SOUNDFIELD SIMULATION

The prediction and validation of acoustical behavior with computer models

by

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Licenciado en Arquitectura
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Submitted to the Department of Architecture in partial fulfillment of the requirements for the degree of Master of Science in Architecture Studies

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ABSTRACT

In the past, acoustical consultants could only try to convince the client/architect that with calculations and geometrical plots they could create an acoustically superb space. Now, by modeling the significant acoustical parameters of a design, we can preview a proposed acoustical solution and it is possible to identify the objective parameters that correspond to certain subjective reactions experienced by listeners. The results of a simulation can be presented not only for the eyes but also for the ears.

This document explains the basics behind acoustic computer simulation. It includes case studies that analyze and validate numerical parameters and create a sound simulation of a space that allows the listener to subjectively "grade" the acoustical qualities. It includes details on how human hearing uses several techniques to localize sound sources, how we can simulate factors that influence human auditory perception with computer software, and how we can reproduce the listening experience for a space that has not been built. The simulation techniques offer the possibility to use the ears and listen to the acoustics of a room during the design process. Several acoustic problems can be detected by the ears, whereas they may be difficult to express with a parameter that can only be calculated. Using these tools the acoustician can communicate the acoustic consequences of a design to the client/architect effectively. This technique can be used very early in the project to achieve exceptional results.

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My greatest debt to The Organization of American States and Melinda Lee for believing in me and sponsoring my studies at MIT.

I would like to express my deepest gratitude to my mother for always encouraging me to follow my dreams. To Mona, my soul mate, for being my inspiration and tolerating my endless talk about sound.

And finally, as I once read on a book by Ian Deary, "An author must always have an audience in his mind's eye" and mine was focused on my amazing father who introduced me to the fascination of acoustics and mentored me in this and many other fields. He has been a true role model and my best friend.

To all of the above, I owe my deepest thanks.

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Acoustical Prediction Software

1.0 Introduction

This thesis will explain the basics of computer modeling as well as investigate and discuss comparisons between measured and calculated acoustic parameters. This document will show how a computer model can be the most important tool of the acoustician.

Chapter 1 will present the basics on how the computer models are generated and the steps the software takes to generate an acoustical prediction. At the end, an example will show that if the given guidelines are followed, the computer model can produce far more accurate results than any of the former acoustical prediction tools.

1.1 A New Acoustic Prediction Tool

Acoustical design was once considered as an art form, solely based on the acoustician's experienced and intuition. In 1895 W.C. Sabine took the first step in quantitatively evaluating the acoustical qualities of concert halls [Egan 1988]. Sabine was the first to bring quantifiable measures to reverberation, absorption, and sound transmittance. The Sabine equation for reverberation (the time in seconds that it takes for a sound to reduce in sound pressure level by 60dB after the sound source has been silenced) has been the most important quantitative tool in architectural acoustics for the last 100 years, predicting how "dry" or "reverberant" a room is.

Determining the acoustics in advance has changed from an art into a controlled and precise design process. The traditional tool in acoustic design was either formulae or the physical scale model, but with the growth in computer power, numerical models started to take over for scale models by the end of the 1980s [Lahti and Möller, 1996].

During the last few years, computer modeling has matured from just being a supplemental tool to become not only a full substitute but actually a superior design method. Building a computer model is considerably faster, and introducing version changes is reasonably easy [Lahti and Möller, 1996]. Once the model is established, measurements over an entire area and subsequent analysis take a relatively short time.

One of many important advantages of computer models over scale models is the ability to calculate several acoustical parameters rapidly. The complete hall can be built in a CAD program, computed and analyzed in under 12 hours. Also, the results can be visualized and analyzed much better because the computer model contains more information than a set of measurements done in a scale model by tiny microphones. Today, computer models have become reliable and efficient design tools for the acoustic consultant and "the results of a simulation can be presented not only for the eyes but also for the ears with techniques for auralization" [Rindel 2000].

1.2 Background on acoustical prediction software and related studies.

The search for a device that could realistically simulate the acoustical properties of a space may have started in the late 1960's at Bolt, Beranek and Newman [BBN]. Before then, there were a number of acoustical simulators in existence such as a two- or four-channel stereo system or a theater surround system, but their limited ability to simulate acoustics realistically and to provide easy control and calibration made these systems undesirable as research tools.

In a 1970 paper presented at the Audio Engineering Society, Thomas Horrall of BBN, described their attempts to "aurally preview a proposed architectural solution" and "to clarify and standardize the subjective vocabulary of music in concert halls". The BBN system was helpful in understanding if a certain (known) reflection in a given direction and delay was desirable to the listener. Determination of the correct directional impulse

response to be programmed into the simulator was accomplished by impulse testing in a physical scale model [Horrall 1970].

