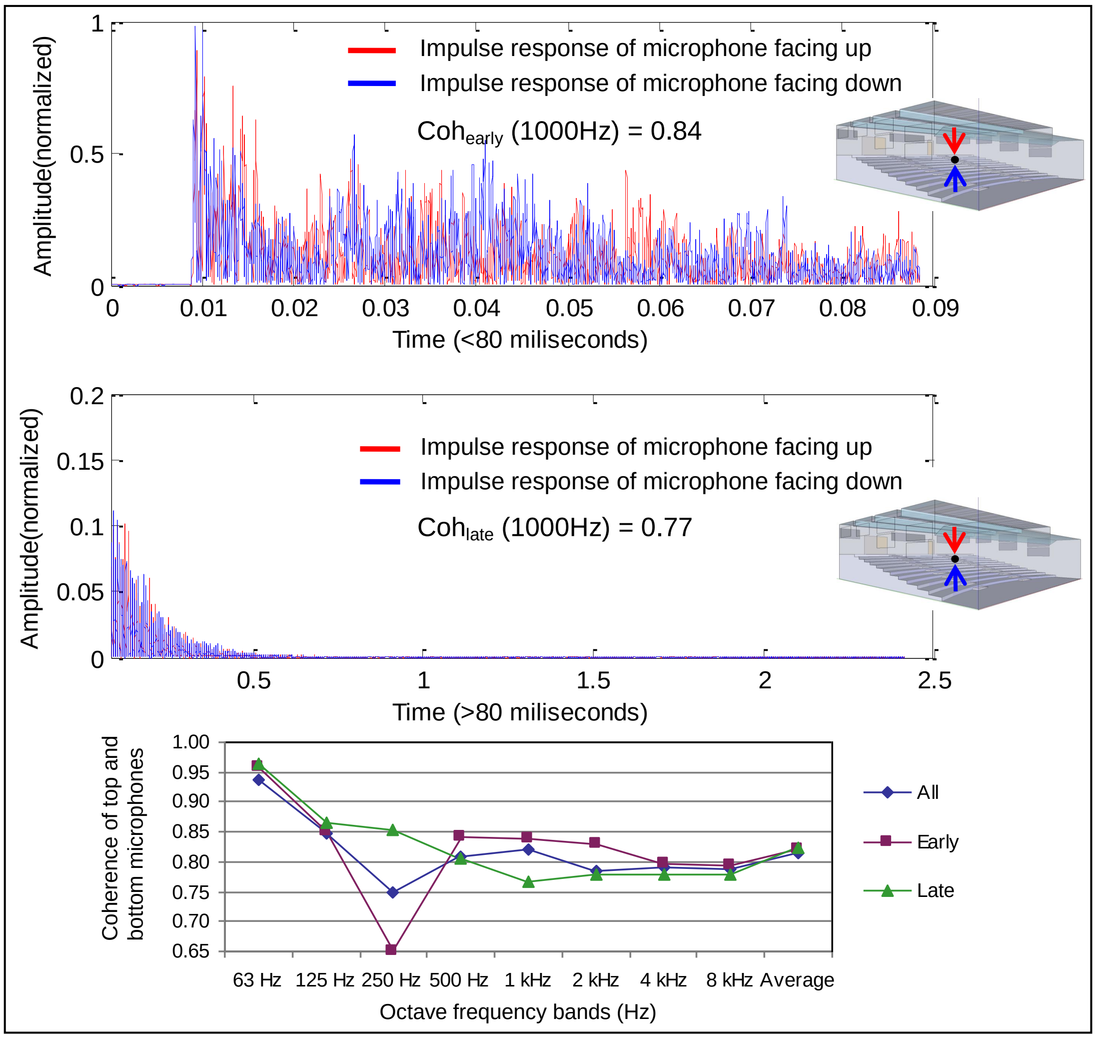

Figure 2.4. Coherence of the early reflections (Coh_{early}) and coherence of the late reflections (Coh_{late}) using the same microphone data in Figure 2.3.

[8] Audacity is a free signal processing tool.
[9] Matlab® (matrix laboratory) is a numerical computing environment and programming language.

A Coh_{early} value of nearly 1 indicated similarity between the early reflections of the two impulse responses, but this value alone is inadequate to describe diffusion occurrences for early reflections. The analysis should proceed with an observation on the possibility of the comb-filtering effect. Parameters that can confirm the findings are the early decay time (EDT) and the clarity index. Meanwhile, a Coh_{late} value of nearly 1 indicates a diffused sound field and, therefore, directly denotes the diffusion occurrences within this space for late reflections.

Every measurement has a starting time of a data recording before the direct impulse is measured or before the impulse sound is emitted. This portion of the measurement contains the information of the ambient noise of the space and equipment. The energy of this portion is often very small, and it can be neglected and considered as zero response (no impulse emitted). Details of the processing in Matlab to identify the boundary point of the useful energy is shown in Figure 2.5 using the entire impulse response, including an example that was illustrated in Figure 2.3.a. Calculation of certain objective parameters requires a technique such as the clarity index (see section 2.1.2).

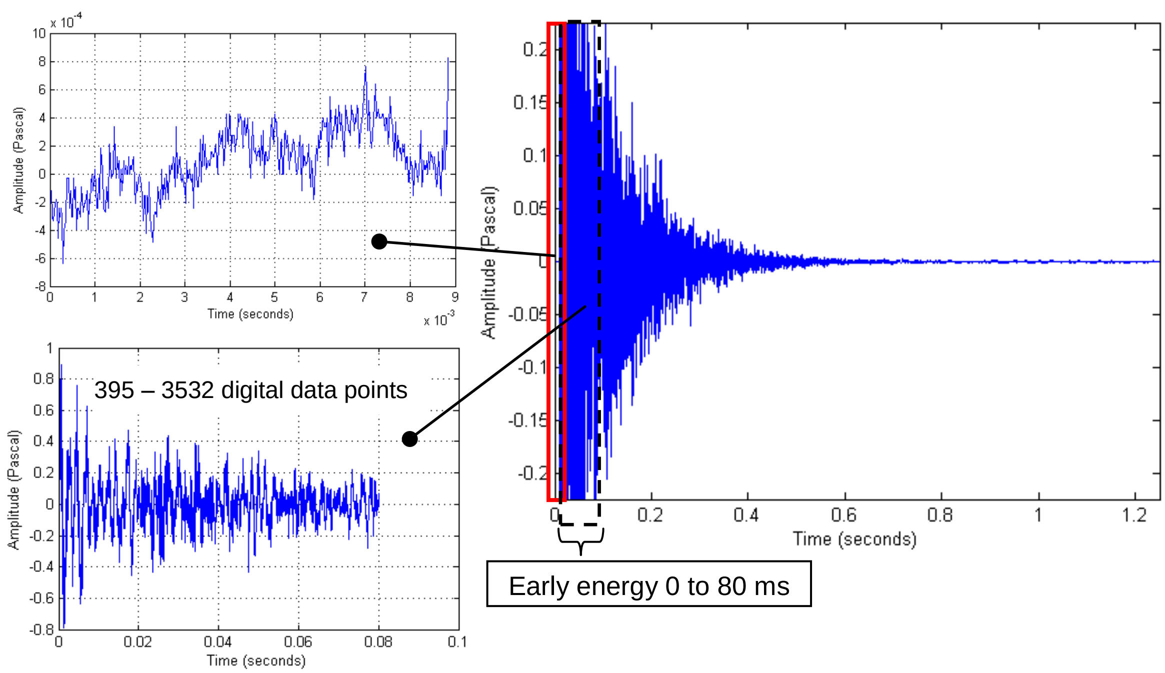

Figure 2.5. Truncation of early reflections of the entire impulse response in Matlab for calculation of Coh_{early} and Coh_{late}.

Most of the objective parameters are calculated for each octave frequency band. An example of octave band filtering using Matlab is shown in Figure 2.6. Details of the technique are provided in another publication (Utami, 2009).

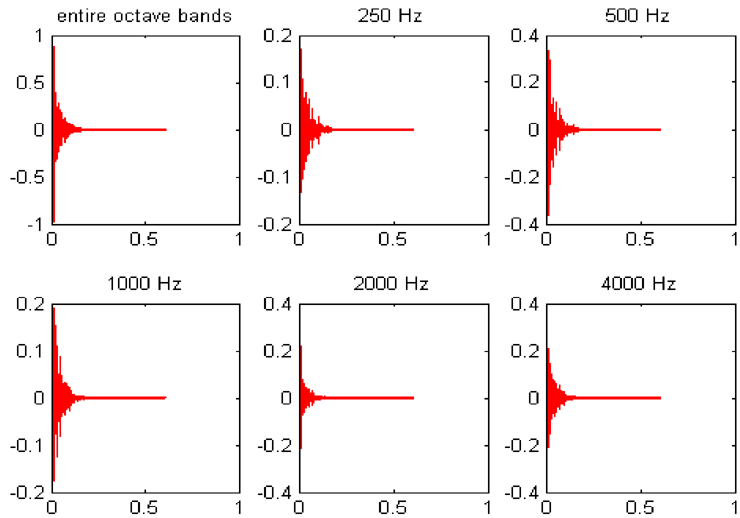

Figure 2.6. Impulse response filtered in octave bands using Matlab.

2.1.1 Total SPL of the Reflected Sound

Total sound pressure level (SPL) measures the intensity level of the incoming sound on the receiver (i.e., microphone) in decibels (dB) due to the direct and reflected sound. The intensity level is relevant to the subjective attribute of loudness. Loudness at a particular listener's position has two components, the early sound and the reverberant sound. Loudness of the early sound is determined by the energy of the sound that comes directly from the source, plus the energy received in the next 80 ms from the early reflections. Loudness of the reverberant sound is defined by the total sound energy that reaches the listener 80 ms after the process of decay has begun.

Sound absorption in a space is predicted by the total Sabin, given the surface area and its absorption coefficient. Therefore, the ratio of the diffuser to the total surface area enables a description of the amount of absorption contributed by the diffuser's surfaces. The ability to isolate the reflected component is essential to evaluate the amount of absorption in different spaces by having the same source-to-receiver distances and source characteristics. The amount of absorption or reflection in a space can be estimated by subtracting the direct SPL from the total SPL (i.e., direct + reflective SPL). The residual is the total reflected SPL.

2.1.2 Clarity of Speech (C50) and Clarity of Music (C80)

Clarity index is defined as the ratio of early sound energy (i.e., mean-square pressure) to later reverberant energy. It is the square of the ratio of the mean-square sound pressure to the reference mean-square sound pressure of 20 μPa, the threshold of human hearing (Kinsler, 2000). In principle, for a room with reverberant condition, the early reflections are useful in improving the auditory quality. On the other hand, late reflections that arrive after the critical delay time are often creating a detrimental effect. The common critical delay time for speech is 50 ms, and for music perception it is 80 ms, which refers to the Clarity of Speech (C_{50}) and Clarity of Music (C_{80}), respectively. Calculation of the clarity index refers to Equation (2-4) by,

$$C_x = 10\log\left\{\frac{\int_0^x \left(p^2(t)dt\right)}{\int_x^\infty \left(p^2(t)dt\right)}\right\};\ C_{80} = 10\log\left\{\frac{\int_{n=1}^{N_1} {p_n}^2}{\int_{n=N_1+1}^{N_2} {p_n}^2}\right\} \tag{2-4}$$

where p_n is the instantaneous pressure in the room impulse response at n sample number of the discrete signal. This formulation is for sampled data, where n is the sample number starting with n = 1 when the direct sound arrives. The value N_1 is the sample number 80 ms after n = 1, i.e., N_1 = 0.08 x *fs*, where *fs* is the sampling frequency. The value N_2 is the total number of samples used in the impulse response. Average values of the octave bands are commonly used as given in the equations below (Marshall, 1994).

$$C_{50} = 0.15 \cdot C_{50}^{500\,Hz} + 0.25 \cdot C_{50}^{1000\,Hz} + 0.35 \cdot C_{50}^{2000\,Hz} + 0.25 \cdot C_{50}^{4000\,Hz} \tag{2-5}$$

$$C_{80} = \frac{1}{3}\left(C_{50}^{500\,Hz} + C_{50}^{1000\,Hz} + C_{50}^{2000\,Hz}\right) \tag{2-6}$$

By comparing the clarity index for speech or music, the differences in the amount of energy of the early reflections within two different spaces can be identified as shown with the example in Figure 2.7.

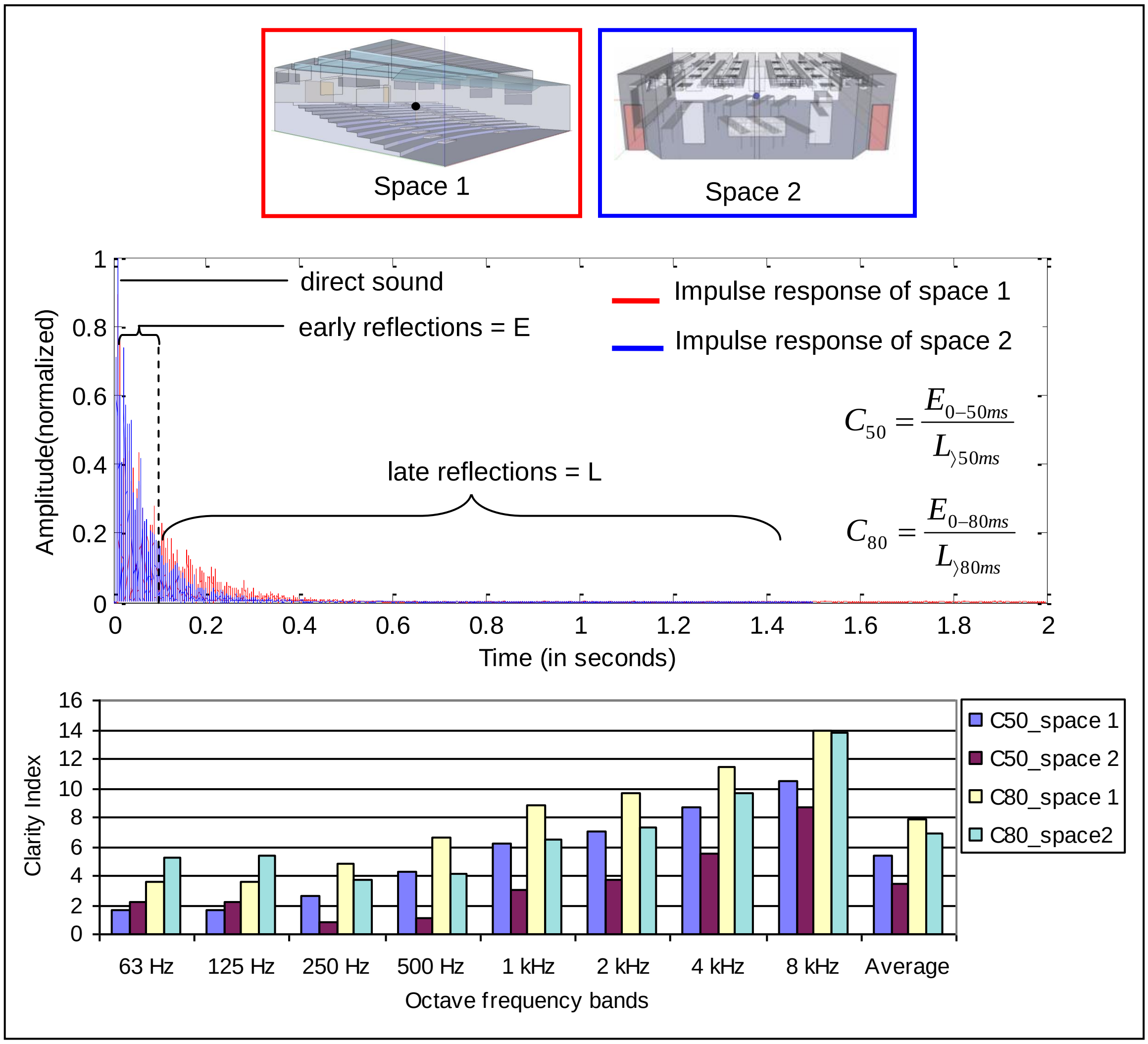

Figure 2.7. A comparison of clarity index for music (C_{50}) and speech (C_{80}) of two different spaces.

2.1.3 Reverberation time (T_{30}) and EDT

After a sound source is turned off in a “live” room, a noticeable time elapses before the noise becomes inaudible. This noticeable time is the reverberation time, defined as the time in seconds required for decaying sound to decrease in level by 60 decibels (dB). The less sound absorbing materials there are in a room, the longer the sound takes to die away.

Reverberation time was first introduced by W.C. Sabine (1898 – 1919) who described the relationship among the reverberation characteristics of a room, the size, and the amount of absorbing material present (Sabine, 1922).

The first reverberation time was calculated with $\overline{\alpha} < 0.2$ and free from pronounced focusing effects given in equation (2-7) (Knudsen, 1932).

$$T_{60} = 0.161 \frac{V}{S\overline{\alpha}} \tag{2-7}$$

where, T_{60} is the reverberation time in seconds, V is the room volume in cubic meters, S is the surface area of the room in square meters, $\overline{\alpha}$ is the area average random incidence energy absorption coefficient.

It is intended to be used for sound fields where all directions of a sound propagation contribute equal sound intensities in steady state conditions and at each moment during the decay of a sound field. The Eyring-Norris formulation was derived by assuming that the intensity of sound in a room, during growth, steady state, or decay, is given by summing up the contributions of radiant sound energy from all possible image sources (Kinsler, 2000). It is given by,

$$T_{60} = -0.161 \frac{V}{S \ln(1 - \overline{\alpha}_E)} \tag{2-8}$$

Equation 2.7 and 2.8 predict the reverberation time, given the acoustic properties of the room. It is clear that the total absorption must depend on the areas and absorption coefficient of all the surface materials within the room, therefore, the form of this dependence is subject to a variety of simplifications and assumptions.

The reverberation time of a room can be derived from the corresponding impulse response measured at a single place. It is based on the analysis of the decay process by evaluating the decay curve. Using the decay curve of the impulse responses, reverberation time of T_{30} was utilized in this research. An octave band filter processed the impulse response to define the frequency discrimination and hence yields the frequency dependence of the reverberation time. An example is shown in Figure 2.8 for calculating T_{30} at 1000 Hz octave band. The T_{30} is the 60dB decay time calculated by a line fit (i.e., line fit is the red line shown in Figure 2.8) to the portion of the decay curve between -5 and -35dB. A sufficient signal-to-noise ratio (SNR) of 35dB or larger should be provided in order to calculate T_{30} (ISO, 2008).

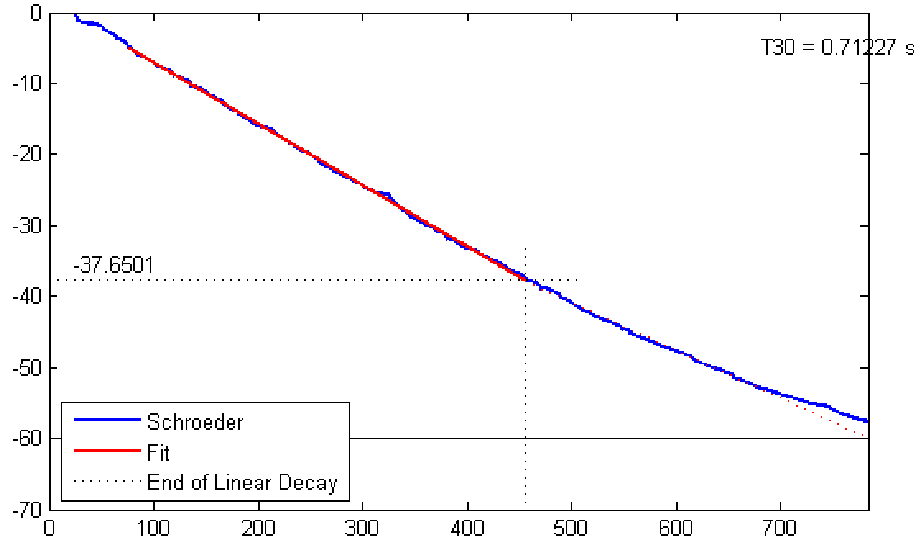

Figure 2.8. Reverberation Time (T_{30}) calculation in Matlab using the Schroeder integrated method.

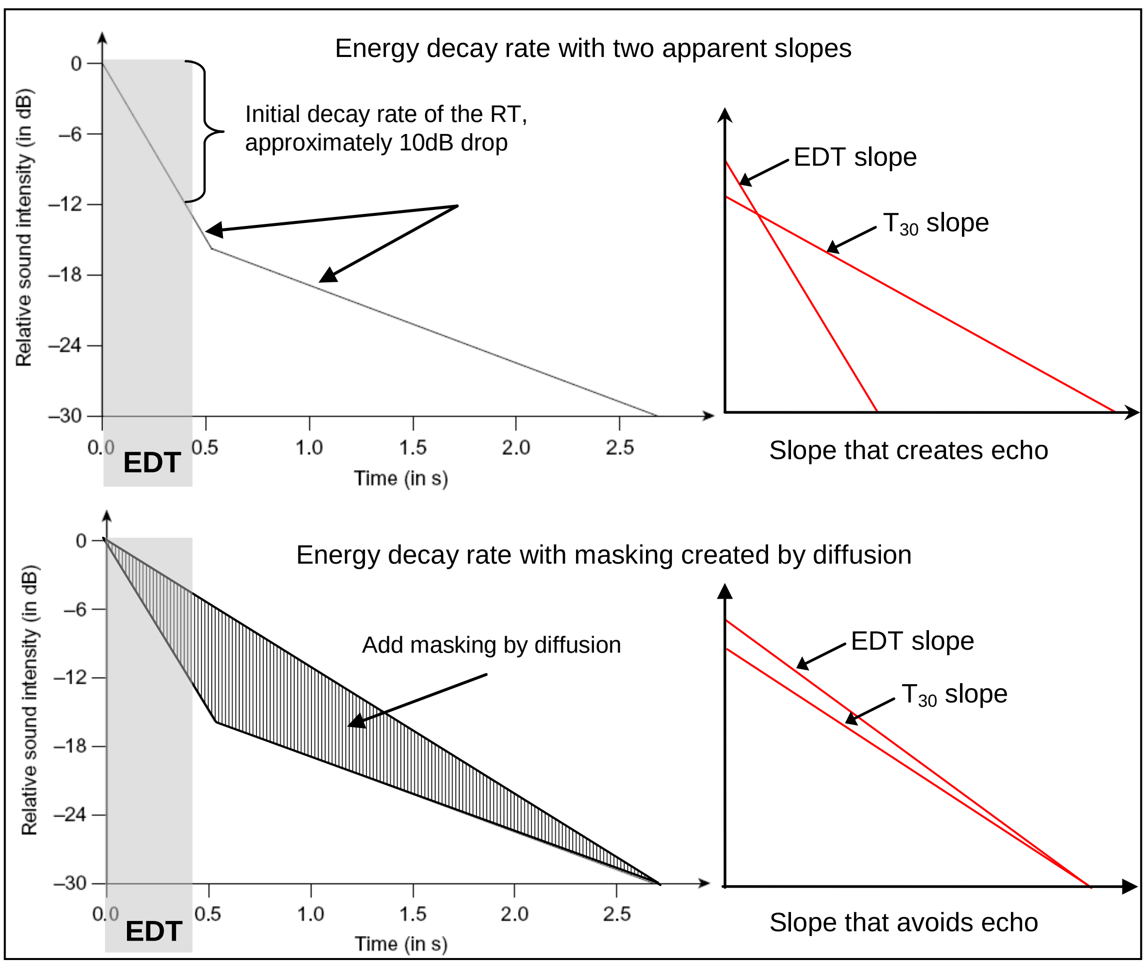

Figure 2.9. Analysis of diffusion occurrences using the slope shape of the Early Decay Time (EDT) and reverberation time (T_{30}).

For the calculation of T_{30}, the Schroeder integrated impulse response function[10] is used since it provides the ability to add a longer reverberation tail for the case of insufficient length of time of impulse data (Schroeder, 1965; 1979b). An example is shown in Figure 2.8 for calculating T_{30} at 1000 Hz octave band. The early decay time (EDT) is the 60 dB decay time calculated by a line fit to the portion of the decay curve between 0 and -10 dB.

2.1.4 Interaural Cross Correlation (IACC) and Listener Envelopment (LEV)

The boundary of sound-field diffusion might be within the microphones' spacing distance. The smallest sound field to be observed is the incoming sound measured at the ear. The degree of similarity of the waveforms at the two ears is the basic cue for binaural processing, detection, and localization of sound in an architectural space.

A physical measure of similarity is the interaural cross-correlation coefficient (IACC), defined as the cross-correlation coefficient of the signals at the two ears. It is calculated using the recording output of two microphones located at the entrances or the ear canals of a person or a dummy head.

This parameter appeared to be valuable for determining the degree of sound-field diffusion along the low-frequency level (G_L) in the frequency range from 100 to 3000 Hz for symphonic music in concert halls (Hidaka *et al.*, 1995). A lower value of IACC indicates that there is less correlation between the sounds at listener's ears than for a higher value of IACC.

The late part of the IACC (i.e., $IACC_L$), to which the late reflections contribute, is used to describe the listener envelopment (LEV). Listener envelopment represents the perception that the reverberant sound seems to arrive at the ears equally from all directions, which is a representation of sound-field diffusion. A high value of LEV indicates the subjective impression of a

[10] Schroeder integration method: using it in a single measurement, yields a decay curve that is identical to the average over infinitely many decay curves that would be obtained from exciting the enclosure with bandpass-filtered noise.

listener that he or she is enveloped by the sound field (Okano *et al.*, 1998). This condition was found in the best halls, giving the listener the feeling of being immersed in the sound, and therefore, is considered as an important subjective attribute by Beranek (Beranek, 1962).

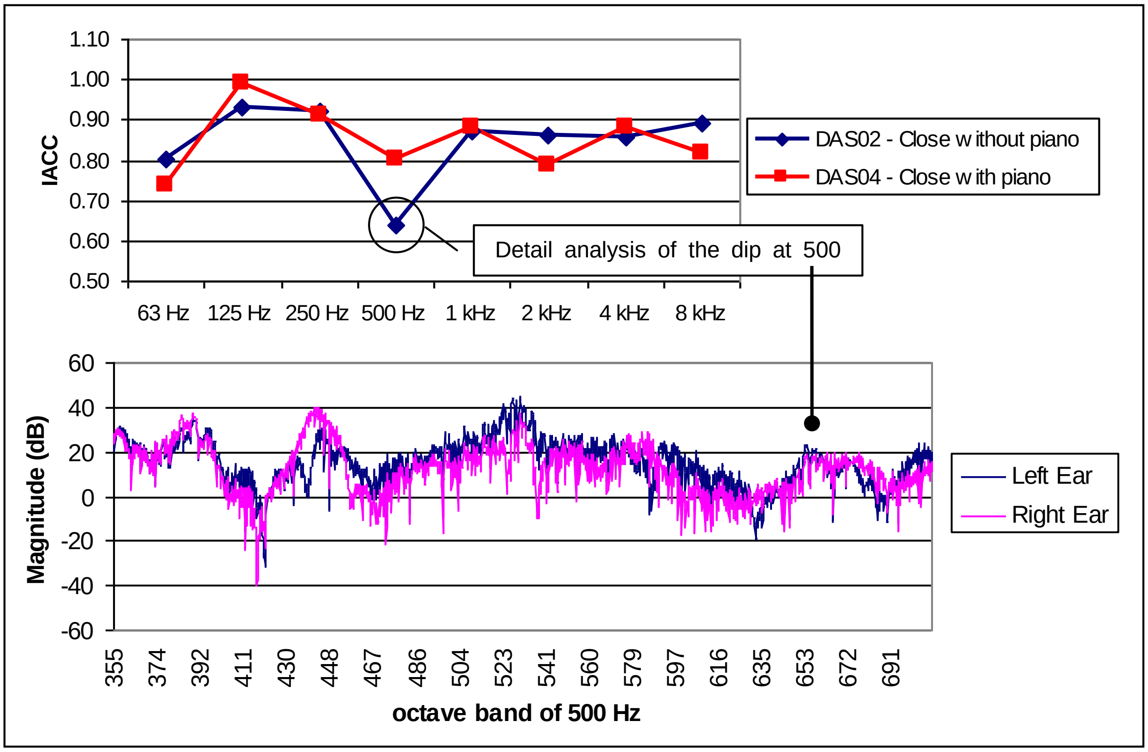

Figure 2.10. Comparison of Interaural Cross-Correlation (IACC) of listeners from the computer model of Duderstadt Audio Studio (DAS) without piano and with piano present.

The way to quantify LEV was first introduced through an experiment conducted at the Communication Research Center in Ottawa, Canada (Soulodre *et al.*, 2003). Along with the $IACC_{L,}$ the LEV is shown for most concert halls to be directly related to the late mid-frequency value of strength factor (G_L). A revised version of their formula is utilized here (Beranek, 2011),

$$LEV_{calc} = 0.5G_{Late,mid} + 10\log\left[1 - IACC_{Late,mid}\right] \tag{2-9}$$

and,

$$G_{Late,mid} = G - 10\log\left(1 + \log^{-1} C_{80}/10\right) \tag{2-10}$$

where, LEV_{calc} and $G_{Late,mid}$ are both in dB while C_{80} is unitless. The middle frequencies (i.e., *mid*) here use the average of the value at 250 to 4000 Hz bands. The strength factor (G) is a measure of the sound pressure level at a point in a hall, with an omni-directional source on stage, minus the sound pressure level that would be measured at a distance of 10 m from the same sound source operating at the same power level, and located in an anechoic chamber (Okano *et al.*, 1998).

2.2 Description and Experimental Setup of Cases Studied

Cases studied within this research are architectural spaces that can be described as a three-dimensional extension of the world around us, the intervals, distances, and relationships between people, people and objects, and between objects (Altman, 1980). The three-dimensional space is created by using a specific material and element to serve certain functions and to be experienced by human senses, which for this study is the sense of hearing.

The cases studied were selected based on the acoustical function, diffusers applied, room size, and room shape. The acoustical function varied from a recording studio, classrooms, and a concert hall to sport facilities. Each type of space requires a unique acoustical condition to support the activities within it. Owing to these activities, acoustical treatment with diffusion is often considered crucial to providing good acoustical quality.

All the cases studied listed in Table 2.1 were located at the University of Michigan, except the Detroit Orchestra Hall.

Table 2.1. Description of the cases studied.

	ID	Room (Acoustical Function)	Wall Shape	Width (m)	Length (m)	Height (m)	Volume (m^3)	Capacity
1.	AA21	Classroom R1221 Art&Architecture	Flat parallel	6.1	10	2.7	165	40
2.	DAS	Duderstadt Audio Studio	Uneven	7.5	11	4	266	-
3.	AA16	Classroom R2216 Art&Architecture	Flat parallel	8.8	11.8	3.4	332	80
4.	DH170	Lecture Hall 170 Dennison	Flat parallel	16.8	18.3	4.8	1513	270
5.	AHA	Lecture Hall A Angell Hall	Curve	17.4	20.7	4.9	1530	275
6.	DOH	Detroit Orchestra Hall	Curve	24	26	14	8895	2014
7.	CRI	Crisler Sport Arena	Ellipse	94.5	113.4	30.3	221559	13751
8.	BH	Big House Football Stadium	Ellipse	237	302.4	30	1999666	106201

The rooms' sizes vary from 165 m^3 to almost 2,000,000 m^3 with a variety of shapes ranging from flat parallel walls of a rectangular room to an ellipse or curved wall, which creates a bowl-shaped space.

The diffusers were present in several existing (as-is) conditions of these spaces. The characteristics of the diffusers in these spaces are described in Figure 2.11. Illustrations of the room geometry indicating the volume ratio from the smallest to largest space are provided in Figure 2.12.

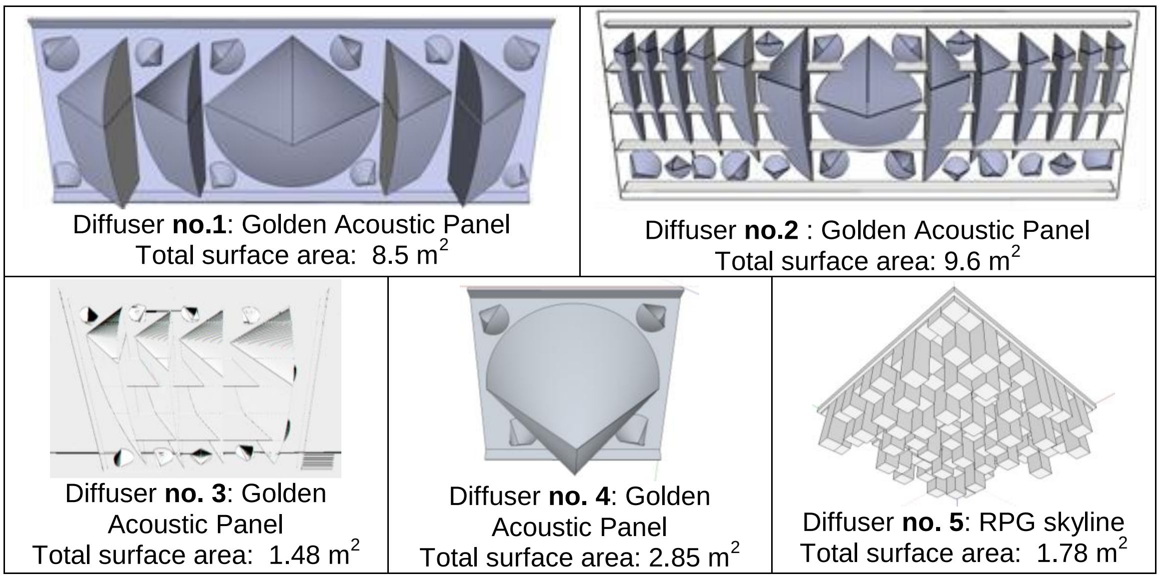

Figure 2.11. Illustration of the diffusers in use in spaces studied in this research.

The room dimensions, surface areas, and materials applied were estimated from an observation within the real space. The information was used as the simulation room data. Simulated impulse responses are obtained from EASE 4.3, which is based on the hybrid method, a combined method of image source models and ray tracing method. Briefly, this method can be described as running a specular ray tracing process to find a receiver hit by a ray. As a result, the corresponding image source must be audible. To observe the room geometry and to find possible reflection sequences, the ray tracing method is applied using Autodesk® Ecotect® Analysis software. The steps for simulation in EASE can be described as follows:

1. Computer aided design (CAD) drawings of the space are made implementing geometrical data from observation of the existing space.

2. Surface detail geometry, absorption coefficient, scattering coefficient, source, and receivers (seats) are inserted as the simulation room data.
3. Acoustics simulation is conducted using the EASEaura module, room impulse response on probe, and binaural impulse response measurement with the head related transfer function (HRTF) as described in section 1.4.5.
4. Data of the objective parameters measured is collected.
5. Auralization of simulated spaces is assessed.

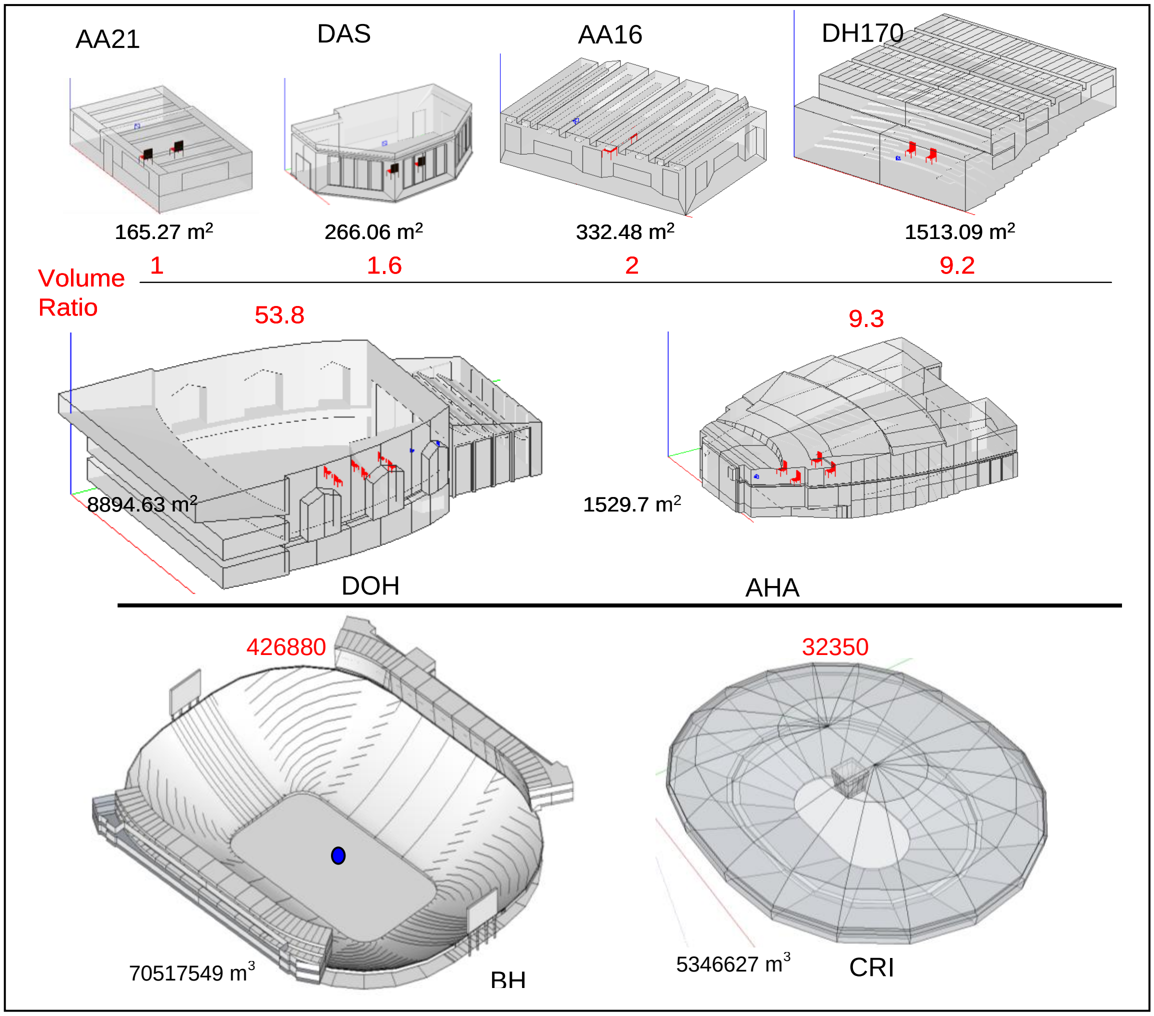

Figure 2.12.Ratio of the estimated room volume of the cases studied in this research.

2.2.1 Computer Modeling of Diffuser

A diffuser's performance is specified by its scattering or diffuse coefficient. It defines the directional characteristics of the diffuse reflections leaving the surfaces. In the computer modeling, scattering coefficients are input variables. Within the study, materials that are not intentionally designated as diffusers are not assigned with scattering coefficients.

A study by Wang and Rathsam (Wang and Rathsam, 2008) defines the factors that influence the scattering coefficient sensitivity in the computer modeling. Surfaces that are sensitive to the scattering coefficient are large mirrored reflective surfaces, surfaces with great disparity of materials, and surfaces with low average absorption (α). This study was limited to a single room with parallel walls, and, therefore, did not include the impact of room shape and volume.

In order to simulate the diffusion from surfaces, a value for the percentage of scattering for surfaces is assigned. It specifies the proportion of reflected energy propagated into non-specular directions. A value of 10% is assigned to the scattering characteristic of smooth flat surfaces and 80% for rough surfaces.

2.2.2 Recording Studio

Recording rooms with a degree of acoustical variability have become a trend since this variability allows the studio to accommodate different types of recording tasks (Newell, 2008). They can be better suited for many musical instruments especially for non-electronic instruments and for vocals. Several strategies are suggested to obtain this acoustical variability by creating a space with multiple acoustic zones.

One of the strategies suggested is to create separate fixed “live” and “dead” zones (sound fields) by having reflective and absorptive surfaces on certain elements of the room. Non-parallel walls and graded surfaces that break the reflections, such as the diffusers, are suggested to have the capacity to eliminate echoes produced by specular reflections. Another strategy is to create an adjustable acoustic zone by inserting a two-fold side wall panel of absorptive and reflective surfaces.

The recording room of the Duderstadt Audio Studio (DAS) has implemented the strategies described in the above paragraphs. There exist two-fold side wall panels with absorptive and reflective materials, which are expected to adjust the liveliness quality (reverberation time) of the space according to the recording task.

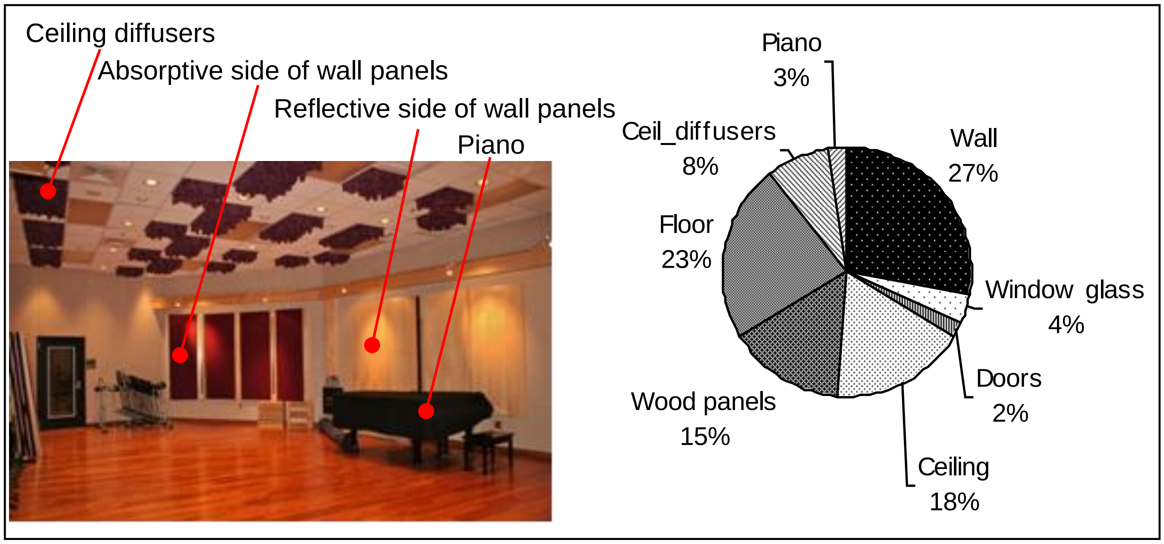

Figure 2.13. Architectural elements observed within the Duderstadt Audio Studio (DAS) and the material properties of its surfaces of existing conditions (as-is).

The largest total absorption (Sabin) is produced by the walls due to the large surface area and the absorptiveness of the material applied. The largest reflective surface area is the wooden floor. The diffusers applied on the ceiling are the RPG skyline diffusers (see Figure 2.11 and Figure 2.13) with a surface area of 8% of the total room surface.

The field measurement in DAS attempted to observe the impact of variation on the positioning of the adjustable two-fold wall panels and the non-parallel walls. The main sound source was a balloon burst. Measurements were taken for four different positions of the adjustable two-fold side walls. Panels were either fully opened, opened to an angle of 45^o, fully closed, or closed to an angle of 45^o. Panels opened are the condition of having absorptive surfaces facing the interior while the closed position is exposing the reflective surfaces.

Positions of the sound source and receivers are shown in Figure 2.14. Measurements were done with the 120-channel Acoustic Camera sphere-microphone array and a stereo microphone placed on the outer ear of a dummy head.[11]

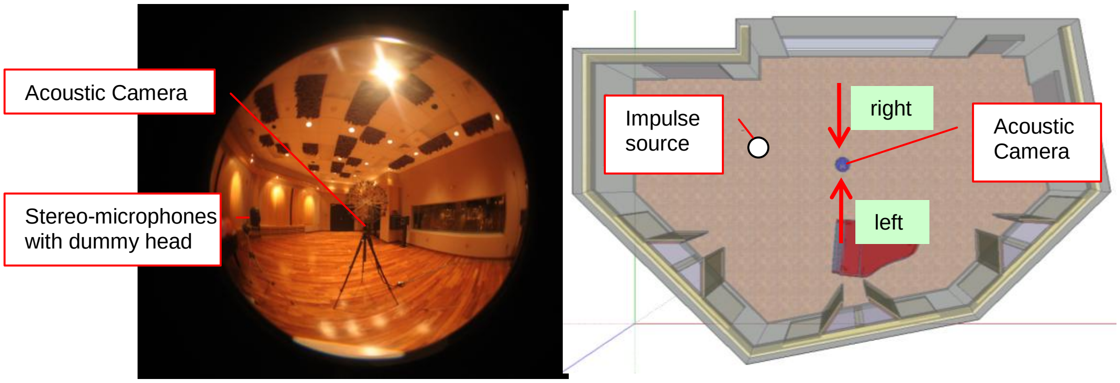

Figure 2.14. Field measurement setup in DAS.

Geometrical configurations for the computer modeling based on variations of the two double-sided adjustable wall panels, ceiling diffusers, and piano are described in Figure 2.15.

There are four main design configurations based on the positioning of the adjustable wall panels. For each variation, the use of the diffusers was also observed. The scattering coefficient referred to the product specification (http://www.rpginc.com/products/skyline/sky_dc.htm).

To replicate the "as-is" condition of DAS, a piano was also modeled. The acoustical properties of the surfaces in the existing condition (as-is) are described by the aggregate plot of the total absorption of the surfaces as shown in Figure 2.16. This information was generated by calculating the total Sabin of each material, which was achieved by multiplying the surface area with the absorption coefficient for each octave band.

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[11] Stereo-phone recording is a technique used in binaural recording, allowing recording of the directional properties at the listener's ears. A dummy head is an artificial model of a human head, built from selected acoustic materials to emulate the sound-transmitting characteristics of a real human head. The stereo-phone recording utilized in this study, however, is mounted at the outer part of the ear and does not take into account the ear canal characteristics.

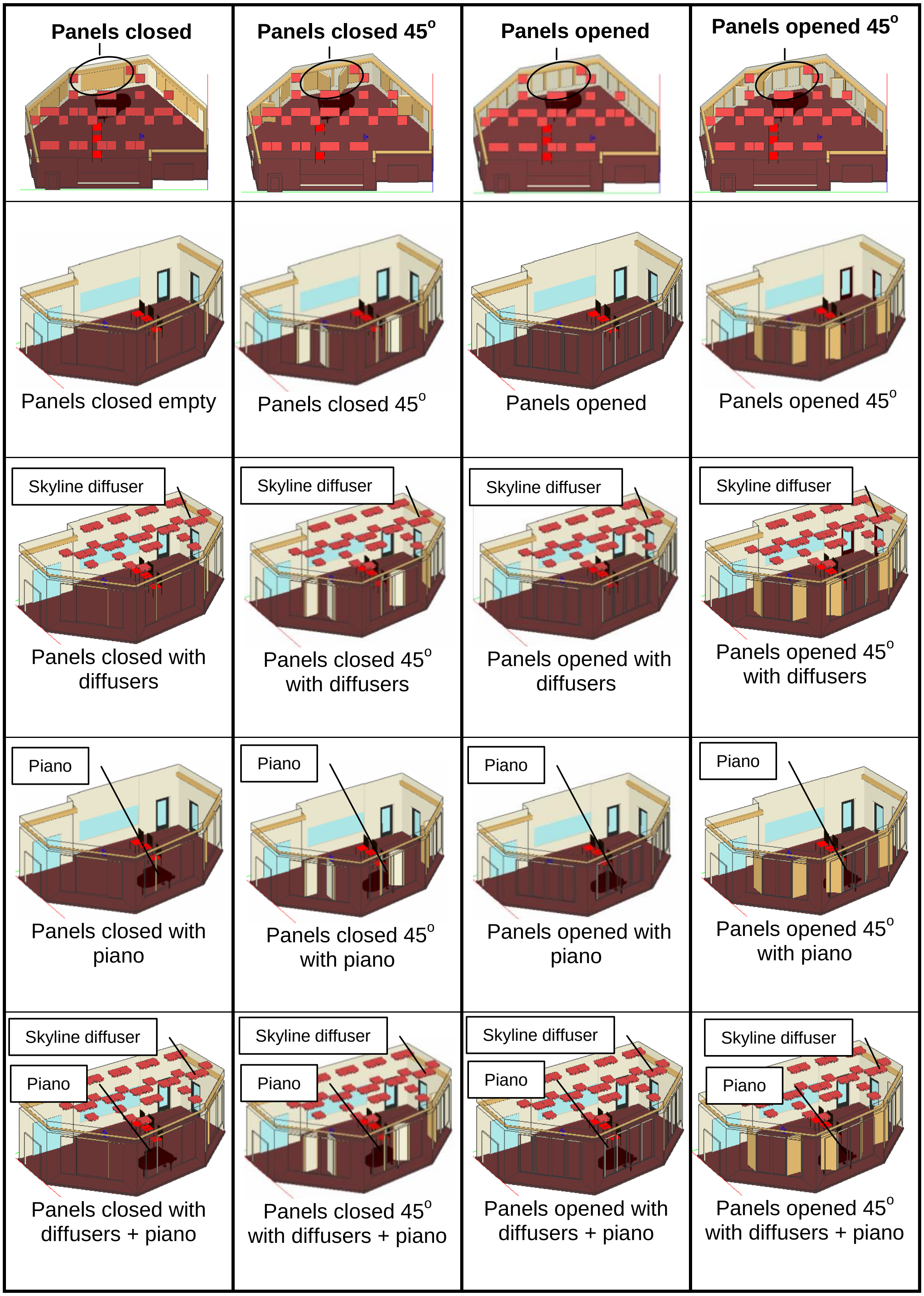

Figure 2.15. Parametric runs and modeling of the computer simulation of DAS.

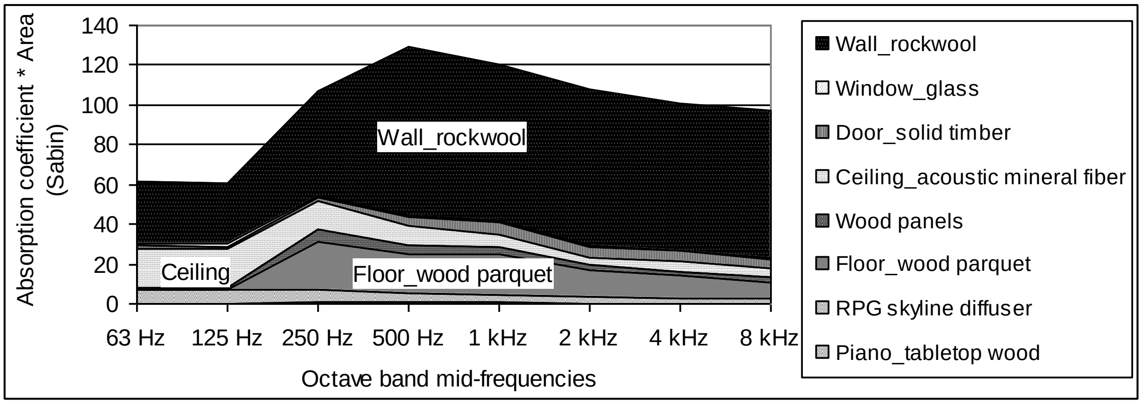

Figure 2.16. Aggregate plot of the total Sabin (i.e. absorption coefficient x surface area) in DAS.

2.2.3 Classrooms and Lecture Halls

An ideal classroom should have the ability to provide a good acoustics quality for communication since speech is the primary activity. The acoustical conditions of classrooms are expected to meet the standard performance criteria, ANSI S12. 60-2002 American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools.

For a given speaker-to-listener distance, speech intelligibility is chiefly degraded by two phenomena: noise and reverberation. Therefore, the criterion for an ideal classroom is based on the reverberation time for different classroom sizes. The design considerations are the ratio of acoustic treatment area over floor area, the ceiling height, the estimated reverberation time (T_{60}), and the absorption coefficient of the acoustic treatment. Listeners in the rear seats are expected to have the same level of speech intelligibility as listeners who are seated in the front rows.

The common designs rely on creating reflections from the ceiling or using lateral reflections (i.e., reflections from the side-walls). This attempt can be supported by the use of diffusers. Four classrooms with a variety of geometrical sizes and shapes were studied as described in Figure 2.17. The two classrooms in the Art & Architecture Building, room 1221 (AA21) and room 2216-2219 (AA16) were representative of rectangular rooms with parallel walls. The acoustical treatments applied in room AA21 are thin polystyrene boards on the side-walls resulting in a reverberant room.