The evolution of mathematical models and the power growth of computers have made it possible to create acoustical software that can simulate as well as quantify the perceptually relevant parameters by which room acoustics can be judged.

Starting in the early 1990's, various companies worldwide began producing programs with which to model concert halls. A series of round-robin test competitions in 1995 proclaimed that the commercial programs ODEON and CATT, in addition to one unspecified research code, were the most accurate out of 16 participants [Vorländer 1995]. Of these three, CATT-Acoustic had the best results for five of the eight measures calculated [Dalenbäck 2004].

In 1992, Graham Naylor and Jens H. Rindel of the Technical University in Denmark wrote a five-page paper comparing results from ODEON with real measurements published by Gade A.C. of the Royal Festival Hall in London. The comparisons were only made at 1000 Hz, labeled as "satisfactory" or "unsatisfactory" and a small explanation of the probable causes of discrepancy was given [Naylor and Rindel 2004].

This document expands on that work. It provides a detailed explanation of acoustical-modeling software and the construction of a model. Detailed data of the absorption and diffusion coefficients used for the prediction will be discussed. The models will be judged against real-life spaces in multiple octave frequency bands, which will be physically and acoustically measured to ensure the reliability of the comparisons. This will also help validate the coefficients to be used by verifying the type of materials currently installed on the facilities. Two different ways of aurally previewing the acoustical qualities of a space, known as auralization, will be explored.

1.3 Choosing the software.

As stated above, on the 1995 "International Round Robin on Room Acoustical Computer Simulation", CATT-Acoustic had the best results. CATT-Acoustic is a hybrid model that combines the best characteristics of the Ray-Tracing and the Image-Source methods, and allows the input of diffusion as well as absorption coefficients for boundary surfaces. These are all essential elements for a reliable prediction. Based on a review of the technology and history in the field of Acoustic-Modeling, CATT-Acoustic was selected for the modeling in this thesis.

1.4 How the software works

CATT-Acoustic works from a description of the room's geometry and the absorption and scattering coefficients of the room's surface materials. The results will vary in accuracy depending on the approximations of such values.

This software allows the acoustician to define plane surfaces in order to build a three-dimensional model of a space in a computer (figure 1). The acoustician then assigns to each surface certain material characteristics: the degree to which it absorbs, reflects and deflects sound waves. A source can then be defined: where it is located, how its directivity is shaped, and how powerful the source signal is. The emphasis in building a model is not on detail, rather on large surfaces that will certainly reflect sound waves.

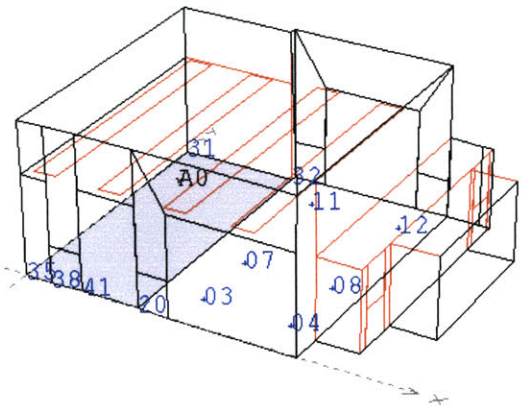


Figure 1, CATT-Acoustic "plane info" screen.

Tens of thousands of sound rays are sent from the sources, and each of these is traced for about the duration of the reverberation time. From this information echograms and a great number of numerical measures can be estimated, e.g. speech intelligibility and reverberation time. It is also possible to post-process the echograms and create room impulse responses for auralization.

CATT-Acoustic is a room acoustic prediction program. Its calculation modes are based on the image source model for "early part echogram" qualitative detail, ray-tracing for "audience area color *mapping*" and randomized tail corrected cone-tracing for "full detailed calculations" [Dalenbäck 2004]. Figure 2 shows the prediction control panel where calculation modes can be selected.

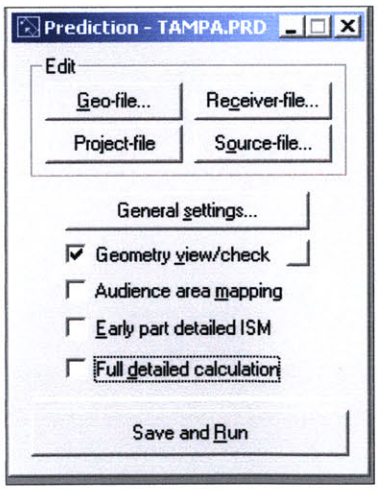


Figure 2, CATT-Acoustic prediction control.