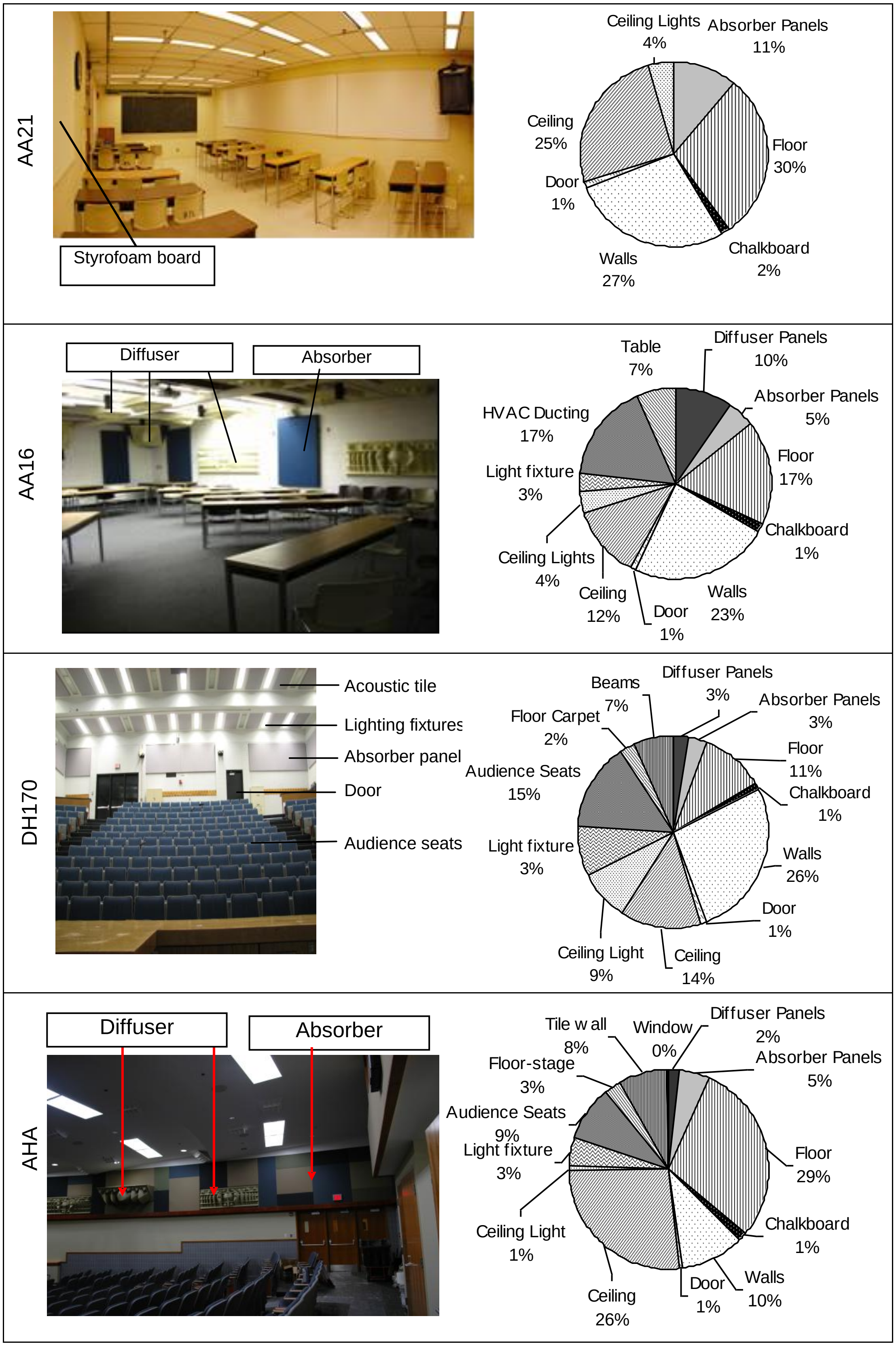

Figure 2.17. Architectural and acoustical properties of four classrooms studied in this research.

The aggregate plot of the total absorption of the surfaces is shown in Figure 2.18.

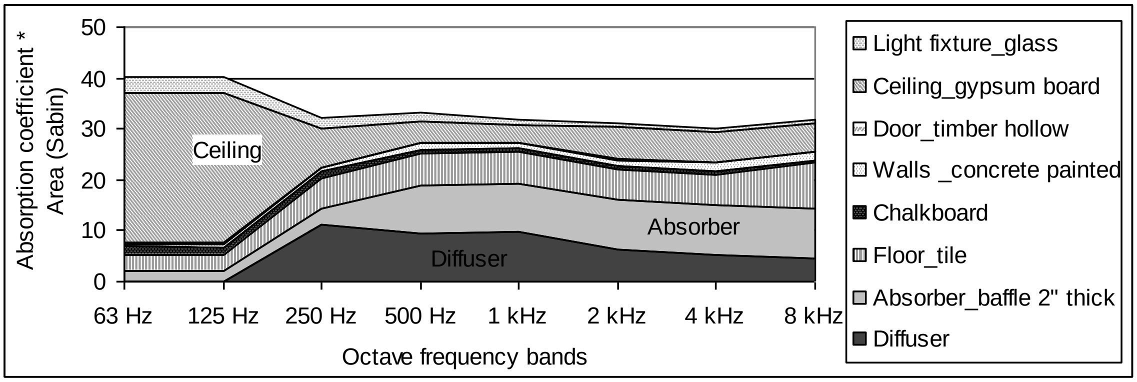

Figure 2.18. Aggregate plot of the total Sabin in room 1221 Art and Architecture Building (AA21).

The different number of diffusers applied and their positioning relative to the walls' normal axis accounted for the variations in the computer modeling (see Figure 2.19). The type of diffuser modeled was diffuser no.1 in Figure 2.11. The diffusers were intentionally positioned asymmetrically to obstruct the parallel walls. This positioning already assures less standing waves occurring within the space. A maximum of six diffusers were applied. In the last two configurations, the diffusers were tilted to the floor with an angle of 15^{o} and 30^{o} from the normal axis.

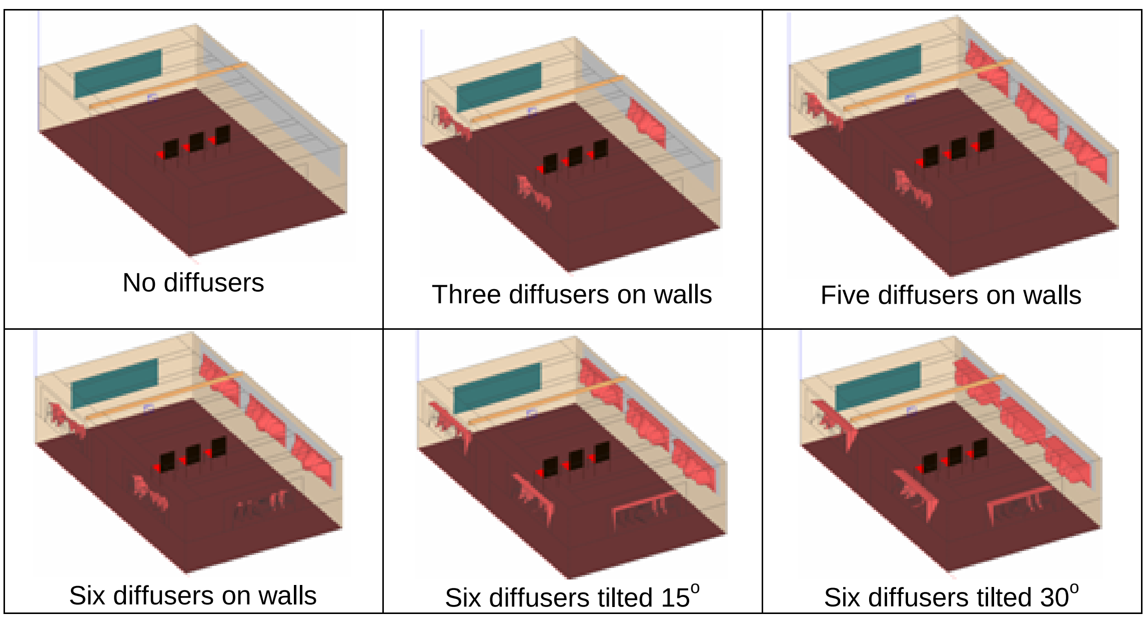

Figure 2.19. Parametric runs and modeling of the computer simulation of AA21.

Room 2216-2219 (AA16) has diffusers applied on the walls and ceilings and is slightly larger than room 1221 (AA21). The diffusers are the product of Golden Acoustics, and details of the diffusers are provided in Figure 2.11. The diffusers were positioned asymmetrically with the attempt to hinder the reflections of the parallel side-walls. After the panels were installed and the space was used, there were positive comments from the occupants where the electrical sound system was no longer needed to obtain speech clarity. Details of the geometrical and acoustical properties of surfaces are also described using the aggregate plot of total Sabin shown in Figure 2.20.

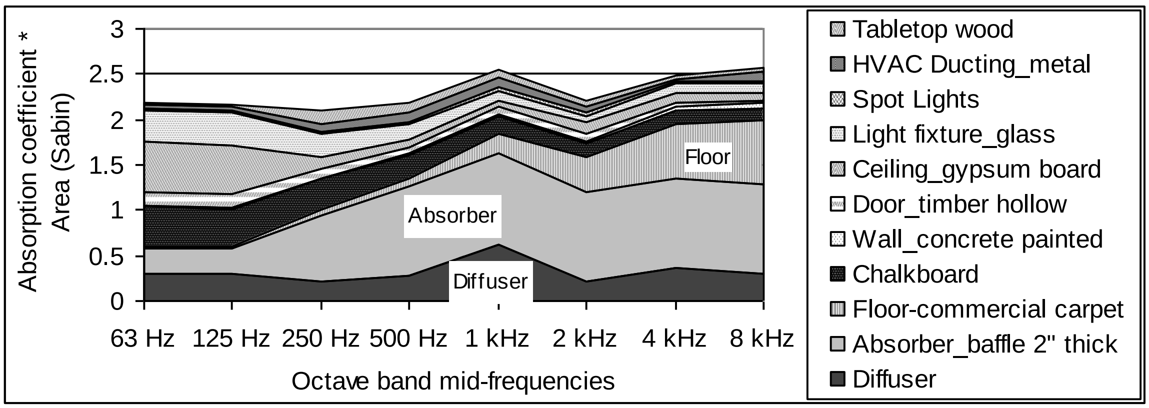

Figure 2.20. Aggregate plot of the total Sabin in room 2216-2219 Art and Architecture Building (AA16).

Several absorber panels are mounted on the walls in between the diffusers, and the impact was also observed in the computer modeling with details described in Figure 2.21. In addition, one of the computer model configurations enlarged all the surfaces twice, except the furniture. This model allowed observation of the diffusion of same-source-to-receiver distances as distances from the room boundaries were increased.

In AA16, the 48-channel Acoustic Camera was positioned at the center of the room with impulse bursts at the front and back side of the room (see Figure 2.22). The position of a human speaker in front of the class is represented by the front source. The source burst at the rear side of the room allowed observation of the rear wall diffuser. Other anechoic sounds were also reproduced through loudspeakers including a piece by Mozart played by a string quartet.[12] Using this source enabled observation of the performance of the loudspeakers that were mounted on the ceiling close to the exposed heating, ventilating, and air conditioning (HVAC) ducting.

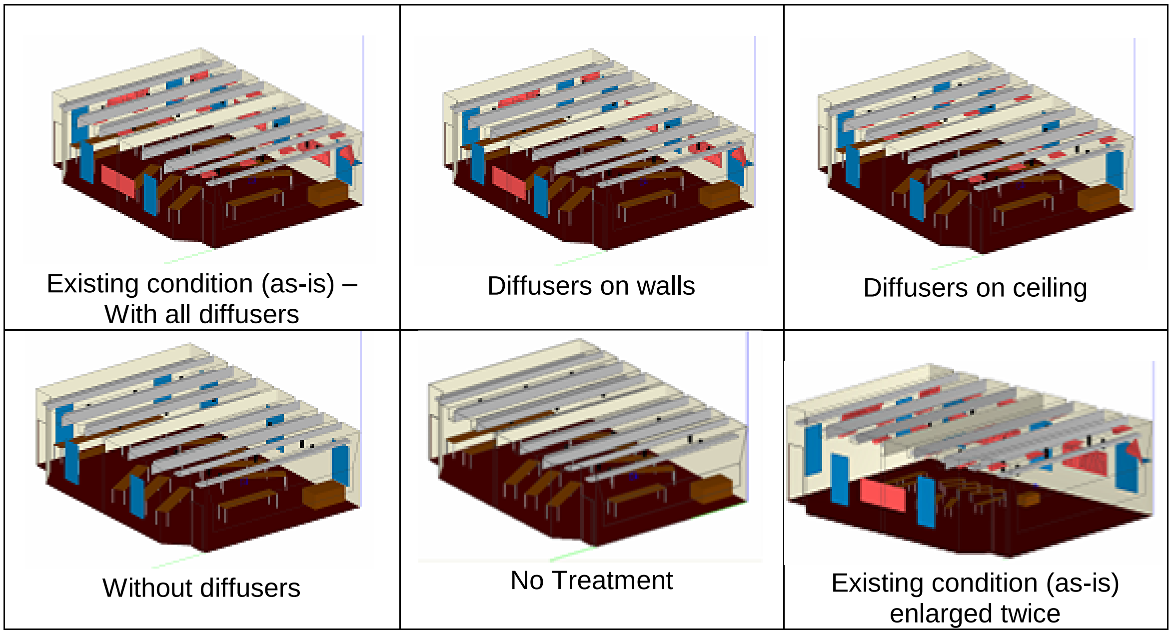

Figure 2.21. Parametric runs and modeling of the computer simulation of AA16.

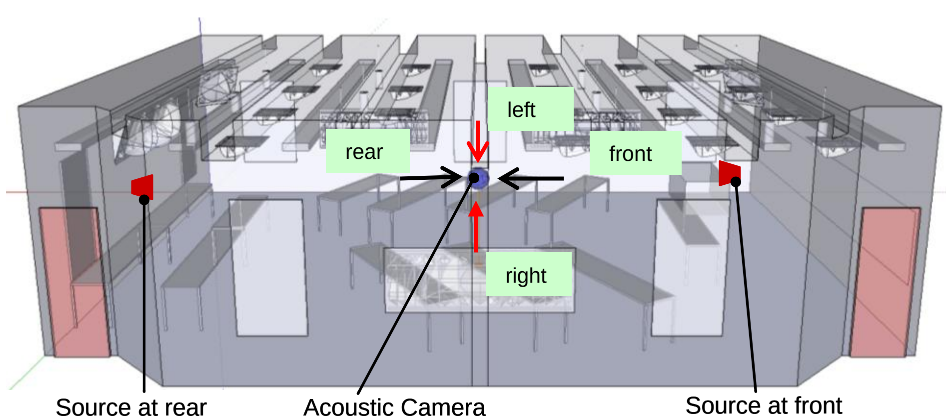

Figure 2.22. Positions of sources and the Acoustic Camera in AA16.

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[12] The music piece by Mozart played by a string quartet was recorded at Michigan State University by the Michigan State Acoustic/Psychoacoustic research group. Special thanks to Prof. William M. Hartmann who provided this anechoic recording.

Dennison Hall room 170 (DH170) was representative of a semi-large room with flat parallel walls. Observations were based on the condition before the room underwent major renovation. The room is an auditorium with a stepped floor audience seating area. Therefore, ceiling and floor are not parallel, and the ceiling height gradually decreases from the front to the rear side of the room. Based on an observation of the existing condition, the room data for the computer modeling was generated with the surface properties described in Figure 2.23. Commercial carpet was applied on the floor audience area, which distinguished it from other floor areas. The largest surface area was the main wall structure.

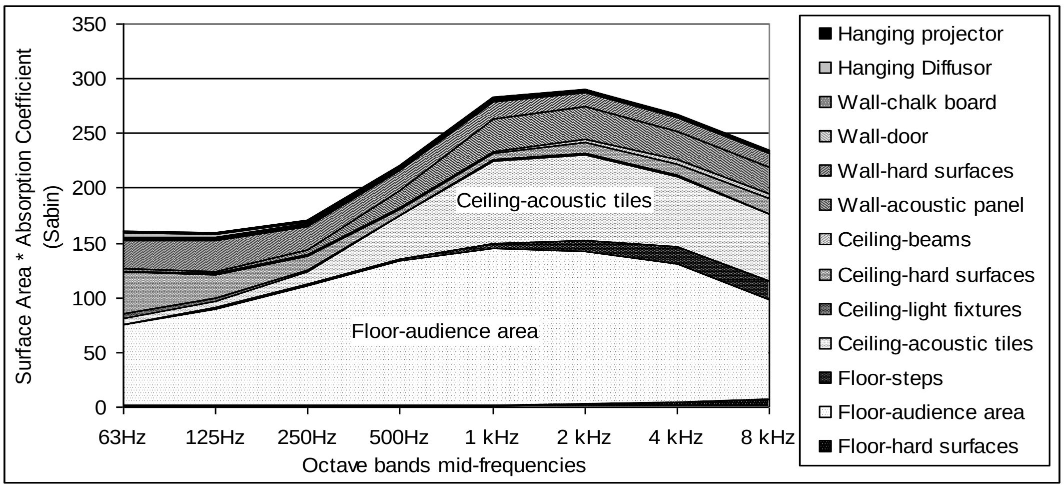


Figure 2.23. Aggregate plot of the total Sabin in room 170 Dennison Hall (DH170).

Three geometrical configurations for the computer modeling were variations on the number of diffusers applied on the side walls as described in Figure 2.24. Audience seats were not included in the computer modeling.

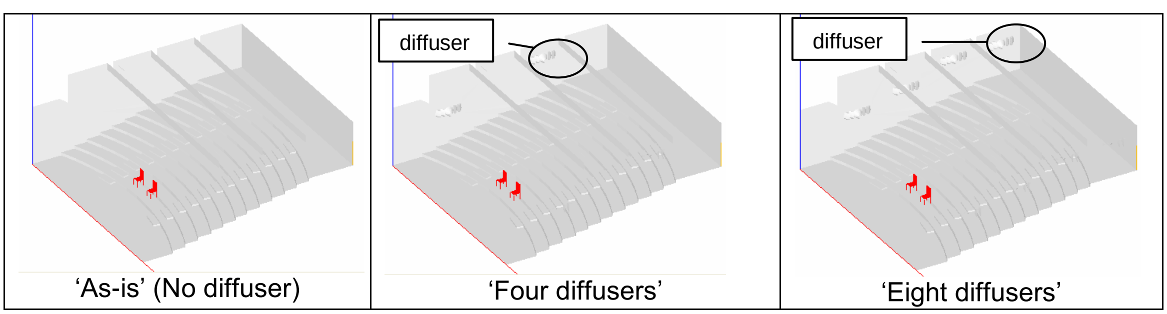


Figure 2.24. Parametric runs and modeling of the computer simulation of DH170.

Impulse sources were excited at five different positions while the Acoustic Camera 48-channel sphere-microphone system was placed at the center as shown in Figure 2.25. Given the source positions, the height from the stepped floor is anticipated to execute different impacts on the sound field.

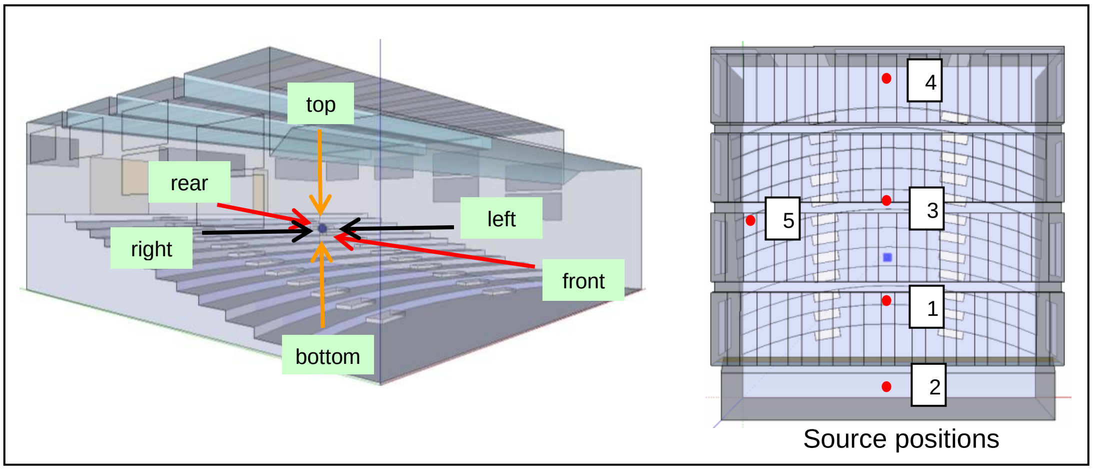


Figure 2.25. Positions of the Acoustic Camera and sources in DH170.

Another lecture hall with space volume similar to room DH170 is the Angell Hall lecture hall – A (AHA). Instead of flat parallel walls, the side walls of AHA have a slight curvature. The observation was based on its recent renovation. The surfaces that are assumed to be highly absorptive were the carpeted floor and the upper section of the walls. Several of the Golden Acoustics diffusers are mounted on the side walls of this space.

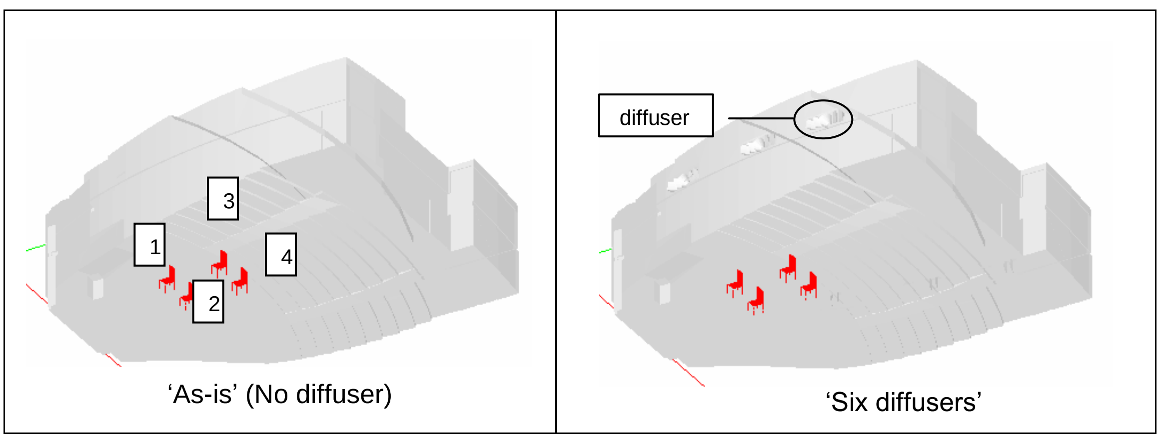


Figure 2.26. Parametric runs and modeling of the computer simulation of Angell Hall auditorium-A (AHA).

The computer models of AHA are shown in Figure 2.26. The effect of diffuser panels applied on the side walls of the existing condition (as-is) were observed. The acoustical conditions at four receiver or seat positions were observed. The first two seats are similar with other cases studied (see Figure 2.1), while two other seats are 3 meters away behind seats 1 and 2. The reason for inclusion of these additional seats was to observe the impact of the curvature of the side walls to a large sound field.

2.2.4 Concert Hall

A diffuse condition of a sound field produced by the surface diffusivity is considered important to achieve good acoustics in concert halls (Haan and Fricke, 1997). According to Beranek, there are five basic acoustical attributes to predict the quality of a concert hall as related to the discussion in section 1.4.2: direct sound, early sound, early sound decay, reverberation time, and loudness (Beranek, 1996). A rating scale for sound diffusion based on irregularities of walls and ceilings in concert halls was suggested by Beranek but without addressing the acoustical attributes above (Beranek, 1962).

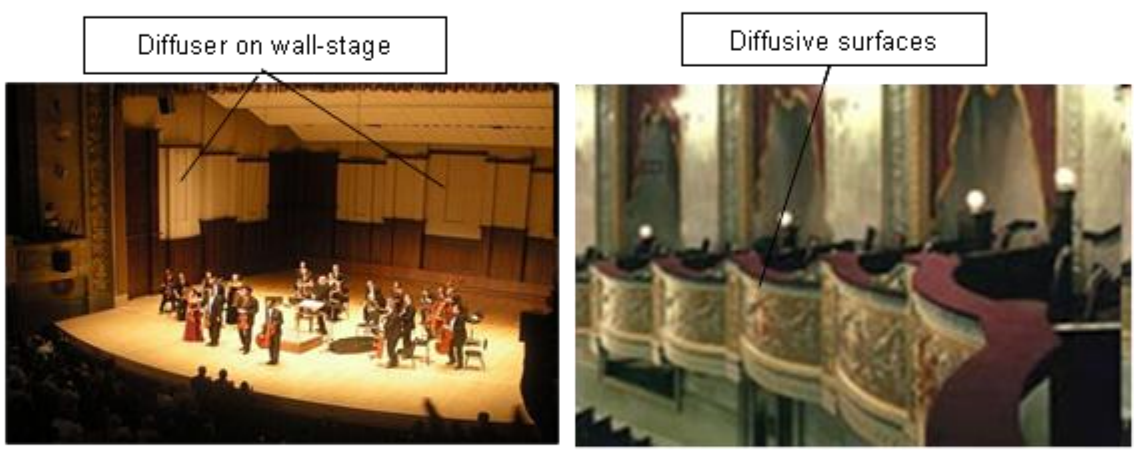


Figure 2.27. Diffuser and diffuse-like surfaces in Detroit Symphony Orchestra Hall.

In order to observe the importance of having diffusion in concert halls, the Detroit Symphony Orchestra Hall (DOH) was selected as one of the cases studied. In the existing condition (as-is), the diffusers are applied on the stage walls. Besides the diffusers, elements that were assumed to behave as diffusive surfaces are the curved balconies and the three-dimensional ornaments on the walls and ceiling. The Acoustic Camera was placed at the center of the audience area during the field measurements. Three types of impulse sources were burst on the stage approximately 2 meters from the front edge. The sources were a clapper, balloon, and yacht cannon.

The geometrical configurations in the computer modeling of DOH were designed to observe the effect of diffuser panels applied on the stage ceiling in the existing condition (as-is) as illustrated in Figure 2.28. The acoustical conditions at six receiver or seat positions were observed.

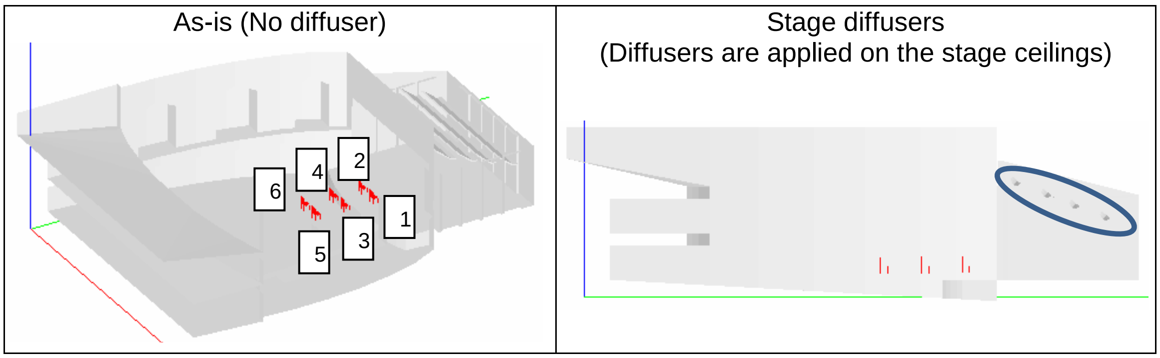


Figure 2.28. Parametric runs and modeling of the computer simulation of Detroit Orchestra Hall (DOH).

2.2.5 Sport Facilities

Sport arenas exist primarily as large spaces with an ellipse or bowl shape. The main acoustical issues in sport arenas are environmental noise impact and sound insulation, reverberation and room reflection, and speech intelligibility. Local codes strictly limit noise exposure in buildings, particularly for spaces with a large number of spectators including the American football stadium. The performance of all building elements has to be predicted to anticipate the noise pollution. It is necessary to carefully control the reverberation in order to achieve the correct ambient condition. A balance must be applied between maintaining spectator excitement and acoustical control (by sound absorbing surfaces) to ensure an optimum performance for the sound system. In accordance with IEC codes of practice, a place of public assembly must have a voice alarm system that achieves a specific minimum speech intelligibility requirement, in this case

0.45 STI (speech transmission index). This involves selecting, locating, and orienting loudspeakers as well as designing and locating the acoustical treatments.

The Crisler arena was constructed in 1967 as a basketball arena with a seating capacity of 13,751 persons using cushioned seats. The largest portion of surface area is the ceiling, which is constructed from corrugated metal roofing, with trusses and beams serving as a structural support. A ray-tracing simulation was done prior to the EASE simulation to visualize the sound propagation path for a given source location. An estimation of the geometrical properties is provided in Figure 2.29 with an acoustical treatment applied on the ceiling to provide an ideal reverberation condition for the arena.

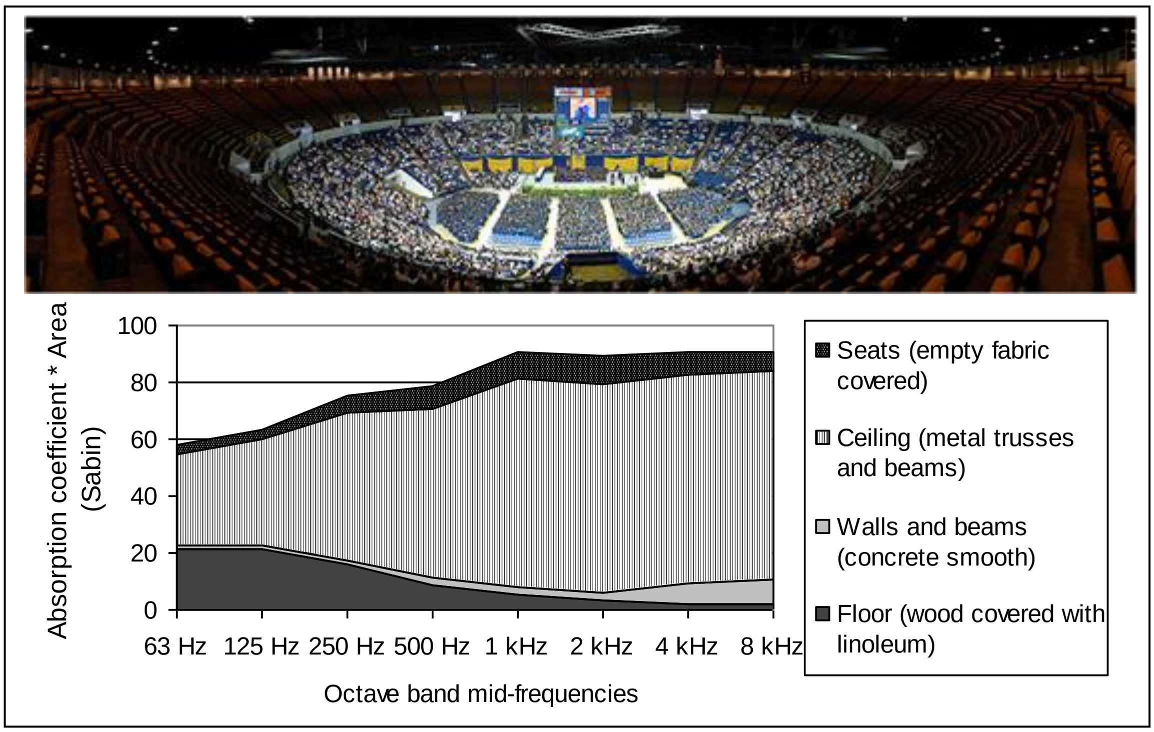


Figure 2.29. Photograph of the interior of Crisler Arena (Source: www.ask.com/wiki/Crisler_arena) and the aggregate plot of the total Sabin estimated from the computer modeling.

The experimental setup in Crisler Arena (CRI) was related to the plan to renovate the sound system. Arrays of loudspeakers were proposed as an integrated element with the scoreboard. Diffusion within this space is mostly contributed by the room shape and size. Any incoming sound is expected to be diffusely reflected by the curvature of the walls. This highly diffuse space benefits from the fact that fewer loudspeakers are required to completely fill the room with sufficient sound energy. Computer simulation supported this hypothesis.

The largest space studied in this research was the Michigan football stadium, also known as the Big House (BH). It was constructed in 1927 and since then has been the home of the University of Michigan football team. Acoustical problems in the stadium are excessive noise or unwanted sound from the spectators.

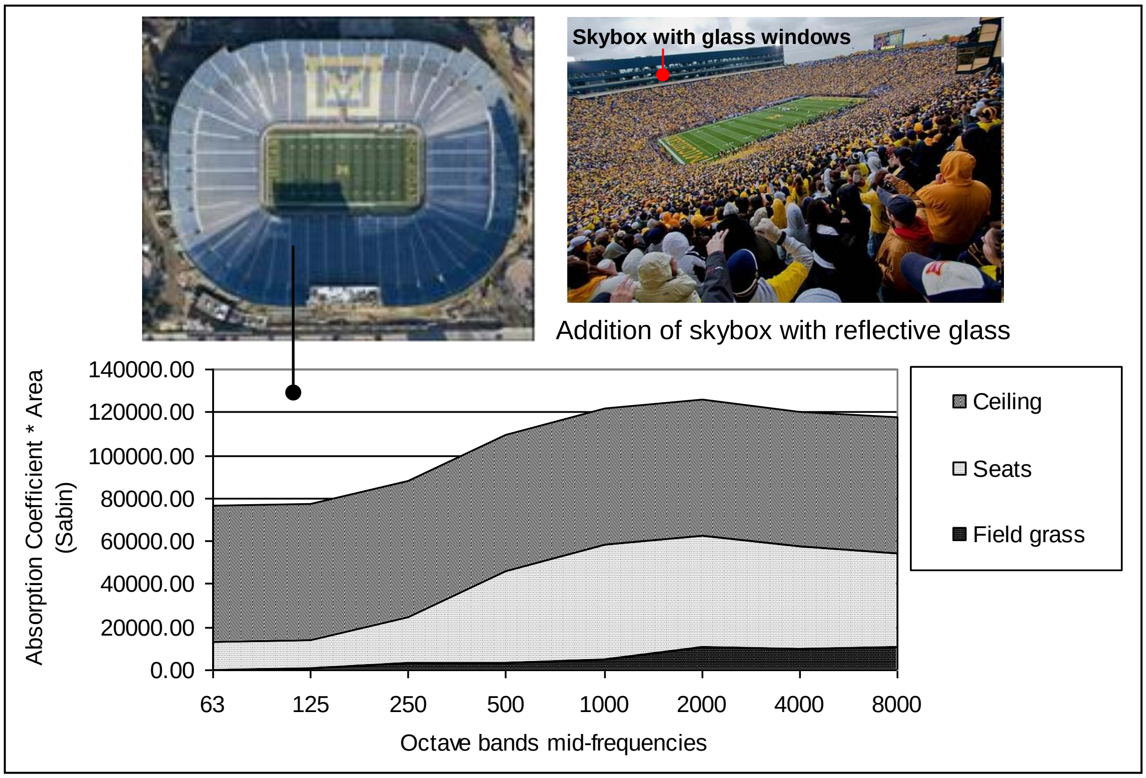


Figure 2.30. “Big House” Football Stadium before and after addition of the skybox and the aggregate plot of the total Sabin.

The standard measurement unit of noise is the decibel (dB), which represents the acoustical energy present. The A-weighted decibel scale (dBA) is commonly used to measure noise levels, because it has been shown to provide a good correlation with the human response to sound. The faintest sound that can be heard by a healthy ear is about 0 dBA (i.e., a sound wave power of 10^{-16} watts/cm^2), while an uncomfortably loud (deafening) sound is about 120 dBA.

A recent renovation was done to the stadium. Skyboxes were added as part of two new structures on the east and west sidelines. These skyboxes have reflective glass windows that are tilted to the field at a certain angle. This additional structure presumably would help to reduce the noise exposure to the surrounding neighborhood by reflecting the crowd noise back into the stadium "bowl."

Similar to the CRI case study, diffused sound fields are most likely to occur in large spaces such as the "Big House," the University of Michigan football stadium. The addition of two skyboxes on the left and right side of the stadium is the main reason why this space was observed. Impulse response measurement of the stadium was obtained from a yacht cannon burst as shown in Figure 2.31.

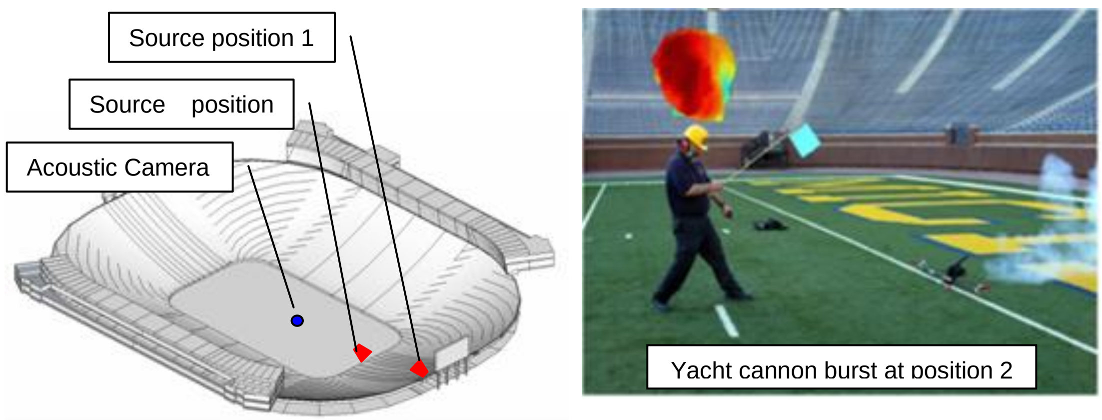


Figure 2.31. Positions of the Acoustic Camera and sources in the "Big House" stadium. Figure at right is the cannon burst for the impulse source in the stadium.

Another impulse was burst at a position of the spectators in the audience seating at the height of the middle seats. The noise from the crowd during a football game was also observed.

2.3 Subjective Assessment

To characterize the sound-field diffuseness, the computer-interface listening test was chosen: subjects were exposed to audio stimuli that were generated from the simulation results using the auralization capability in EASE. In principle, the experimental setup for the subjective assessment can be described as follows: The auralization was generated in simulated spaces with a variety of design alternatives from existing conditions to the best or worst scenarios with the diffusion system applied. After obtaining the responses, design variables that were embedded within the stimuli and that impacted the auditory perception could be traced back. Auditory stimuli were generated from sound-field simulation and auralization in EASE. This technique has a high repeatability so that the survey could be conducted several times to gain sample sizes with significant statistical power (Utami *et al.*, 2011).

Subjective evaluation based on the computer-interface survey was conducted for the Duderstadt Audio Studio (DAS) and room 2216-2219 of the Art and Architecture Building (AA16). Both cases are considered to have the complete parametric simulation since geometry arrangement of the architectural elements includes the following: (1) the amount of diffuser surfaces relative to room surfaces and their positioning, (2) room shape, (3) room size, and (4) interior layout and furniture. These two spaces are also considered as the most common auditory environment where subjects can easily fine-tune their auditory experience.

The binaural impulse response used a KEMAR head-related transfer function available in EASE. Two anechoic recorded sounds, a piece by Mozart played by a string quartet and a male voice, were selected for auralization of the stimuli based on the discussion in section 1.4.5. The time-varying spectral representation of these sounds is presented in Figure 2.32.

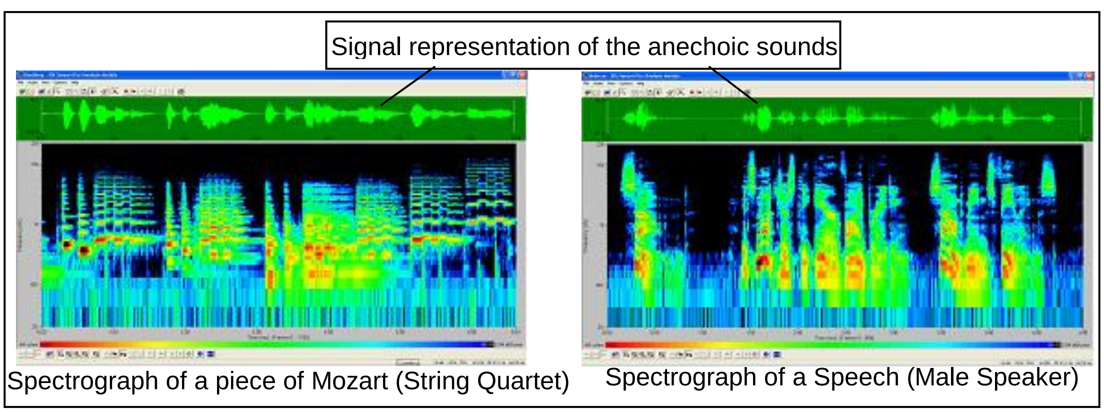


Figure 2.32. Signal representation and spectrograph of the anechoic sounds of a piece by Mozart (string quartet) and a speech by a male speaker.

There were 40 subjects who participated in the survey, ages 18 – 22 years with good hearing condition. Participation was based on the subject's claim of their interest in music and acoustics as part of the environmental technology course material. Each subject received the same treatment and addressed similar questions, and results indicated noticeable differences in auditory perception. Instructions were presented to subjects through a slide presentation using PowerPoint on a computer screen and stereophonic headphones at a level of approximately 60 dB (A-weighted). Auditory stimuli were embedded within the slide presentation. The instructions are discussed in this section while the complete slide presentation presented to subjects is provided in Appendix C,

Stimuli within the questionnaire contained audibility characteristics of an average intensity level (total SPL), C_{50}, C_{80}, and T_{30}. Subjects indicated their responses via a questionnaire sheet. The correct answer for each question was related to the objective values measured. As an example, in the comparison of paired stimuli, the sound with a higher total SPL should have been perceived as louder. Further details on the results and analysis of the subjective assessment are available in sections 3.2 and 4.3.

Sections 2.3.1 through 2.3.5 describe the survey questions and the associated auditory stimuli. The objectives of the subjective assessment were to characterize the following:

1. The impact of the diffusers on clarity, loudness, and liveliness.
2. The impact of the diffusers on sound localization.
3. The impact of architectural elements, other than diffusers, on clarity, loudness, and liveliness.
4. The impact of architectural elements, other than diffusers, on sound localization.
5. The impact of room size on the audibility condition.

2.3.1 Impact of Diffusers on Clarity, Loudness, and Liveliness

Three pairs of stimuli were used to address the impact of the diffusers on clarity, loudness, and liveliness. For all three pairs of stimuli, the question given was to compare which of the stimuli sounded louder, sounded clearer, and sounded livelier. The three pairs of stimuli were:

- Auralization of DAS of the model with all the diffusers and without diffusers on the ceiling with wall panels closed. Graphic of the associated slide is provided in Figure 2.33.
- Auralization of AA16 of the model with all the diffusers and without all diffusers (i.e., on wall or ceiling). Graphic of the associated slide is provided in Appendix C, slide no.7.
- Auralization of AA16 of the model with all the diffusers and without diffusers on the ceiling (see Appendix C, slide no. 9).

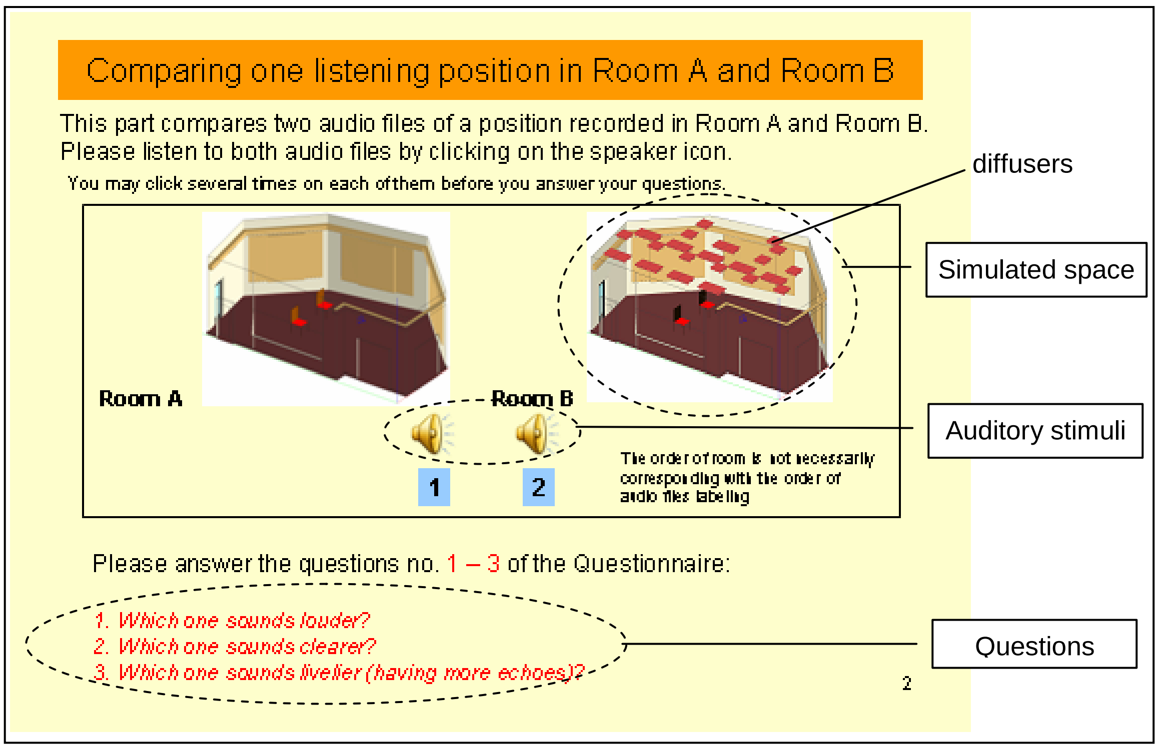


Figure 2.33. A part of the survey interface for observing the impact of the diffusers on clarity, loudness, and liveliness using presentation slides in PowerPoint with embedded auditory stimuli. Stimuli were created by auralization using the EASE capabilities.

2.3.2 Impact of Diffusers on Sound Localization

The pairs of stimuli used to observe the impact of the diffusers on sound localization were:

- Auralization of AA16 at seat 1 and seat 2 of the model with no diffusers. Subjects were asked to identify which stimulus was coming from their left (see Appendix C, slide no. 6). The source and receivers' positions are illustrated in Figure 2.34.
- Auralization of AA16 at seat 1 and seat 2 of the model with all the diffusers. For these stimuli, subjects were asked to identify which stimulus was coming from their right (see Appendix C, slide no. 8).
- Using the aforementioned stimuli, subjects were asked to identify the seat's position of a stimulus they heard (i.e., auralization at seat no.4). A stimulus auralized at seat no. 1 was used as the reference position. The associated slide is shown in Figure 2.34.

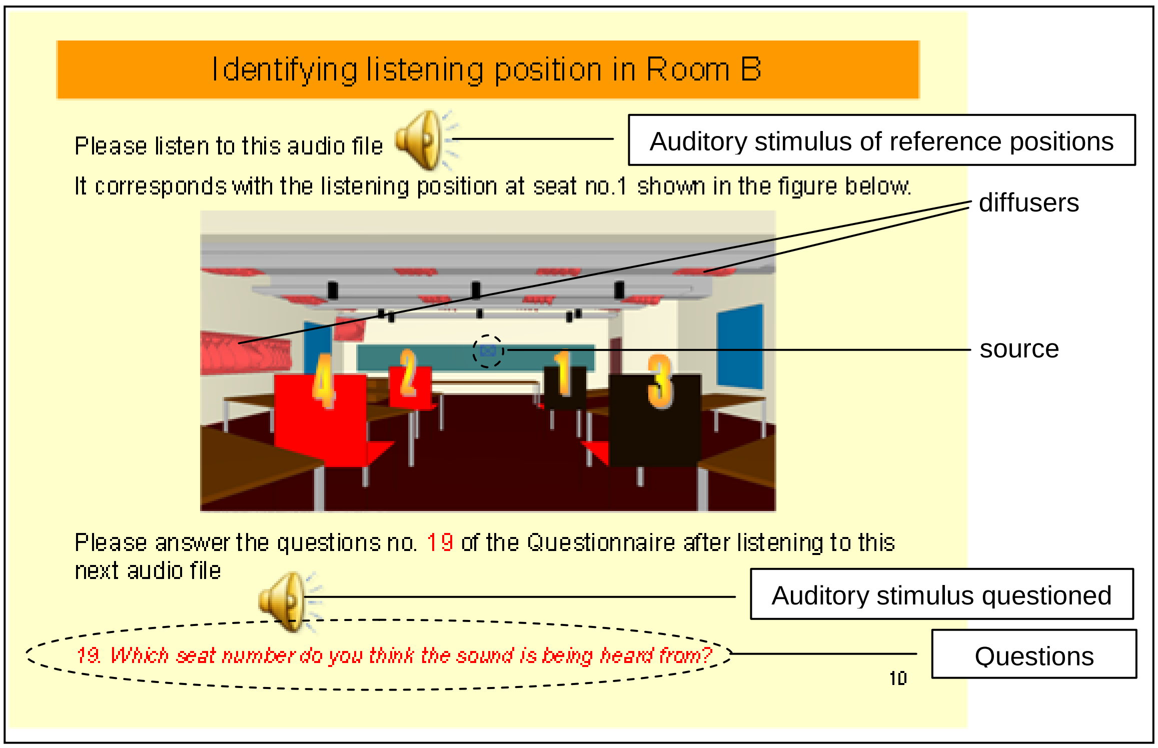


Figure 2.34. Observation on the impact of the diffusers on sound localization using the same survey interface and technique.

2.3.3 Impact of Architectural Elements, other than Diffusers, on Clarity, Loudness, and Liveliness

Other than the diffusers impact on the clarity, loudness, and liveliness in DAS, the impact of the adjustable wall panels' positions were also observed. A question compared the audibility condition in the model with the panels closed and with the panels closed at a 45^o angle; another question compared the panel closed to the panel open. These comparisons relied on paired stimuli of the following:

- Auralization of DAS of the model with diffusers for wall panels closed and opened (see Appendix C, slide no. 4).
- Auralization of DAS of the model with diffusers for wall panels closed and closed 45^o (see Appendix C, slide no. 5).

Details of the results and analyses are provided in section 4.3.

2.3.4 Impact of Architectural Elements, other than Diffusers, on Sound Localization

The impact of a piano's presence in DAS was also observed by using the auralization of DAS of the model with wall panels closed both with and without a piano present at the right seat (i.e., orientation facing the loudspeaker). At the right seat, the auralized sound is expected to be perceived as if coming from the left side, which is the location of the source. Using this paired stimulus, subjects were asked to identify which stimulus had a better perception of the source's direction (see Appendix C, slide no. 3).

2.3.5 Impact of Room Size on the Audibility Condition

In the computer modeling of AA16 with all the diffusers, observations were made for two rooms of different sizes. A paired stimulus embedded within the related slide of the survey interface was the auralization of the AA16 model with all the diffusers for the existing room size and for the room enlarged to twice the original size. After listening, subjects were asked to identify which stimulus was auralized in the larger room (see Appendix C, slide no. 10).

2.4 Preliminary Research

The diffuse reflected energy from a surface is modeled in computer simulations as radiating from the surface with a particular spatial distribution. In most current geometrical room acoustics models, Lambert's law is used to determine this distribution of the diffuse energy (refer to section 1.4.4). Problems would occur if only part of the surface is illuminated, if objects cast shadows on surfaces, or in the case of directional sources. One way to avoid these problems is to provide a complete illumination of sound using omni-directional sources.

An attempt to explore the possibility of limiting these problems in the computer modeling was undertaken, and a detailed explanation of these studies is provided in the following section 2.4.1. Different techniques for subjective assessment in room acoustics studies were explored and applied during the preparation stage of the subjective testing. The techniques explored were on-site listening tests, computer-interface listening tests, Web-survey, and the possibility of using an immersive virtual environment as described in sections 2.4.2 to section 2.4.4.

2.4.1 Visualizing the Sound-field Diffuseness

A preliminary study was conducted using room 1221 of the Art & Architecture Building (AA21) to explore simulation capabilities for visualizing the change in the sound-field diffuseness with a diffuser present. An omni-directional source with an intensity output of 62 dB was positioned at the center of the room. Ten receivers of omni-directional microphones were assigned at certain positions relative to the walls. All the surfaces were assigned to be 20% absorptive (i.e., 0.2 absorption coefficient for the entire octave frequency band).

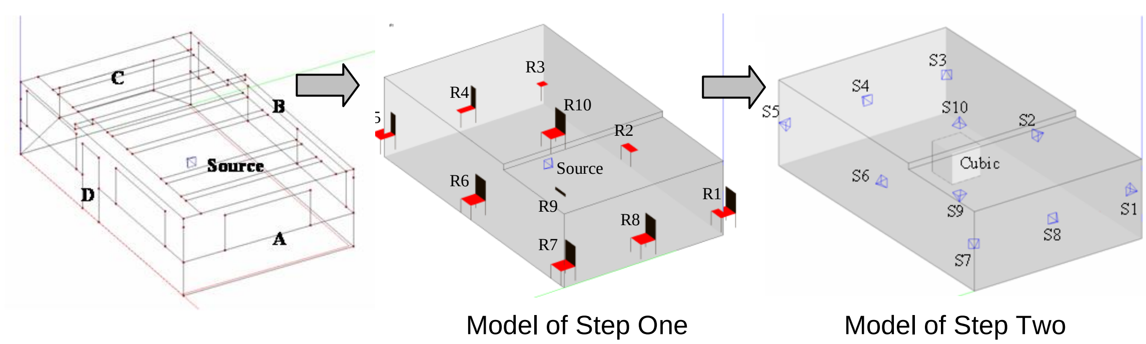


Figure 2.35. Step one and step two of the computer simulation to explore its possibility for diffusion study.