CATT-Acoustic employs a unique and general prediction method, baptized "*Randomized Tail-corrected Cone-tracing*" that combines the best features of the image source model, cone tracing and ray-tracing. All methods take into account frequency dependent diffusion: Walls, or parts of walls, can be assigned a frequency dependant absorption factor as well as frequency dependent surface and edge diffusion factors. Due to the frequency dependence of diffuse reflection, separate ray/cone tracing is performed for each octave. Direct sound, first and second order specular reflections are handled by direct radiation from diffusing surfaces. Higher order diffuse reflections are handled by randomly distributing rays that hit diffusing surfaces [Dalenbäck 2004].

The code computes the path of sound from a source to a receiver. As the sound travels from the source to the receiver, it may reflect off of walls, diffract around edges, or arrive at a receiver position directly. "These paths are utilized to simulate and predict the acoustic qualities of the space" [Markham 2002]. Due to the nature of sound it has been necessary to simulate scattering effects in the models, which define the fraction of the energy that is diffused. This scattering coefficient of rough or structured surfaces is defined as the ratio of non-specularly reflected sound energy and total reflected energy [Dalenbäck 2000]. The energy reflected from a boundary is dispersed over all directions according to Lambert's cosine law. Opposed to a diffuse reflection, a specular reflection can be thought as a "hard" reflection, where the angle of reflection is equal to the angle of incidence.

Lambert's law states that the reflected energy from a small surface area in a particular direction is proportional to the cosine of the angle between that direction and the surface normal. Lambert's law determines how much of the incoming energy is reflected. See figure 3.

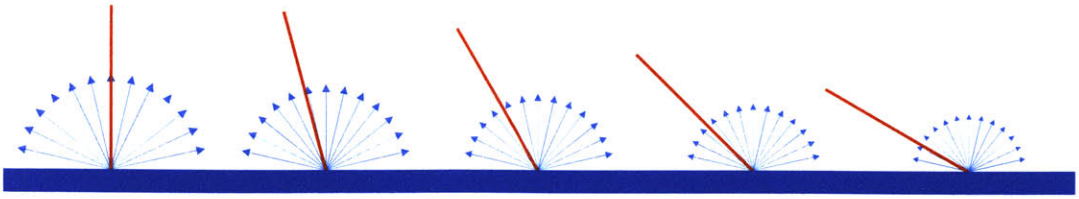


Figure 3, Lambert's Cosine Law diagram.

In this way the reflection from a surface can be modified from pure specular behavior into a more or less diffuse behavior, this introduces the wave nature of sound to the model and has proven to be essential to create reliable results.

Diffuse reflection forces surfaces to be more evenly utilized by redirecting the reflected sound in many directions. Diffuse reflection will let room surfaces be hit by sound in a more uniform manner and absorbing surfaces will be better utilized. The most common ways for numerical modeling programs to handle diffuse reflections is by randomizing the direction of reflected rays according to some distribution. The procedure used by CATT-Acoustic is as follows (figure 4):

"When a ray is to be reflected from a diffusing surface a random number A in the range 0 to 1.0 is generated. If the number is higher than the scattering coefficient S of the surface the reflection is specular. If it is lower, some of the ray's energy will be diffusely reflected (if the scattering coefficient is 0.30, then 30% of the rays that hit the surfaces will be diffusely reflected and 70% will be specularly reflected). If the value A was such that the reflection should be diffuse, two new random numbers, B and C, are generated and decide in which direction the ray should be" reflected according to the direction distribution [Dalenbäck 2000]. The direction of distribution is defined by Lambert's Cosine Law.

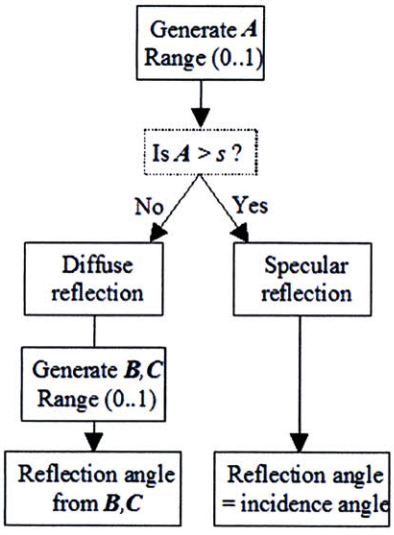


Figure 4, procedure diagram for predicting diffuse reflections, [Dalnebäck 2000].

The scattering of sound from surfaces has proved to be very important in room acoustical simulation techniques, and this has created the need for better information about the scattering properties of materials and structures. Although the model can handle the scattering, the knowledge about which scattering coefficients to use is still very sparse.