In the model for simulation of step two, receivers were replaced with sound sources with intensity output corresponding to the results obtained from the previous step (see Figure 2.35). In this step, a cubical box is inserted at the center of the room with the total SPL mapping the surfaces on this box. This step provides all possible incoming sound from surfaces, given the fact that "objects casting shadows on surfaces" does occur in simulations.

Table 2.2. Total SPL at all receiver positions from simulation step one in the preliminary research of exploring the simulation capabilities.

Receiver	Description	Direct SPL (dB)	Direct to Source Ratio	Total SPL (dB)	Total to Source Ratio
1	Front left corner	46.9	0.81	57.82	0.93
2	Left center	52.4	0.85	59.88	0.97
3	Back left corner	46.68	0.75	55.33	0.89
4	Back center	48	0.77	58.81	0.95
5	Back right corner	46.67	0.75	55.46	0.89
6	Right center	52.43	0.85	59.72	0.96
7	Front right corner	46.67	0.75	55.12	0.89
8	Front center	47.98	0.77	58.61	0.95
9	Floor center	62	1	63.65	1.03
10	Ceiling center	56.89	0.92	60.86	0.98

For instance, the total SPL for receiver no.1 shown in Table 2.2 of 57.82 dB is used as the source output SPL for the sound source at that particular position (i.e., replacing receiver no.1) for simulation of step two. Another use of the direct and total SPL values in Table 2.2 is to learn the effect of the source-to-receiver distance on the amount of direct and reflected sound received.

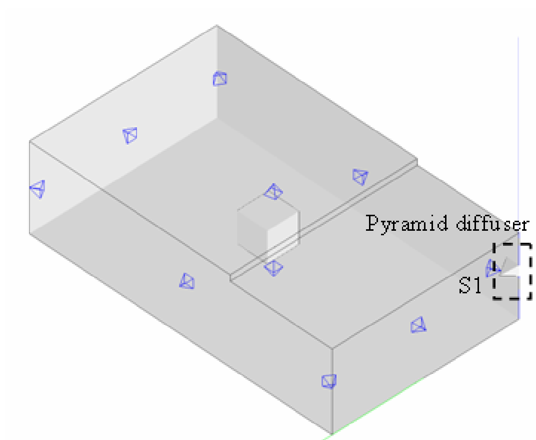


Figure 2.36. Last step of the computer simulation to explore its possibility for diffusion study having a pyramid diffuser presented.

The effect of room absorption characteristics in the computer simulation was observed by using the same procedure or steps as described above. The absorption coefficient for the ceiling was changed to 0.5 while other surfaces remained at 0.2. Mappings of the total SPL on the cubical box surfaces can be seen in Figure 2.37.

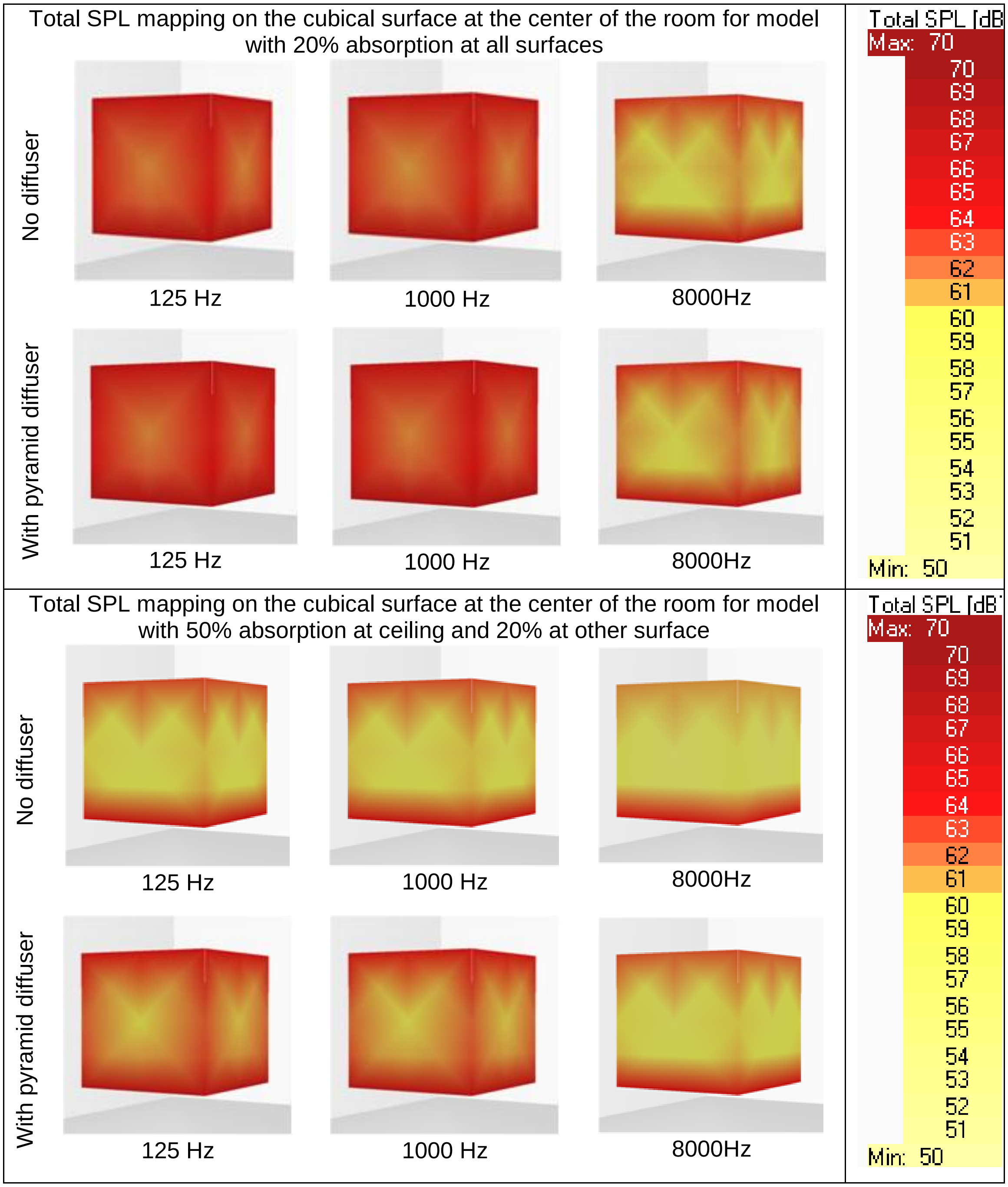


Figure 2.37. Visualization of the diffusion in room 1221 Art and Architecture Building (AA21) with variations on the input variables of the computer modeling in EASE

Assigning a larger absorption coefficient to the ceiling added more absorption into the space and, obviously, reduced the total SPL on the surfaces. Interestingly, the total SPL mapping is more uniform in the space with heterogeneous absorption. This uniformity of sound distribution indicates a diffused sound field.

The last attempt to visualize a sound field with diffusion was accomplished by inserting an element in the shape of a pyramid into the space. A pyramid with the size of 30 x 40 cm^2 for the base and 45 cm in height was attached at the front left corner of the room. The pyramid scattering coefficient is based on the calculation provided within the EASE capability (see section 2.2.1). Variables that defined the estimated scattering coefficient were material's absorption coefficient, shape, and area of the pyramid. Other surfaces were assigned as 20% scattering, which is a slightly rough surface. Comparison of total SPL mappings on the cube surfaces from all simulations with variations on room absorption and with insertion of the pyramid is provided in Figure 2.37.

The impact of the pyramid diffuser is more obvious in the model with 0.5 absorption coefficient at the ceiling as compared to the model with uniform absorption at all surfaces (i.e., 0.2 absorption coefficient). The result indicated an increase in the average clarity by 12% to 19% for frequency ranges between 250 Hz to 4000 Hz. With both absorbers and diffusers applied, the average clarity increased even more by 29% to 36% for the same receiver positions and frequency range.

This preliminary research supports the possibility of using computer modeling and simulation with EASE to visualize and measure the impact of diffusers or other architectural elements. The accuracy and resolution of the simulation mapping can be increased by assigning the room surfaces with non-uniform absorption coefficients.

2.4.2 Preliminary Study Utilizing On-Site Subjective Assessment

An on-site subjective assessment was conducted in the study of AA16 to select the most suitable terms or wording that describes a room's acoustics condition. Terms tested were common wording used by past researchers within this field in addition to the data collection technique that referred to past studies on concert halls by Beranek and others (Egan, 1988; Beranek, 2004).

A questionnaire was used to register the responses using the categorical rating (see also section 1.4.5), which measures people's reactions to some given stimuli in terms of ratings using a scale. The scale is defined with contrasting adjectives at each end (two bipolar adjectives).

The sound source was live music performed at the front of the room. The music was played by an ensemble of different string instruments. Respondents of the survey were students from the School of Music, Theatre and Dance who were seated randomly inside the space as shown in Figure 2.38.

Figure 2.38. An on-site subjective assessment with a string quartet (live music source) and randomly seated respondents. It served as a preliminary research to select the most suitable terms or wording that described the room acoustics conditions.

After listening to the musical performances, subjects were requested to register their auditory experience with a questionnaire. The first part of the questionnaire consisted of questions related to the listener's perception of the musical quality performed in the room using the categorical rating judgment with a seven-point scale. Information of age, gender, and the level of background knowledge of sound were questioned in the second part of the survey, based on the assumption that the responses were related to this information. Details of the questionnaire are provided in Appendix A. The subjective attributes used to judge the musical comfort are listed in the first left column of Table 2.3.

The correlation coefficient values in cells with dark shading in Table 2.3 are correlation coefficients with p-value <0.001; cells with lighter shading are p-values <0.05. Both indicated that the correlation coefficient is statistically significant since the probability is lower than the conventional 5% ($p<0.05$). Among the associated subjective attributes used to judge the musical comfort, the strongest correlations are between "warmth" with "liveliness" and "brilliance" with "liveliness." These correlations are expected since bass warmth is the impression of a room being reverberant for low frequency sound and brilliance is for high frequency, where the objective parameters of bass ratio (reverberation level for low frequency) and treble ratio (reverberation level for high frequency) are calculated from the reverberation time of octave frequency bands (i.e., associated subjective impression is the liveliness).

Table 2.3. Correlation coefficient *r* with the p-values of the subjective impression for musical comfort in room 2216-2219 of the Art and Architecture Building (AA16).

Pearson Correlation			1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Spaciousness	1		**1**														
Clarity	2	*R*	**.204**	**1**													
		p	.24														
Loudness	3	*R*	**.094**	**.098**	**1**												
		p	.592	.574													
Loudness Fluctuation	4	*R*	**.326**	**.123**	**.147**	**1**											
		p	.056	.482	.39												
Ensemble	5	*R*	**.172**	**.108**	**.028**	**.071**	**1**										
		p	.324	.535	.872	.686											
Liveness	6	*R*	**.098**	**.023**	**.252**	**.164**	**.022**	**1**									
		p	.577	.897	.144	.346	.899										
Warmth	7	*R*	**.151**	**.223**	**.122**	**.231**	**.033**	**.678**	**1**								
		p	.387	.199	.485	.182	.85	<.0001									
Brilliance	8	*R*	**.105**	**.175**	**.109**	**.105**	**.031**	**.661**	**.484**	**1**							
		p	.549	.314	.533	.549	.86	<.0001	.003								
Echoes	9	*R*	**.050**	**.090**	**.026**	**.110**	**.289**	**.403**	**.077**	**.127**	**1**						
		p	.778	.61	.883	.523	.092	.016	.661	.468							
Directionality	10	*R*	**.159**	**.130**	**.013**	**.004**	**.155**	**.014**	**.091**	**.085**	**.062**	**1**					
		p	.361	.455	.939	.984	.373	.936	.603	.627	.725						
Balance	11	*R*	**.131**	**.406**	**.234**	**.361**	**.021**	**.038**	**.049**	**.146**	**.191**	**.337**	**1**				
		p	.455	.016	.177	.033	.905	.829	.78	.401	.272	.047					
Dynamic	12	*R*	**.011**	**.004**	**.168**	**.227**	**.149**	**.295**	**.359**	**.352**	**.112**	**.157**	**.111**	**1**			
		p	.951	.983	.335	.189	.394	.086	.034	.038	.521	.367	.526				
Tonal	13	*R*	**.085**	**.345**	**.087**	**.131**	**.389**	**.403**	**.422**	**.449**	**.170**	**.218**	**.305**	**.305**	**1**		
		p	.628	.042	.618	.454	.021	.016	.012	.007	.323	.208	.075	.075			
Ambient noise	14	*R*	**.182**	**.196**	**.050**	**.302**	**.189**	**.094**	**.104**	**.207**	**.220**	**.131**	**.221**	**.127**	**.212**	**1**	
		p	.295	.258	.777	.078	.276	.589	.551	.232	.205	.455	.201	.469	.222		
Overall	15	*R*	**.230**	**.569**	**.336**	**.230**	**.047**	**.104**	**.245**	**.031**	**.153**	**.126**	**.278**	**.003**	**.325**	**.225**	**1**
		p	.184	.0004	.048	.184	.79	.552	.155	.859	.379	.472	.106	.988	.057	.194	

Statistically, the sample size is not sufficient to draw conclusions related to the correlation among the subjective attributes of the acoustical condition being questioned. However, this preliminary study has provided information for selecting familiar subjective attributes or suitable terms to describe the acoustical condition. Terms such as liveliness, brilliance, and bass-warmth were easily understood as shown by the significant correlations. Definition of these terms that were provided within the questionnaire text was easily understood by the subjects. An example of two terms that are similar, but where one is shown to be more difficult to elaborate, is “loudness” and “loudness fluctuation.” A narrower normal curve distribution shown in Figure 2.39 for the response to “loudness” as compared to “loudness fluctuation” indicates a better agreement within the subjects in describing “loudness.” A loud or a weak sound was more easily recognized than the amount of the loudness fluctuation.

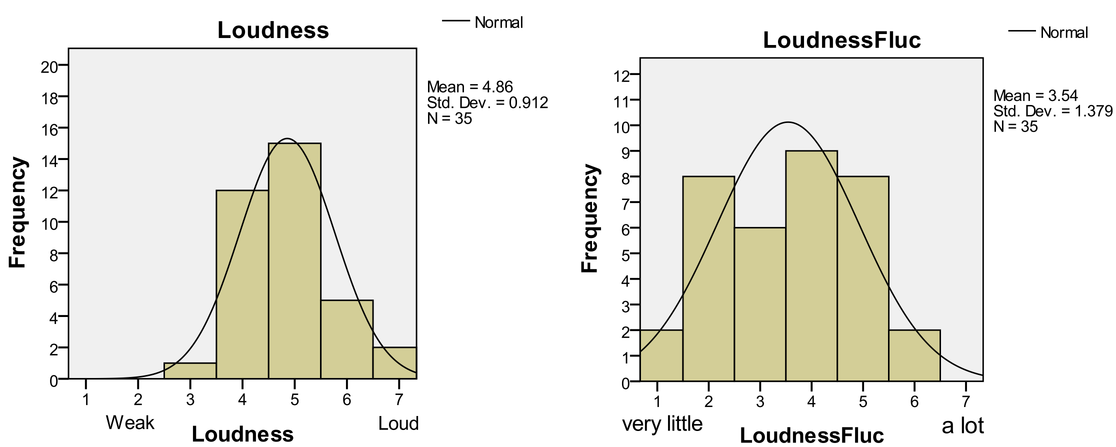


Figure 2.39. Change in the “Loudness” level is better understood as compared to the “Loudness Fluctuation” shown by the narrower normal curve of the response of both subjective attributes.

2.4.3 Preliminary Study Utilizing Web-Survey

A different pilot survey was conducted to evaluate the auditory stimuli quality created by auralization using computer simulation. It utilized the Web-survey technique via the Internet. Internet survey refers simply to any survey in which the data are collected via the Internet. An Internet or Web-survey is a form of data collection using a server-side system where the survey is being completed while connected to the Internet (Couper, 2008). The main difference between client-side surveys and the Web-survey is whether the Internet connection is on while the respondent is completing the survey. A Web-survey is considered more complex than other survey methods due to the data handling that needs to be sent from the client to the server and stored or processed in some way on the server using a Common Gateway Interface (CGI), which requires scripts and other elements in order to run the survey (Couper, 2008).

A signal can be composed of different frequencies, which is known as its frequency component. Owing to the frequency component characteristic, an auditory stimulus of any type of sound, such as pure tone, speech, or music, is perceived differently. This is related to the human hearing sensitivity that varies with frequency. The effectiveness of dispersion of the reflected sound as part of the diffusion mechanism is frequency dependent with a particular directionality. Some portions of the dispersed energy are already attenuated as they reach the receiver. Since the diffuseness of a sound field varies with frequency, the auditory stimuli reproduced for the subjective measurement might be perceived as ‘missing’ some of the frequency components. However, the diffusion might impact at frequencies where human ears are not sensitive, for instance, at frequencies below 200 Hz or above 5000 Hz since most speech is conveyed by sound energy between 200 and 5000 Hz. The first objective of the subjective measurement with the Web-survey was therefore, to observe frequency ranges that are most influential on speech intelligibility for a male and a female speaker.

In the computer simulation, eliminating or filtering certain frequencies theoretically can be done through absorption by applying materials that are totally absorptive (i.e., absorption coefficient $\alpha = 1$). The numerical calculation of the amount of surface absorption (i.e., architectural filtering) during propagation of the sound energy is embedded within the simulation algorithm and based on the hybrid method. In audio engineering of sound reproduction, filtering can be done on the final product of the measured signal at the receiver through signal processing. The purpose of room acoustics design, however, is to manipulate architectural surfaces so they can filter out unwanted frequencies prior to any use of a sound system. Understanding the capability of architectural filtering as compared to signal processing filtering was the second objective of this Web-survey.

Part 1 of the survey addressed the second objective by comparing two auditory stimuli obtained from two different impulse responses. One was an impulse response with frequency filtered through absorption of surfaces (i.e., “architectural filtering”) and the other was impulse response filtered using signal processing.

Auditory stimuli were linear PCM audio (.*wav*) formatted files generated from auralization in EASE. The simulated binaural impulse response (BIR) of model AA21 was convolved with an anechoic recording of speech. Characteristic of the full bandwidth auralization (i.e., Version 1) was the condition “as-is” of the room with 0.2 absorption coefficient assigned to all surfaces. The source was placed at the front of the room representing a teacher position, with the listener or receiver at the middle of the room.

The architectural filtering of the first stimuli in Part 1 was done during the computer modeling. The surfaces were assigned with absorption coefficient of 1 for the frequencies being eliminated. From an audio file with a full frequency bandwidth, filtering was done for four different frequency ranges. This has created four versions of the survey, which are:

1. Version no. 1 was the auralization result with the entire frequency range from 20 Hz – 20 kHz.
2. Version no. 2 was the auralization result with impulse response of 0 – 4000 Hz (i.e., filtered out frequencies >4000 Hz).
3. Version no. 3 filtered out frequencies <250 Hz (i.e., impulse response from of 250 Hz -20 kHz).
4. Version no. 4 was the auralization from impulse response within the range of 250 – 4000 Hz (i.e., a combined filtering of version 2 and 3).

Using the result of auralization, certain frequency ranges were filtered out with the signal processing technique to create the second stimuli of Part 1. Selection of these frequency ranges was based on the theoretical background that humans are most sensitive to frequencies above 250 Hz and below 8000 Hz along with the assumption that human speech is usually within the range of 250 Hz – 4000 Hz for male and female speakers (i.e., frequency below 250 Hz is less important for speech intelligibility).

Details of the interface and the auditory stimuli content of the entire survey are provided in Appendix B. Each subject was only assigned to a version of the survey by randomization as they started the Web-survey. Part 2 addressed only measurements on the sensitivity of frequency components in speech intelligibility using impulse responses that were generated from the “architectural filtering” technique. A schematic of the Web-survey auditory stimuli is shown in Table 2.4.

Statistical inferences were drawn based on the results within each version and between all four versions. From 160 respondents that visited the survey link, only 80 of them completed Part 1 of the survey and 53 completed the entire survey. This high drop-off rate was due to the Internet connectivity problems that created delay or failure during the stimuli streaming process. Further details of this matter are discussed in the following paragraphs. The effect of this problem resulted in fewer respondents completing the Web-survey, which led to a lower statistical power.

Table 2.4. Versions of the Web-survey of four different frequency ranges utilizing “architectural filtering” and signal processing in creating the auditory stimuli.

Survey Sections	Auditory Stimuli	Version			
		1	2	3	4
Part 1	First : Speech 1	Full bandwidth Architectural	Below 4 kHz Architectural	Above 250 Hz Architectural	250 Hz – 4 kHz Architectural
	Second : Speech 1	Full bandwidth Audio Processing	Below 4 kHz Audio Processing	Above 250 Hz Audio Processing	250 Hz – 4 kHz Audio Processing
Part 2	Third : Speech 2	Full bandwidth Architectural	Below 4 kHz Architectural	Above 250 Hz Architectural	250 Hz – 4 kHz Architectural

In Part 1, respondents were asked to identify noticeable differences in two different auditory stimuli of speech. If the subject recognized differences, then they continued to answer the next survey question. Subjects were then asked to define which stimulus sounds were better articulated, more brilliant, and louder. Others who did not recognize any noticeable difference and clicked on the answer “No” automatically proceeded to Part 2 of the survey given a skip logic embedded within the html coding of the survey interface.

It is important to understand that perception of speech being better articulated is related to the speech intelligibility or the ease of understanding the content. Meanwhile a speech that is more brilliant means that the mid-high frequency contents (2000 – 4000 Hz) sounds more vivid than the mid-low frequency components (500 – 1000 Hz).

A noticeable difference on the paired-stimuli of Part 1 due to different filtering techniques was recognized by all respondents of the survey version no. 4. As mentioned earlier, the first stimulus was reproduced with architectural filtering and the second stimulus was filtered through signal processing. This was the speech with frequency content of 250 to 4000 Hz where frequency components were filtered. The result indicated that frequency filtering with the signal processing created a more preferable speech.

Among the 9 subjects, 78% perceived the second stimulus as better articulated and 67% perceived it to be more brilliant. Filtering out the low frequency components (i.e., version no.3) by architectural and signal processing did not affect the speech quality. This is shown by 62% of respondents indicating unnoticeable differences.

Table 2.5. Results from Part I of Web-survey with four versions of frequency filtering.

Version	1 (full bandwidth)		2 (<4000Hz)		3 (>250Hz)		4 (250-4000Hz)	
Stimulus	First	Second	First	Second	First	Second	First	Second
Does ‘First’ sound different than ‘Second’?	34 out of 43 (79%)		9 out of 15 (60%)		5 out of 13 (38%)		9 out of 9 (100%)	
Sounds better articulated	17 (50%)	17 (50%)	3 (33%)	6 (67%)	3 (60%)	2 (40%)	2 (22%)	7 (78%)
Sounds more brilliant	21 (62%)	13 (38%)	5 (56%)	4 (44%)	2 (40%)	3 (60%)	3 (33%)	6 (67%)
Sounds louder	22 (65%)	11 (35%)	7 (78%)	2 (22%)	4 (80%)	1 (20%)	5 (56%)	4 (44%)

Part II served the objective to observe frequency ranges that mostly influence speech intelligibility using the speech of a female speaker. In Part II of the survey, the respondents were given another auditory stimulus of a different speech. The stimuli of the four different versions were processed with the "architectural" filtering. After listening to the passage, subjects were assigned to answer question no. 1, which consists of 7 sentences related to the information given in the passage. Subjects provided a "true" or "false" response (see Appendix B).

Fifty-three respondents completed Part II. Figure 2.40 shows the percentage of subjects who answered correctly in each version. It also provides information about the number of correct answers from zero (i.e., none were answered correctly) to 7 (i.e., all 7 questions of "true" or "false" were answered correctly). The histogram in Figure 2.40 is skewed to the right for all versions indicating that there were no difficulties in understanding the information in the passages given such auralization qualities. It also indicated that the selected frequency ranges filtered within each version were not eliminating the important frequency components of the speech signals, which here was selected within the range of 250 Hz to 4000 Hz.

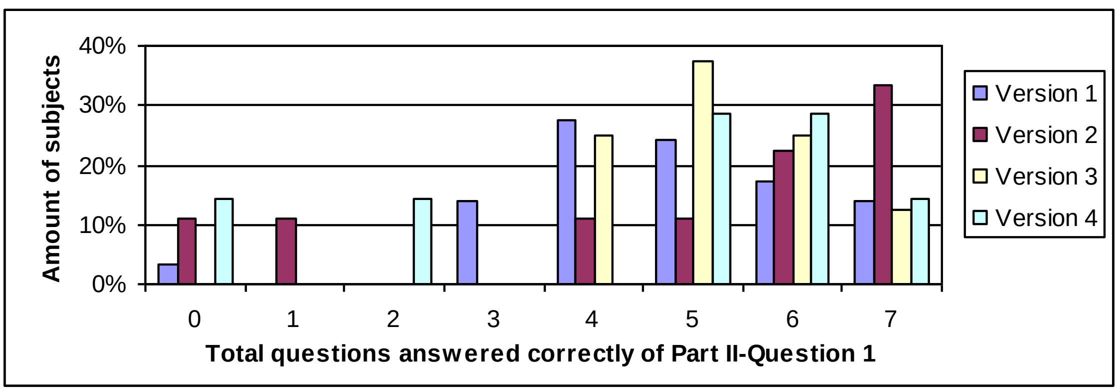


Figure 2.40. Results from Part II – Question 1 of Web-survey.

Following Question no.1 in Part II were several questions to judge the speech quality as related to the room acoustics. Seven categorical rating scales were used with details of questions provided in Appendix B. There is no significant correlation between the survey versions (i.e., variations on the frequency contents of the auditory stimuli) with the acoustical quality judgment.

Whether or not the respondents were native speakers could influence the results, given their ability to understand the speech context, when speech passage was used as the stimuli material. Related information of the respondents was collected using Part III of the survey (see Appendix B). There is a significant difference between native and non-native speakers based on the result of the correct answers in “true” or “false” statements of Part II - Question no 1. This is shown by the *likelihood ratio chi-square* probability value of 0.7.

The result of this preliminary survey also addressed the objective of determining the reliability of Web-surveys for research in room acoustics. Six significant factors that are potential sources of errors governing the implementation of Web-surveys for studying room acoustics were identified, which are: (1) computer hardware and sound reproduction device; (2) operating system; (3) Internet connectivity; (4) how familiar the subjects are with the survey content and terminology; (5) complexity of the survey interface; and (6) space where the survey was completed as related to the background noise.

The first three factors above are related in that a deficiency of performance in one factor affects the ability of the other factors to provide reliable data. Presentation of the auditory stimuli in the Web-survey is an add-on to the basic survey instrument (i.e., basic HTML scripting) known also as active content, which can be loaded automatically when the Web page is loaded as a background to collect parallel data, or loaded when it is being executed as a response to a user action.

The loading and executing process requires additional scripting to the basic HTML scripting (i.e., Dynamic HTML), which without a sufficient internet bandwidth can create delays. Loading an active content requires a plug-in at the client-side. This requires the respondent to be knowledgeable with the computer operating system. The psychological reaction related to unfamiliarity with an operating system might create measurement errors.

Survey results may vary due to the audio quality of the stimuli generated from computers with different capabilities as well as the specific model of headphone or speaker used for playback. Since different operating systems and software environments handle and process auditory stimuli differently, generating the stimuli may require additional software. Also since the auditory stimuli are streamed, the speed and reliability of the subject's Internet connectivity is, therefore, a significant factor in the production of an uninterrupted audio flow.

One of the advantages of Web-survey is the ability to obtain a wide target of populations and to conduct cross-cultural studies. However, the cultural diversity in pattern and style of communication should be considered, such as the characteristic of the used passage for the auditory materials. This characteristic may affect the speech intelligibility assessment.

Types of coverage errors are missing units, ineligible frame population, and duplications or over-coverage. Some potential sources of coverage errors in Web-survey with missing units are having short email address lists and unavailable Internet connectivity during the survey period. There is a lack of experimental control in Web-survey since it is difficult to contact the unit to determine eligibility, and willingness to participate in the survey is based on self selection. Multiple submissions are difficult to avoid in Web-survey. This can be eliminated by providing a user Login system and ID authentication.

The types of non-response errors that might occur in Web-survey are the unit non-response and item non-response. Unit non-response describes the failure to obtain any of the substantive measurements from the sample person, mostly caused by the inability to contact the participant or refusals in survey participation. The drop rate of respondents visiting the survey link to those that respond might indicate refusals. Item non-response is the failure to obtain information for one question from a sample person by not answering the question or when individual questions are skipped. The potential causes of item non-response in Web-survey are the complexity of the task, the self-administered mode that is considered complicated, and the time consuming issue. As mentioned earlier, the impact of the non-response error results in fewer respondents, which leads to less statistical power.

2.4.4 Preliminary Study Utilizing Immersive Virtual Environment

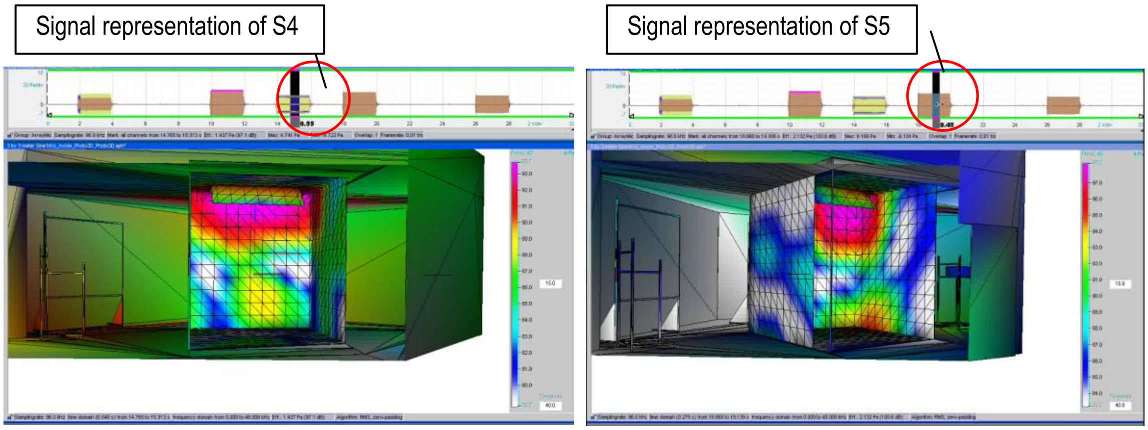
The basic methods and techniques used for auditory representation in a virtual environment and for construction of a spatial perception of the virtual reality (VR) rely on the simulation of sound propagation, auralization, and auditory reproduction. Application of the approach emphasized subjective evaluation using digital data with the Cave Automatic Virtual Environment (CAVE) system. It is an immersive VR environment system provided in the University of Michigan 3D Lab facility (http://um3d.dc.umich.edu/). The projectors are directed to four projector screens including the floor of a room-sized cube.

The auditory stimuli used within the CAVE are *.wav* formatted digital audio files reproduced through the CAVE's sound system. In an attempt to evaluate the auditory representation of a designed space, the subjects are positioned at the same location for their selected visual scenes and auditory scenes. This enables one to experience and interpret the room acoustic conditions before and after the design changed.

The use of a real time feedback data collection system provides a new alternative to capture the user reaction to a given visual and auditory cue simultaneously. The subject recognition of the sound quality and its room acoustics characteristics may be different with and without the visual stimuli. The advantage of this integrated simulating technique within virtual environments helps to accelerate decision making during the design process (Utami and Navvab, 2011).

An objective measurement of an immersive VR environment system provided in the University of Michigan 3D Lab facility (http://um3d.dc.umich.edu/) was conducted using the Acoustic Camera system. The projectors in the CAVE were directed to four projector screens including the floor of a room-sized cube. The background noise level, reverberation time, and loudspeakers' performance were the variables measured.

A noise image mapping of the CAVE surfaces shown in Figure 2.41 was used to observe the reflection paths and the directionality of the sound energy coming out from the loudspeakers. The sounds recorded by the microphones are shown in the upper bar of the noise image mapping. The color mappings on the CAVE surfaces indicate total SPL (in dB) that arrived at the microphones due to the direct and reflected sounds. The legend interprets the range of loudness level.



Sound mapping of an imaginary loudspeaker

Sound mapping of a real loudspeaker

Figure 2.41. Signal representation and the mapping of the sound pressure level (SPL) produced by the Noise Image software within the CAVE space, used to visualize the virtual sources.

The computers, projectors, and other electronic devices produced a high ambient noise level, which exceeded 40 dB during the measurement. The average reverberation time (T_{60}) was in the range of 0.5-0.6 seconds. Performances of the loudspeakers were evaluated by displaying a recorded sound of a Mozart composition played by a string quartet at 8 locations within the virtual space. These positions are shown in Figure 2.42. Sources 1, 3, 5, and 7 in the virtual space were matched to the positions of the loudspeakers in the real space.

For VR applications, the auditory display devices should be able to provide 3D localization cues. The signal received at the ears is influenced by all the signals transmitted from the auditory display device together with the transformation that the signal undergoes as it propagates through the sound path.

Nine subjects were brought into the CAVE and experienced the auditory stimuli that were reproduced in sequence from four loudspeakers (Utami and Navvab, 2011). The subjects were graduate students in Architecture who enrolled in a course of environmental design simulation. Therefore, subjects were already exposed to theoretical background in acoustics and virtual simulation. The recorded sounds used as the stimuli were the same as the ones used in the objective measurement.

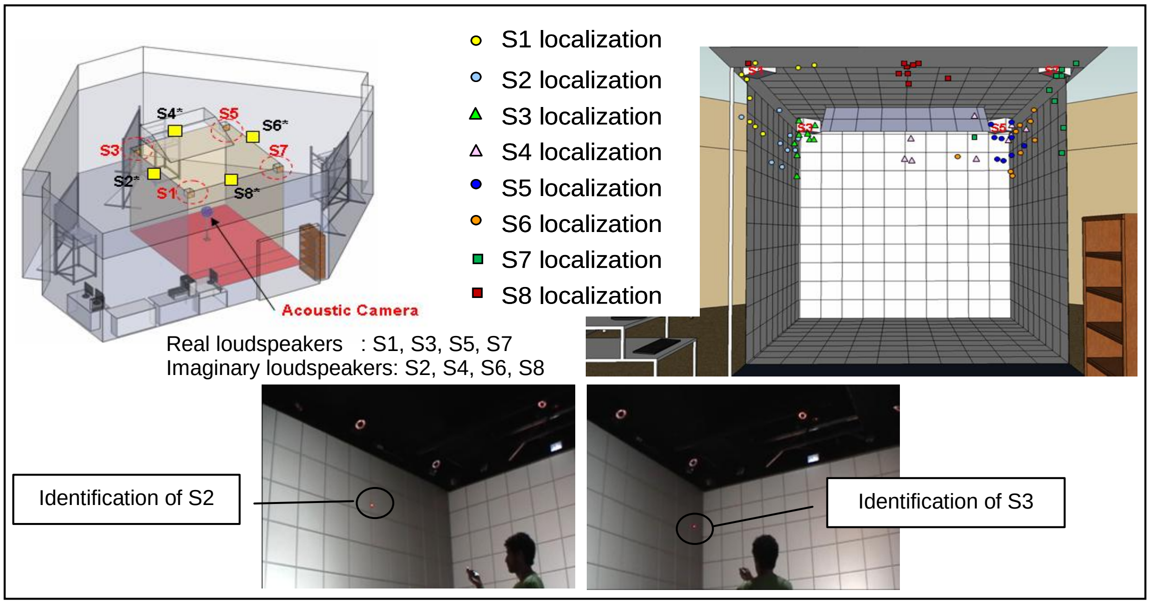


Figure 2.42. Identifying virtual sound source positions in the CAVE through a subjective testing.

By using a laser pointer, the subjects indicated the locations where the auditory sources were perceived. This process was recorded and results of the laser points are represented on a 3D drawing of the CAVE with the grid scene as shown in Figure 2.42. Even though the background noise level was high, the subjects were still able to locate the sound sources as they stood inside the cubical space. The results show that all the sound sources in the virtual space were identified and localized both from the objective measurement and the subjective testing.

2.5 Summary of Technique Details

By obtaining an impulse response either measured or simulated, the sound-field diffuseness can be characterized by using the coherences and other objective parameters as described in section 2.1. The sound-field properties indicated by each parameter are associated with the observation of the early and late reflections of the entire impulse response. Even though subjective attributes associated to these parameters are well defined (see Figure 1.3), it is essential to select a word or term that explicitly describes the hearing impression. Some subjective attributes are highly correlated, which requires them to be used simultaneously as shown in Table 2.3. A subjective assessment similar to the preliminary study described in 2.4.2, which applies well-selected subjective attributes, can provide more assurance that the results will confirm the acoustical condition predicted by the objective parameters.

The sensitivity of a computer model to visualize the impact of a diffuser in a simple rectangular room depends on the absorption properties as shown in section 2.4.1. If auralization is done using this computer model, then the frequency component of the anechoic sound utilized should be observed in advance. A space with high absorption at certain frequencies will affect the auralization result if those frequency components are dominant within the anechoic sound. This was shown in the Web-survey results provided in section 2.4.3.

Besides the Web-survey, this chapter also explored the possibility of using an immersive virtual environment as a new method for subjective assessment in room acoustics. The ability of auralization to synthesize a virtual source inside a virtual space can be validated within the CAVE system capabilities. This was proven by the ability to recognize locations and sound pressure output of the virtual sources with the Acoustic Camera measurement and was confirmed by the subjects' sound localization inside the CAVE.

Overall, this integrated methodology provides the ability to conduct a thorough study of sound-field diffuseness, especially because each of the cases studied required a unique experimental setup that accounted for its architectural and acoustical properties.

Chapter 3

Results and Analysis of the Objective Parameters

The results of objective parameters from field measurements and computer simulations of the cases studied are provided in this chapter. The discussion of the results is provided separately for each case studied based on the geometrical characteristics of the space and a generalization of it for all cases studied.

3.1 Results of Each Case Studied

This section discusses the results of objective parameters of coherence for the degree of diffuseness, the total minus direct SPL (in dB), reverberation time (T_{30}), early decay time (EDT), clarity index (C_{50} for speech and C_{80} for music), and calculated listening envelopment (LEV_{calc}) for all cases studied.

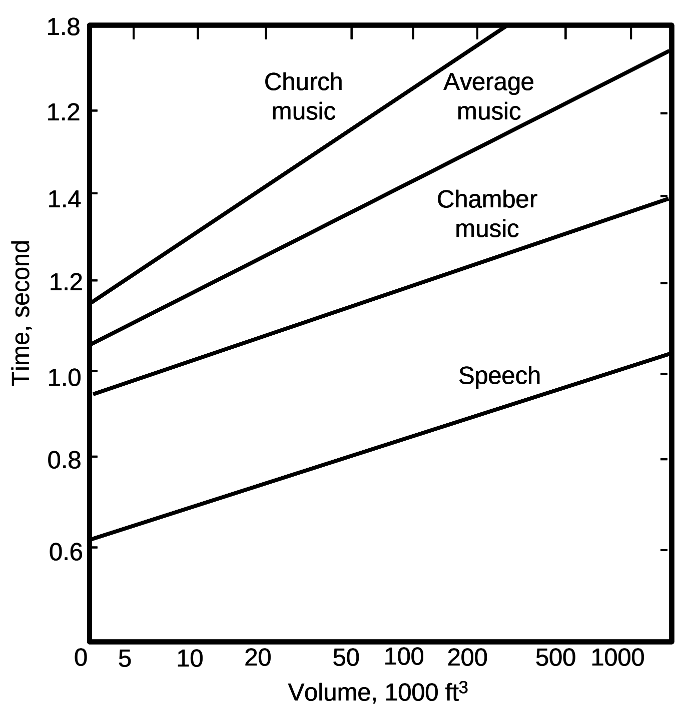


Figure 3.1. Optimum reverberation time for a given room volume (Hemond, 1983).

3.1.1 Small Room with Non-parallel Walls

The three important space elements in the Duderstadt Audio Studio (DAS) are the RPG skyline diffusers (see Figure 2.11) on the ceiling, the adjustable wall-panels, and a piano. To resemble the existing condition (as-is), design configurations using these three elements were explored through parametric runs in the computer simulation. Ray tracing in Ecotect was utilized with 50,000 sound particle rays released into the model; each ray's path was traced until the 10^{th} order reflections. All four panel positions were observed with two of them presented in Figure 3.2.

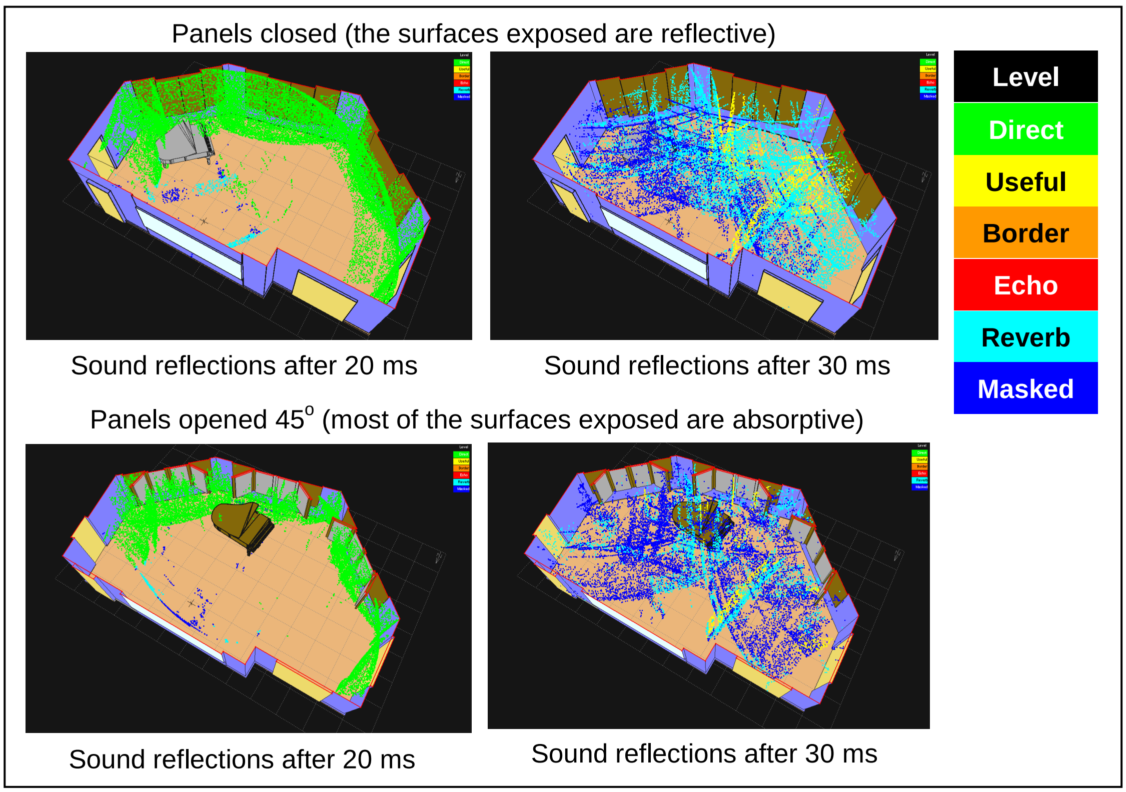


Figure 3.2. Ray tracing in Ecotect to predict sound reflections' path in the Duderstadt Audio Studio (DAS).

By visualizing the amount of reflected sound particle rays, one can see that the panels closed is more reflective than the panels opened 45^{o} with more diffused particles (i.e., light blue color). Utilizing all sixteen design configurations of the computer models described in Section 2.2.2 and illustrated in Figure 2.15, the diffuseness of the simulated sound fields is indicated by the coherences in Figure 3.3.

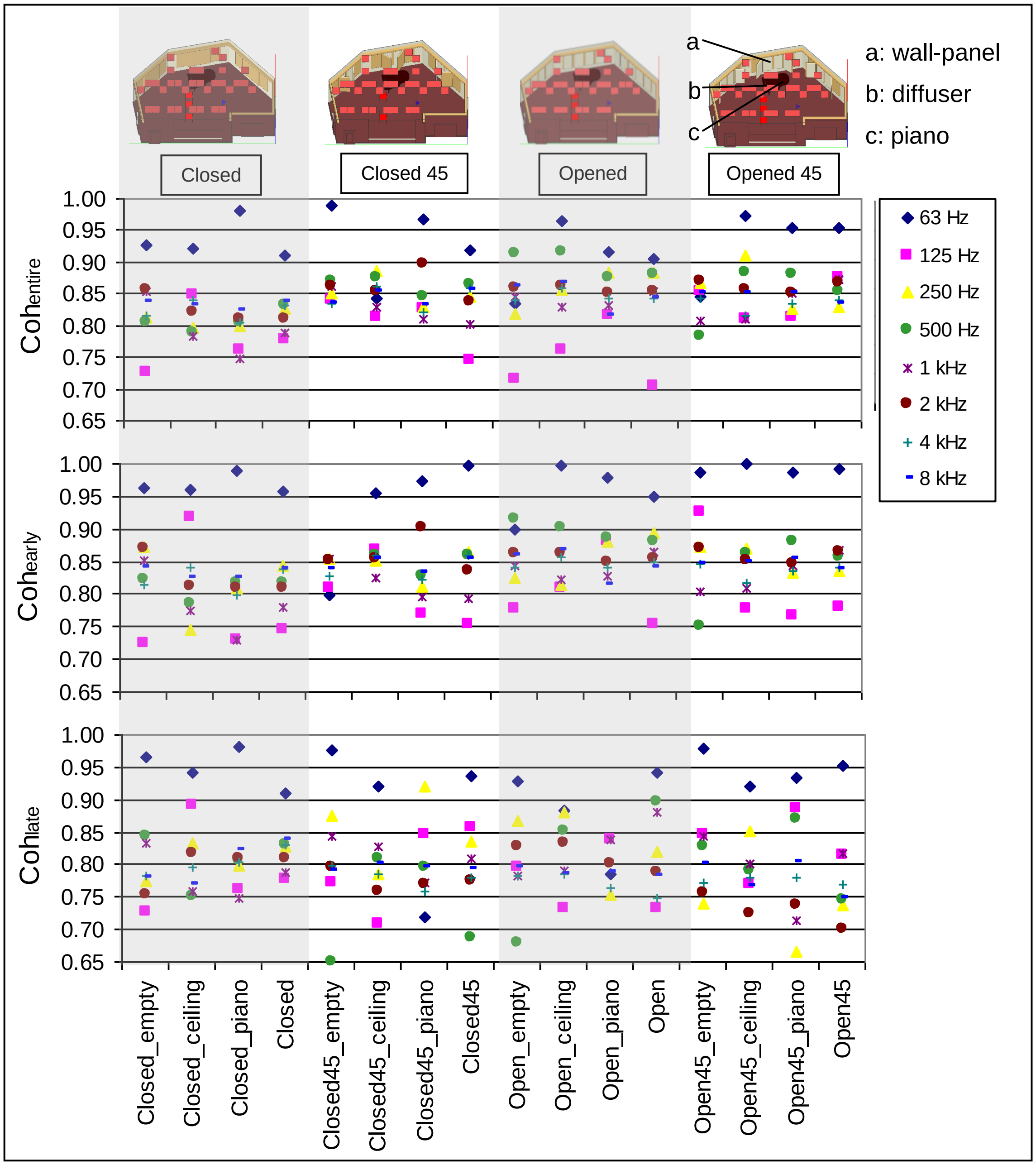


Figure 3.3. Coherences of octave frequency bands of the sixteen design configurations in the computer modeling of DAS.

For average values of the octave frequency bands, the coherences increased by 0.05 or 5% as the wall panels were opened, a position where the absorptive surfaces are exposed. Owing to the room dimensions, values of Coh_{entire} and Coh_{early} are mostly varying at the frequency 125 Hz. Receivers are within the distance of 2-3 meters from the left and right walls, while wavelengths of octave band 125 Hz are within the range of 1.9 – 3.8 meters. As for the late reflections, configurations of the three elements mentioned previously are mostly affecting the octave band of 500 Hz.

The adjustable wall-panels (see Figure 2.13) were the first element observed utilizing the computer simulation (i.e., simulated impulse response) and field measurement (i.e., measured impulse response). The coherences and objective parameters calculated from the simulated impulse response in DAS with variations on the wall-panel positions are provided in Figure 3.4.

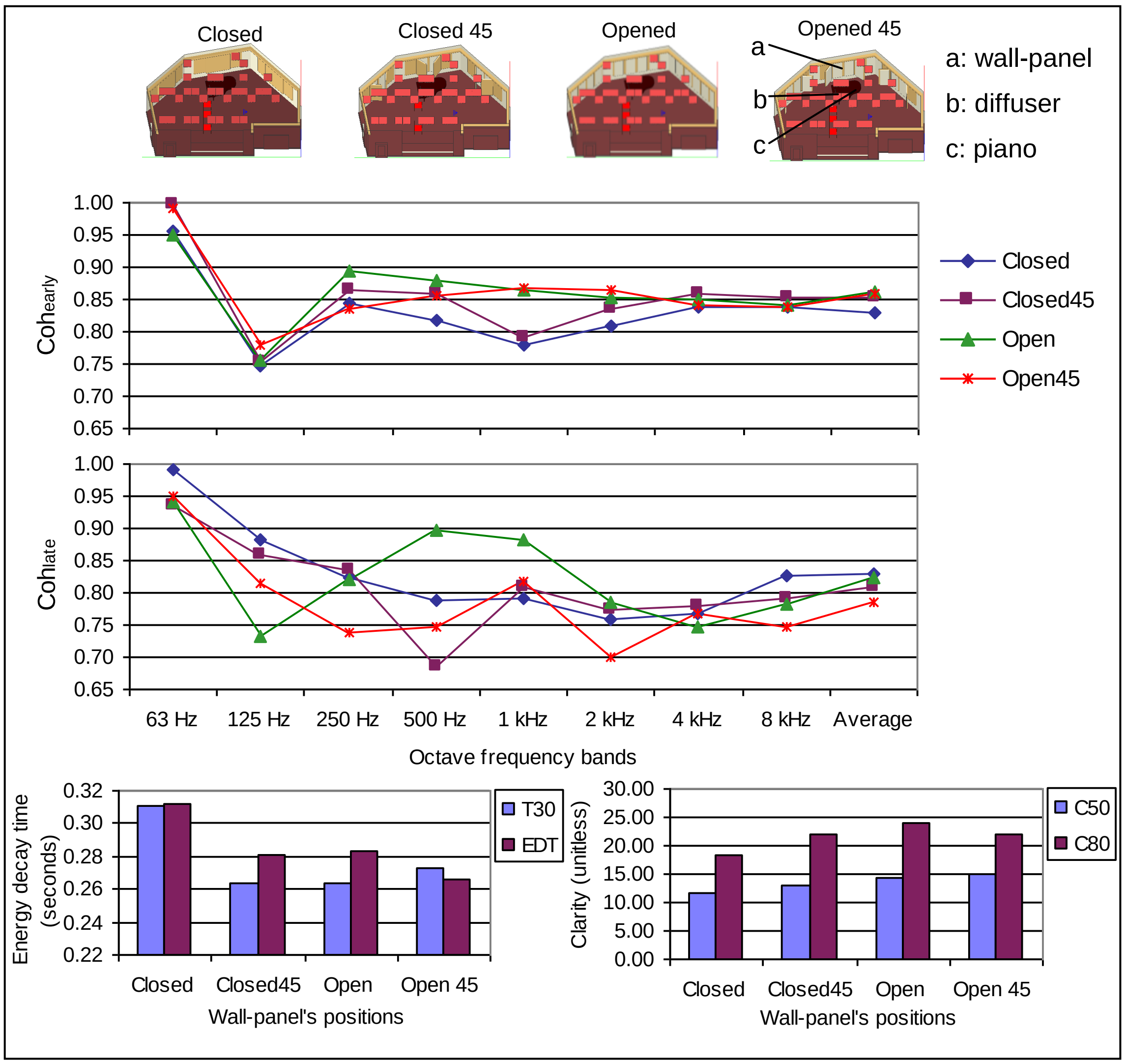


Figure 3.4. Line plots are the Coh_{early} and Coh_{late} of DAS with variations on wall-panel positions obtained from computer simulation results, while bar plots are energy decay time and clarity index.