There exist no tables of data for scattering coefficients that are nearly as widely distributed and accepted as those for absorption coefficients. Data containing scattering coefficients for complex surfaces like a statue or an audience will not be found in the near future. However, a measuring method for scattering coefficients is being developed by the *International Organization for Standardization* or *ISO*. The good news is that for room-acoustics prediction purposes the scattering coefficients do not have to be very detailed and a bit of physical reasoning will in most cases suffice [Rindel 2000]. There are certain recommendations as follows [Dalenbäck 2000]:

- Use 0.10-0.20 minimum on all surfaces, 0.08-0.1 on large flat surfaces
- Use 0.40-0.70 for 125Hz - 4KHz on audience areas
- Use 0.80 for general rough surfaces, where the roughness is of the order or higher than the wavelength, and gradually lower below

Although "reasonable" guessing is the only way of estimating the coefficients, in many cases the ability to "reasonably" guess comes only with experience and through trial and error.

There are spaces that are more sensitive to scattering and have to be treated with caution. To illustrate the extreme case of diffusion sensitivity that a non-mixing room with parallel surfaces might have, an example found on the CATT-Acoustic webpage is shown next (figure 5). The graph shows the impact on reverberation time that different scattering coefficients can have.

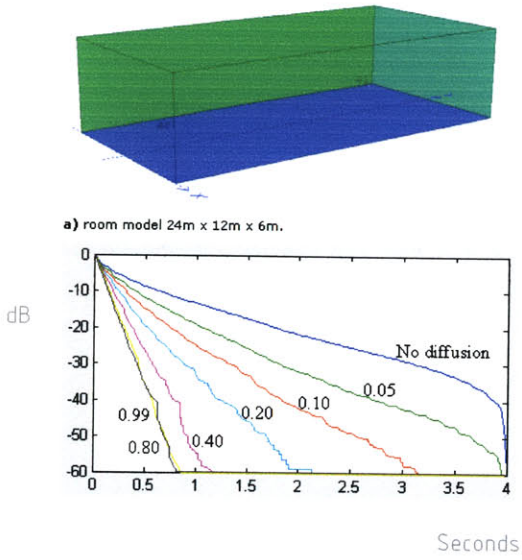


Figure 5, Reverberation prediction sensitivity to scattering [Dalnebäck 2000].

Figure 5 illustrates the effects on the reverberation time for scattering coefficients from 0 to 0.99 using an idealized rectangular room with an absorbing floor.

In an example of a sports hall, shown in a paper written by B.I. Dalenbäck, an acoustic consultant (Akustikon, Sweden) recommended placing acoustic absorbers in the ceiling between every second beam pair, 50% coverage, and in addition high absorption was to be placed on one side wall and one end wall. However, to save money the contractor decided to use only the ceiling absorbers. With only the treated ceiling the calculated Sabine RT @ 1KHz was 1.9 sec but the measured reverberation time was 5.7 sec. This is a clear case of a non-mixing shape with uneven absorption where formulae can not render accurate results. Next the acoustic consultant did a series of computer prediction tests to investigate what scattering coefficients would have given the measured reverberation time in the initial case (5.7 sec) and the result was around 0.10 (in accordance with the general recommendation above of 0.10 for large flat walls). The table shows various predicted RT values for the 1 kHz octave-band where the Sabine and the specular-only computer model cases form the low and high extremes while the

computer model values with diffusion (scattering coefficients) taken into account are close to the measured values [Danlenbäck 2000]. Table 1 summarizes the above.

Method/model	Scattering Coefficent	Reverberation Time (T-30)
Sabine formula	N/A	1.9 sec
Measured	N/A	5.7 sec
Computer Prediction	Specular = 0	13 sec
Computer Prediction	Ceiling: 0.80 Rest: 0.08	5.9 sec (3% error)

Table 1, Measured and predicted RT values @ 1 kHz for the sports hall.

1.5 Conclusions [Chapter 1]

It is clear that although scattering coefficient standards have not been developed, algorithms that take into account diffusion are required for accurate results. As seen in the example above, reasonable guessing of the scattering coefficients can generate extremely accurate results. CATT-Acoustic is one of the few packages that handle diffusion, this is the main characteristic that makes it superior to others and made it the clear choice.

It is now known that a computer model is much more flexible than a scale model. It renders more information in a more accurate manner and can easily be modified in geometry and surface materials. The computer model is fast and results can be analyzed numerically with parameters and even heard through the auralized room response.

Chapter 2
Reverberation Time Prediction

2.0 Introduction

This section will be devoted to the explanation of reverberation time; details on how it is measured in real spaces, how it is calculated using the Sabine equation and how the computer model predicts it will be discussed. At the end, as on an all chapters, a case study will be presented were the accuracy of the different predictions methods will be tested and compared.

2.1 Reverberation time

Reverberation is the buildup of sound within a room, resulting from repeated sound wave reflections off all of its surfaces. Reverberation can be defined as the time in seconds that it takes for a sound to decay in sound pressure level by 60dB after the sound source has been silenced. Most rooms have reverberation times in the range of 0.5 to 6.0 seconds. A very "dead" room has a short reverberation time, while a "live" room has a longer reverberation time. Generally, larger rooms have longer reverberation times simply because it takes longer for the sound to travel around the room. Rooms with shorter reverberation times tend to be better suited for speech, while rooms with reverberation times around 2 to 3 seconds are good for solo or orchestral music. For organ music and chant, Rooms with longer reverberation times are preferred [Acentech and Cowan 2000]. Figure 6 shows a typical reverberation time spectrum.

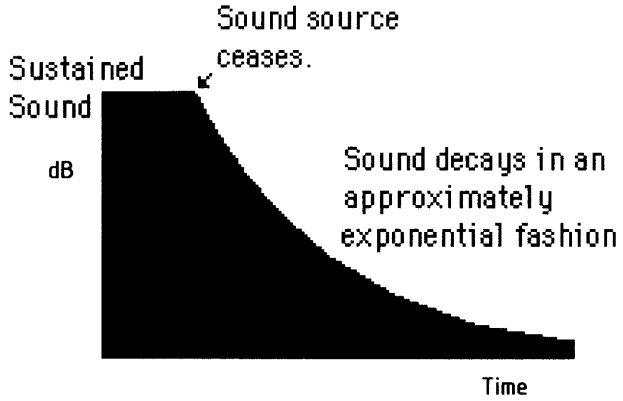


Figure 6, Typical RT measurement [Nave 2004].

Reverberation time (RT) is far from the only metric used to judge the acoustics of spaces. However, the RT is a good starting point since it is a central parameter in many applications of room acoustics. In order for a room to achieve appropriate room acoustics conditions, most acousticians would agree than a room must have an appropriate reverberation time.

2.2 Reverberation time measurements in real spaces

A source and an instrument that can capture sound decay are necessary for measuring reverberation time. The sound source can produce sound by either impulse excitation or noise excitation. This requires a source to produce an even spectrum of sound (usually pink noise) that excites the space. When the source is interrupted the decay is measured.

Although the reverberation time is defined as the time measured for the sound to decay 60dB, it is very difficult to make such a measurement. Measuring the complete spectrum would require the source to produce the background level, plus another 5 dB distance to the background noise and finally the 60dB to drop. This means that a source needs to

produce the background level plus 65dB, a requirement very hard to achieve by practical systems.

Instead of measuring the complete decay, normally a 15, 20, 30 or 40dB decay is measured; afterwards it is extrapolated to 60dB assuming that the part of the decay used is representative for the entire decay. It is common to specify the decay used as T15, T20, T30 or T40 respectively. Figure 7 illustrates and explains the steps of a typical reverberation time measurement [Norsonic 2004].

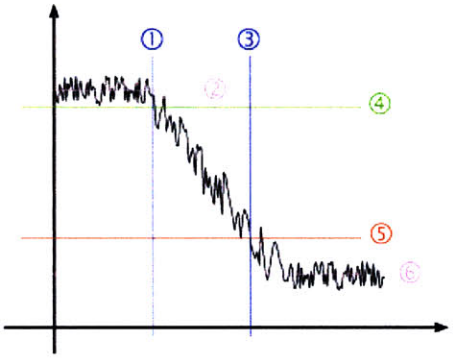


Figure 7, Decay curve detail [Norsonic 2004].

After the source is interrupted (1), the measurement begins when the original level (2) has decayed by 5dB (4). It continues until the desired level (5) is reached at a certain time (3). Note the distance from the background noise (6) [noise floor] [Norsonic 2004].

2.3 Reverberation time prediction with formulae

Sabine and Eyring developed equations that are widely used to calculate RT. These assume that reflections are fully diffuse and that each surface is visible from all other surfaces so that the utilization of absorption of a surface can be considered to be in direct proportion to its relative area. For practical purposes it is generally assumed that the Eyring expression for RT is applicable if the average absorption coefficient is greater than 0.2. In diffuse rooms with less absorption, the Sabine RT formula is generally used [Welsh School of Architecture 2004]. Sabine's equation is the most commonly used formula and could be considered as the cornerstone of acoustic calculations; see equation 1.

$$RT = 0.161\frac{V}{A}$$

V = volume of enclosure (m^3)

A = total absorption (sabins)

Equation 1, Sabine equation.

This means that with even absorption distribution throughout a room, the actual RT is often close to the classical Sabine or Eyring formulas. However, as soon as there is an audience in a room there is bound to be an uneven absorption distribution, especially at higher frequencies where the absorption coefficients from the audience greatly differ from the rest of the surfaces. Acoustical modeling software has been developed through the past three decades as a tool to accurately calculate the RT in spaces where formulae have not been successful.