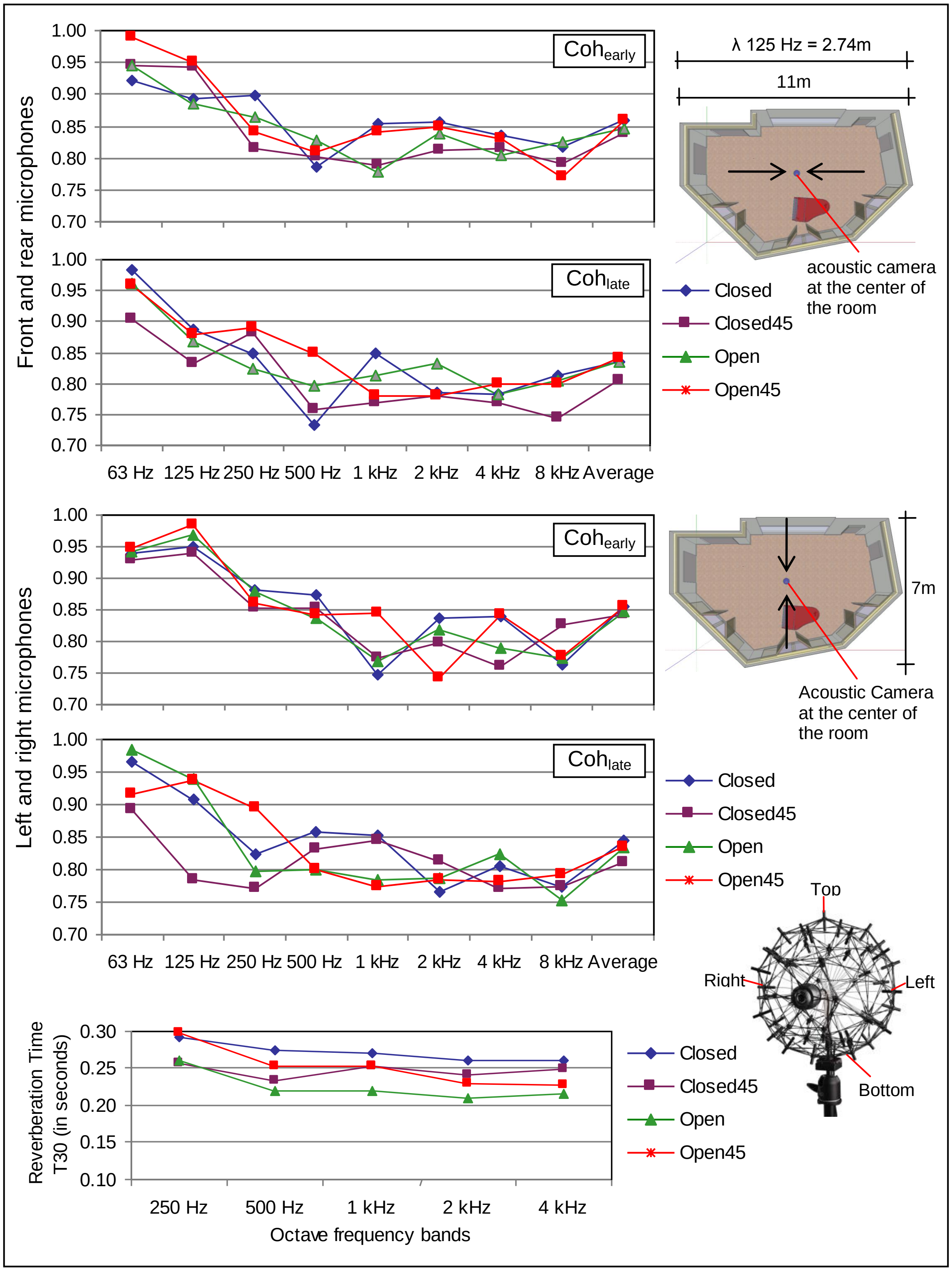


Figure 3.5. The Coh_{early} and Coh_{late} of variations on wall-panel positions in DAS obtained from the measured impulse response at two pairs of microphones (top-bottom and left-right). Bottom plot is the reverberation time (T30) calculated from the top microphone output of the same measurement.

Also, the coherences and objective parameters calculated from the measured impulse response using the Acoustics Camera in DAS with variations on the wall-panel positions are provided in Figure 3.5. It is obvious that exposing the absorptive side of the adjustable wall-panel in the computer simulation reduced the EDT and T_{30} as shown in Figure 3.4. By a closer look at the results for the panel fully opened and opened 45^o, it can be seen that the T_{30} remained relatively the same, while the EDT decreased. Opening the panels at an angle created inter-reflections between the reflective side of the panels and the walls. The early reflections were trapped between these surfaces, and only a few arrived at the microphones. There is a similarity between the computer simulated and field measurement results of Coh_{late} and T_{30} values, particularly for models with the panels closed. Furthermore, models with this panel position were used for auralizing the auditory stimuli of the subjective assessment.

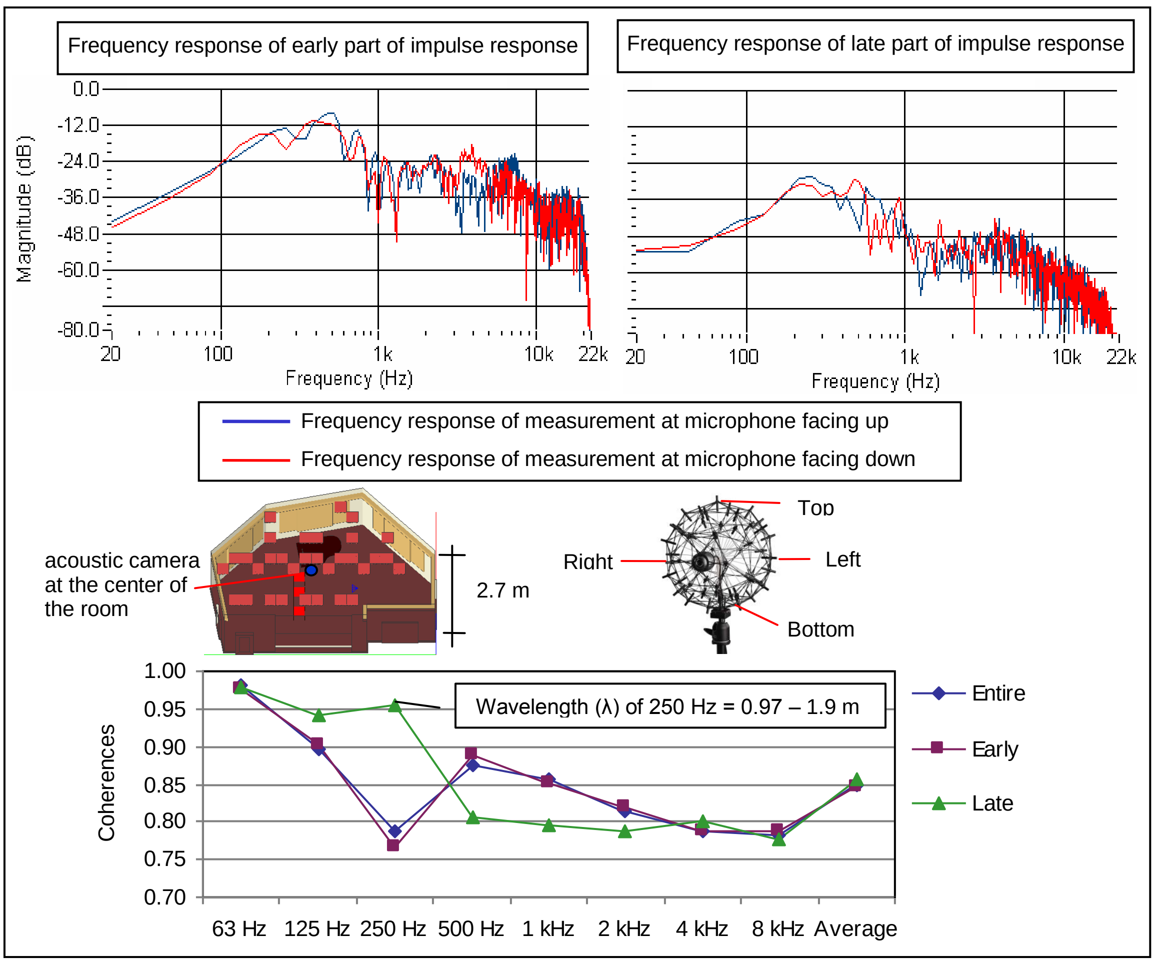


Figure 3.6. Coherences of paired microphones facing the ceiling and floor for panel closed.

The ceiling diffusers (see Figure 2.13) on the ceiling of the Duderstadt Audio Studio (DAS) were the second element observed. Since diffusers were applied on the ceiling, the outputs of the top and bottom microphones of the Acoustic Camera were used to calculate the coherences (see Figure 3.6). The lowest coherence (Coh_{entire}) for all impulse responses is for octave bands 125 Hz, indicating that the diffusers on the ceiling did not contribute diffusion to the sound field. Comb-filtering reduction within the octave band of 500 Hz is depicted in the upper graphs in Figure 3.6, and this observation confirms the high Coh_{early} for that particular octave frequency band as compared to other mid-to-high frequencies. This means that there is a more diffused condition due to the interplay reflections between the ceiling surface with diffusers and the floor. All the coherences, except the Coh_{late} at 250 Hz, have a similar pattern as shown in Figure 3.5, which were obtained from other pairs of microphones.

Meanwhile, in the computer simulation, the models of panels closed with and without the diffusers were observed with the line charts of coherences provided in Figure 3.7. Inserting 3D objects into a space reduced the space volume or the total acoustic volume. For instance, inserting 32 diffusers into the computer model of DAS has reduced the acoustic volume by 0.9 m^3. The Coh_{late} at low frequencies increased by nearly 20% with the diffusers inserted. The Coh_{early} line chart for the space with the diffusers is slightly higher for all frequencies above 63 Hz. However, these disparities in each octave band are insignificant since they are less than 0.05.

A slight change in the C_{50} and C_{80} with the diffusers applied indicated that both spaces are similar in their early and late energies. They may be dissimilar, but since the early and late portions were both impacted, a further analysis using the EDT, $T_{30,}$ and LEV is required. The EDT remained the same, while the T_{30} was reduced by 0.1 seconds, indicating a larger effect on the late reflections. This impact is confirmed by the slight increases in the LEV_{calc} value, which indicates a more diffused condition. Given values of these three parameters, it can be stated that the diffusers on the ceiling of Duderstadt Recording Studio

(DAS) affected the late reflections and, therefore, impacted the sound-field diffuseness.

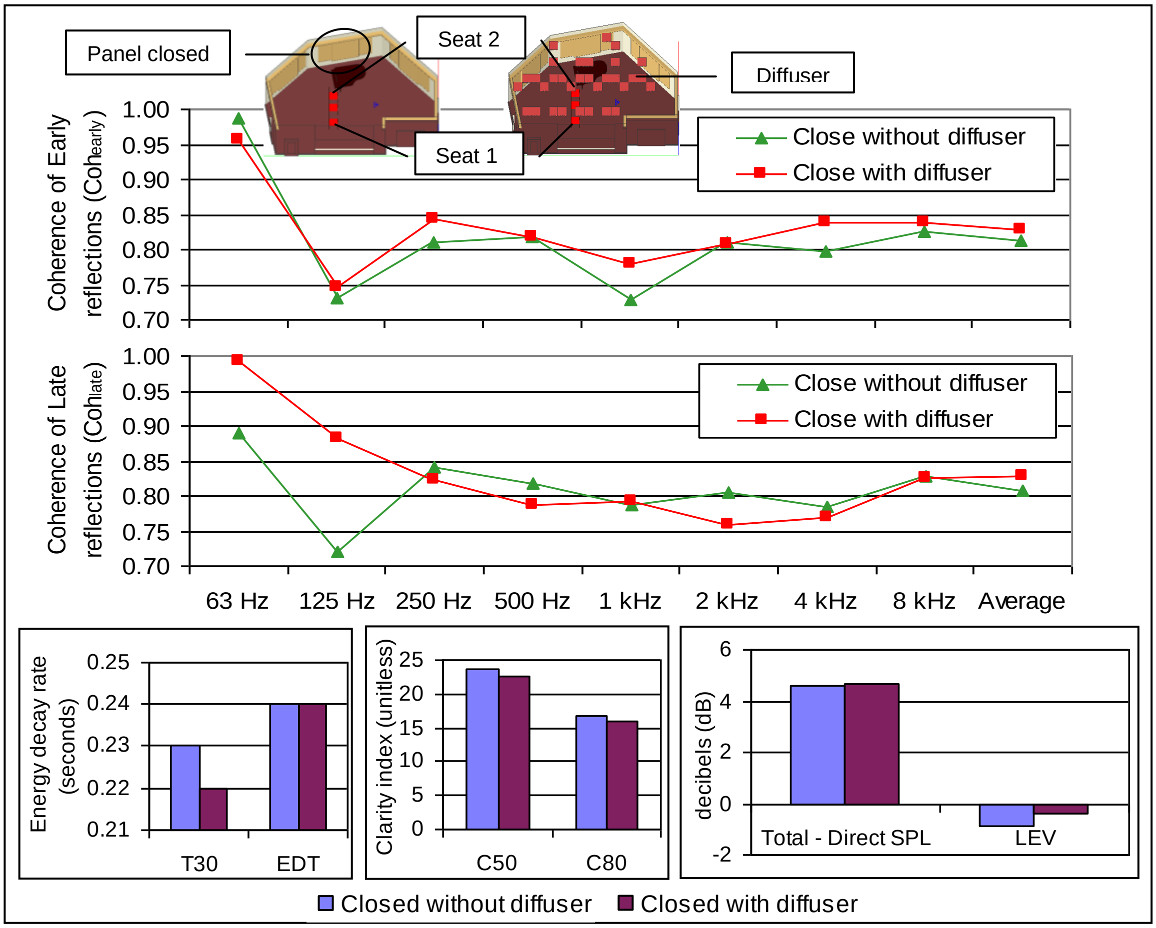


Figure 3.7. Results of observations on the impact of diffusers in DAS using computer simulation: illustration of the computer models, the coherence plots, and the objective parameters.

Visualization of the SPL on early and late reflections inside the DAS for panels opened is shown in Figure 3.8, produced by the Noise Image software.

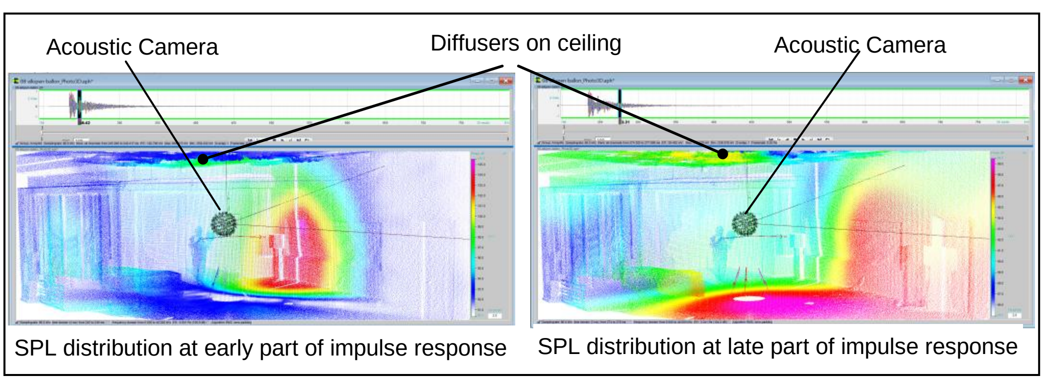


Figure 3.8. SPL distribution for two different time slices of propagation measured with Acoustic Camera and visualized with Noise Image.

The assumption that the diffusers are impacting the late reflections is validated using the right-side figure, showing a more yellow color on the SPL mapping of the diffusers surfaces. This indicated that the sound energy was maintained by the diffusers until the end of the decay.

The Coh_{late} line plots shown in Figure 3.9 are used to explore which element, the diffusers or the piano, played an important role in the sound-field diffuseness. The computer simulation result shows that the impact of a piano alone on the sound-field diffuseness is less than the impact of having both piano and diffusers or the diffusers alone. Within the simulation, the piano has contributed to the absorption as indicated by the decrease in T_{30}. EDT remains the same, which also indicates that the piano did not affect the early reflections. The larger value of the EDT as compared to T_{30} for octave band 500 Hz helped to distinguish better the early reflections component and, therefore, supported sound localization. This is later proven by the subjective assessment results and confirmed by the IACC plot in Figure 2.10.

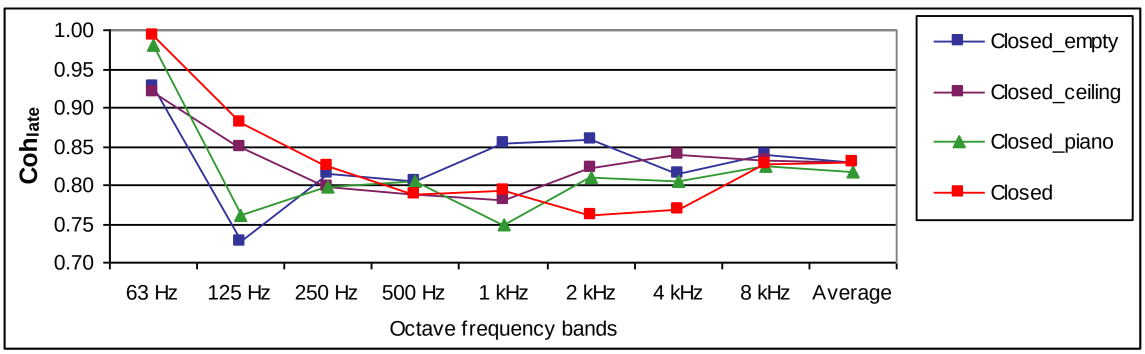


Figure 3.9. Coherences of late reflections (Coh_{late}) from computer simulation of DAS for all model configurations with the wall-panels closed.

3.1.2 Small Room with Parallel Walls

The coherences and objective parameters of the computer simulations of room 1221 Art and Architectural Building (AA21) are shown in Figure 3.10. As described earlier in the experimental settings, the parametric runs were done by initially simulating an empty space (as-is), inserting diffusers, and gradually increasing the number of diffusers.

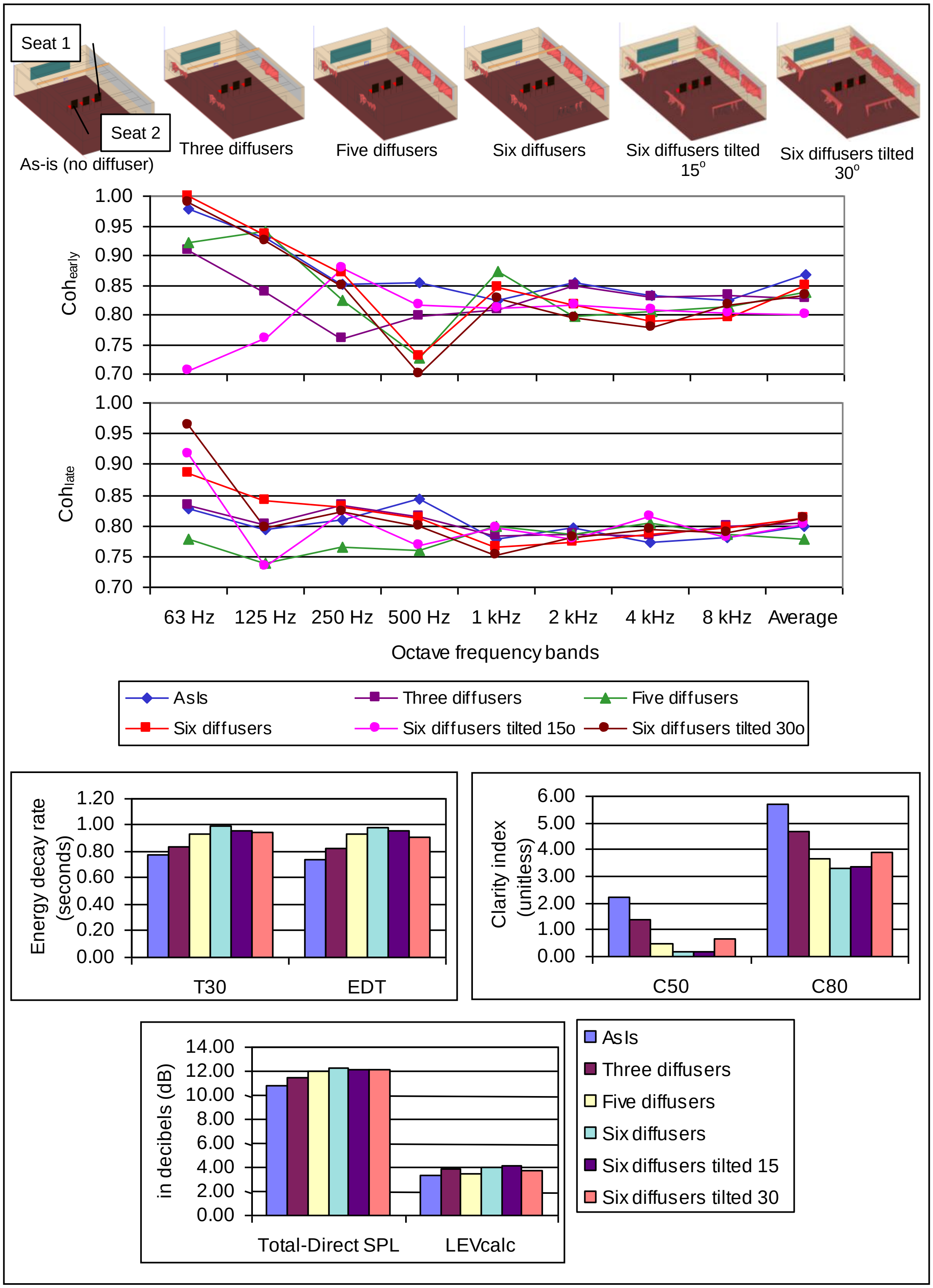


Figure 3.10. Results of observations on the impact of diffusers in AA21 using computer simulation: illustration of the computer models, the coherence plots, and the objective parameters.

The sound-field diffuseness indicated by the Coh_{late} changed as the diffusers were added for low frequencies, especially at octave band 500 Hz. Owing to the diffusers application, there is no improvement in the Coh_{early}. The impact of tilting the diffusers at a certain angle to the early and late reflections for low frequencies should be considered carefully for this size and shape of room. Tilting the six diffusers at an angle of 15^{o} apparently reduced the diffuseness for octave bands 125 – 500 Hz when compared to having six of them mounted parallel to the walls. Furthermore, in the frequency responses, having the diffusers tilted actually created more comb-filtering effect at low to mid-frequencies especially for the receiver at seat 1 as shown in the upper panel of Figure 3.11.

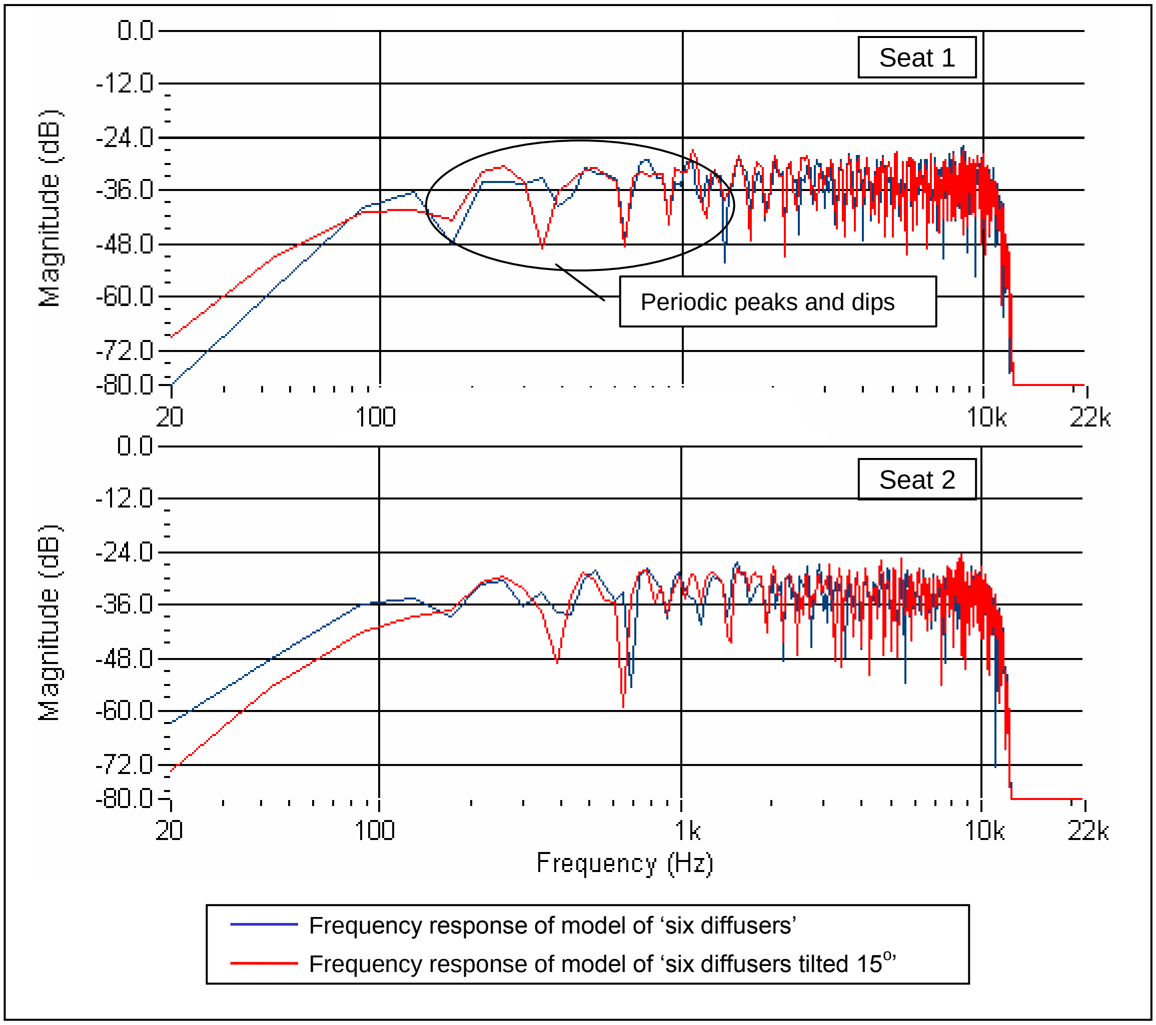


Figure 3.11. Frequency response for model with six diffusers mounted parallel to the wall and tilted 15^{o} showing the comb-filtering effect particularly at seat 1 (upper figure).

Adding diffuser panels increased T_{30} and EDT up to approximately 1.20 seconds on average. For the given room size of 165 m^3, the T_{30} and EDT of more than 1 second, with no diffuser applied, have already indicated acoustical problems for speech activity within the space (see Figure 3.1). For reverberation control, adding the diffusers failed to improve this acoustical condition.

3.1.3 Semi-Small Room with Parallel Walls

There are Golden Acoustics diffusers applied on the walls and ceiling of room 2216 in the Art and Architecture Building (AA16). The diffusers of types 1-4 described in Figure 2.11 existed within this space. Using field measurement results, the highest coherence for octave bands 125 to 500 Hz is given by the top and bottom microphones for both source positions (see Figure 3.12). The relatively low ceiling with the length-to-height ratio of 3.4 created inter-reflections between the ceiling and the floor, which causes the higher Coh_{late} at low to mid-frequencies for the top and bottom microphones. The impact of the diffusers on the ceiling becomes more dominant.

The results from simulation indicate less change in the sound-field diffuseness (i.e., using Coh_{late}), due to diffusers applied, for all frequencies above the octave band 63 Hz (see Figure 3.13). The disparities are between 0.3 – 0.6, far less than compared to 1.7 at 63 Hz.

The early diffusion affected the sound field more than the late reflections. The ability of the diffusers to reduce specularity of early reflections is indicated by the C_{80} value for model “as-is” or existing condition (see Figure 2.12), where the average clarity index at 250Hz - 4000Hz is 1.5 times larger as compared to the model without any diffusers. Using the average for mid-high frequencies of objective parameters, the values supported evidence that the acoustics treatment in model “as-is” provided the best condition.

The diffusers on the ceiling improved clarity and reverberation time as compared to the application of diffusers on the walls, especially for octave bands of 125 to 2000 Hz. Results on the subjective assessment due to the diffuser configurations in this space are provided in section 3.2.

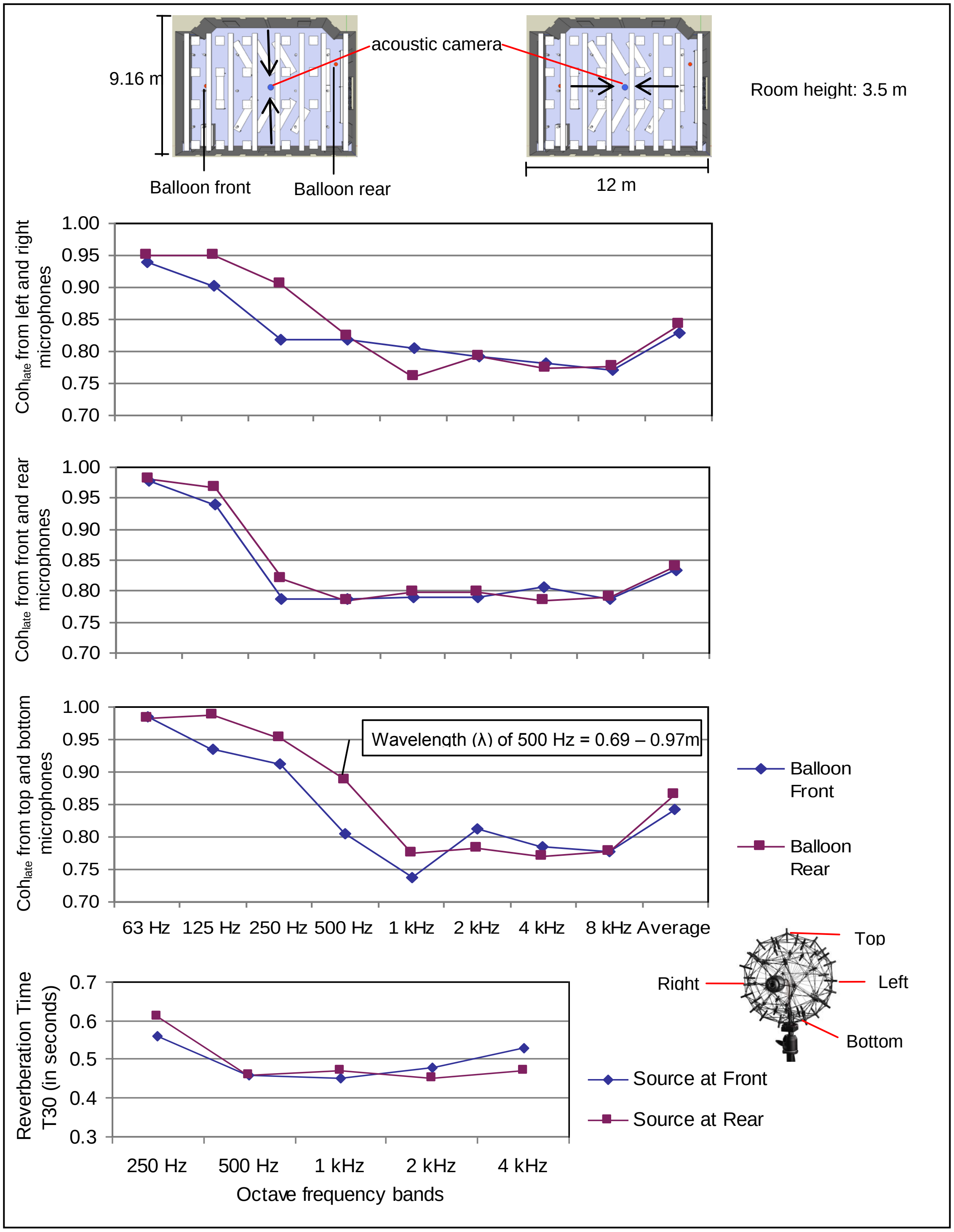


Figure 3.12. The Coh_{early} and Coh_{late} of AA16 obtained from the measured impulse response at three pairs of microphones (top-bottom, left-right, and front-rear of Acoustic Camera). The bottom plot is the reverberation time (T_{30}) calculated from the top microphone.

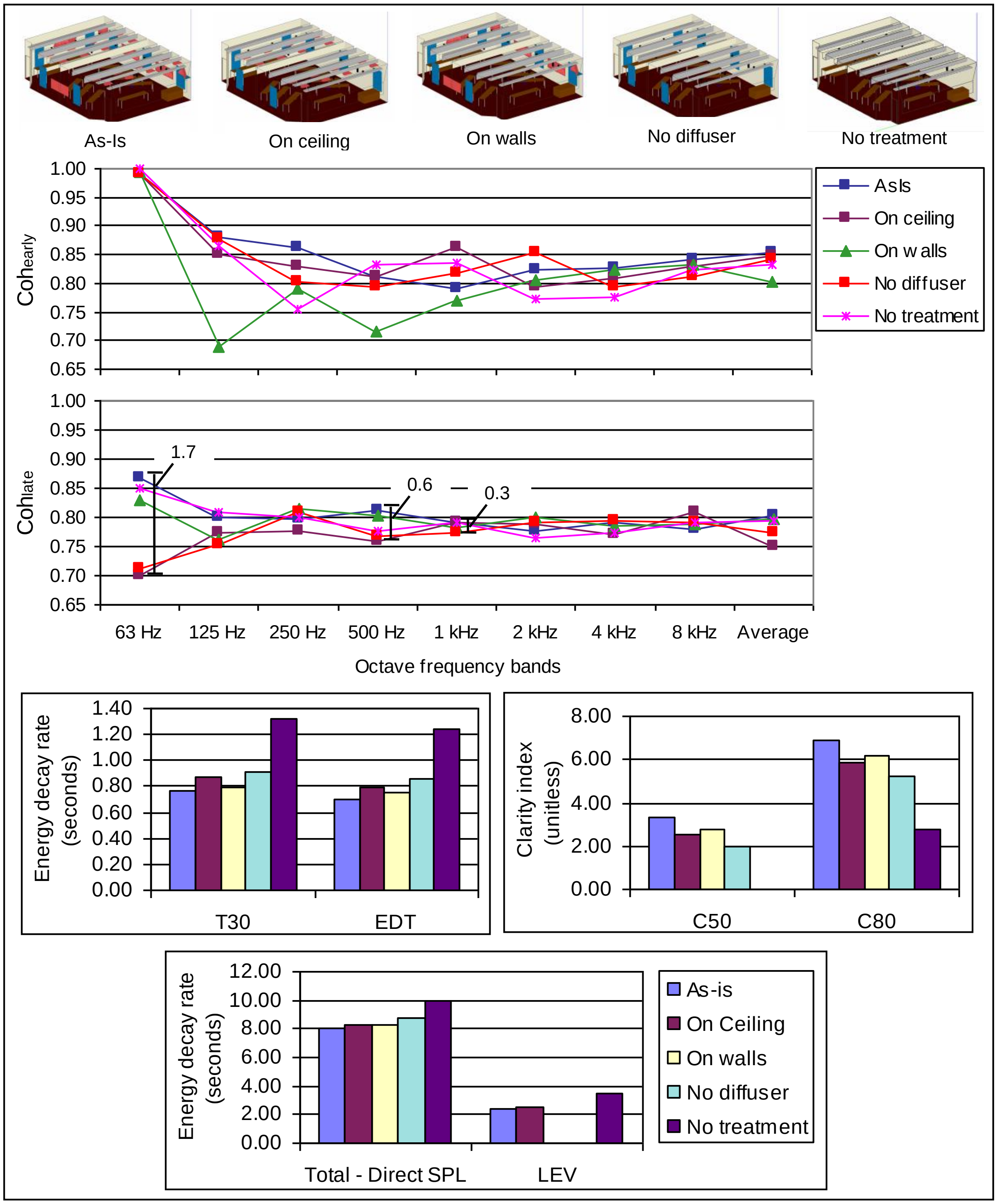


Figure 3.13. Results of observations on the impact of diffusers in AA16 using computer simulation: illustration of the computer models referred to Figure 2.21, the coherence plots, and the objective parameters.

In Figure 3.14, the results obtained from field measurements with the source burst at the front and the source at the rear side of the room are compared. The analysis is based on the data representation of the Noise Image software.

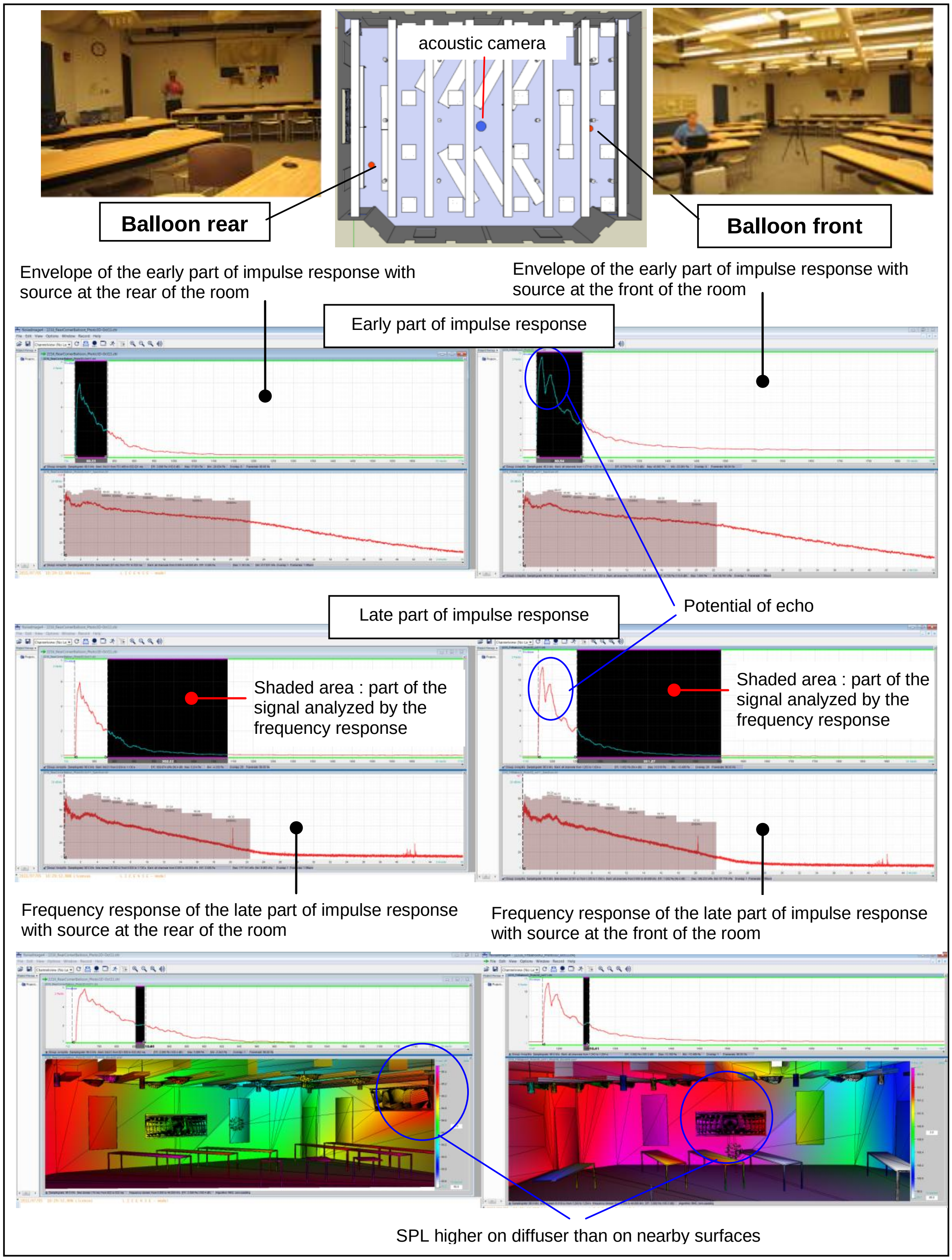


Figure 3.14. SPL distribution in AA16 measured with Acoustic Camera and visualized with Noise Image.

Instead of having a diffuser, the front wall was a flat reflective surface, which created a strong early reflection following the direct sound. This led to the potential of echo even though a comb-filtering effect was not vividly seen in the spectrograph. On the other hand, the diffuser at the rear side of the room reduced the chance of specular reflections from the parallel walls. This is demonstrated by the smooth time decay curve. It also increased the sound-field diffuseness at low frequencies as shown by the flat frequency response of the late reflections. Unsymmetrical positioning of the panels obstructed the possibility of standing waves at low frequencies. Occurrence of diffused reflections on the diffuser is shown by the SPL mapping in the last panels of Figure 3.14. These Noise Image mappings were using a time slice of 80 – 90 ms of the impulse response, which is already within the portion of the late reflections. The diffusion is indicated by a higher SPL (i.e., red to yellow color mapping) on some part of the panel surface, as compared to the nearby surfaces (i.e., green to blue color mapping).

3.1.4 Semi-Large Room with Parallel Walls

Room 170 of Dennison Hall can be described as nearly a rectangular room. In Section 2.2.3 with Figure 2.15, the room is described as an auditorium with a stepped floor audience seating area. The coherences obtained from field measurements are provided in Figure 3.15 along with the reverberation time.

The largest disparities on the Coh_{late} values calculated from the field measurements are values for octave band 63 Hz and 125 Hz. These apply for all source positions measured at left-right and front-rear microphones on the Acoustic Camera. As for the top-bottom microphones, the largest disparity is at 250 Hz. Since the lowest ceiling height was at the rear side of the room, the source burst at that location created the largest degree of diffuseness for low frequencies. The disparities of Coh_{early} due to the sources' positions are not as large as Coh_{late}, except at 250 Hz for the top-bottom microphones. The lowest reverberation time (T_{30}) values were obtained from the source burst at the center of the room (i.e., middle source). This is expected since the source at the middle is the furthest distance from the walls. The seats were the dominant sound absorbers, which reduced a lot of the early reflections.

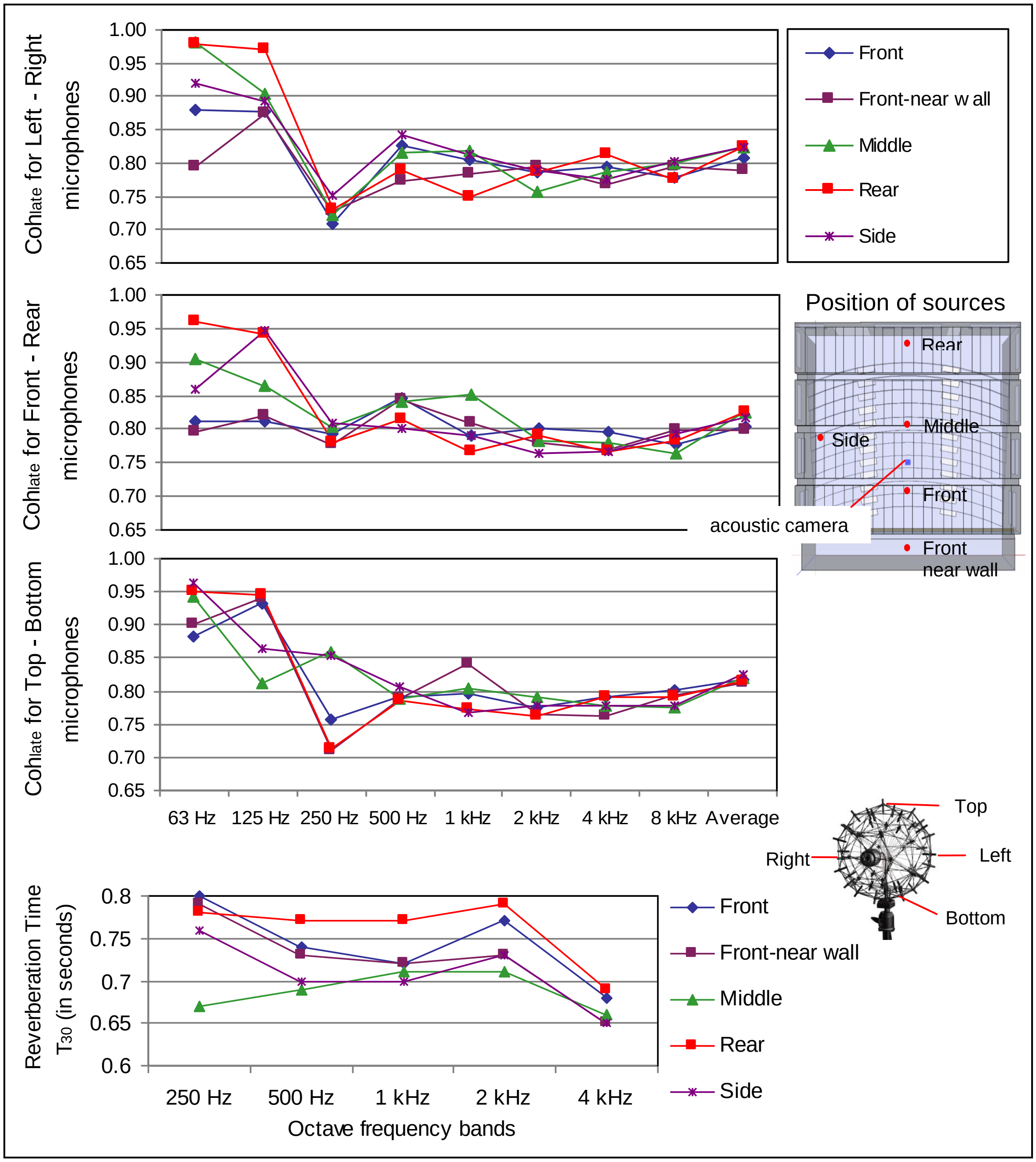


Figure 3.15. The Coh_{early} and Coh_{late} of DH170 obtained from the measured impulse response at three pairs of microphones (top-bottom, left-right, and front-rear on the Acoustic Camera) along with the reverberation time (T_{30}) calculated from the top microphone.

However, the sound-field diffuseness is not the lowest for all pairs of microphones for the source position at the center. At low frequencies, the Coh_{late}

obtained from the source near the side walls (i.e., source and receiver are at the same height) are higher than those obtained from other source positions. This has led to the assumption that the room geometry with the stepped floor also created diffusion.

Utilizing the computer simulation (see Figure 2.24) to observe the impact of a diffuser within DH170 has shown that the average coherences for all three space configurations are very similar (see Figure 3.16).

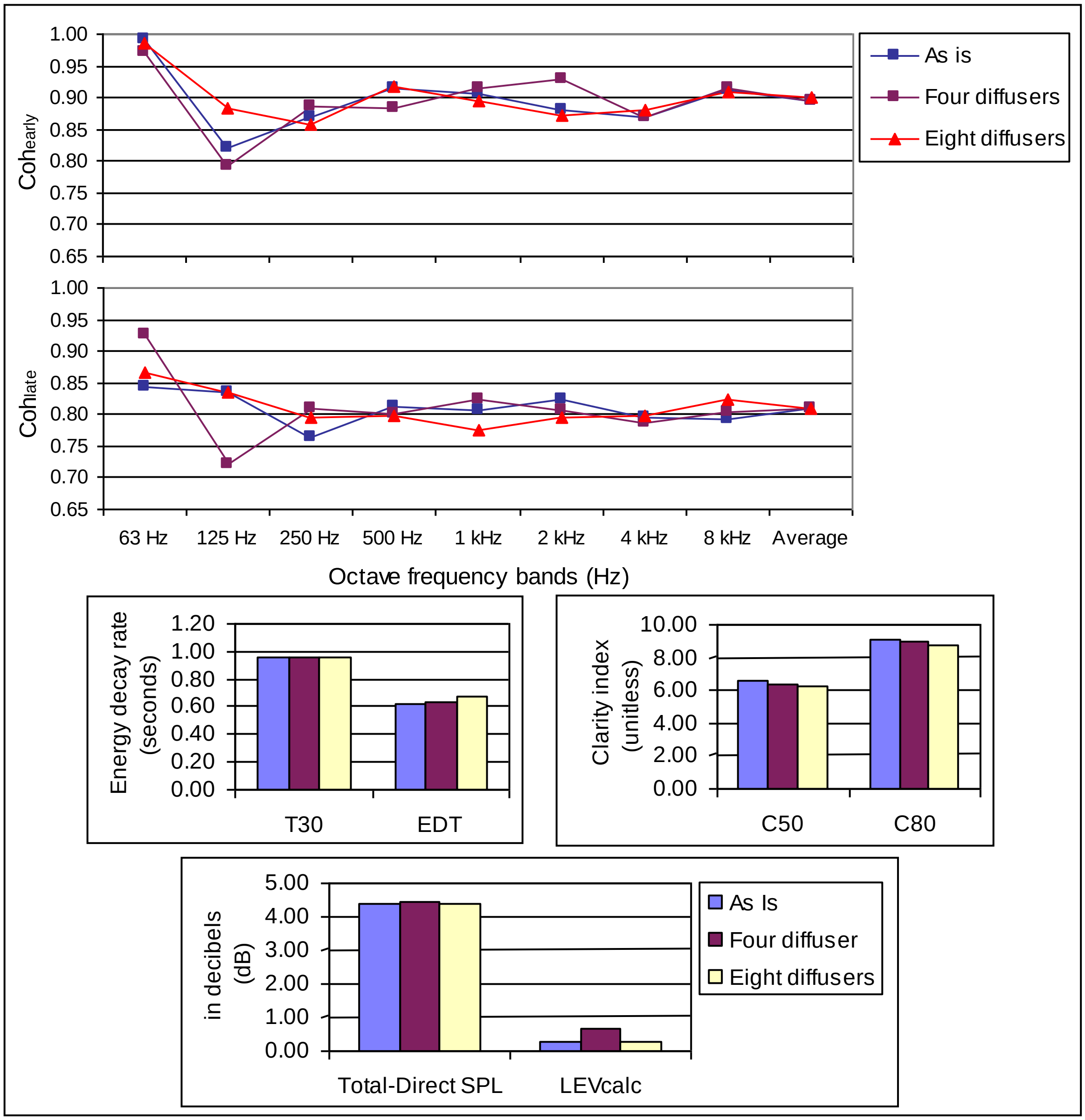


Figure 3.16. Results of observations on the impact of diffusers in DH170 using computer simulation: illustration of the computer models, the coherence plots, and the objective parameters.

Applying eight diffusers has created a uniform sound-field diffuseness for all octave bands above 63 Hz as compared to other space configurations. The total-direct SPL values remain the same with the diffusers added into the space. However, there is an indication that with eight diffusers applied, the amount of early reflections is slightly increased as shown by a slightly larger value of EDT while T_{30} remains the same. There is a potential use of the eight panels to reduce comb-filtering effect. Adding the diffusers did not improve the reverberation time; the optimum value for this space with a size of 1513 m^3 is 0.8 seconds. The average clarity index at mid-frequencies for both speech and music is still within an ideal range (i.e., good clarity is a value above 0) with the addition of the diffusers.

Further observations were done using the frequency response in the model with no diffusers and with eight diffusers applied. The comb-filtering occurring at the early and mid-frequencies due to the parallel walls was not reduced by the diffusers (see Figure 3.17). This has led to the assumption that to increase the sound-field diffuseness of a space similar to DH170, careful design is required as to the number and positioning of the diffusers, especially given the parallel walls.

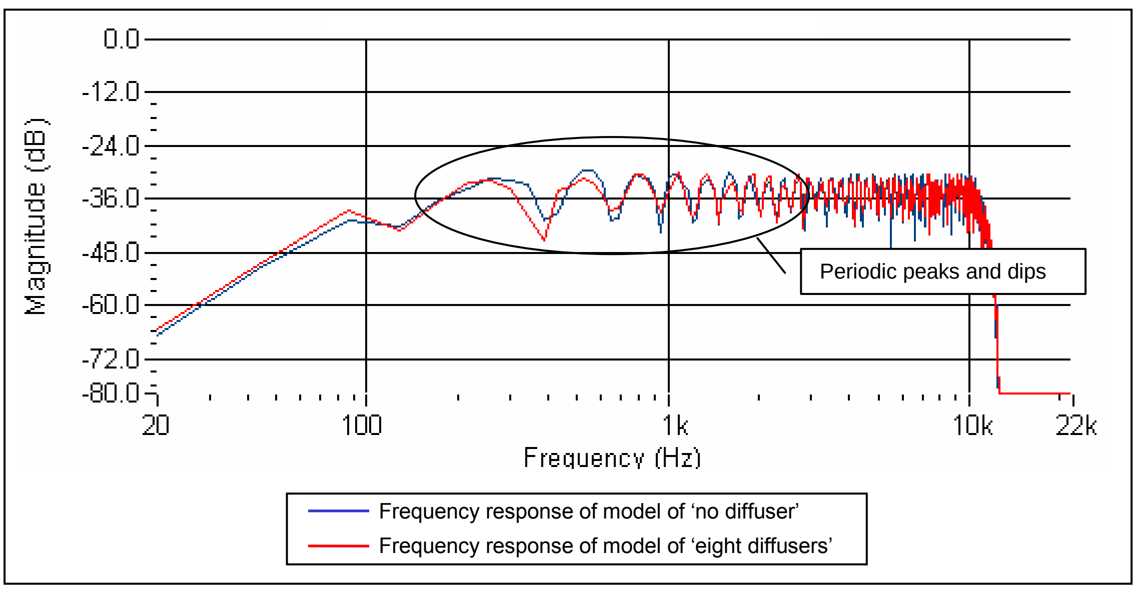


Figure 3.17. Frequency responses of computer models of DH170 to observe the comb-filtering effect.