2.4 Reverberation time computer prediction

The software relies on the real geometric layout of the space and specific properties (sound absorption and diffusion) of materials within that space. The software uses a technique similar to ray tracing known as randomized tail-corrected cone tracing in order to calculate the room response and acoustical signature of the modeled space. Simulations are run with one source sending its signal through the room. Results are exhibited at point receivers. A point receiver gives the values for reverberation time (among other variables) at a sampled listening post. It also illustrates the decay curve for how the sound died out. For a more in-depth explanation see *1.4 "Reverberation time computer prediction".*

2.5 Case study

The Carl and Ruth Shapiro Campus Center Theater

Brandeis University, MA

The new facilities of the Carl and Ruth Shapiro Campus Center, situated in Waltham, Massachusetts, were completed on 2002.

The new Campus Center serves as a center for student life at Brandeis University. This 24-hour building includes a student theater, rehearsal spaces, the University bookstore, cafe, a two-floor library with computer clusters for group and individual study, student organization spaces, function rooms and lounges (figure 8).

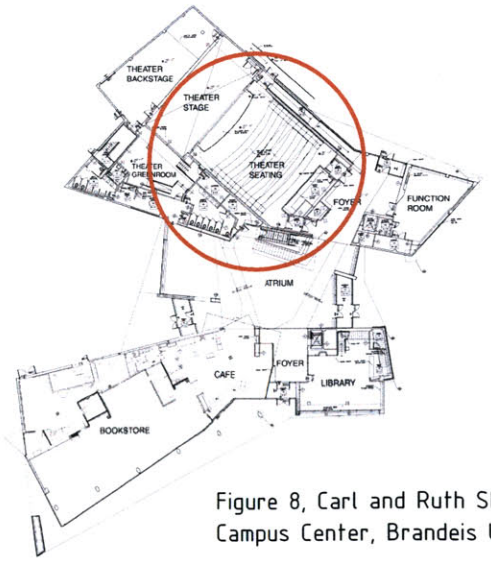


Figure 8, Carl and Ruth Shapiro Campus Center, Brandeis University.

The Ruth Shapiro 249-seat theater will accommodate productions by the Undergraduate Theater Collective and lectures, among other events. By all accounts this hall has taken advantage of all acoustical principles. Although the basic shape of the room is rectangular, the side walls are randomly broken into sections to produce diffuse reflections. The first two sections, closest to the stage, are angled so that they reflect the sound to the audience area (figure 9).

Figure 9, Interior of Ruth Shapiro Theater, [Brandeis University]

A permanent set of wooden reflectors hanging from the ceiling enforces sound reflection on the vertical axis. The floor is a light carpet directly pasted on concrete; the side-walls and ceiling are covered by wood paneling. It seems that for the purpose of avoiding lower frequency absorption the wood panels are very rigidly braced. The only absorptive surfaces are the audience area and the rear wall, consisting of cloth covered glass fiber panels. All other surfaces are kept hard to maximize the Theater's reverberation time.

An advantage of this project was that an acoustic consultant was hired in the design stages; thus the appropriate acoustic considerations were designed into the facility, rather than patched in as an afterthought.

MODELING:

The first part of this project was to model the Ruth Shapiro Theater in 3D using AutoCAD software and transferring it into CATT-Acoustic. There is an interface that can be loaded from CATT-Acoustic that allows for the complete geometry modeled in AutoCAD to be exported and also enables the user to define surface planes, sources, receiver positions, plane names and absorption [material] characteristics. After this process, only some minor editing of absorption data and debugging of the model remains. Figure 10 shows the finished model in CATT-Acoustic's 3D viewer. It is important to state that all the modeling has to be done using only 3D faces (two-dimensional surfaces placed in a three-dimensional virtual environment), due to the fact that the CATT-Acoustic software does not handle 3D solids. The model developed for this project uses the basic geometry of the hall and does not take into account small details. Emphasis is made on larger surfaces that will reflect sound. Receivers and sound sources were added later in CATT-Acoustic.

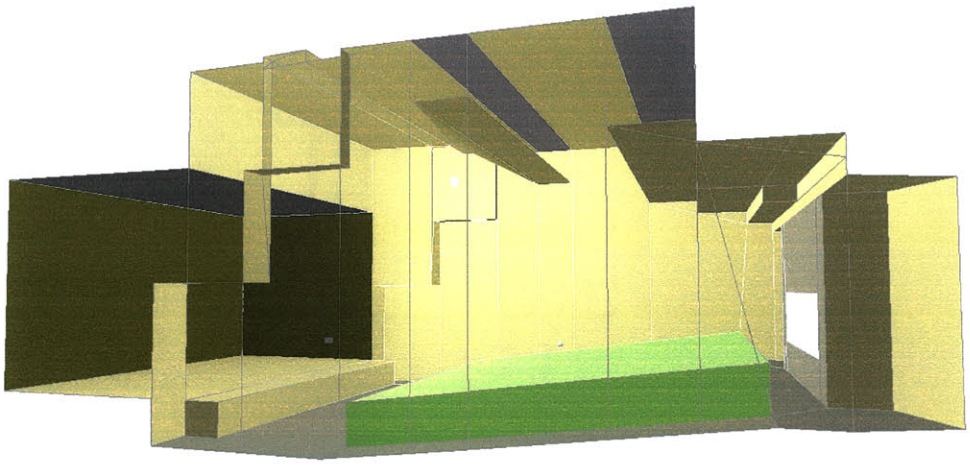


Figure 10, model of Ruth Shapiro Theater.