3.1.5 Semi-Large Room with Non-Parallel Walls

Absorption panels and several diffusers on the side walls were added during recent renovation[13] in Angell Hall Auditorium-A (AHA). Computer modeling of space AHA attempted to resemble the renovated condition, particularly the absorption characteristics (see Figure 2.26).

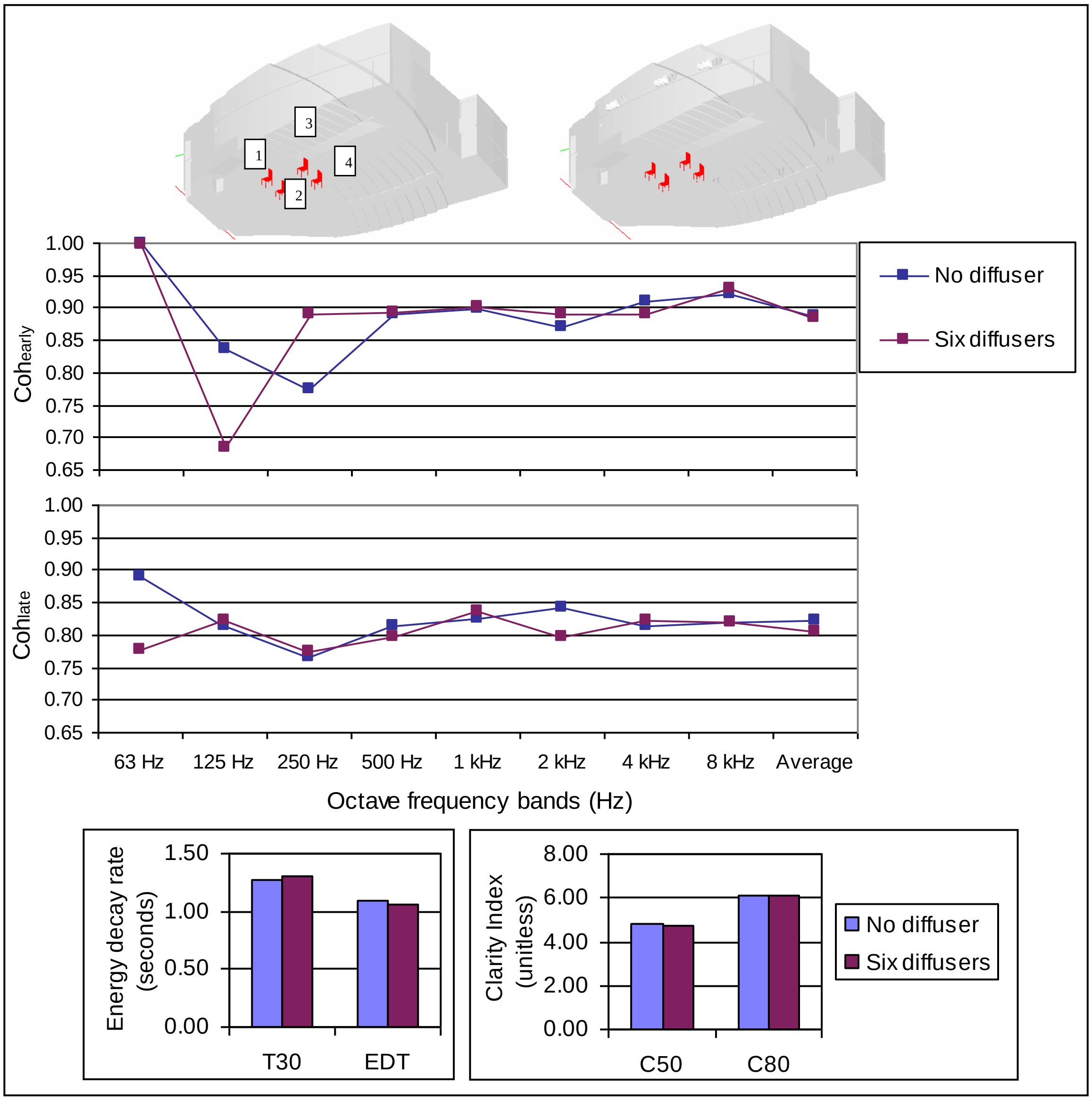


Figure 3.18. Results of observations on the impact of diffusers in AHA using computer simulation: illustration of the computer models, the coherence plots, and the objective parameters.

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[13] There is no result of field measurements in Angell Hall auditorium A (AHA) due to the renovation that was in process during the data collection stage of the study.

Among the three types of Golden Acoustics diffusers that were found in the existing condition, only one type was modeled, which is the diffuser no. 2 in Figure 2.11. The coherences and objective parameters obtained from the computer simulation of AHA are provided in Figure 3.18.

The Coh_{early} are similar to the results for DH170 with low dips at 125 Hz. The line chart pattern of Coh_{late} of AHA without the diffusers throughout the entire octave bands is similar to the pattern of DH170 with eight diffusers and AA16 with all the diffusers and absorbers applied.

This led to the assumption that the curved walls can create a sound-field diffuseness even when a diffuser is not inserted. A slightly lower EDT and slightly higher T_{30} with the diffusers applied indicated that the diffusers created more late reflections than early reflections. These decay rates and the clarity index values are also similar to what was observed in DH170.

3.1.6 Large Room with Semi-Parallel Walls

The acoustic volume of the Detroit Orchestra Hall (DOH) is six times larger than the two spaces discussed previously. Detailed description of the space is provided in Section 2.2.4. There exists a stage with diffusers mounted on the stage walls. These are the geometrical properties that restrained the ability to compare the results of this space with the other cases studied. In theory, if the early reflections are supported by the diffusers on stage, the sound from sources on the stage can be blended before reaching the audience's ears.

The Coh_{early} and Coh_{late} line charts obtained from the computer simulation (see Figure 2.28) in Figure 3.19 indicate that the diffusers did not improve the sound-field diffuseness nor did they create uniform distribution of the early reflections. The diffusers, however, were mounted on the stage's ceiling at a height of 4.5 meters from the receivers. Given this height, the diffused reflections of the diffusers would already decay before they reached the receivers.

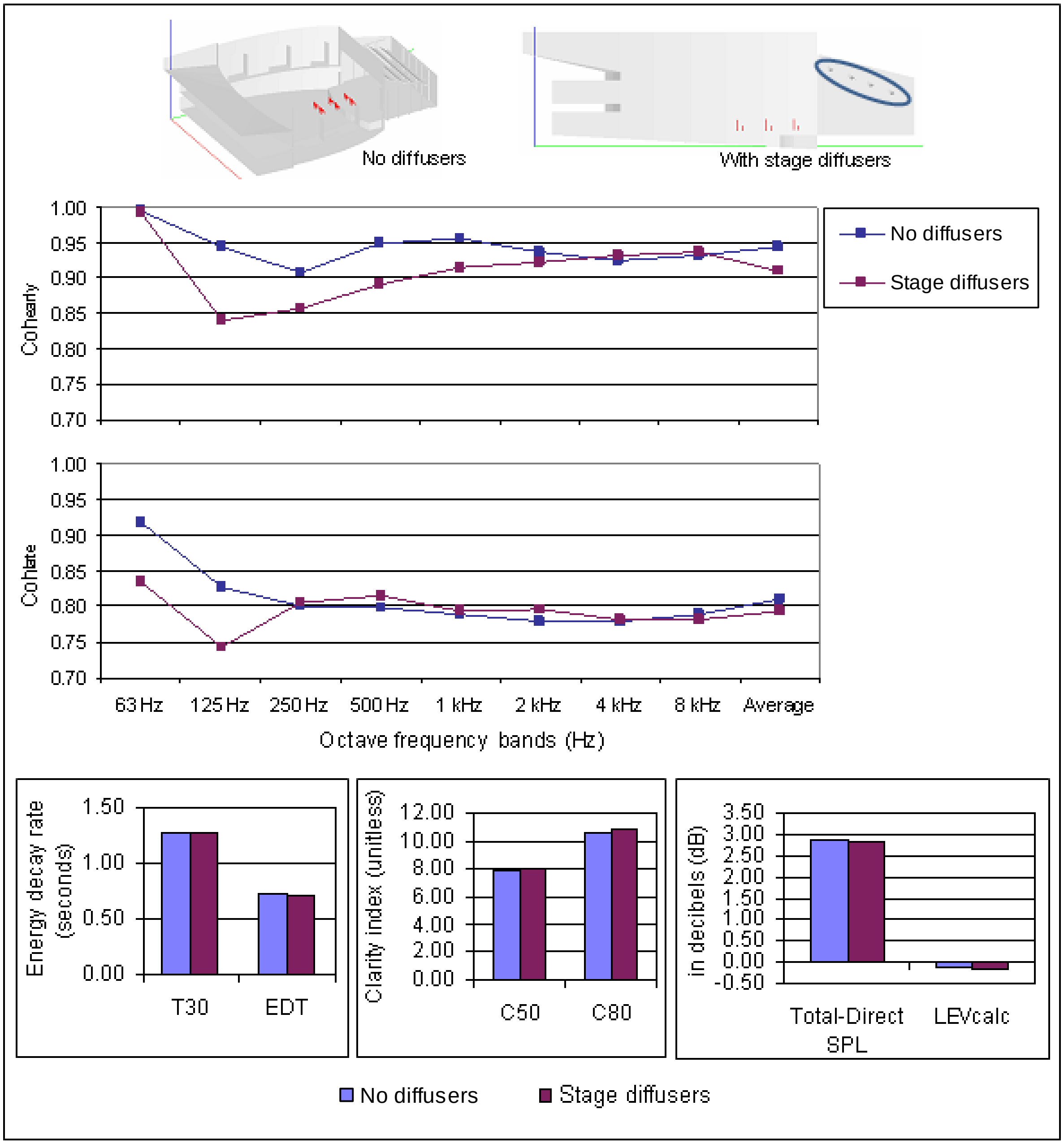


Figure 3.19. Results of observations on the impact of diffusers in DOH using computer simulation: illustration of the computer models, the coherence plots, and the objective parameters.

The envelope of a signal and the spectrum from the Acoustic Camera outputs of three types of sources, balloon, clapper, and yacht cannon, were observed. In Table 1.2, the criteria of a reliable source are listed, where a sufficient sound pressure level for low to mid-frequency bands is one of the criteria. A flat frequency response at low frequencies of a balloon burst in the Detroit Orchestra Hall (DOH) is shown in Figure 3.20.

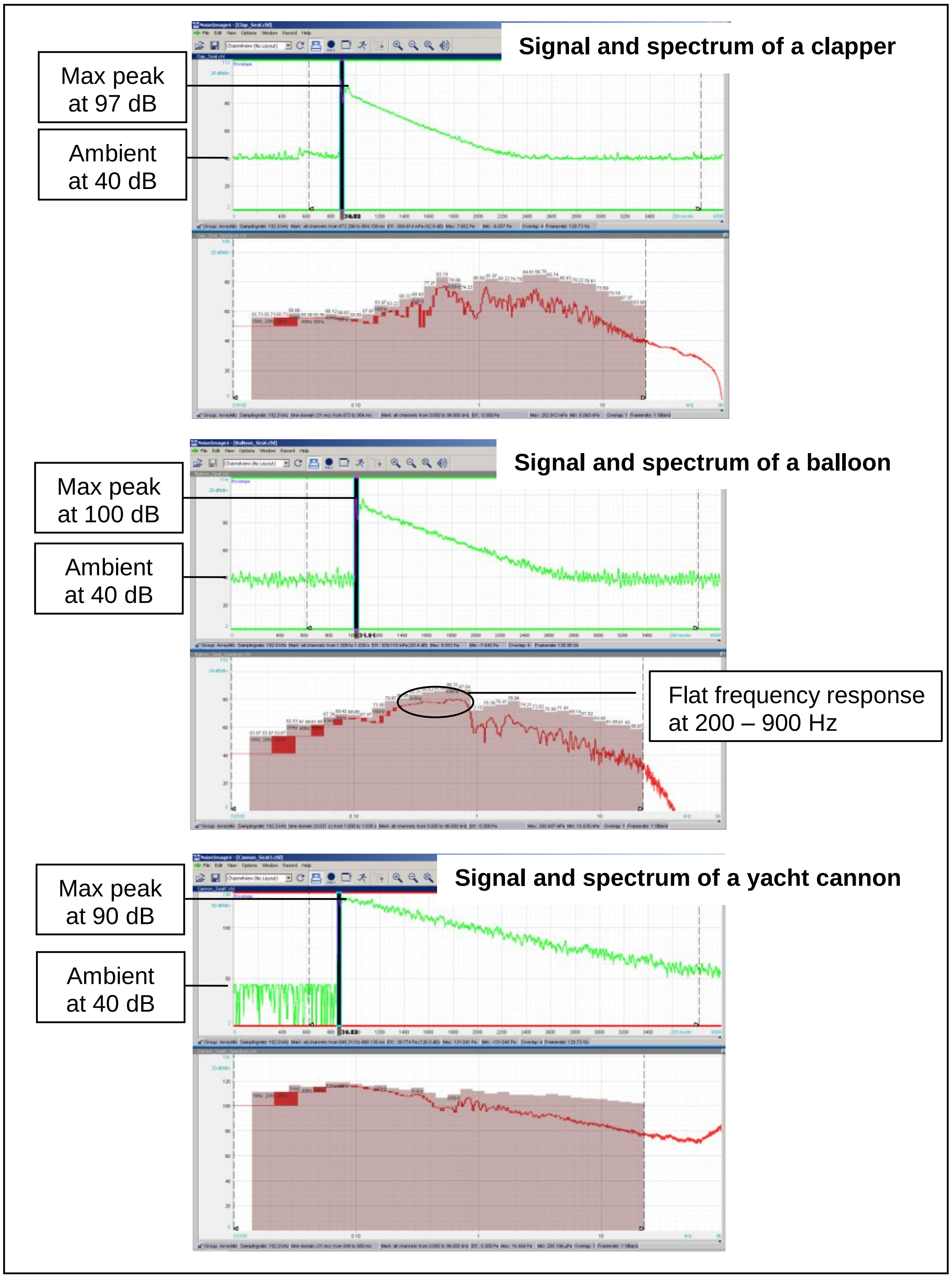


Figure 3.20. The envelope of signal from the Acoustic Camera output and the spectrums using three types of sources; a clapper, balloon, and yacht cannon, in the Detroit Orchestra Hall (DOH).

Meanwhile, the yacht cannon explosion produced a flat frequency response for the entire frequency range. Measurement outputs of the three paired microphones on the Acoustic Camera (front-rear, left-right, and top-down) produced similar coherences, especially the values of Coh_{early} (see Figure 3.21).

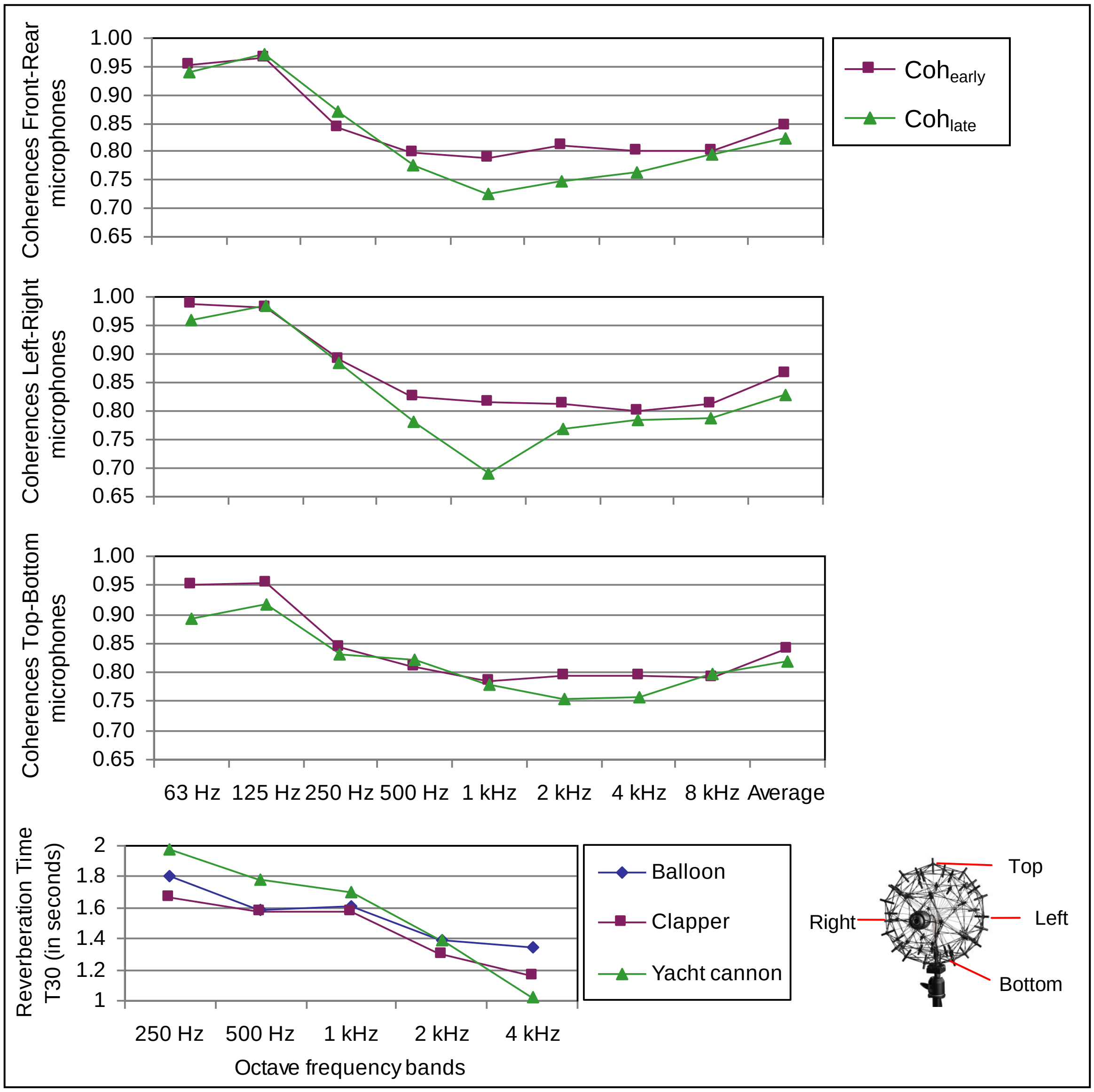


Figure 3.21. The Coh_{early} and Coh_{late} of DOH obtained from the measured impulse response at three pairs of microphones on the Acoustic Camera (top-bottom, left-right, and front-rear) along with the reverberation time (T_{30}) calculated from the top microphone.

Indication of this sound-field diffuseness provides evidence of the effect of the stage diffusers. It is based on the assumption that since the source was positioned on the stage, the majority of the reflections were hitting the stage boundaries before they reached the receivers. Another condition supporting the diffuseness is the length-to-ceiling ratio of 1.8 that created lateral reflections from the side-walls and less impact from the ceiling. With a volume of 8895 m^3, the mid-frequencies reverberation time for a good music performance (i.e., music type in general) is approximately 1.5 seconds, which is already fulfilled by the existing condition of DOH, given the average value of the T_{30} provided in Figure 3.21.

3.1.7 Very Large Room with Non-Parallel Walls

The Crisler arena (CRI) and the "Big House" stadium are sport facilities, categorized as large rooms, with non-parallel walls and an elliptical bowl-like shape. Detailed description is provided in Section 2.2.5 and Figure 2.29.

A music track of a xylophone from "Music from Archimedes" (Bang and Olufsen, 1992) was reproduced through the loudspeakers during the field measurement in Crisler arena. Loudspeakers were mounted above the scoreboard while the Acoustic Camera was positioned at the center of the basketball court. The coherence of the entire output signal (Coh_{entire}) of three paired microphones on the Acoustic Camera is provided in Figure 3.22.

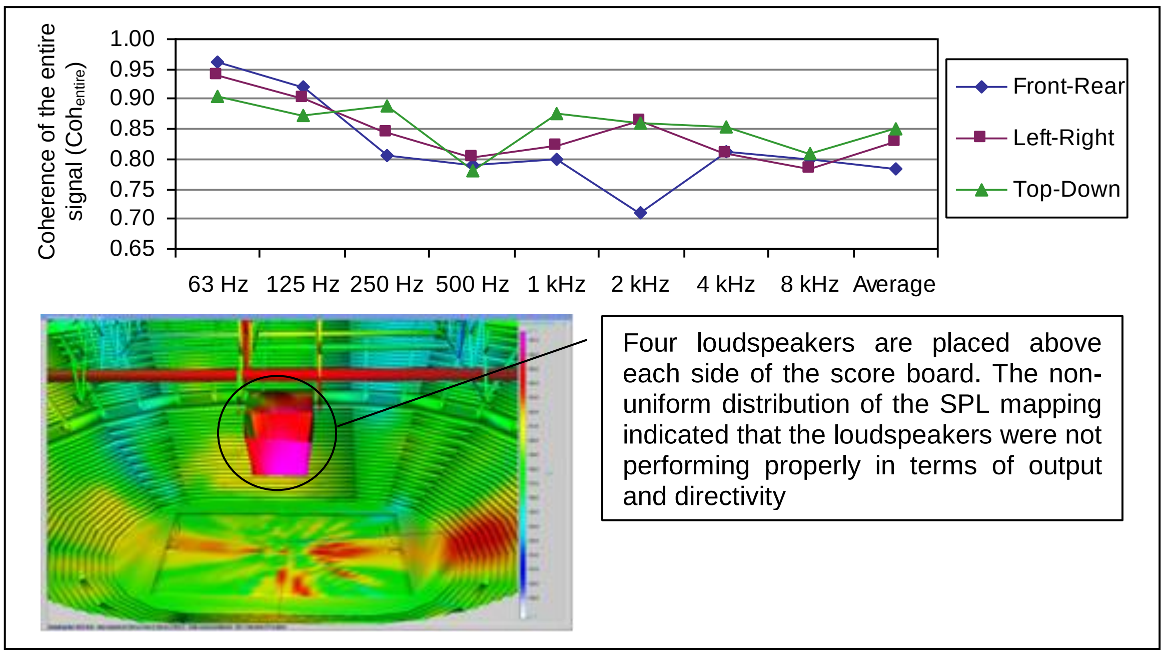


Figure 3.22. Upper panel: the coherence of the entire output signal from three paired microphones on the Acoustic Camera observed in CRI. Lower panel: the SPL mapping from the Noise Image software to identify loudspeakers' output quality and directivity pattern.

By comparing the coherences obtained from simulation to those calculated from the field measurement data, it is obvious that the material selected for the ceiling within the computer simulation was over-estimated for its absorption coefficient. This is also demonstrated by the very low T_{30} values for this size of a space as plotted in Figure 3.23.

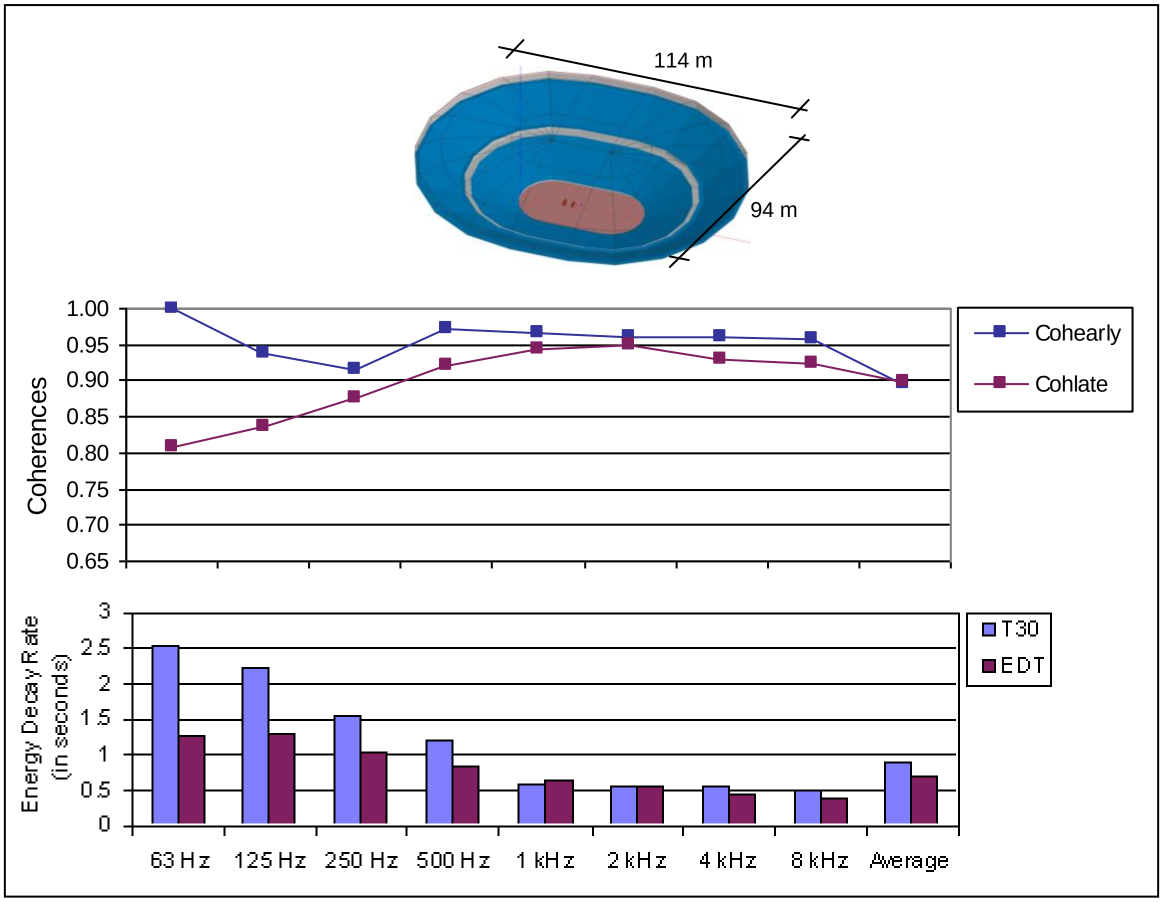


Figure 3.23. Objective parameters in CRI from computer simulation.

Consequently, the large difference between the absorption of the ceiling compared to the other surfaces created Coh_{late} values that diverged from other cases studied. The values at low frequencies are less than the Coh_{late} at high frequencies. An observation with ray tracing using Autodesk Ecotect™ also identified the ceiling as the critical element for the performance of the scoreboard sound system.

Coherences in Figure 3.24 were calculated with source (i.e. cannon burst) positioned on the field at the center of the “Big House” football stadium (BH). The line charts of left-right and top-down paired microphones on the Acoustic Camera are very similar with an exception on the octave band 250Hz for the Coh_{early}.

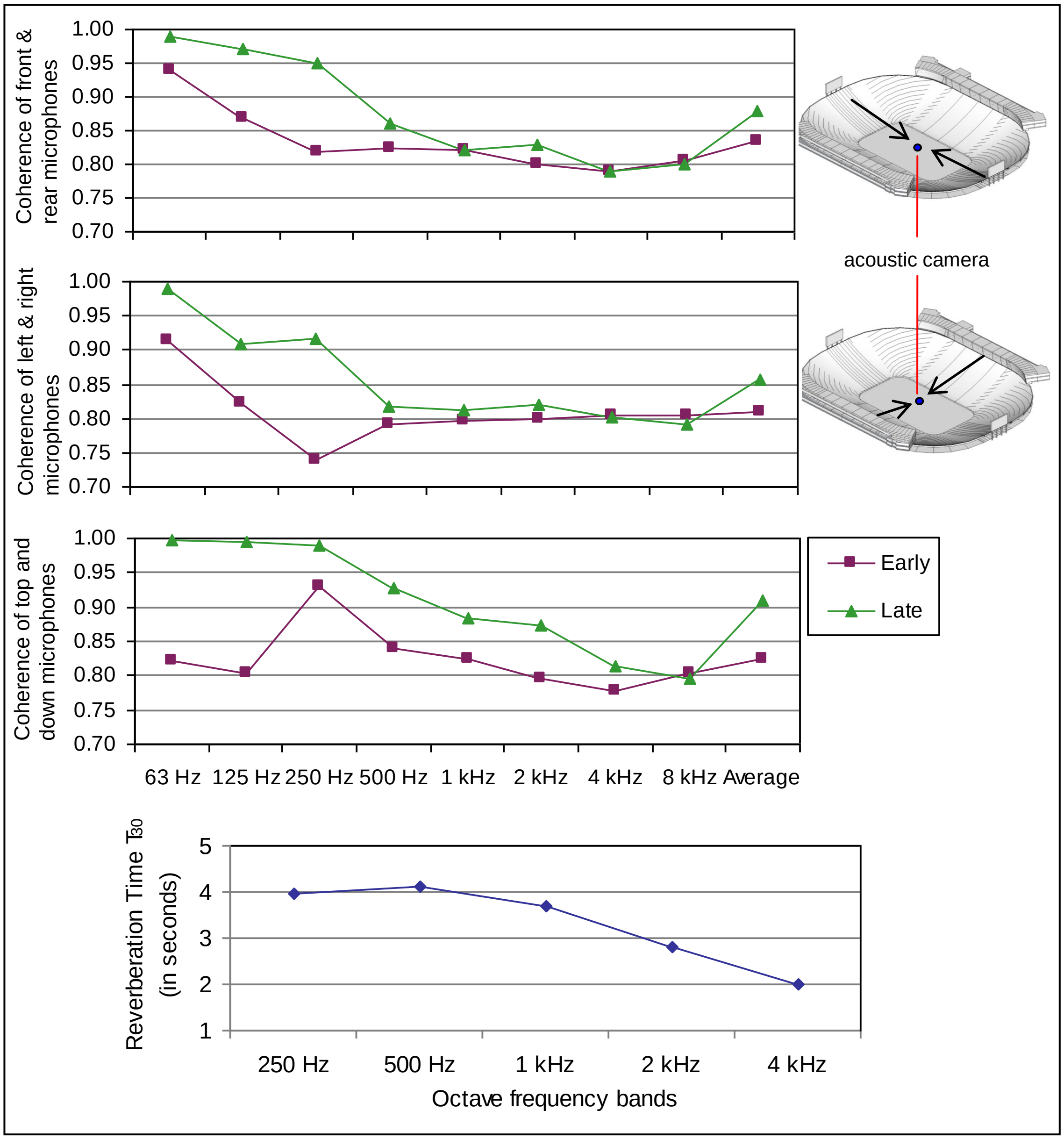


Figure 3.24. Objective parameters in BH stadium from field measurement.

This smaller correlation between the left and right microphone outputs is due to interferences with the early reflections created by the skybox structures that were still under construction when the measurement was conducted. The skyboxes are placed at the East and West (left and right) sidelines of the stadium and have a glass window surface. The uniqueness of the BH coherences is the larger Coh_{late} values (compared to the Coh_{early} values) at low to mid-frequency octave bands.

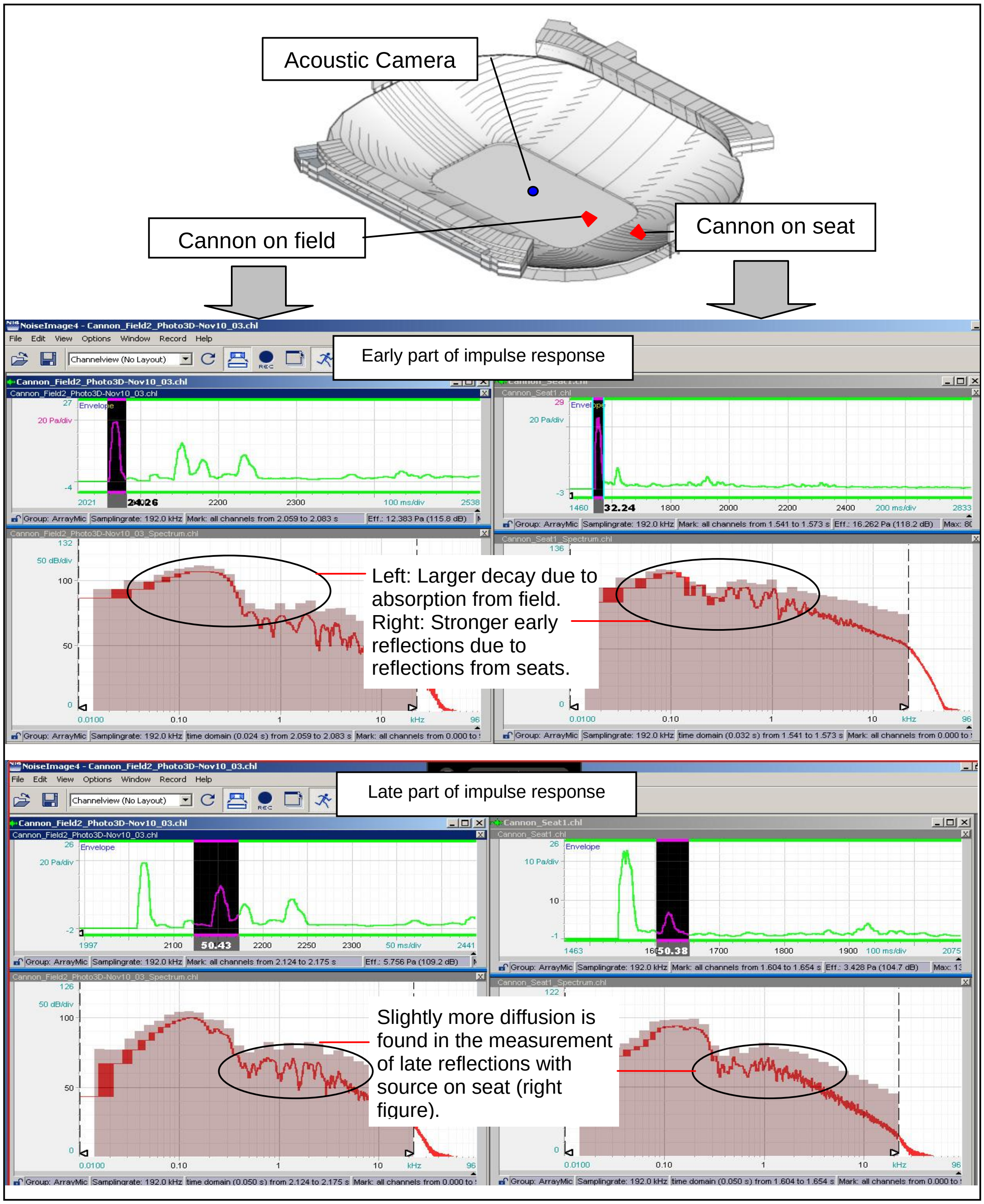


Figure 3.25. SPL distribution in BH measured with Acoustic Camera.

At the very early reflections, the source burst on the field is already losing more of its frequency component as compared to the source burst in the seating area (see Figure 3.25). The open and large space created a high atmospheric absorption. However, the geometric “bowl” shape of the seats created a longer sound decay, and the reflections were expected to be diffused. In a smaller room with this shape, a sound might creep from one side to the other through the reflections of the curved-walls, which is known as the “cocktail party” effect.

3.2 Generalizing the Results from All Cases Studied

Owing to the sphere-microphone array radius of the Acoustic Camera, the receiver (microphone) spacing is less than the incoming wavelengths for certain frequencies of sound waves. The closer spacing leads to a sound field being more diffused at low frequencies than at high frequencies. The line charts of coherences in Figure 3.26 show the tendency of a down slide slope with the high values nearly reaching the maximum value of 1 at the low frequency octave bands. At frequencies above 500Hz, the Coh_{early} line charts for all three paired microphones turned into flat lines with similar values ranging between 0.75 to 0.85 and a disparity ≤ 0.10.

Careful attention should be given during the measurement of impulse responses for spaces with a stepped floor, such as Dennison Hall R170 (DH170). The stepped floor is an obstacle for the incoming sound, especially for the sound waves from sources with heights lower than the receiver. It creates a sound field with diffuseness less than other spaces for low frequencies indicated by the Coh_{late} line that diverges from the others for the front-rear paired microphones (i.e., walls that are the longest distance apart), as shown in the second plot in Figure 3.26.

The coherences for all the cases studied for left and right paired microphones (i.e., walls that are the shortest distance apart) at octave band 63Hz are within the range of 0.9 to 1. Given the variations on room shape and size, the Coh_{early} are very similar (i.e., disparity ≤ 0.10) throughout the entire frequency except for the stadium. The largest disparity of the coherences is for Coh_{early} obtained from top and bottom paired-microphone output at the 250Hz octave band with the smallest value for DH170. This provided evidence that the early reflections were also affected by the stepped floor.

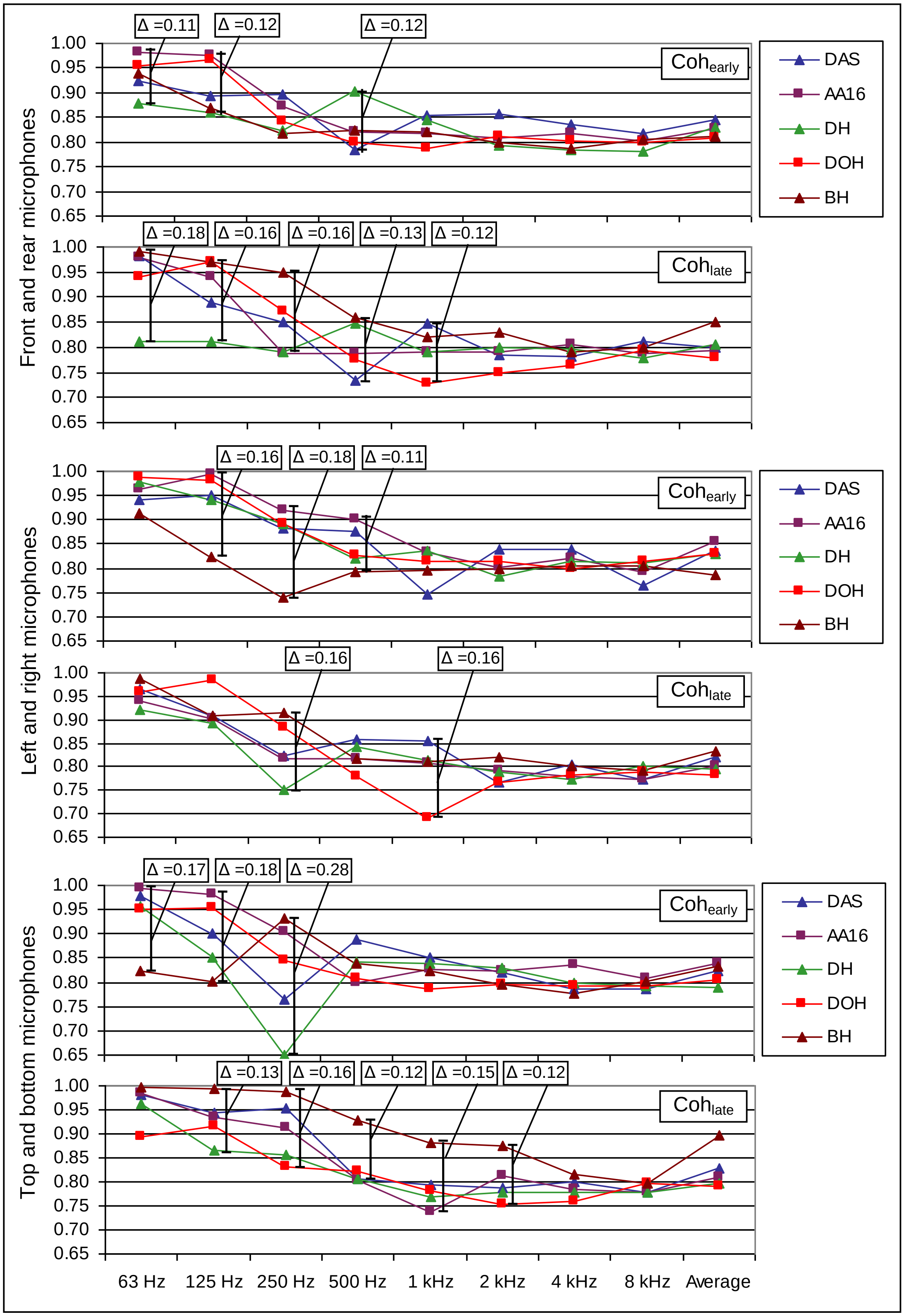


Figure 3.26. Comparison of the Coh_{late} using impulse responses of 5 cases studies.

In the stadium, the Coh_{late} values obtained from the top and bottom paired microphones are highly impacted by the atmospheric absorption within this large open space. Consequently, the Coh_{late} line chart diverged from other spaces, especially the line curve at mid-frequencies (500 Hz – 2000 Hz octave bands).

The distinct results from the stadium have led to the assumption that this space is not comparable to the other spaces due to its size, openness, and bowl-shaped boundary. Therefore, generalization of the calculated coherence trend for all spaces using the computer simulation results excluded the Crisler Arena and Stadium (i.e., very large spaces). Thus, in the computer models, these greater distances for microphone spacing in very large spaces makes the task more difficult since the sound fields become more sensitive to architectural impacts. To address this issue, the source-to-receiver distances in the cases studied were all the same (see Figure 2.1).

Two sets of Coh_{early} and Coh_{late} line charts are used for the comparison. The first set in Figure 3.27 is obtained from the computer models without diffusers. The second set in Figure 3.28 represents values for models with the greatest number of diffusers applied. In general, the diffusers did not significantly change the diffuseness of the sound field at frequencies above 125 Hz, given the disparity of the $Coh_{late} \leq 0.05$ for all spaces. Interestingly, the significant increase of sound-field diffuseness (i.e., disparity ≥ 0.05) at low frequencies is only seen in the small and semi-small rooms (volume < 400 m^3) with a more vivid change in the objective parameters. However, an exception to this condition occurs in a highly absorptive space, such as the recording studio (DAS with all diffusers and closed wall-panels) where the objective parameters are less impacted by the addition of the diffuser‘s surfaces. As for semi-large and large rooms, the diffuseness at low frequencies decreased with hardly any change in the objective parameter values.

The effect of the diffusers on the early reflections that are considered significant can be seen in certain spaces for certain frequencies indicated by the Coh_{early} that have differences ≥ 0.05. For instance, in model AA21 having six diffusers applied and tilted 30° (AA21-six diff 30°) reduced the Coh_{early} by 0.17 at the 500 Hz octave band.

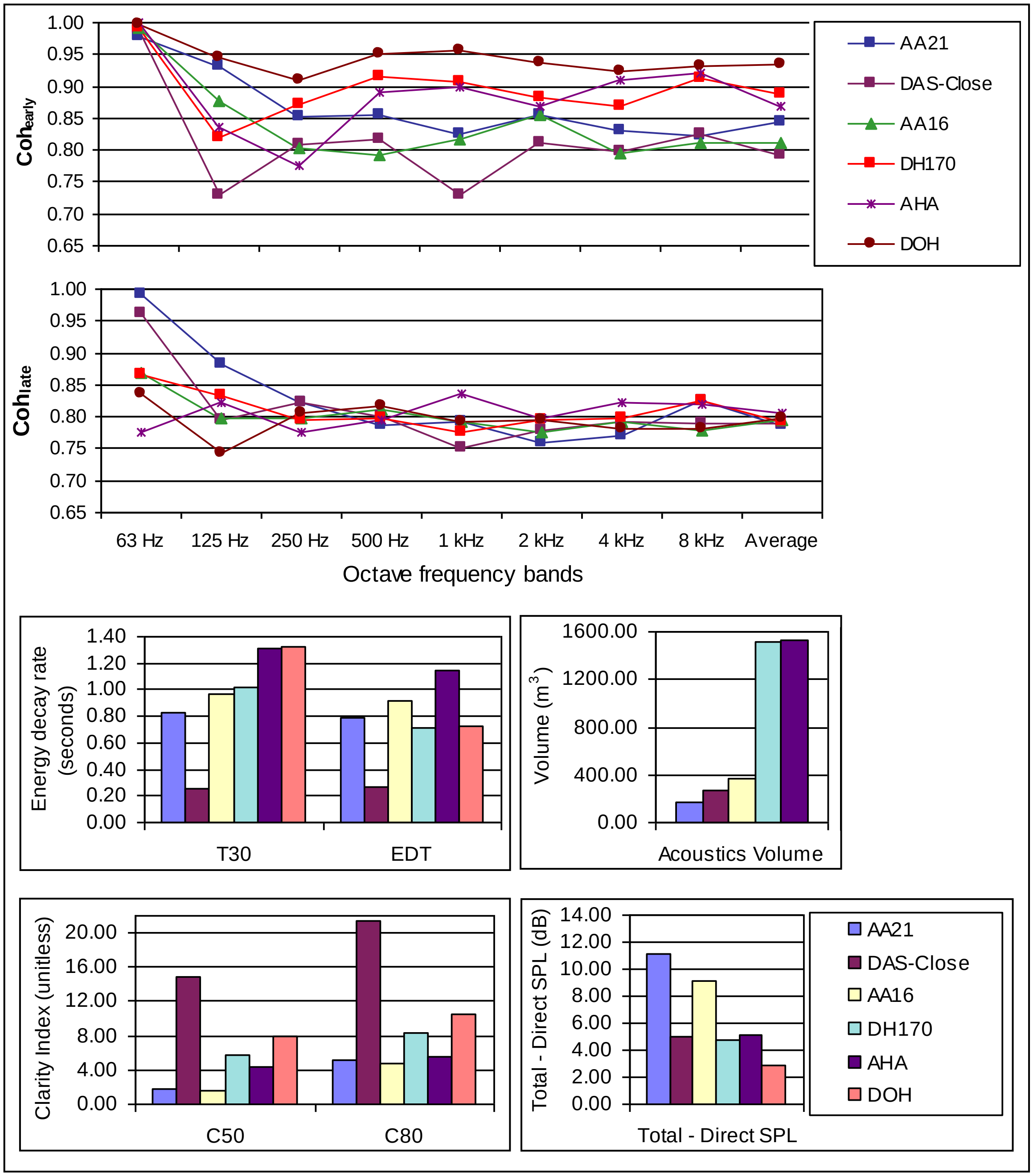


Figure 3.27. Comparison of coherences for computer models without any diffusers applied.

An example of comparison between spaces based on the geometrical properties only is Dennison Hall Room-170 (DH170) compared with Angell Hall Auditorium-A (AHA). Both spaces have similar acoustics volume (i.e., effective room volume). The total minus direct SPL (i.e., SPL of the reflected energy) indicates the amount of absorption in which these two spaces are also similar. What differs in these spaces is the geometrical shape where DH170 is a rectangular shaped room with parallel-walls while AHA has curved side-walls.

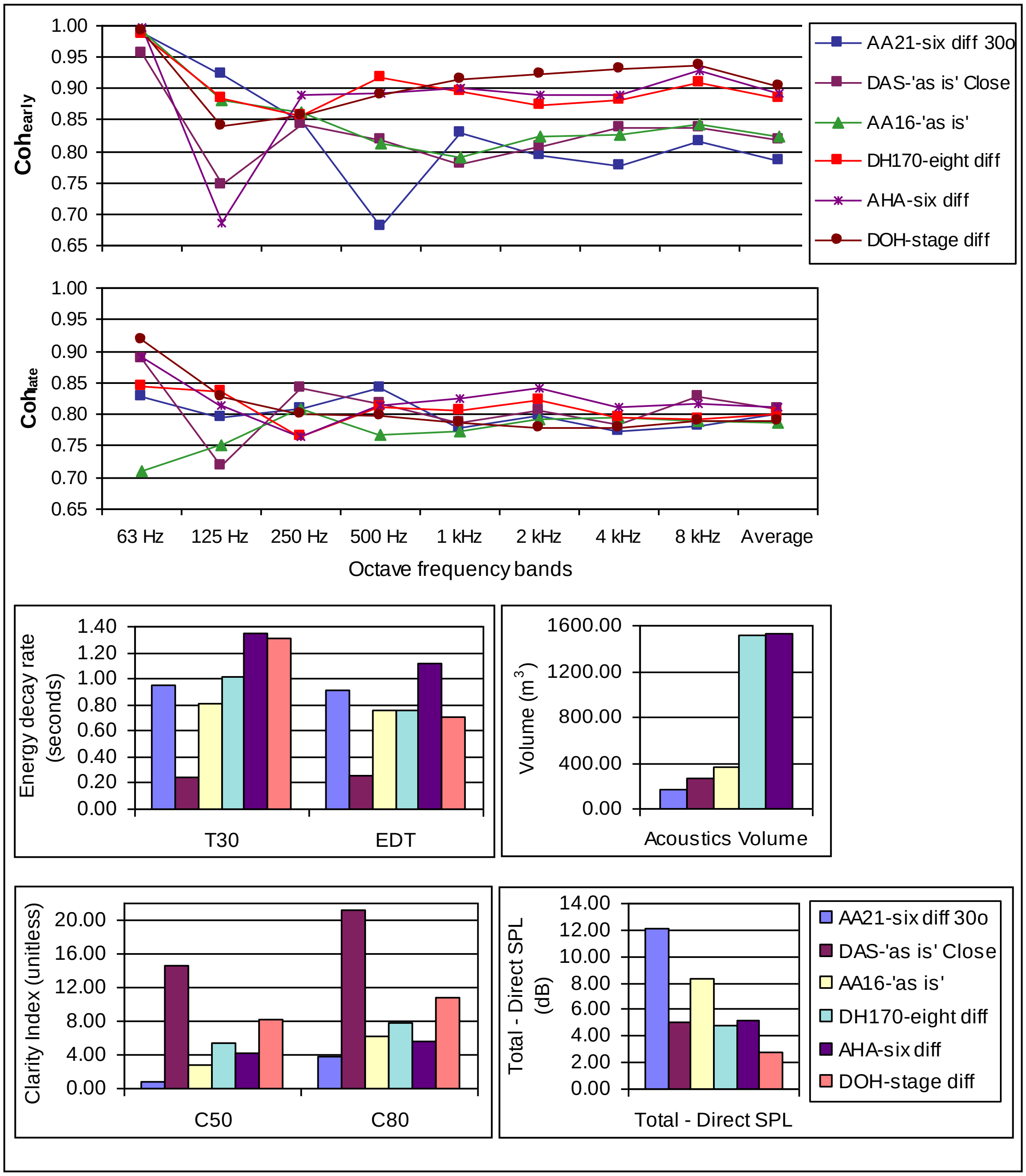


Figure 3.28. Comparison of coherences for computer models with the greatest number of diffusers applied.

Values of all the parameters generated from the impulse responses of computer simulation of all cases studied are listed in Table 6.4.

The eight diffusers in model DH170 (DH170-eight diff) did not create diffusion of the early reflections (see Figure 3.27). Meanwhile, the diffusers in model AHA (AHA-six diff) reduced the Coh_{early} significantly even though the ratio of diffuser surface to total surface area is 30% more in DH170. Furthermore, for frequencies above 500 Hz, the diffuseness is slightly higher in AHA due to the effect of the curved side walls.

3.3 Summary of Results

The results were classified as objective parameters or subjective parameters. Discussion of the objective parameters was presented for each case studied in terms of the space geometry and architectural elements. The logic of the data processing to measure and calculate the objective parameters in section 2.1 was applied. The discussion of each case, however, emphasized different objectives given the uniqueness in architectural and acoustical characteristics of each space.

In the Duderstadt Audio Studio (DAS), observations highlight the impact of the adjustable wall panels, the ceiling diffusers, and a piano. The non-parallel wall was also an interesting architectural property that was explored. The impact of these elements was clearly recognized, both with the field measurements and computer simulation. The similarity between the results in DAS with those of room 2216 Art and Architecture Building (AA16) is the higher Coh_{late} from the top and bottom paired-microphones condition. Interestingly, in both spaces, diffusers were applied on the ceiling and both had relatively the same ceiling height.

This led to the assumption that the ceiling diffusers make a greater contribution to the diffused sound field. The presence of ceiling diffusers also improved the performances of the diffusers on the walls in AA16; in the case without wall diffusers, a different acoustical condition was seen in the computer simulation results. The results of these two spaces, DAS and AA16, led to further study of the audibility conditions utilizing subjective assessment with the results provided in Chapter 4.

In other rectangular rooms, simply adding diffusers on the wall did not significantly change the degree of diffuseness. In small and semi-small rooms, impacts on early reflections were still recognized. As the room became larger, the impact became less critical both in the early and late reflections, and the diffusive surfaces contributed more to the sound absorption. The degree of diffuseness was influenced more by the room shape, which is formed by the wall configuration. Comparison of room size and shape is provided in more detail in Chapter 4 with further analysis on the relationship between the objective parameters and the subjective attributes.

Chapter 4
Characterizing the Audibility of the Sound Field

Results from computer simulation and field measurements show that the degree of diffuseness and other acoustical conditions vary for each space. In some cases, the numerical values do not clearly describe the acoustical condition. Therefore, a subjective assessment is needed to identify noticeable differences in the auditory experience due to the variation of the geometrical and acoustical properties of the space. Further analysis is needed to determine actual auditory impact, given changes in acoustic volume and the area of the absorptive and diffusive surfaces.

The analysis focuses on the audibility conditions, and is based on the relationships among objective parameters and subjective attributes. Parameters observed are the coherences, total minus direct sound pressure level (SPL), reverberation time (T_{30}), early decay time (EDT), clarity index (C_{50} and C_{80}), listener envelopment (LEV_{calc}), and the audibility quality of clarity, loudness, and liveliness. The analysis uses the average of the values at 250 Hz to 4000 Hz octave bands. Calculations of the LEV_{calc} use G (strength factor) and C_{80} in Equations (2-9) and (2-10). The values of these components within the LEV_{calc} are provided.

Owing to the need to vary the architectural configuration, the majority of the data utilized within the subjective assessment was obtained from computer simulation. The analyses consist of:

1. Analysis of the sound field diffuseness and the associated objective parameters measured.
2. Analysis of the audibility condition of a space given the sound-field diffuseness and its objective parameter values.

3. Characterization of the architectural elements (diffuser and non-diffuser), room size, and room shape, which strongly impact the sound-field diffuseness and the audibility conditions.

4.1 Results of the Subjective Assessment

Auditory stimuli used in the subjective assessment were generated from simulation and auralization of the Duderstadt Audio Studio (DAS) and room 2216 in Art and Architecture (AA16). Stimuli consisted of two types, which were a brief male-voice speech (provided within EASE database) and a string quartet playing Mozart (see Figure 2.32). Details of the experimental setup for the subjective assessment are provided in section 2.3 with the entire survey interface and questionnaire provided in Appendix C.

Forty subjects (18-22 years old) enrolled in the environmental technology course participated in the survey. Instructions and the stimuli were presented to subjects using a slide presentation on a computer screen and stereophonic headphones at a level of approximately 60 dB (A-weighted). Subjects indicated their responses via a questionnaire sheet. The correct answers were based on the values of the associated objective parameters. For instance, in the comparison of paired stimuli in slide no. 1 or row no. 1 of Table 4.1, the sound with a higher total SPL (i.e., the total SPL value of stimulus 1 is larger than the total SPL value of stimulus 2) should be perceived as louder. The questions and results of the subjective assessment are shown in Table 4.1.

Table 4.1 Questions (left column) and Results (right column) of the Subjective Assessment.

NOTE:

1 : stimulus 1 or first stimulus of each slide.

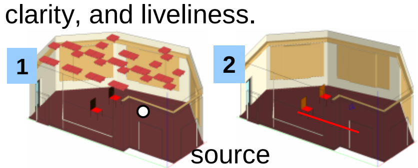
2 : stimulus 2 or second stimulus of each slide.

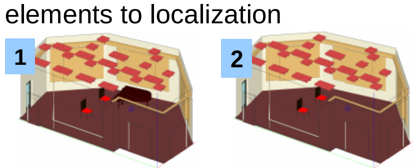
N/A : Not applicable, is an option if the subject does not recognize any differences between the stimuli.

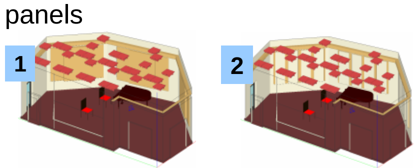
Total SPL : the objective parameter that indicates the loudness

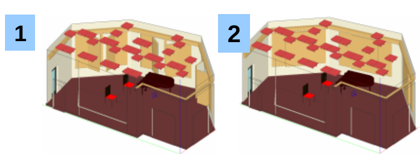
C_{50} : the objective parameter that indicates the clarity

T_{30} : the objective parameter that indicates the reverberation or liveliness

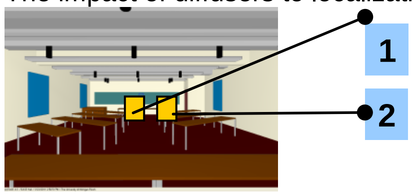
<table>
<tr><th>No.</th><th>Slide Representation of the Survey (Research Questions)</th><th>Results and supporting data</th></tr>
<tr><td>1</td><td>The impact of diffusers on loudness, clarity, and liveliness.


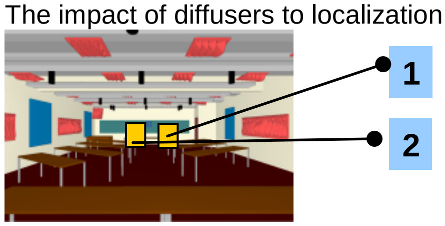
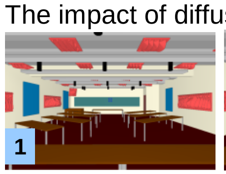
With diffusers Without diffuser</td><td>
<table>
<tr><th>Question</th><th>1</th><th>2</th><th>N/A</th></tr>
<tr><td>Which speech sounds louder?
Total SPL of 1> Total SPL of 2</td><td>52.5</td><td>25</td><td>22.5</td></tr>
<tr><td>Which one sounds clearer?
C_{50} of 1 < C_{50} of 2</td><td>42.5</td><td>37.5</td><td>20</td></tr>
<tr><td>Which one sounds livelier?
T_{30} of 1> T_{30} of 2</td><td>40</td><td>45</td><td>15</td></tr>
</table></td></tr>
<tr><td>2</td><td>The impact of architectural elements to localization


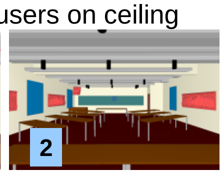
With piano Without piano</td><td>Which sound do you hear is coming further left from you?
<table>
<tr><th>Answer</th><th>Associated stimuli</th><th>Percentage (%)</th></tr>
<tr><td>1</td><td>Music_As-is seat 1</td><td>7.5</td></tr>
<tr><td>2</td><td>Music_CloseC seat 1</td><td>92.5</td></tr>
<tr><td>3</td><td>N/A</td><td>0</td></tr>
</table></td></tr>
<tr><td>3</td><td>The impact of adjustable wall panels


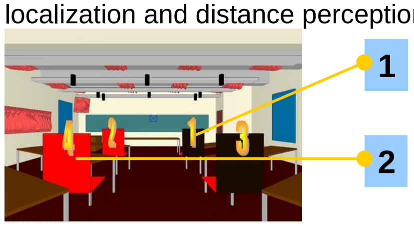
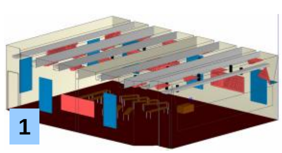
Closed panels Open panels</td><td>
<table>
<tr><th>Question</th><th>1</th><th>2</th><th>N/A</th></tr>
<tr><td>Which one sounds louder?
Total SPL of 1> Total SPL of 2</td><td>82.5</td><td>5</td><td>12.5</td></tr>
<tr><td>Which one sounds clearer?
C_{50} of 1 < C_{50} of 2</td><td>7.5</td><td>77.5</td><td>15</td></tr>
<tr><td>Which one sounds livelier?
T_{30} of 1> T_{30} of 2</td><td>82.5</td><td>5</td><td>12.5</td></tr>
</table></td></tr>
<tr><td>4</td><td>The impact of adjustable wall panels


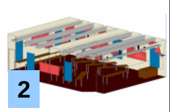
Closed 45^0 panels Closed panels</td><td>
<table>
<tr><th>Question</th><th>1</th><th>2</th><th>N/A</th></tr>
<tr><td>Which one sounds louder?
Total SPL of 1< Total SPL of 2</td><td>5</td><td>87.5</td><td>7.5</td></tr>
<tr><td>Which one sounds clearer?
C_{50} of 1 > C_{50} of 2</td><td>65</td><td>15</td><td>20</td></tr>
<tr><td>Which one sounds livelier?
T_{30} of 1< T_{30} of 2</td><td>15</td><td>72.5</td><td>12.5</td></tr>
</table></td></tr>
</table>

<table>
<tr><th>No.</th><th>Slide Representation of the Survey (Research Questions)</th><th>Results and supporting data</th></tr>
<tr><td>5</td><td>The impact of diffusers to localization

</td><td>Which sound do you hear is coming from your left?
Associated stimuli: No. 2
<table>
<tr><td>Answer</td><td>Associated stimuli</td><td>Percentage (%)</td></tr>
<tr><td>1</td><td>Music at left seat</td><td>7.5</td></tr>
<tr><td>2</td><td>Music at right seat</td><td>80</td></tr>
<tr><td>3</td><td>N/A</td><td>12.5</td></tr>
</table></td></tr>
<tr><td>6</td><td>The impact of diffusers


With all diffusers Without diffusers</td><td>
<table>
<tr><td>Question</td><td>1</td><td>2</td><td>N/A</td></tr>
<tr><td>Which one sounds louder?
Total SPL of 1< Total SPL of 2</td><td>27.5</td><td>45</td><td>27.5</td></tr>
<tr><td>Which one sounds clearer?
C_{50} of 1 > C_{50} of 2</td><td>97.5</td><td>0</td><td>2.5</td></tr>
<tr><td>Which one sounds livelier?
T_{30} of 1< T_{30} of 2</td><td>15</td><td>85</td><td>0</td></tr>
</table></td></tr>
<tr><td>7</td><td>The impact of diffusers to localization

</td><td>Which sound do you hear is coming from your right?
Associated stimuli: No. 2
<table>
<tr><td>Answer</td><td>Description of Stimuli</td><td>Percentage (%)</td></tr>
<tr><td>1</td><td>Music at right seat</td><td>2.5</td></tr>
<tr><td>2</td><td>Music at left seat</td><td>92.5</td></tr>
<tr><td>3</td><td>N/A</td><td>5</td></tr>
</table></td></tr>
<tr><td>8</td><td>The impact of diffusers on ceiling




Without diffusers on ceiling With diffusers on ceiling</td><td>
<table>
<tr><td>Question</td><td>1</td><td>2</td><td>N/A</td></tr>
<tr><td>Which one sounds louder?
Total SPL of 1< Total SPL of 2</td><td>20</td><td>50</td><td>30</td></tr>
<tr><td>Which one sounds clearer?
C_{50} of 1 > C_{50} of 2</td><td>92.5</td><td>2.5</td><td>5</td></tr>
<tr><td>Which one sounds livelier?
T_{30} of 1< T_{30} of 2</td><td>17.5</td><td>80</td><td>2.5</td></tr>
</table></td></tr>
<tr><td>9</td><td>The impact of diffuser on localization and distance perception

</td><td>Sound 1 is as if you are sitting at seat no. 1
Which seat number do you think sound no. 2 is being heard from?
Correct answer : 4
<table>
<tr><td>Answer</td><td>Description of Stimuli</td><td>Percentage (%)</td></tr>
<tr><td>1</td><td>Music at seat no.1</td><td>15</td></tr>
<tr><td>2</td><td>No sound assigned</td><td>45</td></tr>
<tr><td>3</td><td>No sound assigned</td><td>22.5</td></tr>
<tr><td>4</td><td>Music at seat no.4</td><td>17.5</td></tr>
</table></td></tr>
<tr><td>10</td><td>The impact of room size and diffusers




Larger room (doubled) Actual room</td><td>Which sound do you hear is coming from the larger size room?
Associated stimuli: No. 1
<table>
<tr><td>Answer</td><td>Description of Stimuli</td><td>Percentage (%)</td></tr>
<tr><td>1</td><td>Music in As-is Large</td><td>85</td></tr>
<tr><td>2</td><td>Music in As-is</td><td>15</td></tr>
<tr><td>3</td><td>N/A</td><td>0</td></tr>
</table></td></tr>
</table>

4.2 Diffusers' Impact on Clarity, Loudness, and Liveliness

Details of the experimental setup to observe the impact of the diffusers on clarity, loudness, and liveliness is provided in section 2.3.1, where three pairs of stimuli were used to address this objective.

The first observation is on the impact of diffusers on the ceiling of Duderstadt Audio Studio (DAS) using slide no. 2 in Appendix C. Referring to the results from the objective measurement (see Figure 3.8), an increase of the Coh_{late} value, which indicates an increase of diffuseness due to the diffusers, is only significant at octave band 125 Hz. The only objective parameter that changed was the reverberation time (T_{30}). It decreased by 12% due to the addition of 8% of the diffusers' total absorption (Sabin). This difference, however, did not impact the auditory conditions given the unnoticeable difference in the liveliness perception, with only 40% of the subjects recognizing the higher T_{30} for the "without diffusers" model shown in Figure 4.1.

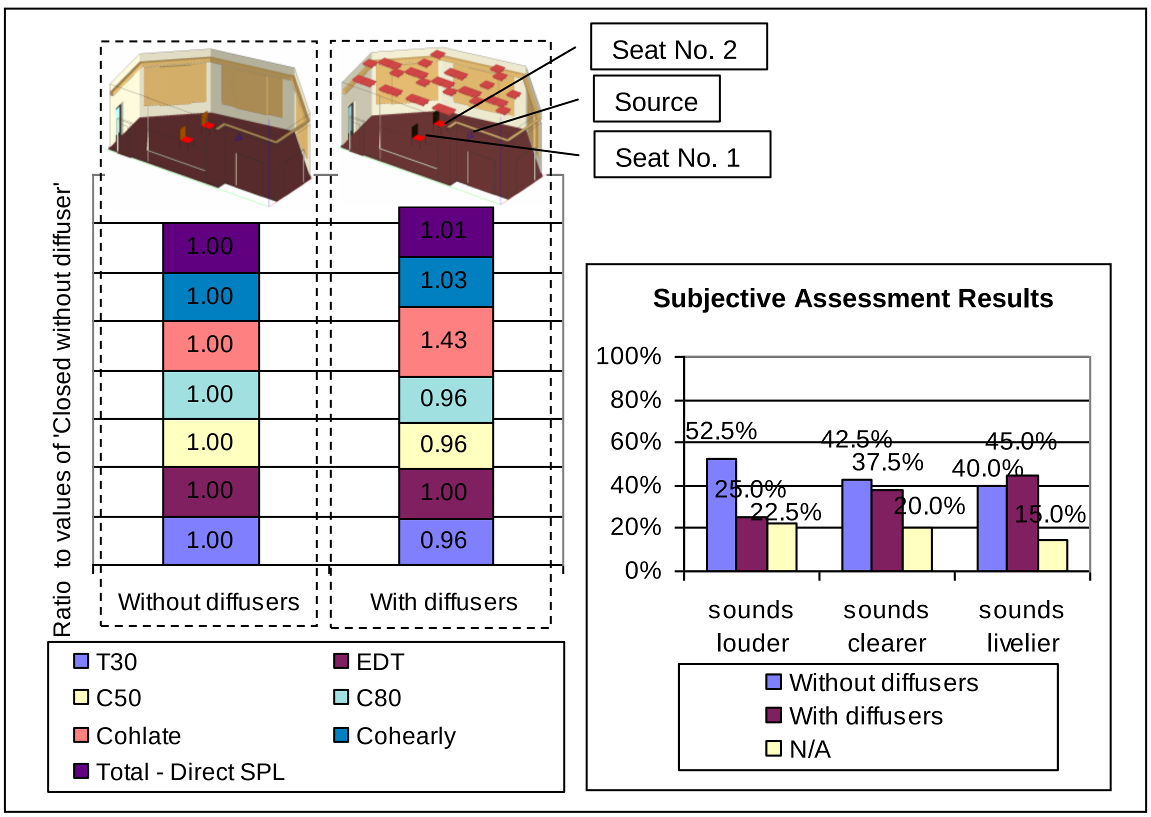


Figure 4.1. Results from the subjective assessment to compare audibility impact of the ceiling diffusers in DAS between model with diffusers and model without diffusers.

Even though the only objective parameter that was affected by adding the diffusers on the ceiling of DAS is the Coh_{late} (i.e., a value larger by 0.43 with the diffusers as compared without diffusers), the result in Figure 4.1 is showing that there is a noticeable difference in the loudness and clarity perception. Therefore, these results confirmed the conclusion that adding the diffusers affected the auditory perception.

A closer observation using the listener envelopment (LEV_{calc}) is provided in Table 4.2. A more diffused sensation on the listeners' ears is indicated by a larger LEV_{calc} value for the model with diffusers. Moreover, this indicated a more diffused sound field. It also verified the more significant impact of the ceiling diffusers on the late instead of on the early reflections (i.e., a larger Coh_{late} with the diffusers). The values of $IACC_{late}$ also confirmed this finding where a larger number indicates less diffuseness.

Table 4.2. Variables of Listeners Envelopment of model DAS "without diffusers" and "with diffusers."

Objective parameters (average at 250 - 4000Hz)	**Model of DAS 'without diffusers'**	**Model of DAS 'with diffusers'**
G_{late} (source strength)	14.81	14.81
C_{80} (clarity for music)	22.17	23.27
$IACC_{late}$	0.85	0.82
LEV_{calc} (Listener Envelopment calculated)	-0.76	-0.13

Owing to the application of the ceiling diffusers in room 2216-2219 in the Art and Architecture Building (AA16), the stimulus from auralization of the model with the ceiling diffusers was compared to the stimulus from the model without the ceiling diffusers using slide no. 9 in Appendix C as the survey interface. The Coh_{late} line charts in Figure 3.13 indicate the similarity of sound-field diffuseness for the frequencies above 63 Hz for all the diffusers' variations.

The subjective assessment result in Figure 4.2 shows that there is a noticeable difference in the clarity and liveliness perception for speech. The increase of the clarity index for speech (C_{50}) in the model with ceiling diffusers and T_{30}, which slightly decreased, supports the subjective assessment result. It can be concluded that the diffusion contributed to the early reflections and enhanced the speech clarity.

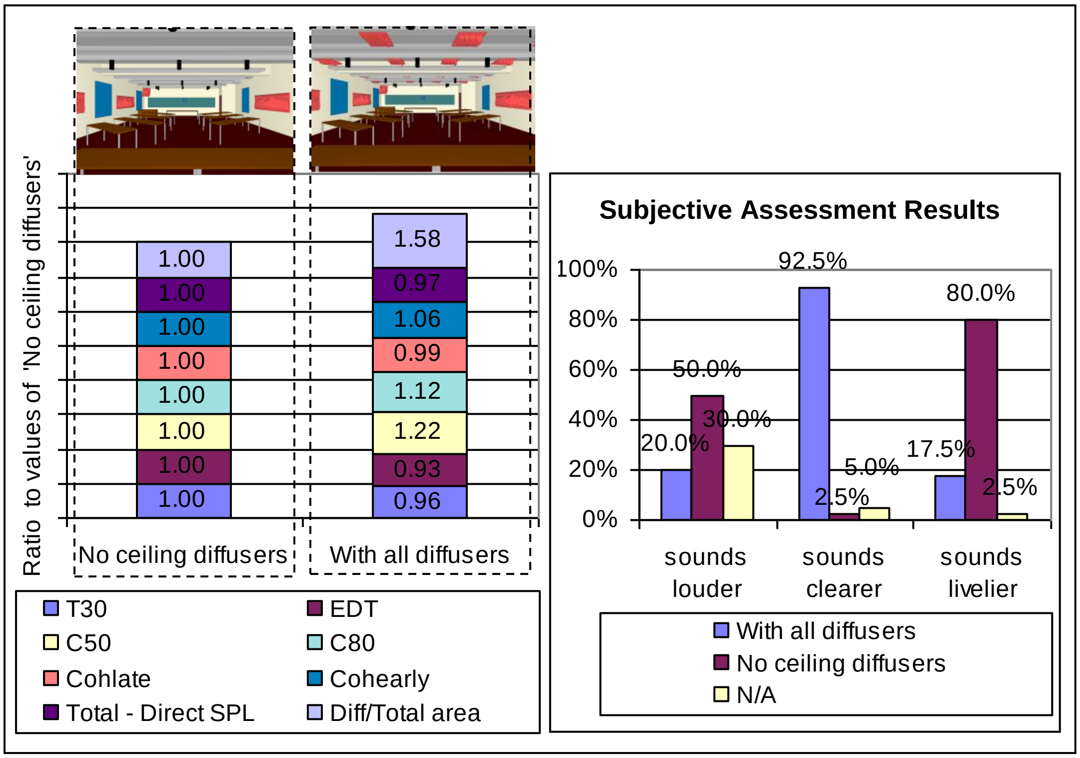


Figure 4.2. Results from the subjective assessment to compare audibility impact of the ceiling diffusers in AA16 between model "with all diffusers" and "no ceiling diffusers."

The third observation is the impact of all diffusers (on walls and ceiling) to the audibility condition in AA16 (see Appendix C, slide no. 7). As mentioned earlier, the diffusers did not impact the late reflections (Coh_{late}). Special attention is given for the 250 Hz octave band for early reflections, since at this frequency the Coh_{early} has the largest difference between values for model "with all diffusers" and "no diffusers."

More diffusion of the early reflections increased the clarity index values, which was confirmed by the subjective assessment result. There were 97.5% of the subjects that recognized the sound in "with all diffusers" to be clearer. Furthermore, 85% of the subjects indicated the space of "no diffusers" created a livelier sound, which was also confirmed by the larger reverberation time value. Adding diffusers added 10% total absorption within the space and reduced the SPL of the reflected sound by 8%. This reduction created insignificant differences in the loudness perception as only 40% of the subjects were able to recognize the sound in the "no diffusers" model as being louder.

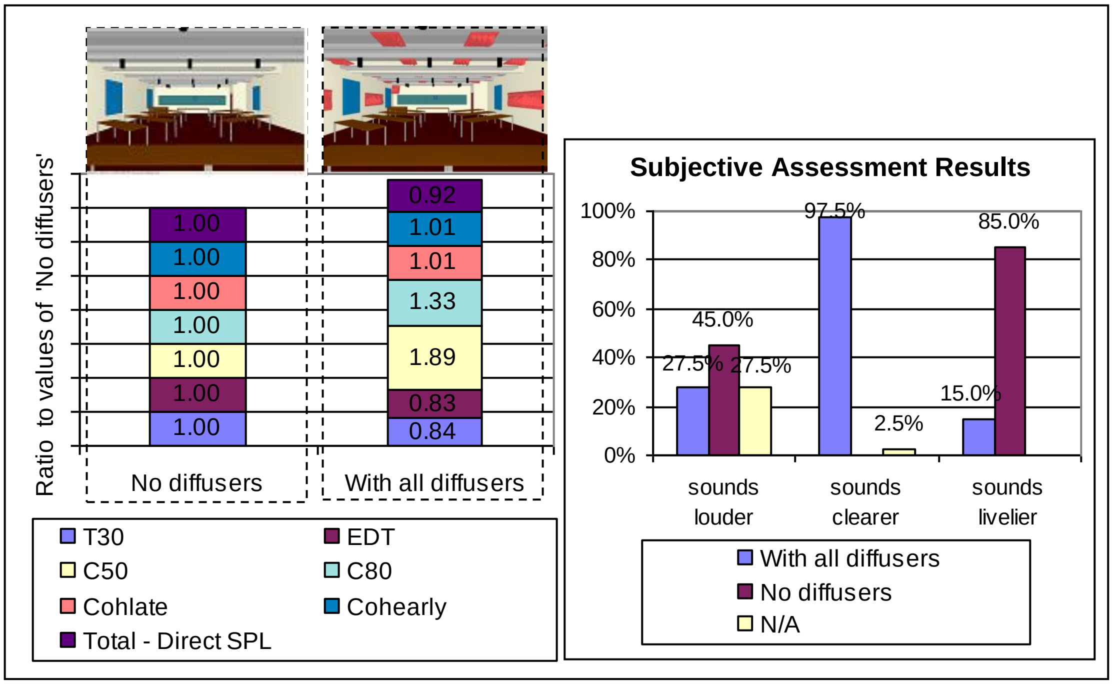


Figure 4.3. Results from the subjective assessment to compare audibility impact of all of the diffusers in AA16 between model "with all diffusers" and model "no diffusers".

At the listeners' ears, the model of "no diffusers" was perceived as more diffuse given the larger LEV_{calc}. In addition, the $IACC_{late}$ for both spaces are the same, which confirmed the finding that the diffusers insignificantly impacted the late reflections even at the listeners' ear. Without the diffusers, the total absorption within the space is lower. Consequently, this condition created a larger G_{late} value. Since enhancement of reflections is on the early energies, the clarity index of the model with all the diffusers applied is larger.

Using this relationship of variables, it can be concluded that a larger LEV_{calc} indicates more diffusion in the sound field but not necessarily a more diffused sound field (i.e., diffusion at the late reflections). The result depends on the portion of the reflected sound being impacted by the diffusers or other architectural elements.

To evaluate the diffused sound field at a listener's ear, the $IACC_{late}$ parameter is a better predictor. To evaluate the diffusion on the early reflections, the clarity index is used in addition to the LEV_{calc}. On the contrary, the LEV_{calc} in the model of "no ceiling diffuser" is 0.4 dB less than the value for the "with all diffusers” model, as shown in Table 4.3 with $IACC_{late}$ slightly larger, indicating that the sound field in the “with all diffusers” model is more diffused.

Table 4.3. Variables of Listeners’ Envelopment of model AA16 “no ceiling diffusers,” “no diffusers,” and “with all diffusers.”

Objective parameters (average at 250 - 4000Hz)	**Model of AA16 “no ceiling diffusers”**	**Model of AA16 “no diffusers”**	**Model of AA16 “with all diffusers”**
G_{late} (source strength)	18.14	19.37	18.08
C_{80} (clarity for music)	5.6	2.41	6.43
$IACC_{late}$	0.83	0.82	0.82
LEV_{calc} (Listener Envelopment calculated)	1.27	2.26	1.67

There is evidence that the ceiling diffusers are also contributing diffusion to the late reflections. The diffusion, however, did not impact the loudness condition given the subjective assessment result that only 50% of the subjects were able to recognize the louder sound in the “no ceiling diffusers” model (i.e., a larger total – direct SPL value). Identification of the SPL of a sound field beyond the critical distance as one of the characteristics of a diffused sound field is clearly demonstrated by this result. Increasing the diffuseness of a diffused sound field will not change the intensity level.

4.3 Diffusers’ Impact on Localizing Sound Direction

In this section, the diffusers’ impact on source localization ability is evaluated using the same space configurations for AA16 as in the previous section (see Appendix C, slides no. 6 and no. 8). From the subjective assessment, 80% of the subjects and 92.5% of the subjects were able to locate the correct source direction in the “no diffusers” model and in the “with all diffusers” model, respectively. The fact that subjects had less difficulty identifying the source direction in the “with all diffusers” model is assumed to be due to the dominant impact of the diffusers on the early reflections.

To evaluate the similarity of the diffuseness at the listener’s ear between two receivers, the interaural cross correlation of the entire signal (IACCA) of the left and right ears is used. The better ability to localize the sound direction shown by 92.5% of the subjects with a correct answer in Figure 4.4 confirmed the smaller value of $IACC_A$ at seat-2, which indicated less dissimilarity between response at the left and right ear. It is the result of the unsymmetrical positioning of the diffusers and absorber panels in the "with all diffusers" model that created more diffusion. The $IACC_A$ for seat-1 and seat-2 in "with all diffusers" with differences of 0.16 and 0.02 for model "no diffusers" has led to the conclusion that a larger difference in the $IACC_A$ values of the two receivers created a more optimal condition for subjects to identify the source direction.

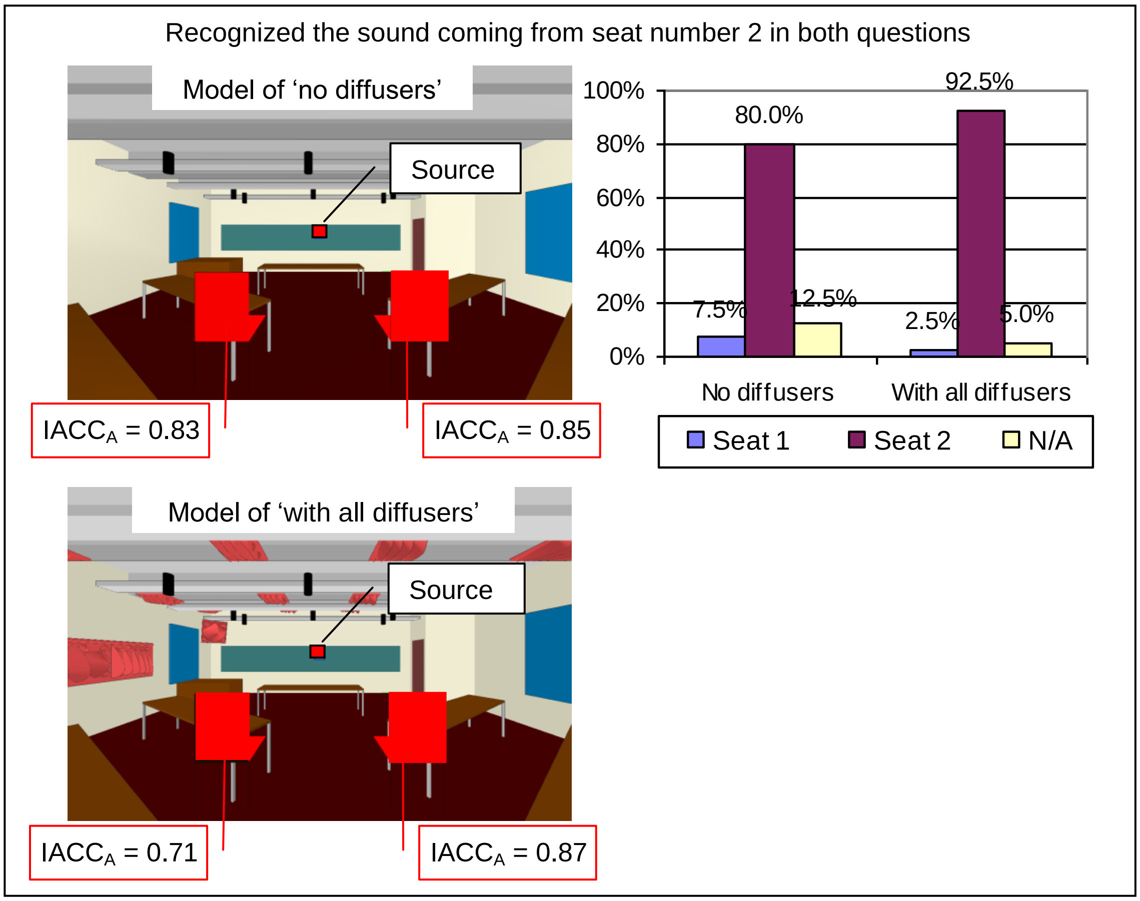


Figure 4.4. Identification of source direction inside AA16 of "with all diffusers" and "no diffusers."

The last paragraph in section 4.3 provided evidence of unnoticeable differences in the loudness perception as distances increased in a diffused sound field. Further observation was done with the subjective assessment by questioning the ability to identify a listening position of a given audio stimulus. The figure of the space modeled with the four different auralization positions or seats was provided in the survey interface (see Appendix C, slide no. 10). The auralization at seat no. 1 was given to the subjects as the first stimulus where subjects were notified of the corresponding listening position. It was used as a reference for the listening position in order to determine the listening position of the second stimulus, which corresponded to seat 4 (details in Figure 2.34 and Figure 4.5). By increasing the source-to-receiver distance, the ability to localize the sound tends to be more difficult in a sound field with diffusion.

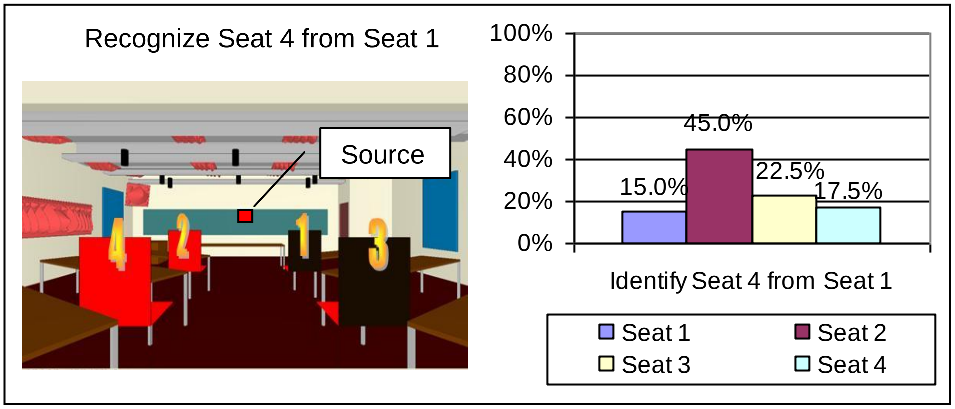


Figure 4.5. Identifying the source position relevant to another source.

Only 17.5% of the subjects were able to locate the correct listening position. More than half of the subjects recognized that the source was coming from their right side (i.e., seat 2 or seat 4). The result also provided evidence that distance perception was not as easy as direction perception. Moreover, it can be concluded that the loudness level as compared to sound clarity is more critical for supporting sound localization.

4.4 Architectural Element Impact on Clarity, Loudness, and Liveliness

This section analyzes the impact of the adjustable wall panels on the loudness, clarity, and liveliness in Duderstadt Audio Studio (DAS). The subjective assessment for this research question is demonstrated in slides no. 4 and no. 5 in Appendix C.

In Figure 3.4 and Figure 3.5, coherences obtained from computer simulation and field measurements in DAS with variations on the panels' positions are provided. This data indicates that the sound field with the panels opened is more diffused. Apparently, the impact is greater on the early reflections since the differences of Coh_{early} are larger than the differences of Coh_{late}. This degree of diffuseness has also increased the clarity index and reduced both the T_{30} and the total SPL, which is confirmed by the subjective assessments results. As shown in Figure 4.6, 82.5% of the subjects perceived the speech to be louder and livelier in the "closed" panel model while 77.5% confirmed speech to be clearer in the "open" panel model.

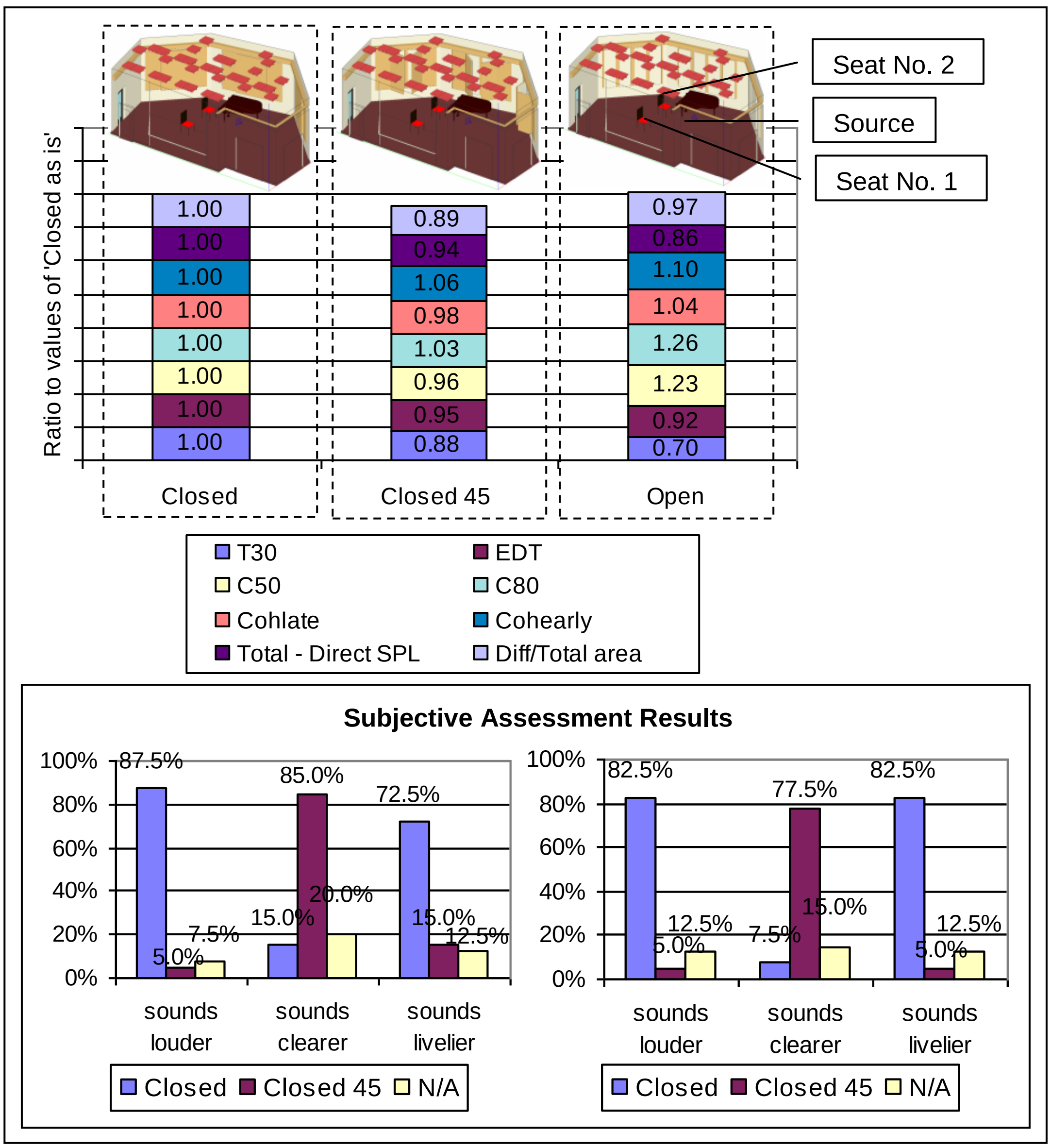


Figure 4.6. Objective Parameters and Audibility in DAS with different wall panel positions.

4.5 Architectural Element Impact on Localizing Sound Direction

The impact of a piano on the diffuseness of the sound field in Duderstadt Audio Studio (DAS) was investigated (see Appendix C, slide no. 3). The coherence for the model with a piano is higher than the model with no piano only at the mid-frequencies. The subjective assessment provides the information that identification of the source location (localization) was easier in the space without a piano.

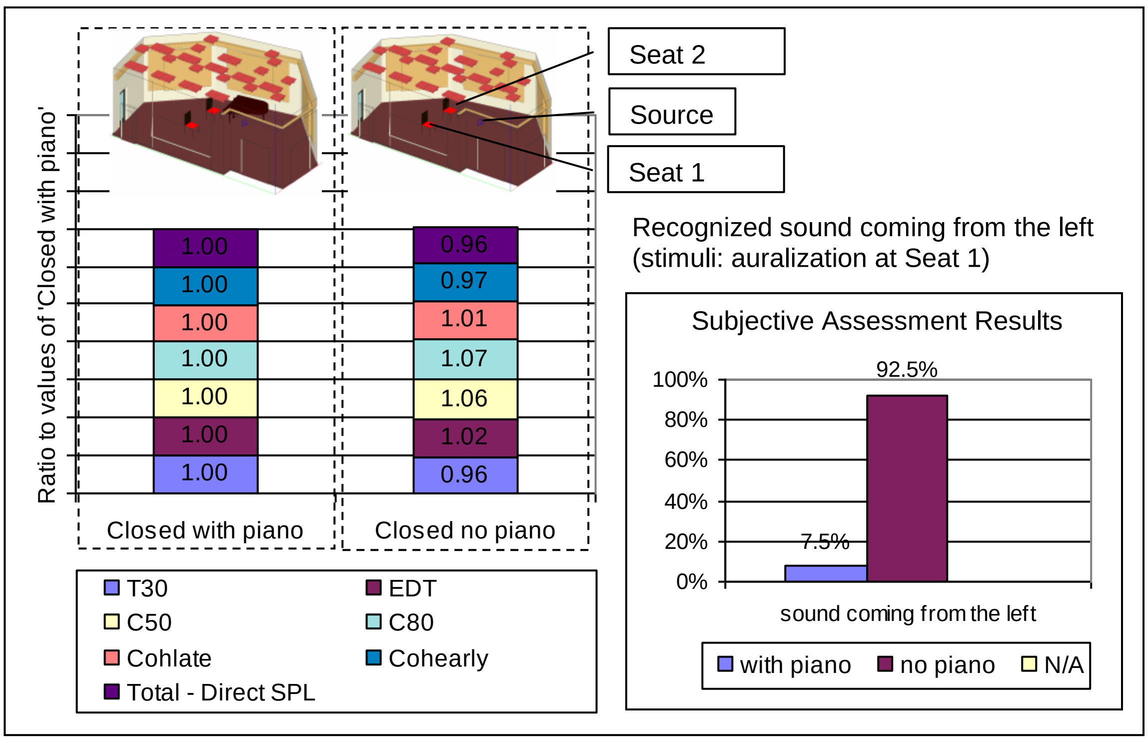


Figure 4.7. Identification of source direction inside DAS of models "With piano" and "No piano."

The result of a higher coherence at mid-frequencies for the "closed no piano" model indicates that a higher degree of diffuseness reduced the ability of sound localization. However, the same values of LEV_{calc} and $IACC_{late}$ for both spaces indicate that there is no difference in sound diffuseness at the left and right ear. A slightly higher clarity index for the "closed no piano" model shown in Figure 4.8 is another objective parameter that supports the subjective assessment results.

4.6 Room Sizes Impact on the Audibility

As described in section 2.3.5, observations of different sizes of rooms were made for the AA16 existing condition. Two room sizes observed were 1) the existing room AA16 with a volume of 332 m^3 labeled "as is" and 2) a model that had twice the volume of "as is" referred to as "large as is."

The audibility condition of a space with two different volumes was compared using the simulation and auralization of room 2216-2219 in the Art and Architecture Building (AA16 "as is"). The subjective assessment for this research question is demonstrated in Appendix C, slide no. 11.

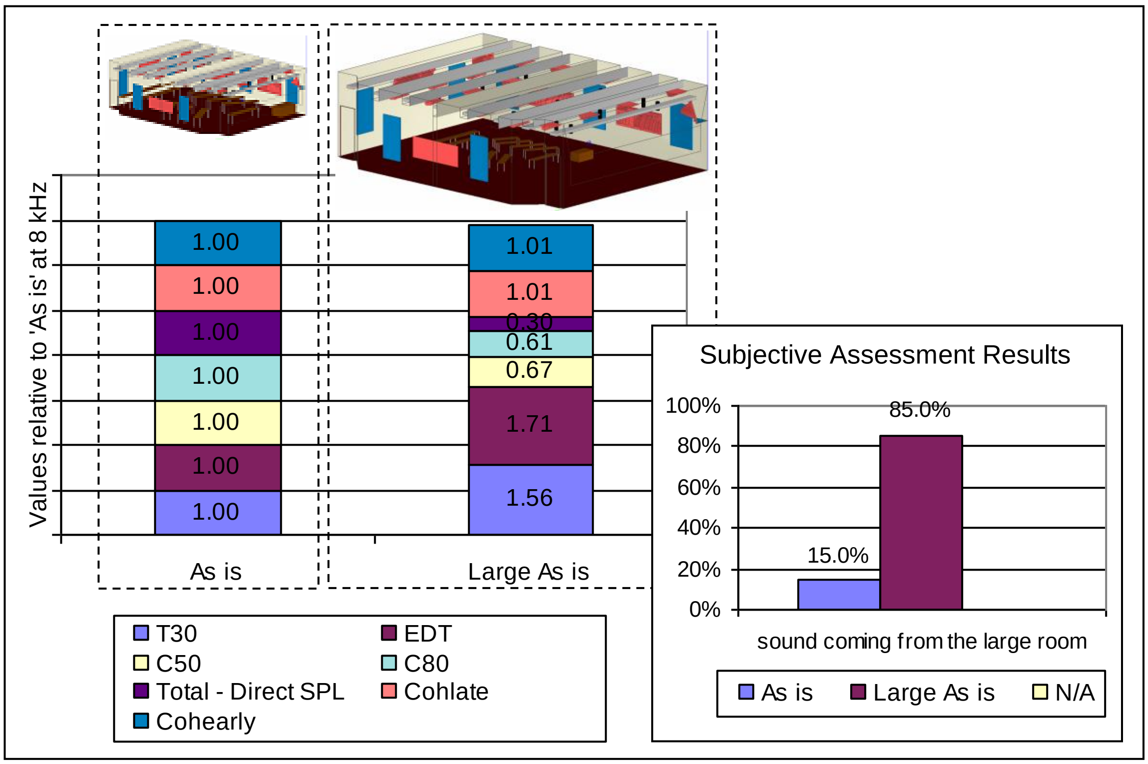


Figure 4.8. Results from the subjective assessment to compare audibility impact of different room size for room 2216-2219 Art and Architecture Building (AA16 "as-is" closed).

A difference between listening in the small and in the large space was perceived by 85% of the subjects. This subjective perception is supported by the change in the clarity index, which decreased by 60% in the enlarged room and which is also related to the increase in T_{30} and EDT. The sound-field diffuseness however, did not change as indicated by the Coh_{late} values. Therefore, doubling the space volume affected the auditory conditions, but not the diffuseness.

4.7 Summary of the Audibility of the Sound Field

The results in general indicated that there are relationships between the objective parameters and the subjective responses with noticeable differences in the loudness, clarity, reverberation perception, and the ability of sound localization.

In Chapter 3, the coherences at late reflections (Coh_{late}) indicated that the ceiling diffusers in DAS increased the sound-field diffuseness. However, ceiling diffusers did not affect the audibility condition. Meanwhile, the application of the ceiling diffusers in the AA16 increased the Coh_{early} as well as the clarity index and reduced the reverberation time. Overall, this created a better audibility condition since the speech was perceived as clearer and less reverberant.

In spaces where the sound-field diffuseness was indicated by the coherences, no noticeable differences were observed in the loudness perception at different listening positions. This is expected based on the theory that the intensity level stays constant beyond the critical distance.

Results from the subjective assessment indicated that diffusion of late energy reduces sound localization ability as the distance increases, since loudness perception remains the same. Meanwhile, the diffusion of the early energy enhances sound localization ability if the amount of diffusion is occurring more on the early energy than the late energy. This was confirmed by observing the EDT values, which in this case were expected to be larger than the T_{30} values. Sound field diffusion and ability to localize sound and distance perception due to architectural elements showed that the more diffuse the sound field is, the more difficult it is for distance perception.

Another parameter used to characterize sound-field diffuseness, especially at the listener's ear, is the LEV_{calc} using variables of G_{late} (i.e., derived from the source strength (G) using equation 2.10), C_{80}, and $IACC_{late}$. An example of analysis using this parameter is found in the observation of the ceiling diffusers in AA16. The LEV_{calc} indicated that the impact was also on the late reflections, given the higher LEV_{calc} with the ceiling diffusers applied. However, applying all the ceiling and wall diffusers created a lower LEV_{calc} as compared to the "no diffusers" model. If more diffusers are applied and the LEV_{calc} decreases, then the parameter which should be evaluated is the C_{80}. A higher C_{80} in the case of a lower LEV_{calc} value indicates that the diffusers' impact is on the early reflections. The subjective assessment results have shown that when this condition occurred, subjects were able to recognize the noticeable differences in clarity, loudness, and liveliness perception.

Given the results and analysis obtained in Chapters 3 and 4, some examples of applications of the key findings in architectural design are provided in Chapter 5. The first section describes principles for room acoustics design guidelines using the ability to control excessive reverberation and to maintain the needed sound energy with a combined strategy of absorption and diffusion application.

Chapter 5
Guidelines for Architectural Design Applications

Similar to any architecture design process, the acoustical design requires a certain course of action to ensure that objectives are reached in order of priority. The highest priorities are addressed, and the largest design solutions are arrived at first, then the details fall into place in concert with the larger issues and ideas (Marshall, 1990). To achieve a systematic room acoustics design process, the highest priority is to define the acoustical objectives of the room.

In practice, architects rely heavily on room geometry and surface characteristics to obtain a desirable hearing condition. The only way to manipulate this condition is by altering the path of sound, which is based on identification of each surface's contribution using field measurements and computer simulation. Failure to translate the acoustical indicators into an appropriate design solution is a major challenge, which is due to the deficiency of available room acoustics design guidelines. Furthermore, guidelines that account for diffusion within a sound field are currently unavailable. This chapter provides the principles pertaining to the use of diffusion for a design solution based on the findings in Chapters 3 and 4.

Each space requires a specific design solution, which depends on its acoustical function and geometrical properties. Owing to these factors, the use of diffusion may not contribute to the room acoustics manipulation. This is obvious in large spaces as shown within the results in Chapter 3. However, large spaces are commonly found to be complex sound fields with multi-zones, in which each sound field or zone requires its own acoustical condition with diffusion as the solution. Three large spaces are used as an example of the application of the principles for the room acoustics design guidelines offered within this study, including a hospital patient care-unit, an atrium of an office, and an ice hockey arena.

5.1 Principles for the Guidelines

Both the desirable and undesirable acoustical conditions that exist in a space can be categorized as bad, nondescript, or excellent acoustics. Nondescript acoustics is an acoustical condition that has no errors and satisfies the listeners. It can be seen as the optimum design achievement for any type of space. This section is intended to help designers and sound engineers provide an acoustical quality within spaces that achieves a nondescript acoustic, without the use of electronic amplification systems, based on diffusion and absorption.

During the design process, observations on sound-field diffuseness should be conducted simultaneously with the reverberation control, not only in an attempt to select the suitable materials for surfaces, but also to position the absorptive and diffusive surfaces.

5.1.1 Acoustical Function

Identification of the design challenges of a room is accomplished by understanding the acoustical function. In the case of concert halls, the primary function of the space is to provide musical communication between performer and audience. The typical concert hall consists of a stage area for the performers and an audience seating area. Design challenges of the stage area depend on the characteristics of the music being performed. In order to accommodate a variety of musical types, the current trend is to create a multi-purpose concert hall by implementing adjustable acoustical panels.

In classrooms and other learning spaces, the primary function is to provide the acoustical qualities for good speech communication between students and teachers without the use of electronic amplification systems (American National Standards Institute, 2002). Reducing energy consumption and promoting an adequate manner of using natural resources are the ecological considerations when addressing environmental comfort in classrooms (Krüger and Zannin, 2004). Thus, it is necessary to create a good design that incorporates an integrated view of multiple environmental variables: acoustics, heating-ventilating-and-air-conditioning (HVAC), and lighting.

In arenas and large sport facilities, such as a football stadium, noise exposure becomes the major concern. Noise exposure is generated from the large number of audience members and other sound reproduction, such as musical performances. The total sound level produced is commonly still being enhanced by electronic amplification systems. Reducing and redirecting the noise propagation by using acoustics barriers are common solutions.

Basically, the aim in using diffusion for sound path manipulation is to satisfy the acoustical function of the space by creating multiple acoustic zones or by creating a single acoustic zone with uniform sound-field characteristics.

5.1.2 Controlling Excessive Reverberation

Techniques for controlling reverberation include the application of a sufficient amount of room absorption. Representation of the data uses the aggregate total room absorption (Sabin), which requires the information of surface areas and the absorption coefficient. Within the computer simulation this information will be treated as the room data.

Given the room data, reverberation time is the next estimator, which requires identification of the room volume. The expected design result is a space with an estimated ideal reverberation time for a particular acoustical function. In large spaces, this design challenge requires careful attention to the amount of sound energy absorption. It is necessary to assure adequate sound levels in listening positions that are farther away from the source. This becomes the main role of diffusers for maintaining and redirecting the sound reflections within the frequencies of interest. The appropriate materials used for diffusers should be carefully selected since all types of materials would have a certain amount of absorption.

Identifying the main source location for the activity within the space and the occupants' area are key elements that significantly create the overall acoustical condition at the human ear. Based on this information, decisions can then be made about the location of absorptive materials in conjunction with the shape of the room. The most critical surfaces are those that are parallel to one another. The aim is to reduce the chance of specular reflections by applying non-uniform amounts of absorption on wide parallel surfaces. Room size, length-to-width ratio, height-to-width ratio, and height-to-length ratio should also be considered. Parallel surfaces that are farther apart should have absorption less than the ones that are closer together in order to provide the occupants with the required sound level for the most desirable listening condition.

5.1.3 Maintaining Sound Energy and Redirecting Reflections

Aside from the diffuse or scattering coefficient of the acoustical treatment product claimed by the industry, it is most important to consider the size, roughness, and absorption coefficient of the diffusive surfaces. Size and roughness can be determined by the depth of the surface roughness relative to the wavelength of the sound being controlled as illustrated below in Figure 5.1.

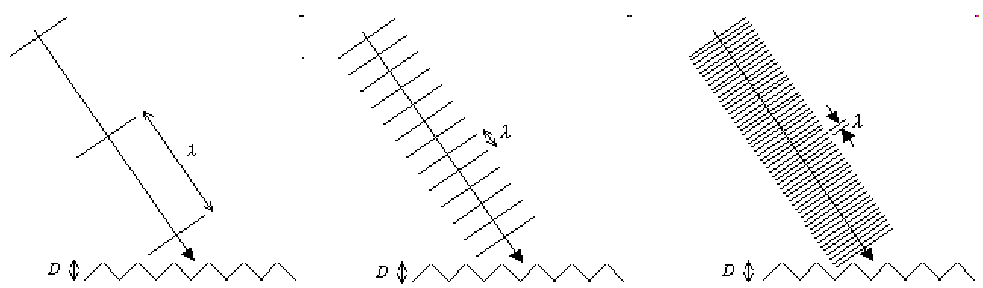


Figure 5.1. Proportion of surface roughness depth relative to the wavelength of interest.

Effectiveness of the acoustical treatments in maintaining the sound energy can be first predicted by the calculation of reverberation time within a variety of design configurations. Using the computer simulation, the ratio of the diffuser surfaces relative to the total room surfaces in the computer models can be traced relying on sets of parametric runs. Contributions to the early and late reflections of the sound decay utilizing EDT and T_{30} of these alternative designs should be compared. Maintaining the sound energy ideally can be achieved by maintaining the EDT value, while enhancement of sound energy is expected by creating a longer reverberation time. Along with the EDT and T_{30} is the use of Coh_{late}, Coh_{early}, and clarity index, which can predict the proportionality of the sound energy distributed by the early and late reflections.