After exporting the geometry to CATT-Acoustics, an omni-directional sound source and a receiver were situated within the space. The location was approximately the same as in the field measurement (figure 11).

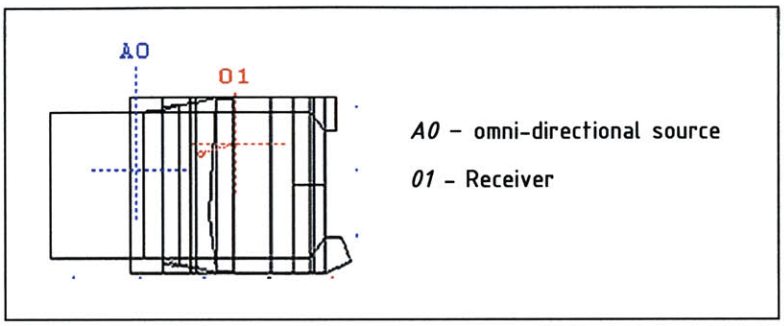


Figure 11, locations of source and receiver.

Each surface is assigned certain material characteristics; the degree to which it absorbs, reflects and deflects sound waves. The absorption performance data were taken from Cavanaugh [Cavanaugh 1999] "Acoustical control in Buildings" and have been verified in a variety of sources including the "Acoustical Material Association" (table 2). The data on diffusion were deduced from physical reasoning following the scattering coefficient recommendations in section 1.4 of this document; the values used derive from my own judgment from visual analysis of the surfaces (table 3).

ABSORPTION COEFFICIENTS

	125	250	500	1K	2K	4K
Walls and ceiling - 3\8" wood	0.28	0.22	0.17	0.09	0.1	0.11
Stage Floor - Wood	0.15	0.11	0.1	0.07	0.06	0.07
Back Wall - 1" Glass Fiber no cavity	0.05	0.08	0.6	0.93	0.99	0.96
Back Wall - common glass	0.35	0.25	0.18	0.12	0.07	0.04
Audience / m²	0.49	0.66	0.8	0.88	0.82	0.7
Light Carpet Floor-glued on concrete	0.01	0.02	0.1	0.25	0.35	0.4
Stage Walls - Curtain	0.14	0.35	0.55	0.75	0.7	0.6
Cavity on ceiling - Hole	0.7	0.7	0.7	0.7	0.7	0.7

Table 2, Octave bands center frequencies, Hz.

SCATTERING COEFFICIENTS

	125	250	500	1K	2K	4K
Walls and ceiling - 3\8" wood	0.1	0.12	0.14	0.18	0.2	0.2
Stage Floor - Wood	0.15	0.15	0.15	0.15	0.15	0.15
Back Wall - 1" Glass Fiber no cavity	0.1	0.1	0.1	0.1	0.1	0.1
Back Wall - common glass	0.1	0.1	0.1	0.1	0.1	0.1
Audience / m^2	0.3	0.4	0.5	0.6	0.7	0.8
Light Carpet Floor-glued on concrete	0.1	0.1	0.1	0.1	0.1	0.1
Stage Walls - Curtain	0.3	0.4	0.5	0.55	0.6	0.7
Cavity on ceiling - Hole	0.8	0.8	0.8	0.8	0.8	0.8

Table 3, Octave bands center frequencies, Hz.

The program calculates a series of parameters using the input mentioned above. This study did several analyses using the Sabine reverberation time, two variables of Eyring reverberation time, the reverberation time from -5 dB to -20 dB (T-15) and the reverberation time from -5 dB to -35 dB (T-30), all calculated by CATT-Acoustic's processes explained in Chapter 1. The results are as follow:

	125	250	500	1k	2k	4k	
EyrT	0.75	0.71	0.59	0.55	0.54	0.54	s
EyrTg	0.75	0.71	0.59	0.55	0.54	0.54	s
SabT	0.89	0.85	0.74	0.69	0.68	0.66	s
T-15	0.98	0.91	0.79	0.77	0.75	0.69	s
T-30	1.01	0.95	0.83	0.97	0.80	0.75	s

Table 4, RT calculation; CATT-Acoustic "Full Detailed Calculation".