Meanwhile, effectiveness of the diffusers in intruding upon the sound reflection's directivity is predicted by the observation on the Coh_{early} and Coh_{late}. In small to semi-large rectangular rooms, positioning of the diffusers is a critical decision, with the goal being to intrude upon inter-reflections between opposite parallel surfaces. Diffusers are shown to be effective on room corners. Often, tilting the diffusers to a certain angle improved the results. Two cases of rectangular rooms that were studied can be used as an example of this result: Dennison Hall (DH) and room 1221 of the Art and Architecture Building (AA21). The results and analysis in Chapter 3 have shown the impact of tilted diffusers in AA21. Sound-field diffuseness increased with noticeable changes in the objective parameters. Meanwhile, without being tilted, the use of diffusers deteriorates the acoustics condition due to its reflective surfaces. The space DH was compared with Angell Hall Auditorium A (AHA). The curved walls in AHA created the same degree of diffuseness as the sound field in DH even though the number of diffusers was only half as many as in DH. This supports the conclusion that geometrical shape of the room contributes to the sound-field diffuseness.

In order to obtain an effective design solution, the number of diffusers in large rooms mainly depends on the sound source and the listener's position. This requires the entire sound field within the room to be divided into small sound-field regions, where each may require a unique solution with a greater number of observation positions.

5.2 Example of a Complex Sound Field: Hospital Patient Care Unit

The acoustical design challenges in hospitals are to provide better communication that will reduce medical errors, and to assure speech privacy that has also become a legal issue, according to the Health Information Portability and Accountability Act (HIPAA). Some industrial product performance criteria have been found to exaggerate claims concerning solutions for better room acoustics in healthcare spaces.

No matter what the desired acoustical function of any room is, good speech intelligibility is important in any activity. Parameters for best speech predictors are discussed in section 2.1. Speech intelligibility is a measure that indicates the ease of understanding speech. Speech intelligibility depends on the signal-to-noise ratio (SNR) and reverberation time (RT). The SNR is determined by the speech sound level pressure and also the A-weighted noise level. The importance of controlling these two variables is equivalent to the idea of diffusion control in reducing excessive reverberation, while maintaining the energy required for a sufficient SNR.

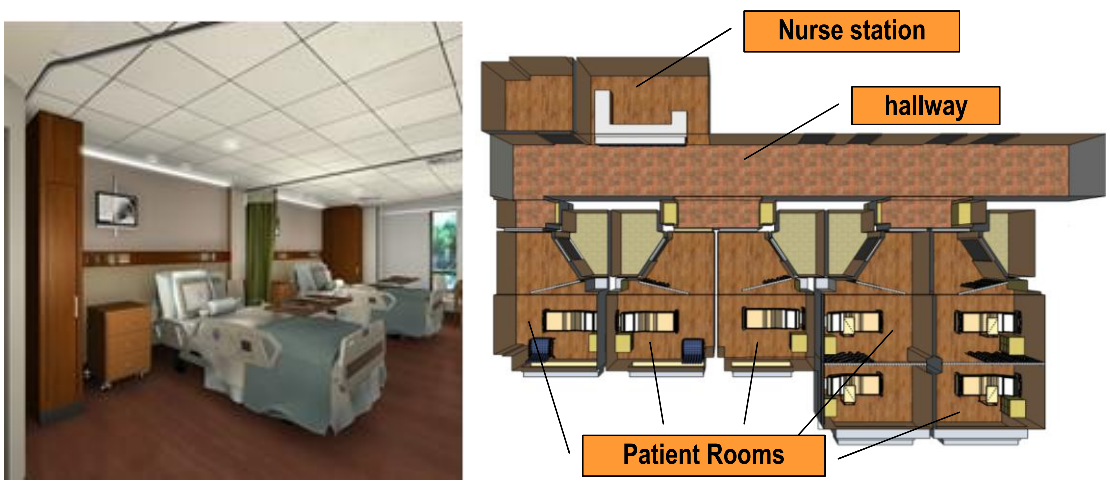


Figure 5.2. Space layout of the existing condition and the patient room layout prior to study.

The modeled space consisted of adjacent patient rooms and a nurse station connected by a hallway. Building regulation codes, health service activities, space openness, and confined space layout due to occupancy activities were used as parametric studies for an effective room acoustics design solution. Acoustical treatments with characteristics adopting the industrial products were applied.

The computer models were based on the condition of having patient-room doors opened, a scenario often required to support intensive care and emergency access. As a consequence, the noise from the adjacent hallway will leak into the patient room, interfere with the ambient noise, and reduce the signal-to-noise ratio within the room. To provide speech intelligibility, the adjustable hanging curtain became the critical element for sound insulation to reduce the ambient noise.

Applying more absorption in the patient room increased the speech intelligibility, but with the drawback of decreasing speech privacy. This is due to the high signal-to-noise ratio (i.e., given the low noise level as a result of highly absorptive space).

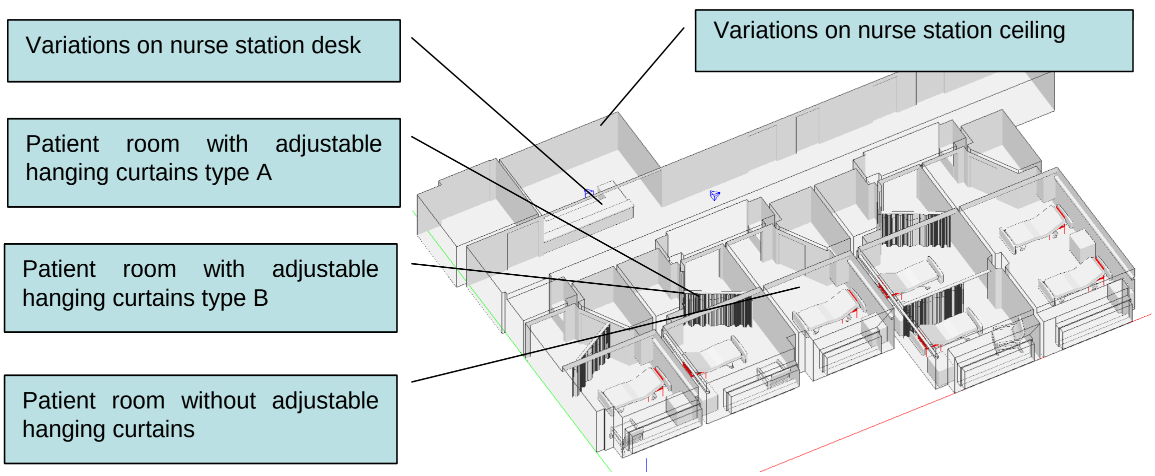


Figure 5.3. Parametric runs in the computer simulation of a hospital patient-care unit.

Speech privacy in hospital is required to avoid breaching patient confidentiality agreements and, as well, to prevent patients from overhearing information that would cause them stress or create anxiety. The values of the objective parameters, which were a result of design configuration of hanging curtains in the patient rooms, are provided in Figure 5.4.

As the listener gets farther away from the source (i.e., source no. 2), a sufficient amount of sound level is required to achieve good clarity. If the intensity level of the source output remains the same, the good clarity can be achieved by reducing the amount of sound attenuation, which gives a high reverberation time (T_{30}). However, the T_{30} of more than 1 second at receiver no. 5 did not support the clarity shown by the drop of C_{50} value, while at receiver no. 2 (i.e., closer to the source), the high T_{30} was supported with a good clarity (i.e., a positive C_{50}).

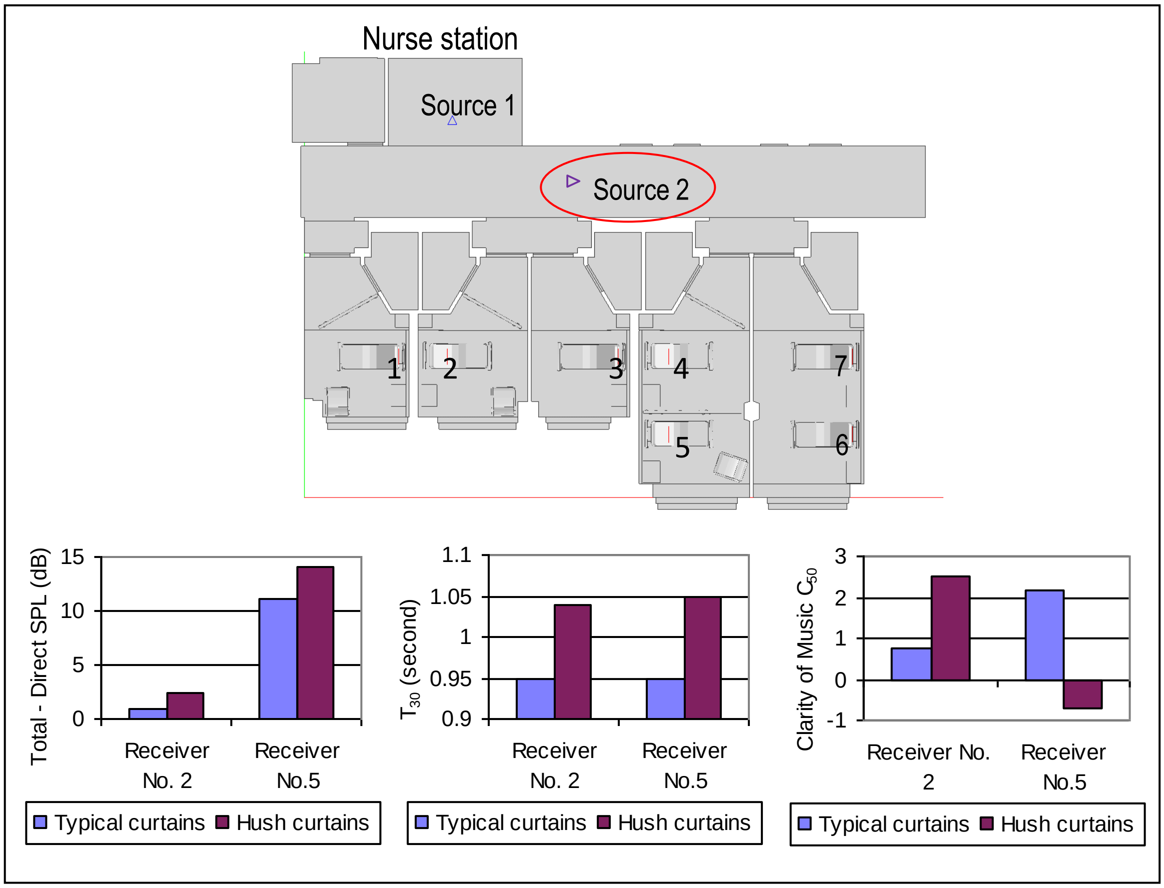


Figure 5.4. Results in patient rooms due to curtain variations based on sound source 2.

By comparing the intensity of the reflected sound using the total SPL minus the direct SPL (i.e., delta SPL), the amount of absorption by two different types of curtains and the air, given the different source to receiver distances, can be observed.

The direct SPL depends on the distance between source and receiver. Receiver no. 4 has a larger direct SPL than receiver no. 2 since it is closer to the source. However, receiver no. 4 has a lower delta SPL (see Figure 5.5) due to the space layout and the second curtain dividing receiver no. 4 and no. 5.

It affected the sound insulation result (i.e., reverberation control). In the case of receiver no. 4, the use of hanging curtains becomes less necessary to reduce noise from the hallway, given the space layout.

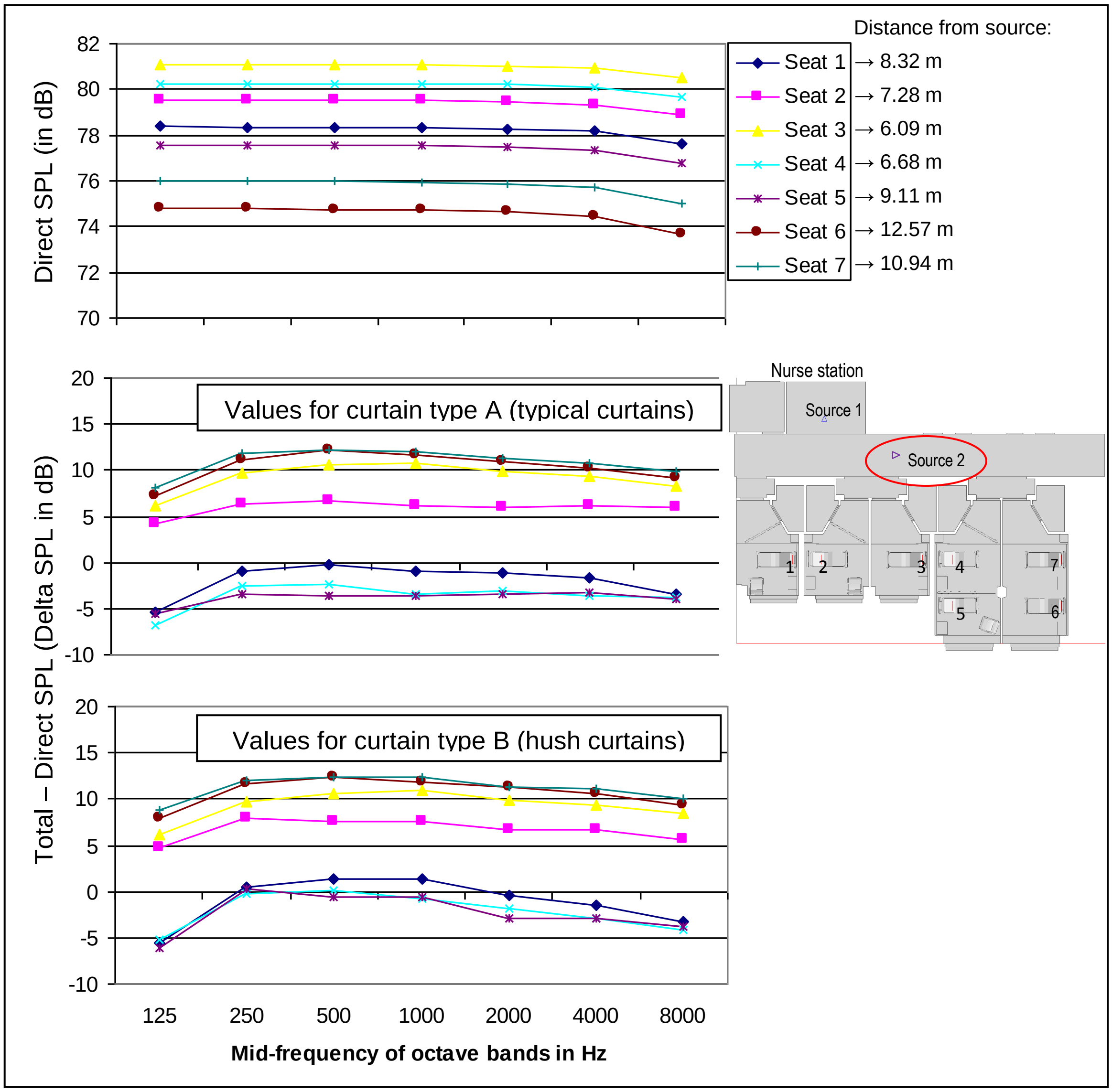


Figure 5.5. The intensity of reflected sound due to absorption by curtains and air at all 7 receivers with source no. 2, indicated by the total minus direct SPL values.

Speech activity among physicians and nurses for medical information exchanges is the major communication activity in the nurse station. Therefore, it is considered to be the most important space to be acoustically well designed so that medical errors are avoided and speech privacy is maintained.

The attempt to maintain the sound energy within the nurse station only and avoid leaking sound to patient rooms was the hypothetical design solution offered in this study. Source no. 1 at the nurse station was used in this observation.

The discussion of the results focuses on the values of parameters measured at receivers no. 2 and no. 5. Using the results in Figure 5.6, adding absorber and diffuser panels on the ceiling reduced the reverberation time (T_{30}) values. Adding diffusers on the ceiling that already had absorber applied (i.e., diffusers and *ultima* ceiling) increased the T_{30} at receiver no. 5 but created a negative C_{50} or an indication of a condition of poor speech clarity.

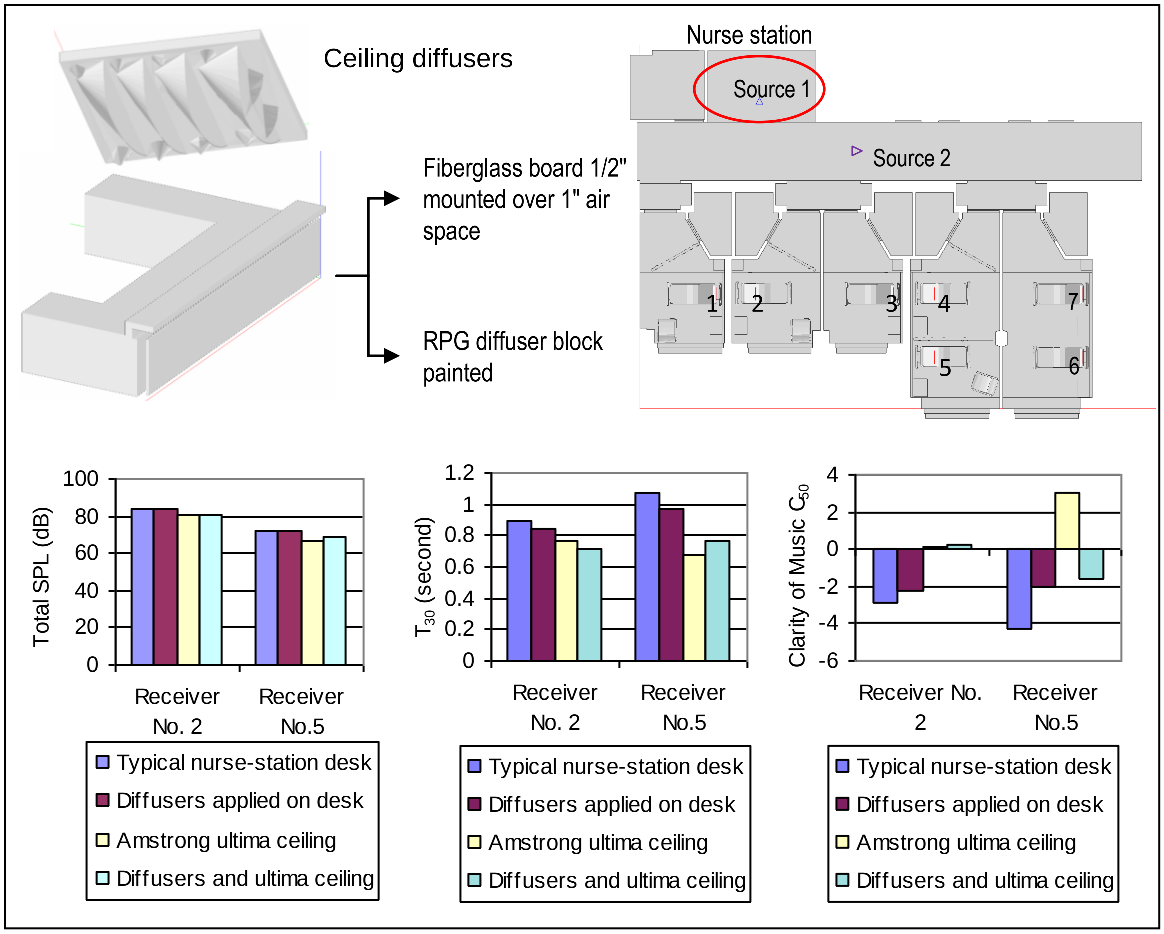


Figure 5.6. Results in patient rooms due to nurse station design configurations based on sound source 1.

Although adding diffusers slightly increased T_{30} (see Figure 5.6) and the intensity of the reflected sound (see Figure 5.7), it did not always create better speech clarity for the sound fields that are highly absorptive.

The important variable here is the direct sound characteristic that propagates within the sound field, demonstrated by the results at receivers no. 2 and no. 5, which represent two different source-to-receiver distances.

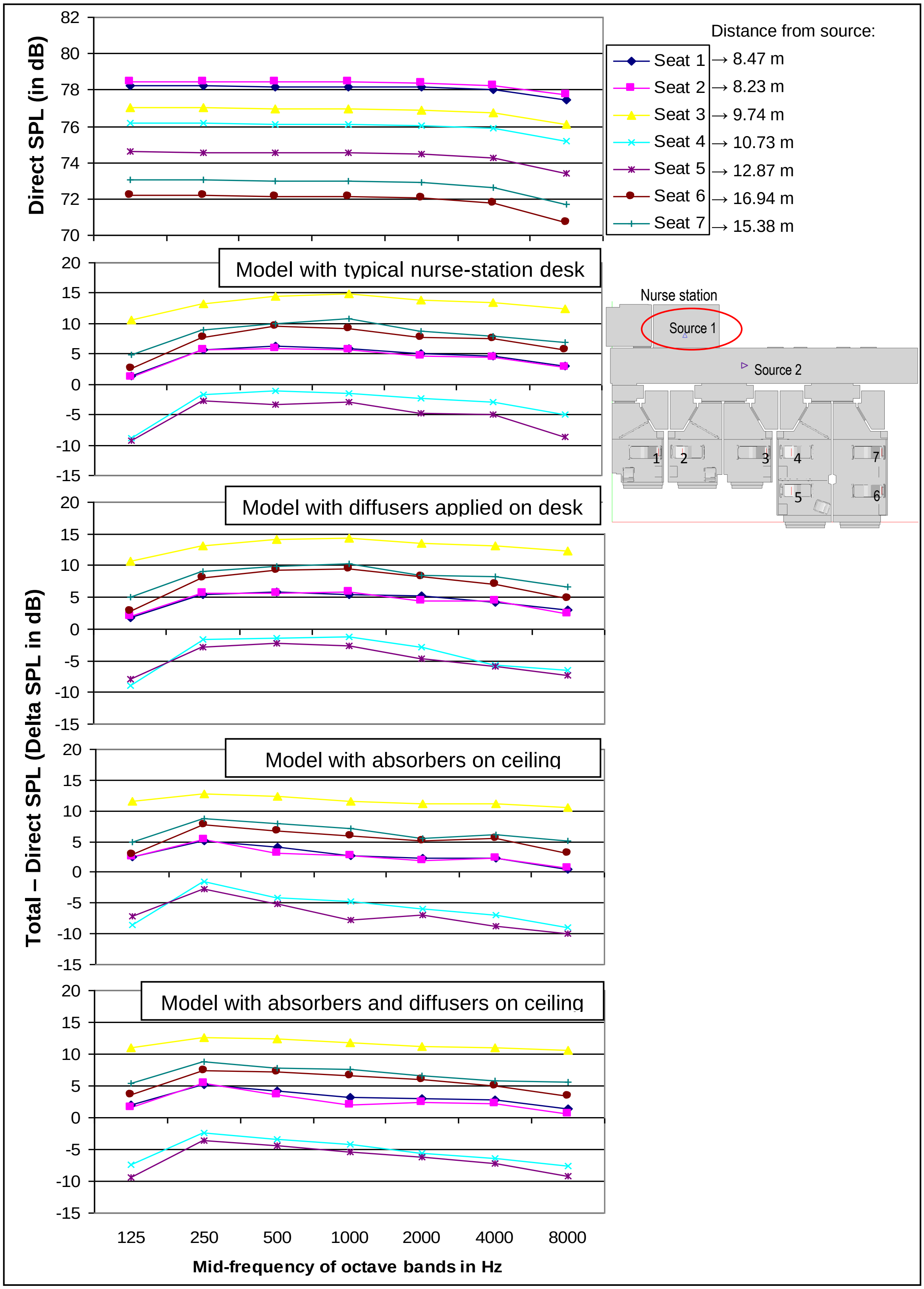


Figure 5.7. The delta SPL resulting from the acoustical design of the nurse station based on source no. 1.

Controlling the reverberation can bring about better speech clarity, and therefore better speech intelligibility. Adding diffusers enables an increase in the intensity of reflected sound and an amplification of the early reflections. If noise masking is applied by introducing a background noise into the sound field, the noise level will also increase, and therefore support the speech privacy by decreasing the signal-to-noise ratio.

Increasing source-to-receiver distance will decrease the signal strength, and therefore improve speech privacy. In order to maintain the speech intelligibility by having sufficient signal strength, the amount of room absorption should be considered carefully (e.g., results at receiver no. 5).

The hospital patient-care unit is an example of a multi-acoustic zone. Each patient room is a sound field with different design solutions. The study requires further observation using more source positions to first estimate the optimum baseline condition for the entire care unit.

Further design alternatives should be observed in future studies particularly to observe sound insulation at the nurse station by adding receivers within the sound field. The goal is to isolate this sound field where medical information is dominantly being exchanged from other parts of the semi-open-space layout.

5.3 Example of a Vertical Sound Field: Atrium of an Office

An atrium is a circulation space within an office that allows access into the various adjacent work spaces. Flexibility of the space function should ensure that there is the correct balance of controlling excessive reverberation while maintaining some energy for ease of communication.

The main acoustical issues impacting the design are 1) noise leak from the atrium into adjacent rooms or offices, 2) reverberation and room reflections, which have to be carefully controlled to achieve the correct ambient condition, and 3) speech intelligibility and speech privacy, which are major problems in an open space layout.

Table 5.1. Materials assigned in the original design.

Surface	Item	Material	Total Surface [m²]
A	Wall A	Brick unglazed	367.16
B	Wall B	Dry wall	756.88
C	Wall+ column+ overhang ceilings	Concrete smooth	298.12
D	Window glass	Window Glass	238.81
E	Ceiling	Gypsum board	303.62
F	Wood panels	Wood finish	107.63
G	Floor	Tile Floor	318.19
H	Plant Bowl	Marble	45.20
Total surface area			2435.61

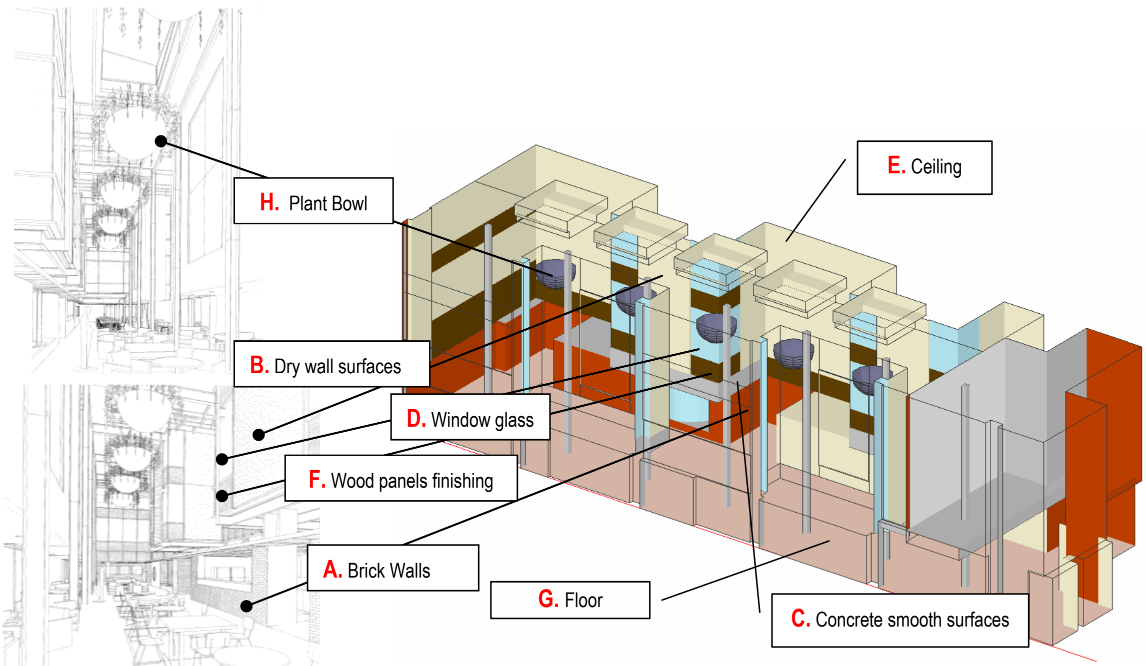


Figure 5.8. Physical properties of the office atrium.

The acoustical design is based on the given architectural design, and the most critical surface is the window glass due to the highly reflective characteristics of glass. The largest proportion of the surface area is the ”wall B” assigned in the original design as drywall. These two surfaces are among others that were first taken into consideration during the room acoustics design.

The other critical element is the enhancement of daylight penetration into the atrium. Alternatives for room acoustics design solutions are based on the installation technology and materials, while taking into consideration the effectiveness of daylighting from the skylights. In a conventional installation the acoustic panels are usually designed to attach parallel to the ceiling. To avoid blocking the skylight ceiling, and thus reducing the daylight, the acoustic panels can be installed vertically, as proposed in Figure 5.9. Installing them in a row would diffuse some portion of the sound energy that is coming from any direction. The panels presumably can also behave as daylight diffusers to create more uniform daylighting throughout the space underneath them.

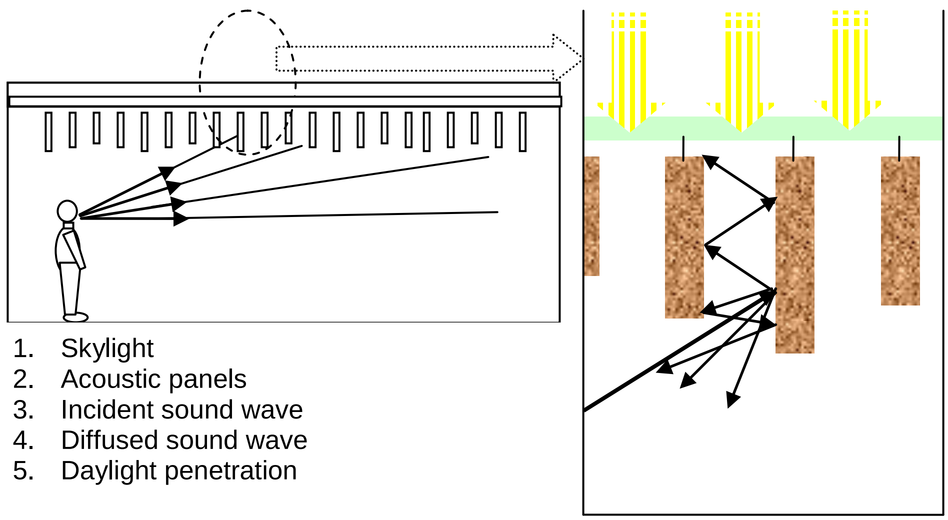
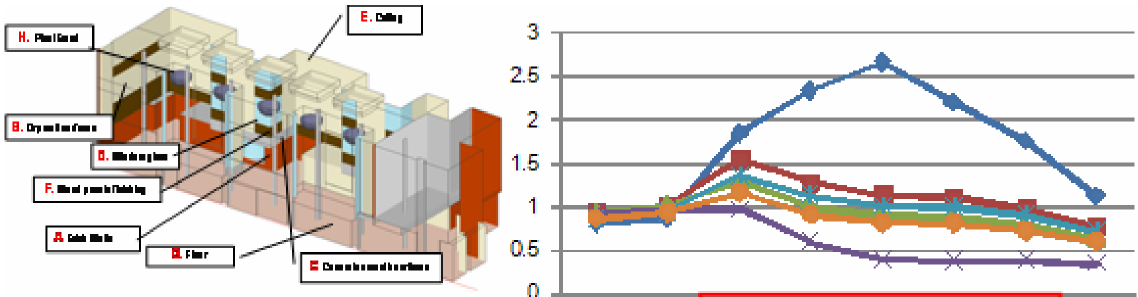


Figure 5.9. Vertical acoustic panels suspended from a skylight structure.

The material used for suspended horizontal panels that have large surface areas should take into consideration the transmittance coefficient for light to avoid low efficiency of daylighting obtained from the skylight. A well-selected material must be chosen to serve this purpose. The alternative designs addressed here are categorized into two types of design: decorative design and integrated design, which are panels installed as part of the ceiling element. The key factor is the proportionality between the area for daylighting and the room acoustics treatment. The hanging plant bowls present in the design prior to the study were utilized as decorative acoustic diffuser panels. The decorative design, in this case, is the ”panels” installed as the decorative elements of the room ceiling. Alternative designs of this type of ”panel” are unlimited, as they may be constructed in an infinite variety of shapes, sizes, and installation types. In principle, the installation does not need to block the whole ceiling area, but may be more scattered throughout the space with sufficient distance from the skylight.



	63 Hz	125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8 kHz
As designed	0.82	0.87	1.84	2.33	2.66	2.19	1.75	1.12
With Ultima ceiling	0.93	1.01	1.54	1.28	1.13	1.11	0.98	0.77
Absorption panels on surface A	0.94	1.02	1.29	1	0.9	0.88	0.79	0.64
Acoustics plaster on surface B	0.94	0.96	0.97	0.6	0.4	0.38	0.39	0.35
Abbfusors on plant bowls	0.86	0.94	1.36	1.14	1.01	1	0.9	0.71
Abbfusors on plant bowls + Absorption panels on surface A	0.87	0.94	1.16	0.91	0.82	0.81	0.73	0.6

Figure 5.10. Reverberation time (T_{30}) of different acoustical treatments.

The reverberation time at the mid-frequency of the design alternative in which acoustic plaster was applied to the surface of ”wall B” was significantly reduced by more than 2 seconds. This design element created an ideal reverberation time for speech intelligibility with the range of 0.7 – 0.6 seconds. The reverberation time modeled above has rendered a listening condition that is “too dead” for this relatively large open-space.

The atrium is designed to serve several activities, which include circulation and seating area. A large projector screen will be displayed on the upper wall of the auditorium entrance. In addition to efforts that reduce reverberation for speech intelligibility within the space, the atrium design should also have sufficient background noise. Without a certain level of background noise and an increase of signal strength due to room absorption, problems related to speech privacy may occur within the seating area. A group of people might overhear the conversation of a different group.

The effect of plant bowls modeled as diffusers using acoustical characteristics (i.e., absorption and scattering coefficient) of the RPG Harmonix™ K material on the sound-field condition can be observed with the intensity of the reflected sound (i.e., delta SPL). The hanging bowls slightly increased the delta SPL, especially at seat no. 2 at low frequencies up to 500 Hz as shown in Figure 5.11. The delta SPL only indicated the amount of reflections, but not the change of the diffuseness.

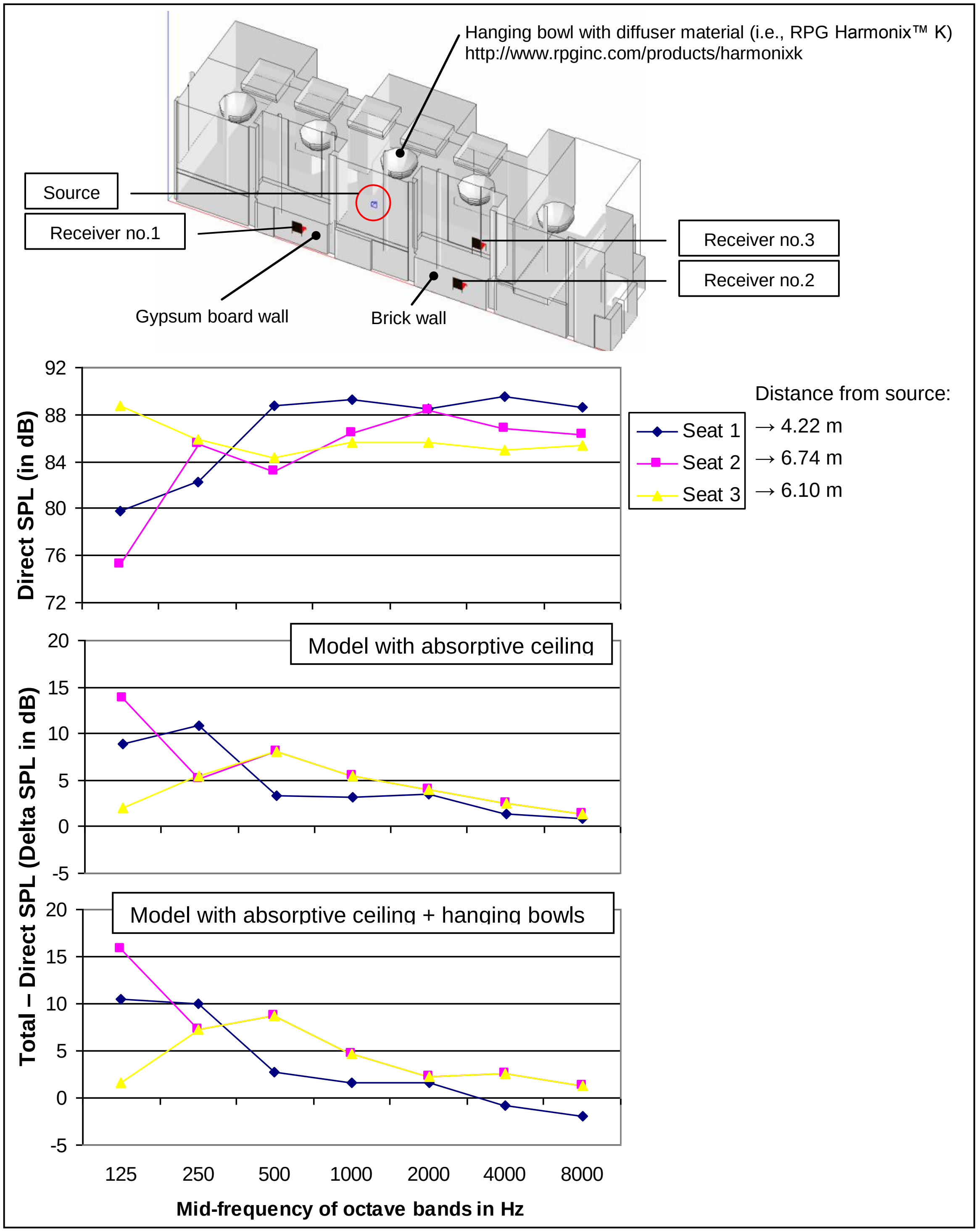


Figure 5.11. A comparison of delta SPL with and without the hanging bowl.

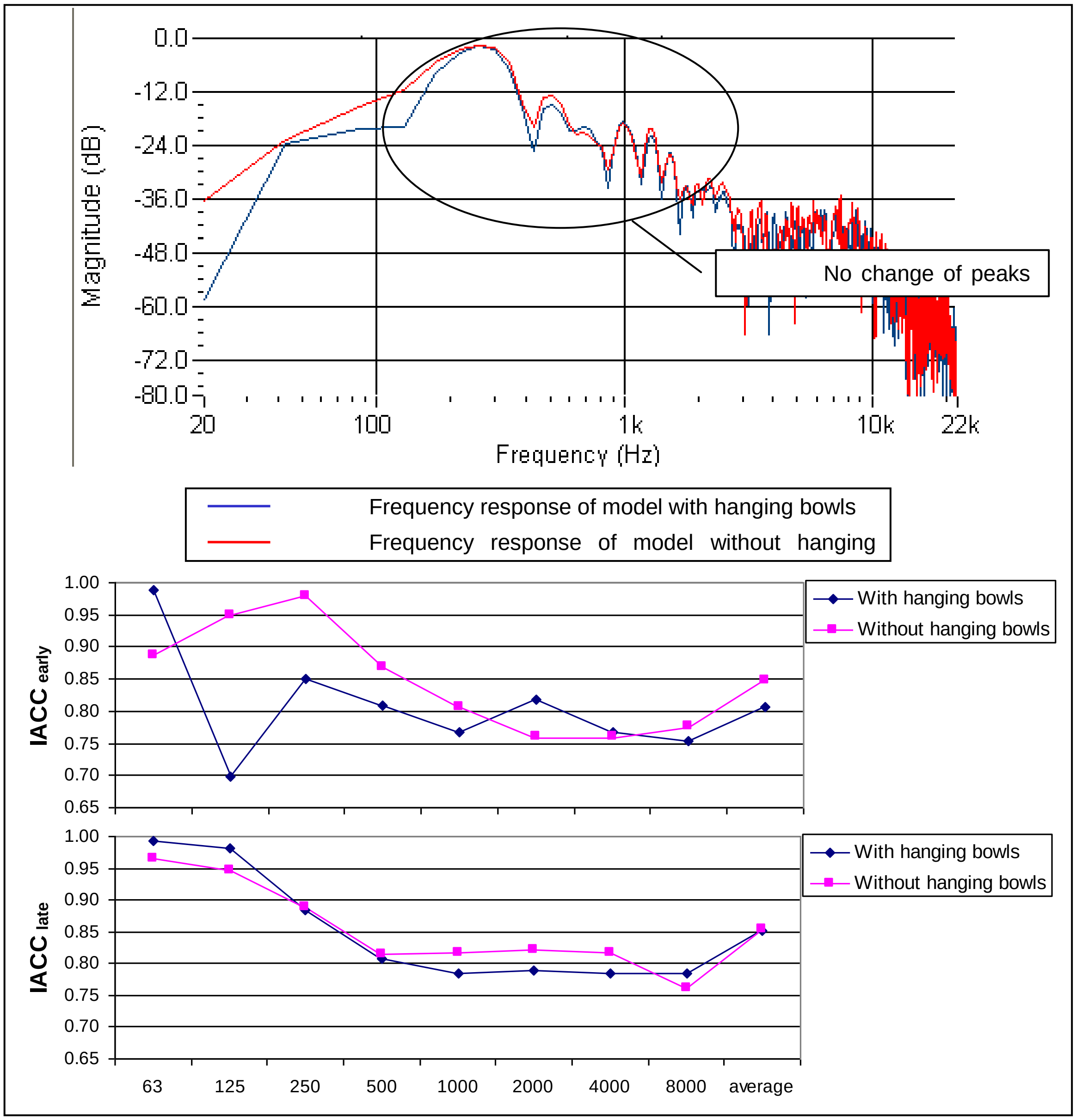


Figure 5.12. The $IACC_{early}$ and $IACC_{late}$ at seat no. 2 of computer models with and without the hanging bowl.

Observations of the early reflections indicated that the hanging bowls did not reduce comb-filtering (see Figure 5.12). There is a drop in the IACC values of early reflections ($IACC_{early}$) for seat no. 2 with the hanging bowls applied, indicating that early reflections did not improve the diffuseness. As for the late reflections ($IACC_{late}$), the hanging diffusers indicated no change in the diffuseness.

Since multimedia will be projected into the space, a public announcement (PA) surround sound system was modeled, and simulation of the performances was modeled. The study on optimization of the sound system was done after the optimum room acoustics was obtained based on architectural solutions only.

Two types of sound sources were utilized in the simulation. Sound with directivity of human speakers was positioned at the seating area to simulate the condition of having occupants sitting or walking through the atrium. The surround sound system is expected to provide ambient sound into the space, which is used as a sound masking effect to support speech privacy.

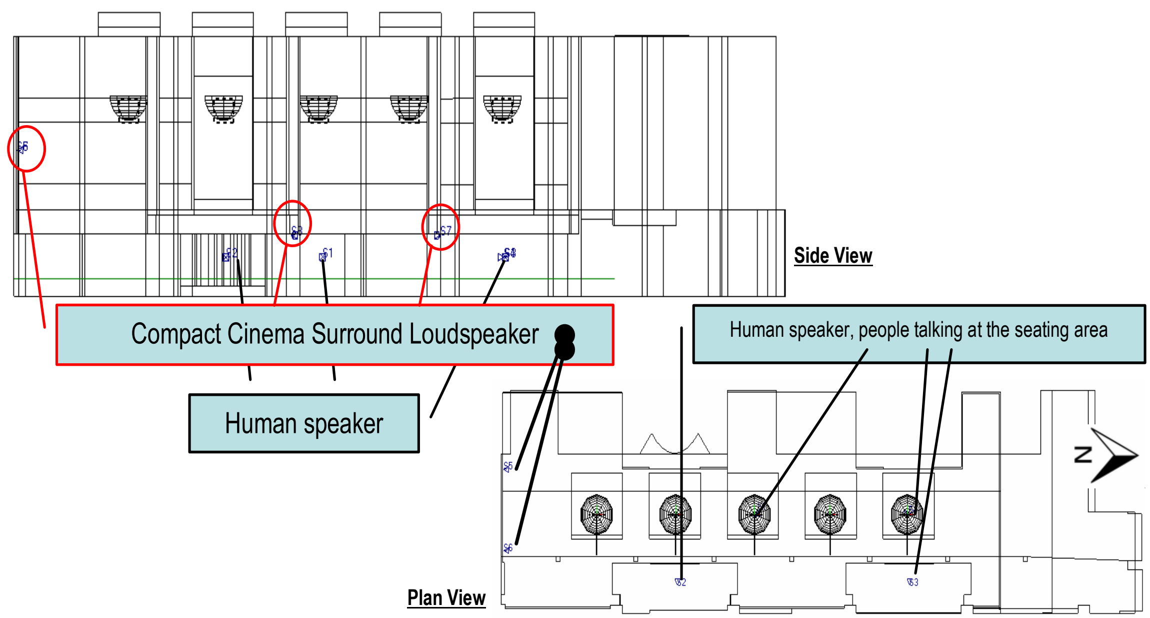


Figure 5.13. Positions of sound sources within the computer model.

5.4 Example of a Sport Arena: Ice Hockey Arena

Environmental noise impact and sound insulation is one of the main concerns in acoustical design. Local codes strictly limit noise exposure in buildings, particularly for sport arenas with crowds of spectators. Design of the sound insulation is related to the amount of reverberation and room reflections. It is necessary to carefully control the reverberation to achieve the correct ambient condition. A large arena cannot rely on the room acoustics alone without the use of a sound system. A balance must be applied between maintaining spectator excitement and acoustical control (by sound absorbing surfaces) to ensure optimum performance for the sound system.

In accordance with International Electrotechnical Commission (IEC) Standard 60268-16 codes of practice, a place of public assembly must have a voice alarm system achieving a specific minimum speech intelligibility requirement, in this case 0.45 STI (speech transmission index). This involves selecting, locating, and orienting the loudspeakers as well as designing and locating acoustic treatments.

Figure 5.14. Existing condition of the ice hockey arena.

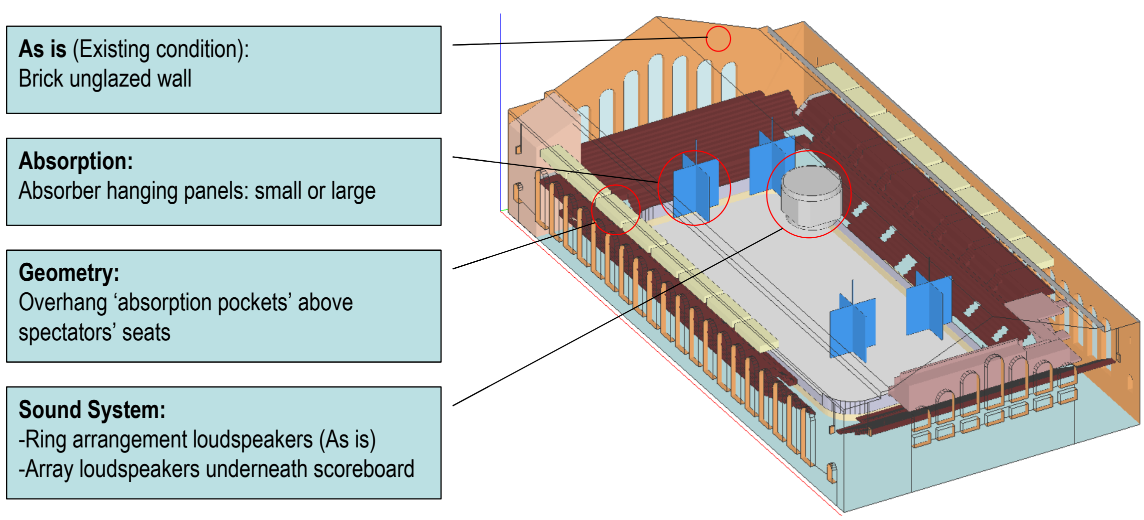


Figure 5.15. Elements of design for room acoustics computer simulation of the ice hockey arena.

Four elements of design were considered within the parametric runs of the Yost ice hockey arena, which were the main wall, hanging panels, absorption "pockets" above spectators' seats, and arrangement of the sound system as described in Figure 5.15. A major renovation was planned for this place with a particular focus on the audience seats and the sound system.

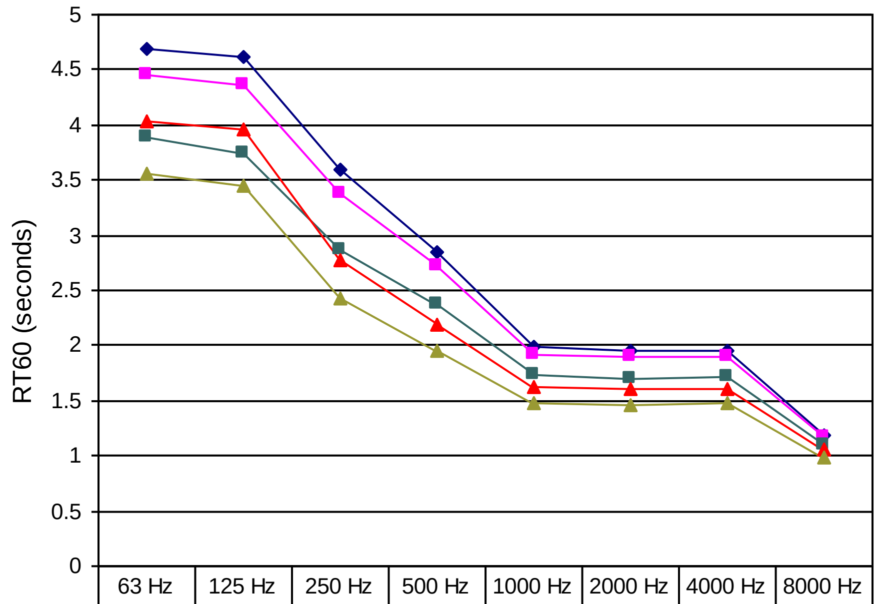
The acoustical design goal was to replace the existing loudspeaker arrangement with arrays of loudspeakers, a new scoreboard, and embedded loudspeakers as a whole sound system unit hanging at the center of the arena. The final sound system design options are not provided in this text since the scope of the research was to explore the amount of absorption and reverberation time given the preliminary design drawings.

To define the baseline or the model of "as is" (i.e., existing condition), the first element explored was the material of the main wall given its large surface area with the possibility for design alteration. The main wall has an area of approximately 4641 m^2. Unglazed and unpainted brick was chosen as the "as is" condition for the main wall since it has a larger absorption coefficient as compared to painted brick. The Eyring reverberation time (RT60) values for model "as is" are shown in Figure 5.16.

The existing condition (as is) with the brick walls and steel frame ceiling structure creates a high reverberation time (RT_{60}), more than 3 seconds for low octave band frequencies. More sound absorption is required to control the excessive reverberation. However, the possibility for physical changes was limited by the geometrical property of the space. This led to an attempt in which hanging absorber panels were inserted into the space. Two different sizes were modeled and the RT_{60} are shown in Figure 5.15. Applying the large hanging panels only reduced the RT_{60} by 0.20 seconds at 1000 Hz.

The final attempt to achieve a smaller RT_{60} was the addition of absorption pockets above the spectators' seats. This design alternative was based on the identification of the most critical reflective surface with the scoreboard as the source.

The observation was done using the ray-tracing method in Ecotect (see Figure 5.17). The most effective design solutions were the application of both the absorptive hanging panels and the absorption pockets.



	63 Hz	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	8000 Hz
As is	4.69	4.62	3.59	2.85	1.99	1.96	1.95	1.18
Add small hanging panels	4.46	4.37	3.38	2.71	1.92	1.89	1.89	1.16
Add large hanging panels	3.89	3.75	2.87	2.37	1.73	1.7	1.71	1.09
Add small panels + absorption 'pockets'	4.04	3.96	2.78	2.19	1.63	1.6	1.61	1.05
Add large panels + absorption 'pockets'	3.56	3.45	2.42	1.95	1.48	1.46	1.47	0.99

Figure 5.16. Reverberation time obtained from computer simulation of the ice hockey arena.

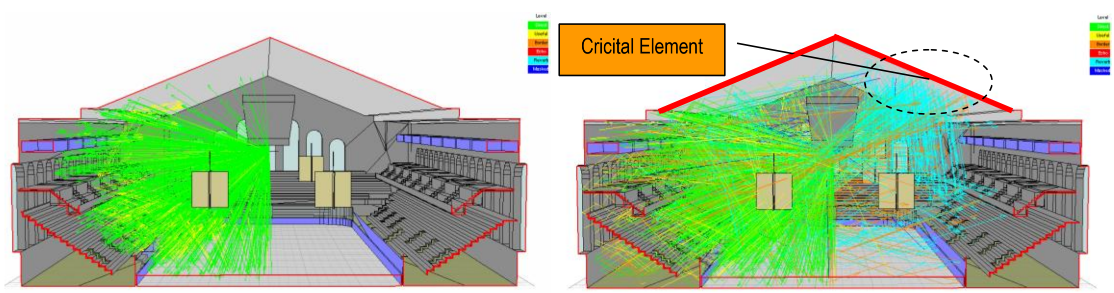


Figure 5.17. Ray tracing with Ecotect of the Yost ice arena.

Using this preliminary result, two other design elements will be further investigated, which are the sound system and the hanging panels. The hanging panels will serve not only to absorb the excessive reflections that reach the ceiling but will also behave as diffuser panels. With careful adjustments between the positioning of the panels, the shape and grating of the diffusive surfaces, and the directivity of the loudspeaker output, this solution will fulfill the required codes of practice described in the beginning of this section.

5.5 Summary of the Application within the Examples

The principles for the guidelines for architectural design application defined in this chapter are intended to help designers and sound engineers provide an acoustical quality within spaces based on diffusion and absorption.

There are three important principles to understand, owing to the combined use of diffusive and absorptive panels in a space. First is to understand that every space has a unique acoustical function, where common design solutions are applied with a different state of the art approach in the use of diffusion. The room acoustics manipulation is to satisfy the acoustical function of the space by creating multiple acoustic zones or by creating a single acoustic zone with uniform sound-field characteristics.

The second principle is to control excessive reverberation by using absorptive panels and engaging the potential of the diffuser's absorption given the characteristics of the material. The placement and positioning of the absorption treatment is based on the considerations of parallel walls, room size, length-to-width ratio, height-to-width ratio, and height-to-length ratio. The amount of reflections as a result of the absorptions is indicated with the delta SPL or the total minus direct SPL.

Characterizing the scattering of the diffusive surfaces by considering the size, roughness, and absorption coefficient, is the first step in application of the third principle, maintaining sound energy and redirecting reflections. The energy decay curve (EDT and T_{30}) obtained from room acoustics measurement is the first indicator on the effectiveness of the design configuration. Along with the EDT and T_{30} is the use of Coh_{late}, Coh_{early}, and clarity index, which can predict the proportionality of the sound energy distributed by the early and late reflections and also the diffuseness of the reflections (i.e., related to the directionality).

A hospital patient care-unit, an atrium of an office, and an ice hockey arena are the spaces used as example applications of the principles for the room acoustics design guidelines. These spaces are representations of complex sound fields with multi-zones where the sound fields or zones require certain acoustical conditions with diffusion as the solution.

Chapter 6
Conclusion

In practice, an anechoic (i.e., not having echoes) condition within architectural spaces is hardly found. The unabsorbed portion of the sound energy will be reflected. A diffusion control system has the ability to manipulate a portion of reflected energy by changing its directionality and energy distribution within a frequency content. Some spaces rely on this manipulation to fulfill the required room acoustics, such as spaces with activities that need to eliminate excessive reverberation while maintaining a certain amount of the sound energy.

A standardized method to measure the impact of diffusion on a sound field is not yet available, nor is the ability to characterize the audibility conditions. This is due to the lack of measurement procedures, effective equipment, and appropriate parameters to predict the acoustic conditions of the space. This study aims to characterize the audibility of a sound field with diffusion, and to identify the geometrical arrangement and architectural elements of the space that significantly contribute diffusion within the sound field. Characterization is based on relationships among objective parameters and subjective attributes describing the auditory perception.

6.1 Summary of Findings and Conclusions

Diffusion can be created by diffusive surfaces in the form of diffusers or other architectural elements within an enclosed space. The sound-field diffuseness depends on the interplay of diffusers with other architectural elements, room size/volume, and room shape. Diffusion is a dispersion of reflected sound into its frequency components, and it impacts different wavelengths in spaces with a homogenous atmospheric condition. Since, the distance of the sound path is also determined by the room dimension, different room volumes create different degrees of impact on the sound-field diffuseness for the same frequency range.

The room volume with the absorption characteristics of the surfaces defines the classification of a sound field based on the frequency of the propagating sound waves, the specularity, and the diffuseness using the "Schroeder frequency," a crossover frequency that marks the transition from individual, well-separated resonances to many overlapping normal modes. It is also known as the cut-off frequency of a diffused field, f_S, and can be calculated using the reverberation time (*T*) and room volume (*V*) given the equation of $f_s = 2000\sqrt{\frac{T}{V}}$.

Among the cases studied and described in Table 2.1, the "Schroeder frequency" of six spaces that were categorized as small, semi-small, and semi-large rooms were theoretically observed. The cut-off frequency of the three spaces with a volume above 1500 m^3 in Table 6.1 indicated that the frequency range being observed (i.e., speech or music with the spectrum shown in Figure 2.32 and frequency range of human voice and musical instrument in Figure 1.26) within the propagating sound waves created diffused sound fields. For smaller spaces (i.e., volume <350 m^3) the occurrences of diffused sound field are less expected for low frequencies up to the 125 Hz octave band.

Table 6.1. The room shape, volume, RT60, and Schroeder frequencies of the cases studied.

	ID	Room (Acoustical Function)	Wall Shape	Volume (m^3)	RT60 (average)	Schroeder frequency (f_S)
1.	AA21	Classroom R1221 Art&Architecture	Flat parallel	165	1.06	159.92
2.	DAS	Duderstadt Audio Studio	Uneven	266	0.41	78.52
3.	AA16	Classroom R2216 Art&Architecture	Flat parallel	332	0.97	103.67
4.	DH170	Lecture Hall 170 Dennison	Flat parallel	1513	0.81	46.28
5.	AHA	Lecture Hall A Angell Hall	Curve	1530	0.89	48.24
6.	DOH	Detroit Orchestra Hall	Curve	8895	1.13	22.54

The theoretical cut-off frequency of a diffused sound field (Table 6.1) was used to predict the critical distance and define distance between receivers and between source and receivers. These distances defined the boundary of the sound field observed as shown in Figure 2.1 for the computer simulations. Owing to the longer wavelengths, low frequencies are received by the microphones earlier before the high frequencies are detected. Therefore, propagation of the early reflections and late reflections is also frequency dependent. Observations for the early and late portions of the reflected sound define the characteristics of the sound field diffuseness. In order to support the audibility for listening to speech and music in a room, the degree of diffusion was observed within octave frequency bands.

Meanwhile, the majority of available acoustical treatment products are not designed to attend to the low frequency component, while sound engineers, on the other hand, are primarily concerned with the control of sound in low frequencies given the difficulty of enhancing the capabilities of the sound system within that region. Selection of diffusers, therefore, should take into account the characteristics for the entire frequency response and for the entire length of sound propagation.

Observations of the effect of design configurations to reduce the comb-filtering effect were used to evaluate the assumption by past researchers that diffusion primarily occurs at the early reflections. These design configurations of optimizing the impact of diffusion from diffusers on early reflections, however, only benefit spaces with parallel walls that are short distances apart and have homogenous absorption characteristics, a condition where comb-filtering is most likely to occur.

The close-by diffused early reflections created by the diffusers reduce comb-filtering and eliminate specular reflections, which then reduces the possibility of echo. Without applying asymmetric absorption on the parallel surfaces of a small space, the diffusers attached will increase reverberation, as proven by the results obtained from simulation in room 1221 of the Art and Architecture Building (AA21) in Figure 3.10, which is a small room with parallel walls. The increase in reverberation is unwanted for spaces with reverberation time already exceeding the ideal condition.

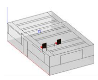
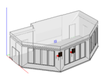
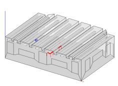
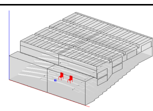
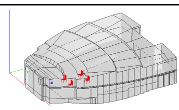
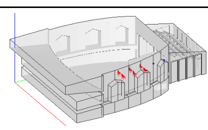
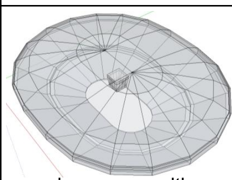
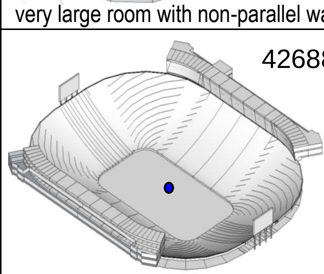
The assumption that tilting the diffuser panel creates more diffusion of early reflections was not demonstrated in the results of the current studies (i.e., using the Coh_{early} values for similarity of responses measured at receivers). The large cones, which jut from the surface of the particular Golden Acoustics panel observed, used for diffusing low frequencies, were seen as plane surfaces and became parallel to the walls as these diffusers were tilted. Furthermore, it can be concluded that the effectiveness of a diffuser to reduce comb-filtering of early reflections while controlling the increase in reverberation is also determined by the positioning of the diffuser.

If reduction of the comb-filtering effect was not shown during the observation of the early reflections, then the observation of the entire signal should be the next step. Observations of the entire signal, which include the late reflections, enable the identification of the actual role of a diffuser or architectural elements that behave as diffusers in creating a diffused sound field.

The coherence values for all measurements conducted in this research are within the range of 0.65 to 1, a range of difference in diffuseness of 35%. For Coh_{early}, the disparity is larger than Coh_{late}, especially for low to mid-frequencies, which is the frequency range of speech and most music pieces. The average coherences from octave bands of 63 to 8000 Hz, for all cases studied using computer simulation and field measurement are listed in Table 6.2.

The delta SPL (the total SPL minus the direct SPL) for octave bands of 125 Hz to 8000 Hz is used to evaluate the amount of reflection or absorption by diffusers as shown in Figure 6.1.

Table 6.2. Values of T30, Coh_{early}, Coh_{late} from computer simulation and field measurement of all cases studied.

	Description and Ratio of Volume	Frequency range	T30		Coh_{early}		Coh_{late}	
			CS	FM	CS	FM	CS	FM
AA21	1  small room with parallel walls	average	1.03	-	0.87	-	0.80	-
			1.20	-	0.83	-	0.81	-
		low to mid	1.22	-	0.89	-	0.81	-
			1.40	-	0.85	-	0.83	-
		mid- to high	0.73	-	0.84	-	0.79	-
			0.89	-	0.78	-	0.78	-
DAS-Close	1.6  small room with non-parallel walls	average	0.33	-	0.81	-	0.81	-
			0.31	0.27	0.83	0.85	0.83	0.86
		low to mid	0.37	-	0.83	-	0.86	-
			0.38	0.28	0.81	0.88	0.81	0.79
		mid- to high	0.22		0.82		0.79	
			0.23	0.27	0.80	0.90	0.8	0.79
AA16	2  semi-small room with parallel walls	average	0.96	-	0.84	-	0.77	-
			0.79	0.50	0.85	0.88	0.80	0.83
		low to mid	1.14	-	0.86	-	0.76	-
			0.91	0.51	0.87	0.90	0.81	0.87
		mid- to high	0.82	-	0.81	-	0.78	-
			0.69	0.46	0.82	0.82	0.79	0.78
DH170	9.2  semi-large room with parallel walls	average	1.06	0.72	0.90	0.84	0.81	0.82
			1.09	-	0.90	-	0.81	-
		low to mid	1.25	0.75	0.90	0.87	0.81	0.83
			1.27	-	0.91	-	0.81	-
		mid- to high	0.88	0.7	0.90	0.81	0.81	0.79
			0.88	-	0.90	-	0.80	-
AHA	9.3  semi-large room with non-parallel walls	average	1.34	-	0.89	-	0.82	-
			1.33	-	0.88	-	0.81	-
		low to mid	1.52	-	0.88	-	0.82	-
			1.48	-	0.87	-	0.80	-
		mid- to high	1.16	-	0.90	-	0.82	-
			1.19	-	0.90	-	0.81	-
DOH	53.8  large room with semi-parallel walls	average	1.42	1.55	0.94	0.84	0.81	0.82
			1.42	-	0.91	-	0.79	-
		low to mid	1.69	1.67	0.95	0.87	0.83	0.85
			1.68	-	0.90	-	0.80	-
		mid- to high	1.16	1.45	0.94	0.80	0.79	0.78
			1.15	-	0.92	-	0.79	-
CRI	32350  very large room with non-parallel walls	average	0.90	-	0.9	-	0.9	-
		low to mid	1.63	-	0.96	-	0.88	-
		mid- to high	0.69	-	0.96	-	0.93	-
BH	426880  very large space with non-parallel walls	average	-	3.3	-	0.93	-	0.86
		low to mid	-	3.92	-	0.95	-	0.89
		mid- to high	-	2.83	-	0.91	-	0.81

NOTE: CS = Computer Simulation
FM = Field Measurement

Without diffuser | With diffuser

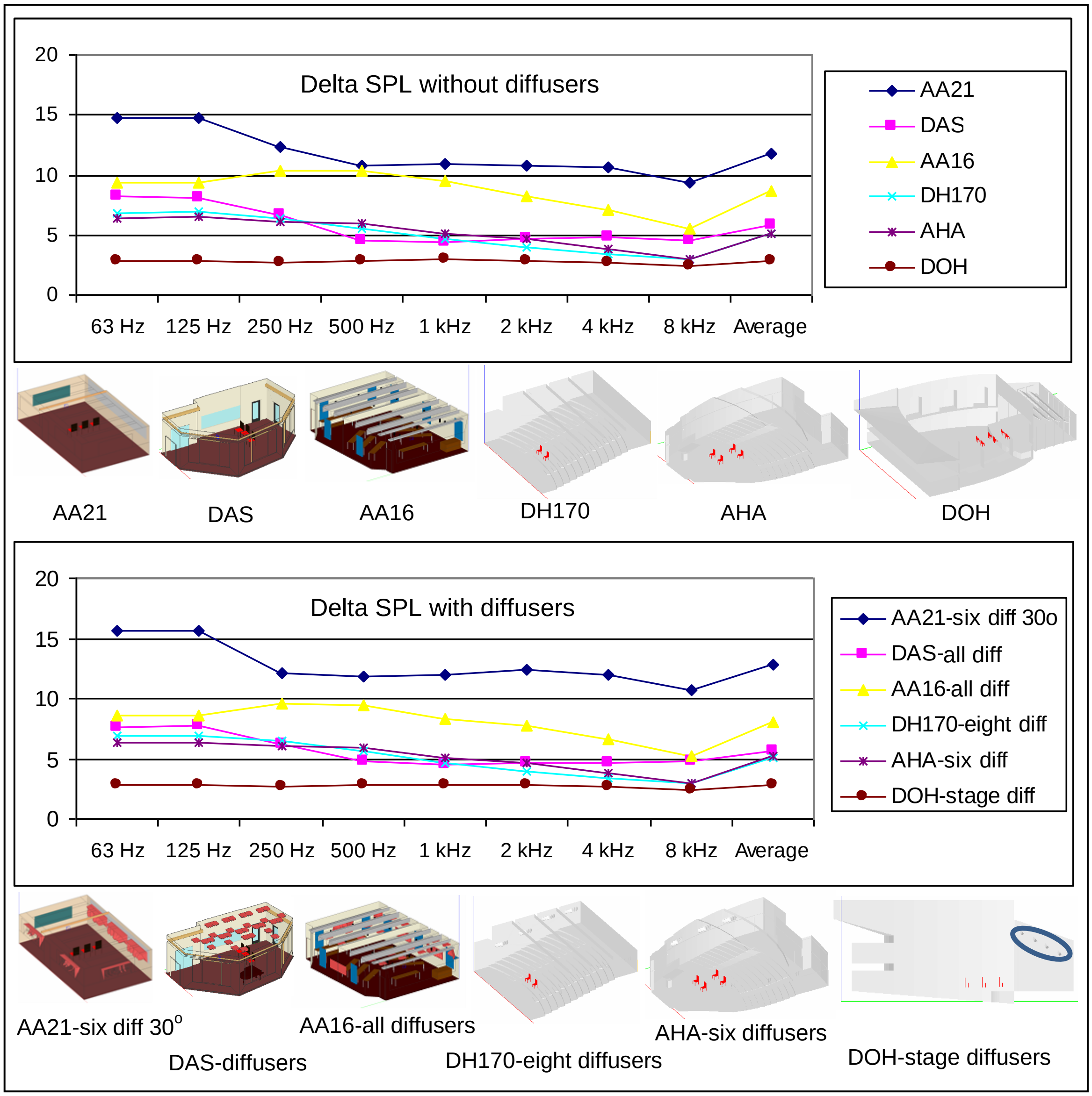


Figure 6.1. Comparison of the total SPL of reflected sound within the cases studied for simulated spaces without and with the diffusers applied.

An increase of the delta SPL is a first indication that there are more reflections. However, it does not provide any information concerning the reflection directivity, whether it is diffused or not.

Details of the related component (i.e., the diffusers in Figure 2.11) that contributed to the reflections within each computer model observed in Figure 6.1 are listed in Table 6.3. It is indicated that the diffusers managed to change the delta SPL values for the spaces where the ratio of the diffuser surface area to the total surface area is ≥ 0.10.

Table 6.3. Ratio of diffusers' surface to total surface area within the cases studied.

	ID	Diffusers applied	Volume without diffusers (m^3)	Volume with diffusers (m^3)	Ratio of diffuser surface to total surface area
1.	AA21	6 of diffuser no.1 tilted 30^o	165.27	161.01	0.14
2.	DAS	32 of diffuser no. 5 (skyline diffusers)	264.89	264.59	0.08
3.	AA16	3 of diffuser no.2 1 of diffuser no.1 2 of diffuser no.4	362.50	360.67	0.10
4.	DH170	8 of diffuser no.1	1511.64	1513.09	0.05
5.	AHA	6 of diffuser no.1	1529.70	1528.37	0.02
6.	DOH	16 of diffuser no.1	8891.64	8894.63	0.02

It is not necessary to conduct an analysis of sound-field diffuseness in every architectural space by using all the objective parameters described within this research. Results from the current studies demonstrated that some spaces already possessed diffused sound fields, given the high values of coherences of late reflections (Coh_{late}) throughout the entire impulse response. This condition is expected to occur in large spaces. Once it is determined that the coherence values do not indicate any difference for different design configurations, observations rely on EDT and T_{30} to predict the amount of absorption and reflection on early and late portions of the energy decay, respectively. If one of these parameters increases, observations of the clarity index enable the depiction of which portion of the sound decay is actually receiving more reflections due to diffusion.

Quantification of the diffuseness using Coh_{early} and Coh_{late} is limited to the pair of receivers, which defined the sound field observed. For further investigation, using SPL mapping on surfaces at early and late portions of the sound decay will enable the identification of their contribution to the diffusion. It provides the ability to observe in more detail the actual directionality and distribution of the diffusion.

Values of SPL, clarity index, and reverberation time were observed subjectively. These were the selected subjective parameters that can be quantified with a corresponding objective parameter for speech and music. The attempt to provide speech intelligibility is impacted more by the diffusion than the effort to create a good acoustic condition for music, given the narrow band for speech content. The human hearing system and audibility perception were also considered during the experimental setup and analysis of results obtained in the subjective evaluation. Through an understanding of the human hearing system itself, it can be concluded that the use of diffusion for frequencies above 5000 Hz becomes less important, since the scattering from ear pinna, head, and upper torso already supports audibility for these high frequencies.

Subjective evaluation in the current studies focused on comparing the audibility of one receiver to another receiver position within the sound field of interest. These audibility conditions are sensitive to early reflections proven by the correlation between changes in the Coh_{early} values with noticeable differences in loudness, clarity, and liveliness perception provided in Chapter 4. In the condition of a diffused sound field, the loudness perception remained the same. The audibility condition in general remained the same as the diffusion was altered.

The LEV_{calc} proposed by Beranek is shown to be effective as a predictor of a diffused condition, if the change of diffusion is observed alone while the amount of absorption is maintained. In practice, this condition is hardly achieved since architectural elements that create a diffused sound field may also contribute to absorption. Therefore, the effectiveness of the use of LEV_{calc} as a diffuse sound-field index depends on the material characteristics.

Values of all the parameters generated from the impulse responses of computer simulation of all cases studied are tabulated as a summary in Table 6.4.

For room acoustics design application purposes, it is important to consider the type of the acoustical function of a space in order for the diffusion control to satisfy the acoustical demand. For instance, the procedure for modeling and the steps of observation are different for spaces considered to have multiple acoustic zones as opposed to a single acoustic zone.

Table 6.4. Values of all parameters generated from impulse responses of computer simulation of all cases studied.

Average values of all frequencies	AA21		DAS-Close		AA16		DH170		AHA		DOH	
	None	Diffuser	None	Diffuser	None	Diffuser	None	Diffuser	None	Diffuser	None	Diffuser
Delta SPL	11.80	12.79	5.75	5.63	8.71	8.01	5.08	5.12	5.17	5.17	2.75	2.84
T30	1.03	1.20	0.33	0.31	0.96	0.79	1.06	1.09	1.34	1.33	1.42	1.42
EDT	1.00	1.18	0.34	0.31	0.87	0.73	0.81	0.86	1.17	1.12	0.80	0.82
C50	0.97	-0.18	12.60	12.55	2.19	3.39	5.33	5.03	4.30	4.35	8.23	8.03
C80	4.27	2.84	18.41	18.56	5.54	6.96	7.81	7.41	5.57	5.69	10.90	10.64
Cohearly	0.87	0.81	0.81	0.83	0.84	0.85	0.90	0.90	0.89	0.88	0.91	0.94
Cohlate	0.80	0.83	0.81	0.83	0.77	0.80	0.81	0.81	0.82	0.81	0.79	0.81
LEVcalc	3.28	3.79	-0.23	-0.28	3.53	2.42	0.30	0.28	-0.50	-0.53	-0.17	-0.13

6.2 Contributions

This study presents an integrated methodology in the room acoustics design process that enables the characterization of the room acoustics condition resulting from diffusion control at any stage in the design process. A single impulse measurement on multiple microphones or receivers, both with the use of an Acoustic Camera and computer simulation, facilitates the evaluation and quantification of the diffusion occurrences within real and simulated spaces.

Based on the findings, certain positions and orientations of diffuser or diffusive-like surfaces that can obtain an effective design solution with diffusion control are identified as listed in Table 6.2. In a small space, the sound field was obviously more sensitive to changes in the boundary properties. Furthermore, changes in the boundary shape and acoustical properties had a greater effect on the late reflection than on the early reflections if absorption and diffusion were applied simultaneously, such as in the Duderstadt Audio Studio (DAS). Therefore, non-parallel walls within a small room are capable of improving the diffuseness at low frequencies as seen for coherence of early reflections (Coh_{early}) of room DAS (i.e., non-parallel walls) in Figure 3.27 with the largest Coh_{early} values.

Related to the finding above, in rectangular spaces, the application of diffusers not only creates diffusion, but selectively contributes to room absorption for frequencies that are sensitive to the distances between the parallel walls. In non-rectangular rooms, especially in rooms with curved walls, the use of diffusers to create diffusion can be optimized with the curved walls. This conclusion was drawn from the findings based on observations in room 2216 in the Art and Architecture Building (AA16), Dennison Hall room 170 (DH), and in Angell Hall Auditorium A (AHA).

Table 6.5. Position and orientation of diffuser that provides an effective design solution in a variety of geometrical room shapes.

Geometrical properties	**Positioning of diffusers to create diffusion of early reflections**	**Positioning of diffusers to create diffusion of late reflections**
Small rooms with parallel walls	- On the parallel surfaces with the smallest dimension ratio (width-to-length or height-to-length ratio)	- On parallel surfaces with a sufficient amount of inhomogeneous absorption.
Small rooms with non-parallel walls	- On the parallel surfaces with absorption applied on non-parallel surfaces	- On the parallel surfaces with the smallest dimension ratio (width-to-length or height-to-length ratio)
Semi-small rooms with parallel walls	- On the parallel surfaces with the smallest dimension ratio (width-to-length or height-to-length ratio) - On surfaces that are close to the source position	- On the parallel surfaces with the smallest dimension ratio (width-to-length or height-to-length ratio) - The close-by diffusers are tilted to a certain angle
Semi-large rooms with parallel walls	- On surfaces that are close to the source position	- On the parallel surfaces with the smallest dimension ratio (width-to-length or height-to-length ratio) with different heights from the floor and tilted
Semi-large rooms with non-parallel walls	- On surfaces that are close to the source position	- On the non-parallel surfaces with absorption that maintains the signal-to-noise ratio
Large rooms with semi-parallel walls	- On surfaces that are close to the source position with inhomogeneous absorption	- No impact
Very large rooms with curved walls	- On surfaces that are close to the source position	- No impact

Evaluation of the effectiveness of architectural elements to absorb sound energy can be conducted along with the evaluation of the ability to distribute equally in all directions the reflected portion of the propagating sound. This contributes to the methods for room acoustics design, which in the past, were based on reverberation as the main design solution emphasized.

Selecting the most sensitive indicators to measure the sound-field diffuseness and to characterize its audibility is one of the important achievements of this study. As mentioned earlier, the process included literature review of parameters (see Figure 1.3), review of standard measurements in room acoustics, and preliminary research on subjective assessment. The selected parameters are listed in Table 1.1.

The research provided a clear relationship between objective and subjective parameters in order to characterize the diffuseness using selected parameters that are widely used in the research of room acoustics. For instance, this study explored the effectiveness of LEV_{calc} as a diffuse sound-field index. This led to the ability to provide indices for characterizing a sound field with diffusion. In section 2.1, further discussion as to the logic of using these objective parameters and subjective attributes is provided.

An important contribution of this study was the identification of the objective parameters and the subjective attributes that can characterize the diffuseness in a space, which then allowed the exploration of a variety of new methods for subjective assessment in room acoustics. This included the use of Cave Automatic Virtual Environment (CAVE) system capabilities with the ability of auralization to synthesize a virtual source inside a virtual space and the use of Web-survey with embedded audio stimuli.

More understanding of the relationships between the subjective attributes that describe auditory perception with noticeable differences in the audibility (i.e., audible quality) can help to identify the architectural elements that are responsible for creating the acoustical condition, given the capabilities of simulation and auralization. Given the findings, a brief description on the effect of position and orientation of diffusers in a variety of geometrical room shapes is described in Table 6.1. This allows designers to identify architectural elements, including diffusers, that most effectively impact the room acoustics characteristics during the early stages of the design process.

The findings that led to identification of specific principles for guidelines in room acoustics design presented in Chapter 5 provide the research outcomes that can assist architects in creating a better auditory space. Implementing the principles into a real design practice can be accomplished by understanding the acoustical function of the space observed. One then has the ability to control excessive reverberation along with the ability to manipulate the sound path using diffusion control based on the identification of the sound field characteristics. This study served this purpose well, since it observed spaces with a variety of acoustical functions: a recording studio, classroom, auditorium, concert hall, and sport facilities.

The detailed description of the techniques used within the integrated methodology provides the opportunity for further studies to develop instrumentation for investigating sound-field diffuseness. The use of the Acoustic Camera with the multiple microphones and the delay-and-sum beamforming method for measurement in enclosed spaces is still new. This research has initiated the development of this instrument, with the related data acquisition and signal processing tools, to better serve the purpose of room acoustics research.

6.3 Recommendation and Future Work

Quantifying and characterizing the diffuseness of a sound field requires the use of a multi-microphone system to observe the directivity of sound reflections. Therefore, there is a need for the availability of an affordable and sensitive acoustic sensor with the signal processing method that can be used in a variety of room dimensions.

With the rapid development of techniques to measure diffusion, focus will no longer be on the efficiency of diffuser performance, but rather on the impact of the room geometry and architectural properties. As the use of diffusion control becomes an important room acoustics design solution, it will also be important to develop or revisit new parameters and indices, which in the past were based on absorption control. Evaluation of the acoustical quality of a room, given the design results, is as important as the evaluation of the acoustical treatments' performance. Other parameters that might show stronger evidence of diffuseness should be explored in future work.

Cases studied within this research have investigated a diversity of room shapes and sizes as well as representations of unique architectural configurations and acoustical functions. Available databases of more cases should be considered in future studies to improve the statistical significance of the results. Each edifice with a unique acoustical function, such as concert halls of various sizes and shapes, should be investigated using several halls as case studies for each category of size or shape. An ideal condition would be if future cases studied had a large number of similarities in their architectural properties. This would enable the research to have a greater focus on a particular variable, such as room size for a larger number of rectangular rooms.

Several techniques of subjective assessment demonstrated within this research provide new possibilities to improve subjective evaluation methods in room acoustics and psychoacoustics. The use of Web-survey provides the ability to engage a large number of subjects using a single experimental setup. Further study to reduce the possibilities of measurement errors and to solve technical problems related to audio streaming through the Internet is needed. An immersive virtual environment is also shown to be a promising technique for subjective assessment. Compared to studies in real spaces, this approach benefits from having the ability to control environmental variables, repeatability, and to prevent subjects from being exposed to hazardous conditions from the uncontrolled stimuli within real spaces. Further study is required in the development of auralization techniques in order to create a more realistic virtual source with an improved real-time immersive experience within the virtual space.

Appendix A. Questionnaire of the On-Site Subjective Assessment

The questionnaire for the preliminary research of the on-site subjective assessment is shown below.

PART I. (21 Questions)

1/21 How would you rate the size of the space where the sound is being delivered, based on what you heard?

Small	1	2	3	4	5	6	7	Large

2/21 How clearly can you distinguish the sound of one instrument from another?

Blurred	1	2	3	4	5	6	7	Clear

3/21 Please rate the loudness of the overall sound.

Weak	1	2	3	4	5	6	7	Loud

4/21 How much does the loudness fluctuate?

Very little	1	2	3	4	5	6	7	A lot

5/21 How would you rate the quality of the ensemble (the musicians starts and ends the note together)?

Poorly	1	2	3	4	5	6	7	Excellent

Please rate these following conditions related with reverberation.

6/21 **Reverberation** or liveness of mid-frequency sounds

Dead	1	2	3	4	5	6	7	Live

7/21 **Relative liveness of bass** or longer duration of reverberation at low compared to mid-and high frequencies

Cold	1	2	3	4	5	6	7	Warmth

8/21 **Relative liveness of treble** or longer duration of reverberation at high compared to mid-and low frequencies

Brilliant	1	2	3	4	5	6	7	Dull

9/21 To what degree does the room produces echoes?

No echoes	1	2	3	4	5	6	7	Very echo

10/21 How do you perceive the distribution of the sound in the room when it reaches you?

One direction	1	2	3	4	5	6	7	Many direction

11/21 Balance indicates if there is no instrument being dominant throughout the whole music. How would you rate the balance quality?

Poor	1	2	3	4	5	6	7	Excellent

12/21 How would you rate the dynamic range of the sound?

Small differences	1	2	3	4	5	6	7	Large differences

Dynamic Range: Differences between the loudest sound (fortissimo) that can be produced in the hall to the quietest sound (pianissimo) that can be heard.

13/21 How would you rate the tonal quality of the sound?

Poor	1	2	3	4	5	6	7	Excellent

14/21 How would you rate the ambient noise of the room?

Very noisy	1	2	3	4	5	6	7	Inaudible

15/21 **Overall,** how is the acoustics quality of this room?

Poor	1	2	3	4	5	6	7	Excellent

16/21 How would you rate the acoustic quality of the room compared to other typical classrooms/lecture halls?

Worse	1	2	3	4	5	6	7	Better

GENERAL REMARKS

Do you feel that these room factors influence the acoustics of this room?

		Yes	No
17/21	light	1	2
18/21	temperature	1	2
19/21	humidity	1	2
20/21	room size	1	2
21/21	interior layout	1	2

PART II. (11 questions)

1/11 Gender :

Male	Female
1	2

2/11 Age :

15-20	21-25	26-30	>30
1	2	3	4

3/11 Can you play a musical instrument? **(if *No*, go to number 8/11)**

Yes	No
1	2

What musical instruments do you play?

4/11 Primary : …………………

5/11 How well do you play this musical instrument?

Beginner	Moderate	Good	Advanced
1	2	3	4

6/11 Other : ....

7/11 How well do you play this musical instrument?

Beginner	Moderate	Good	Advanced
1	2	3	4

8/11 Do you play an instrument or sing in a music group?

Yes	No
1	2

9/11 What is your level of understanding about ***sound***? (e.g. frequency, reverberation, acoustic)

Very little	Moderate	Advanced
1	2	3

10/11 How do you consider a ***good*** sound quality in a room is important to enhance a ***good*** music performance?

Not Important	1	2	3	4	5	6	7	Important

11/11 Please mark your listening position.

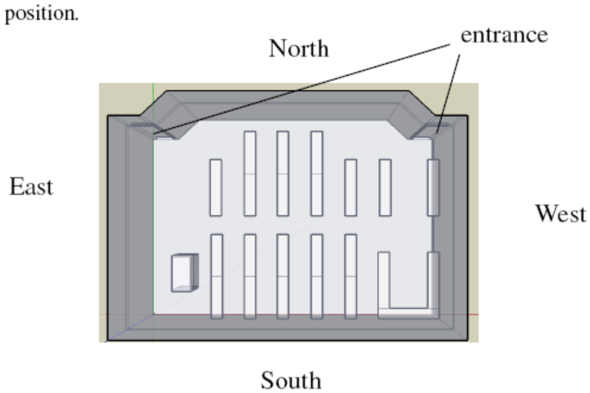
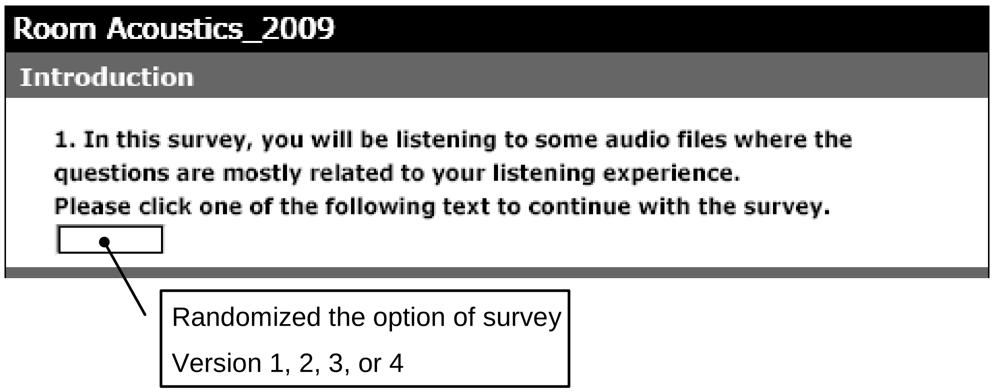
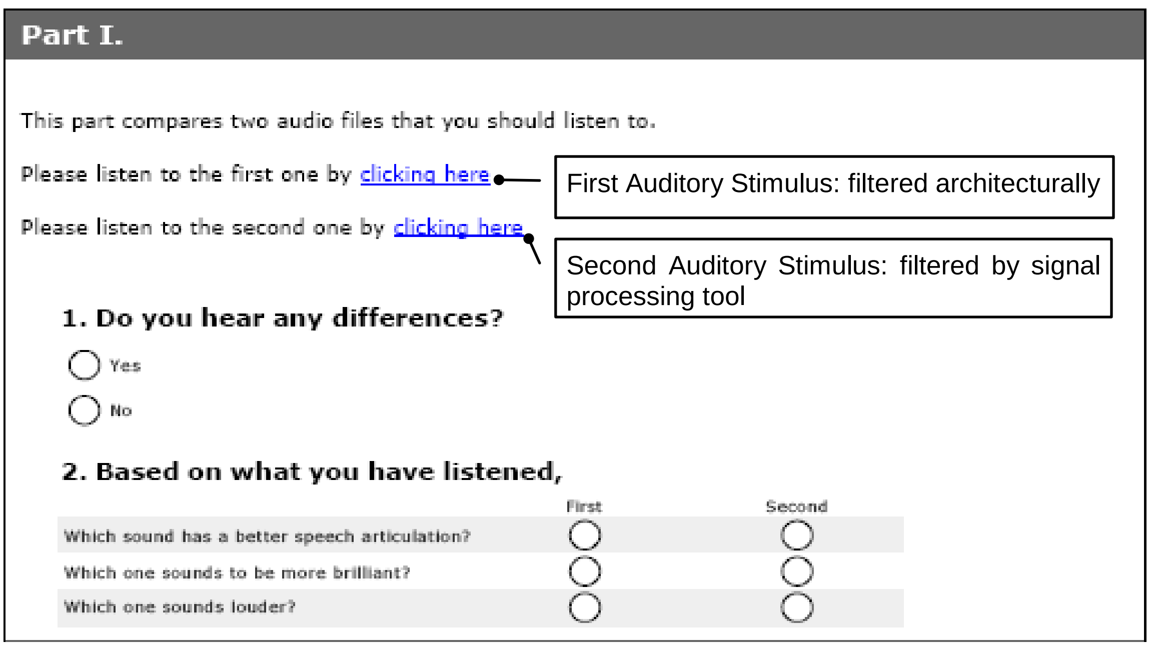


Figure 1. Top View of Room 2216 TCAUP

Appendix B. Questionnaire of Subjective Assessment using Web-Survey





Speech passage of the auditory stimuli in part 1:

"Sound, it's not just the air vibrating, sound means feeling. It refreshes our minds, soothes our heart. It can make us happy or sad or excited. It also is essential to the communication of ideas and the exchange of information. It is vital to daily life."

Part II.

This part concerns a passage that you should listen to.

Please do this by clicking here

Third auditory stimulus: filtered architecturally, different speech content, a female speaker

1. Please answer the following questions.

	True	False
Infinitely many numbers in arithmetic can only be composed by many digits.	◯	◯
The decimal system is an Arabic numbering system.	◯	◯
The decimal system has ten numerals, the digits zero, one, two, three, four, five, six, seven, eight, nine, and ten.	◯	◯
The decimal is actually a base five notation	◯	◯
The first country in Europe that used decimal notation in monetary system was Belgium	◯	◯
Italy and Switzerland started to use base ten notation in their monetary systems in 1865	◯	◯
Infinitely many numbers can be composed from just a few digits, with the help of the symbol, zero, the principal of positions and the concept of base.	◯	◯

True and False Questions based on the third stimulus

2. How would you rate the size of the space where the speech is being delivered, given your hearing impression?

	Small						Large
Hearing Space	◯	◯	◯	◯	◯	◯	◯

3. How clearly was each word, was articulated in the speech?

	Blurred						Clear
Clarity	◯	◯	◯	◯	◯	◯	◯

4. How loud is the overall speech?

	Quiet						Loud
Loudness	◯	◯	◯	◯	◯	◯	◯

5. How much does the loudness of the speech fluctuate?

	Very little						A lot
Loudness fluctuation	◯	◯	◯	◯	◯	◯	◯

6. How well can you hear the individual words in the speech?

	Poorly						Excellent
Ability to hear words	◯	◯	◯	◯	◯	◯	◯

7. To what degree does the speech produces echoes?

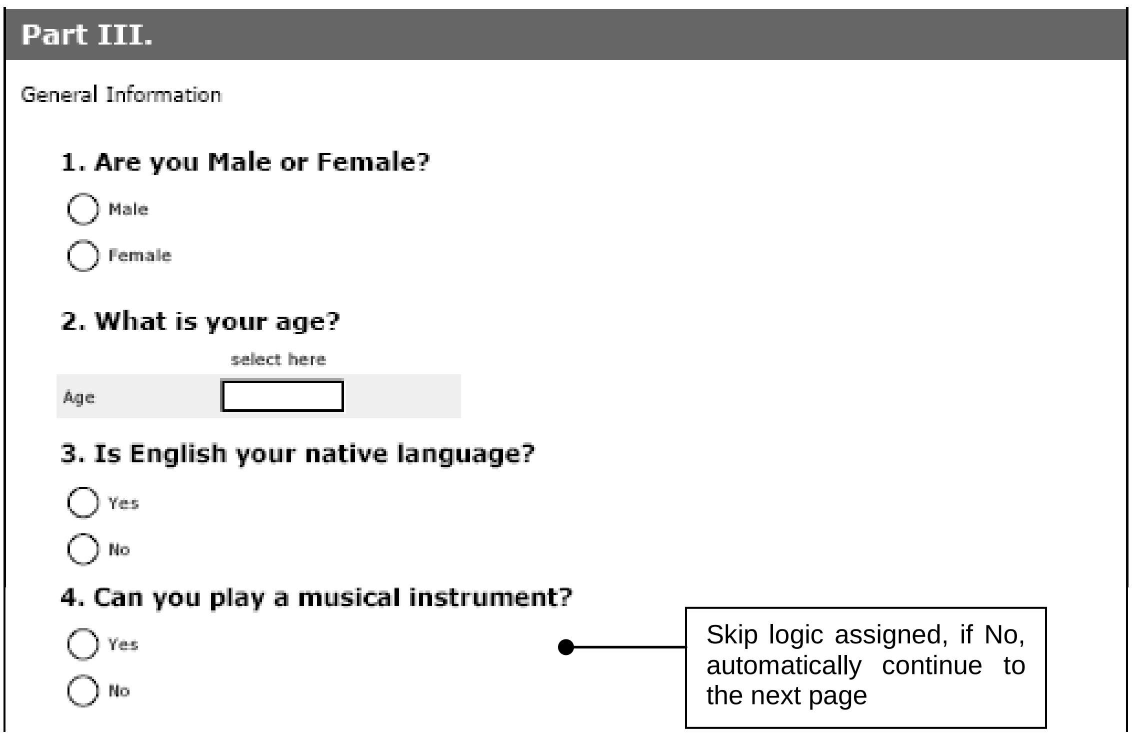
	No echoes						A lot of echoes
Echoes	◯	◯	◯	◯	◯	◯	◯

8. Overall, how would you rate the acoustic quality of the space where the speech is being delivered?

	Poor						Excellent
Acoustic quality	◯	◯	◯	◯	◯	◯	◯

Speech passage of the auditory stimuli in part 2:

“In language, infinitely many words can be written with a small set of letters. In arithmetic, infinitely many numbers can be composed from just a few digits, with the help of the symbol, zero, the principal of positions and the concept of base. Pure systems with base 5 and 6 are set to be very rare. But base 16 occurs in English when we use score, as in 4 score and seven. Eventually, no system could keep phase with the decimal or Arabic numbering system which has ten numerals, the digits zero, one, two, three, four, five, six, seven, eight, nine, and a decimal point. The numerals take different place values, depending on positions, so the number 819.65 can be shown by (8 x 102) + (1 x 101) + (9 x 100) + (6 x 10-1) + (1 x 10-2). Monetary systems have evolved, to make use of this base ten notation. France became the first country in Europe in 1799, joined by Belgium, Italy and Switzerland in 1865. Germany’s decision followed 8 years later and the Scandinavian and states in Russia, changed in 1875.”


Part III.

General Information

1. Are you Male or Female?

- ◯ Male
- ◯ Female

2. What is your age?

select here

Age [ ]

3. Is English your native language?

- ◯ Yes
- ◯ No

4. Can you play a musical instrument?

- ◯ Yes
- ◯ No

Skip logic assigned, if No, automatically continue to the next page

1. What musical instruments do you play?

2. How well do you play this musical instrument?

- ◯ Beginner
- ◯ Moderate
- ◯ Good
- ◯ Advanced

3. What other musical instruments do you play?

4. How well do you play this musical instrument?

- ◯ Beginner
- ◯ Moderate
- ◯ Good
- ◯ Advanced

5. Do you play an instrument in a music group?

- ◯ Yes
- ◯ No

1. Do you sing in a musical group?

- ◯ Yes
- ◯ No

2. How much do you understand about sound? (e.g. frequency, reverberation, acoustic)

- ◯ Some
- ◯ Moderate
- ◯ A lot

3. What is the audio device you used in completing this questionnaire?

- ◯ Headsets
- ◯ Built in computer Speakers
- ◯ Desk speakers
- ◯ Room Speakers (home theater)
- ◯ Other

4. How often do you listen to loud music with headsets in a week?

- ◯ <5 hours
- ◯ 5 - 15 hours
- ◯ 16 - 25 hours
- ◯ >26 hours

5. Have you ever had a hearing problem?

- ◯ Yes
- ◯ No

6. How important is a good sound quality in a room for understanding speech?

	Not important						Very important
Importance	◯	◯	◯	◯	◯	◯	◯

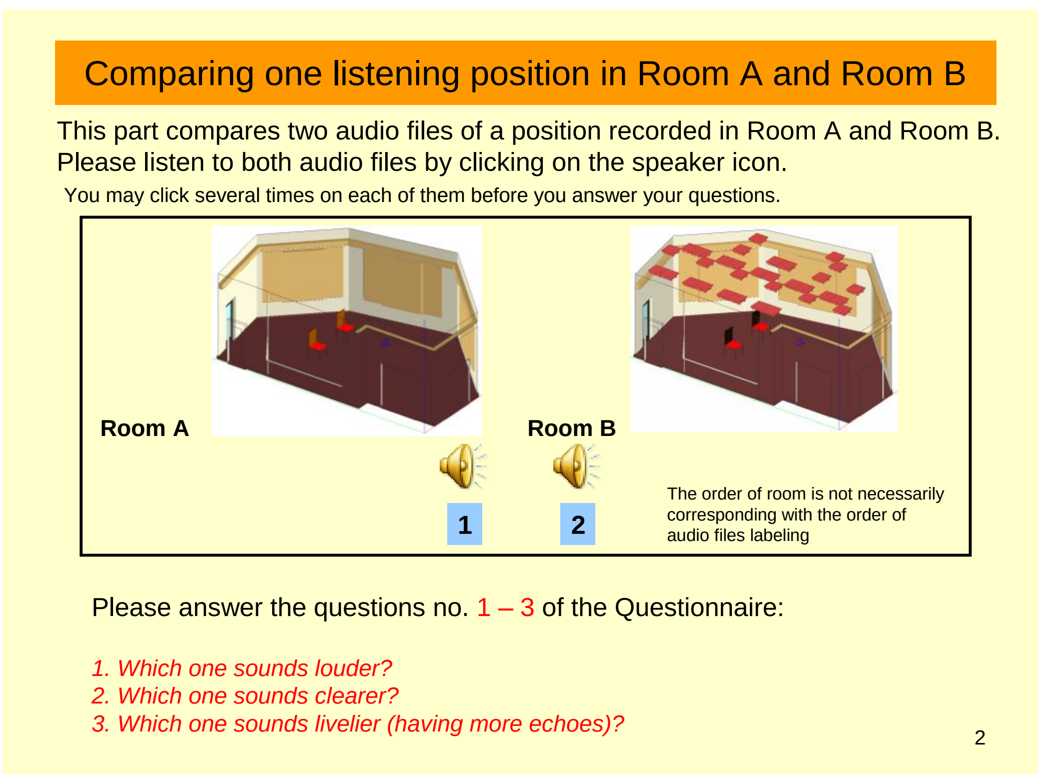
Appendix C. Questionnaire of Subjective Assessment using Slide Presentation

Slide no. 1

The survey instruction is provided in Slide no. 1 with the following text:

> *In this survey, you will be listening to some audio files. Questions are related to your listening experience.*
>
> *Please follow the steps within this presentation and use the printed questionnaire to provide the answers.*

Slide no. 2

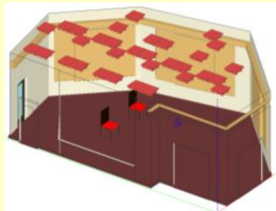


Slide no. 3

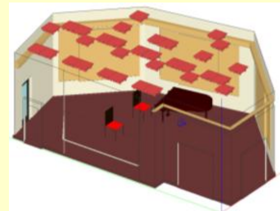
Comparing one listening position in Room A and Room B

This part compares two audio files of a position recorded in Room A and Room B.
Please listen to both audio files by clicking on the speaker icon.

Room A



Room B



1 **2**

The order of room labeling is not necessarily corresponding with the order of audio files labeling

Please answer the questions no. 4 of the Questionnaire:

4. Which sound indicates better as if it is coming from your left?

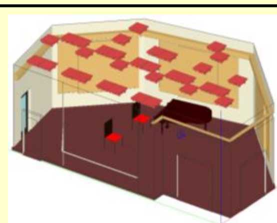
3

Slide no. 4

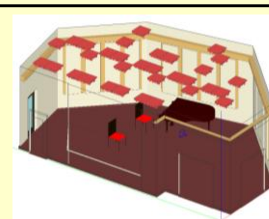
Comparing one listening position in Room A and Room B

This part compares two audio files of a position recorded in Room A and Room B.
Please listen to both audio files by clicking on the speaker icon.

Room A



Room B



1 **2**

The order of room labeling is not necessarily corresponding with the order of audio files labeling

Please answer the questions no. 5 – 7 of the Questionnaire:

5. Which one sounds louder?
6. Which one sounds clearer?
7. Which one sounds livelier (having more echoes)?

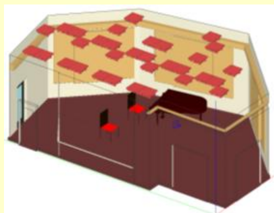
4

Slide no. 5

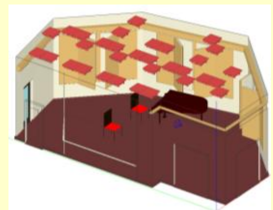
Comparing one listening position in Room A and Room B

This part compares two audio files of a position recorded in Room A and Room B.
Please listen to both audio files by clicking on the speaker icon.

Room A



Room B



1 **2**

The order of room labeling is not necessarily corresponding with the order of audio files labeling

Please answer the questions no. 8 – 10 of the Questionnaire:

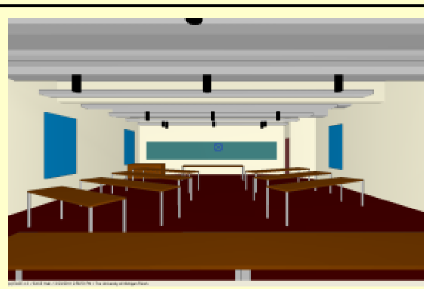
8. Which one sounds louder?
9. Which one sounds clearer?
10. Which one sounds livelier (having more echoes)?

5

Slide 6

Comparing two listening positions in Room A

This part compares two audio files of two different positions in Room A.
Please listen to both audio files by clicking on the speaker icon.
You may click several times on each of them before you answer your questions.



Room A

1 **2**

Please answer the question no. 11 of the Questionnaire.

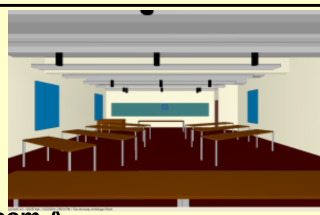
11. Which sound do you hear is coming from your left?

6

Slide 7

Comparing one listening position in Room A and Room B

This part compares two audio files of a position recorded in Room A and Room B. Please listen to both audio files by clicking on the speaker icon.



Room A

Room B

1 2

The order of room labeling is not necessarily corresponding with the order of audio files labeling

Please answer the questions no. 12 – 14 of the Questionnaire:

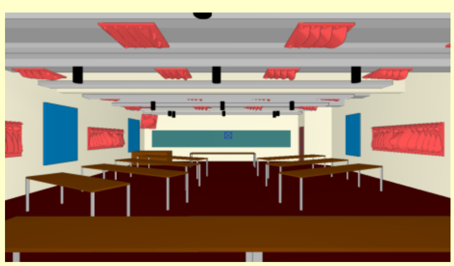
12. Which one sounds louder?
13. Which one sounds clearer?
14. Which one sounds livelier (having more echoes)?

7

Slide 8

Comparing two listening positions in Room B

This part compares two audio files of two different positions in Room B. Please listen to both audio files by clicking on the speaker icon.



Room B

1 2

Please answer the questions no. 15 of the Questionnaire:

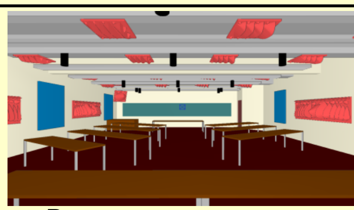
15. Which sound do you hear is coming from your right?

8

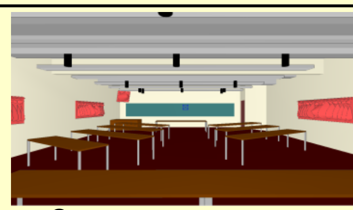
Slide 9

Comparing one listening position in Room B and Room C

This part compares two audio files of a position recorded in Room A and Room B. Please listen to both audio files by clicking on the speaker icon.



Room B



Room C

1 **2**

The order of room labeling is not necessarily corresponding with the order of audio files labeling

Please answer the questions no. 16 – 18 of the Questionnaire:

16. Which one sounds louder?
17. Which one sounds clearer?
18. Which one sounds livelier (having more echoes)?

9

Slide 10

Identifying listening position in Room B

Please listen to this audio file

It corresponds with the listening position at seat no.1 shown in the figure below.

Please answer the questions no. 19 of the Questionnaire after listening to this next audio file

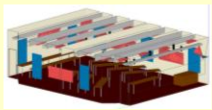
19. Which seat number do you think the sound is being heard from?

10

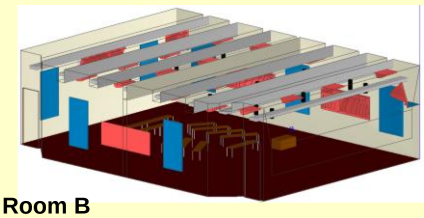
Slide 11

Identifying large and small spaces

This part compares two audio files of a position recorded in Room A and Room B. Please listen to both audio files by clicking on the speaker icon.



Room A



Room B

1

2

The order of room labeling is not necessarily corresponding with the order of audio files labeling

Please answer the questions no. 20 of the Questionnaire:

20. Which sound do you hear is coming from the larger size room?

11

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