MEASUREMENTS:

The next step was to take RT measurements in the existing space and compare how closely the available tools for prediction reproduce the real acoustical characteristics of the hall.

RT measurements of the Ruth Shapiro Theater were done using a RION NA-27 sound level meter and a sound box equipped with a pink-noise generator. Pink noise is used

for acoustic applications because it has a flat spectrum when viewed on a third-octave spectrum analyzer, or any constant percentage bandwidth analyzer.

Figure 12 Sound Level Meter; Rion NA-27.

The sound source (sound box) and the receiver (sound level meter) were approximately located on the same spots as on the CATT-Acoustic model. Pink noise was generated for approximately 10 seconds to excite with sound the entire space; then the sound was cut off and the sound level meter captured the decay of sound, measuring sound level against time in 2 millisecond intervals. The next figure shows an example of how sound is measured and the decay visualized.

The recorded data were then processed and a reverberation time for each octave band was obtained with the decay from -5dB to -35dB (T-30). The results are as follows [table 5]:

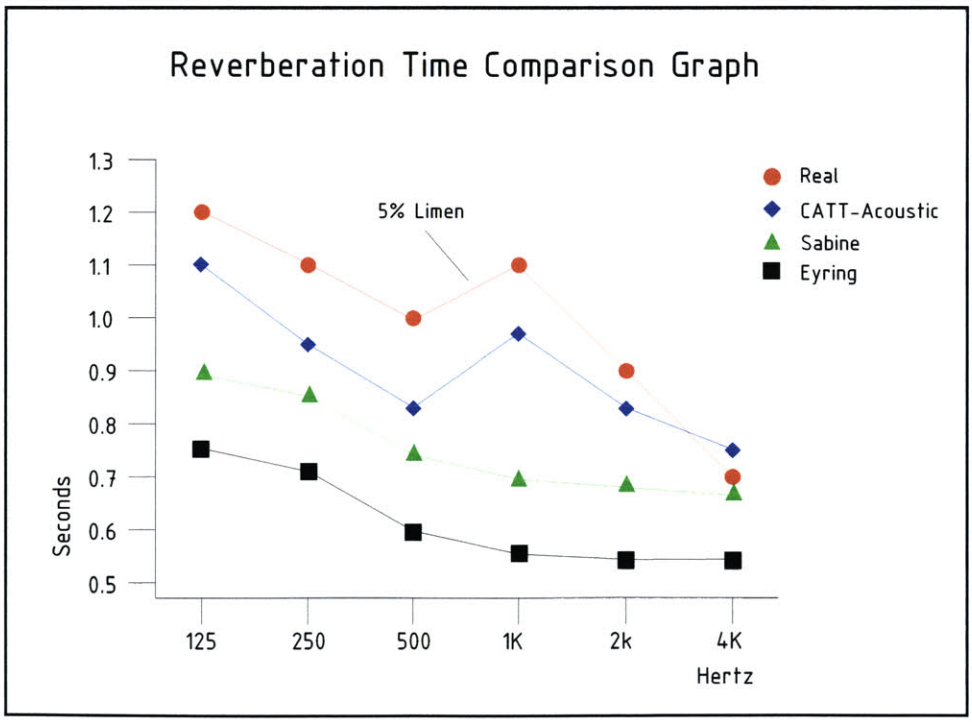
	125	250	500	1K	2K	4K
T-30 (sec.)	1.2	1.1	1.0	1.1	0.9	0.7

Table 5, reverberation time; octave bands center frequencies, Hz.

2.6 Conclusions [Chapter 2]

In comparing the measurement results from the Ruth Shapiro Theater to the CATT-Acoustic model, the Sabine Equation and the Eyring equation are relevant in order to see the usefulness of the software. It is clear that the CATT-Acoustic software is an important step toward predicting the acoustical characteristics of a space. On the following figures and tables [1] it is possible to compare the results between the traditional tools (Sabine and Eyring), the software and the measurements done in the hall with the RION sound level meter.

Frequency [Hz]	125	250	500	1K	2K	4K
Real	1.2	1.1	1	1.1	0.9	0.7
CATT-Acoustic	1.1	0.95	0.83	0.97	0.8	0.75
Sabine	0.89	0.85	0.74	0.69	0.68	0.66
Eyring	0.75	0.71	0.59	0.55	0.54	0.54



Graph 1, reverberation time comparison.

It is clear that the CATT-Acoustic software comes very close to the measured reverberation time. It appears that the difference can be explained by the 3/8" wood absorption coefficients used in the model; in the actual building the wood is thicker and extremely rigidly braced. Depending how rigid the bracing is and the thickness of the material, sound can be absorbed to a different degree, thus causing the gap between measured and predicted sound.

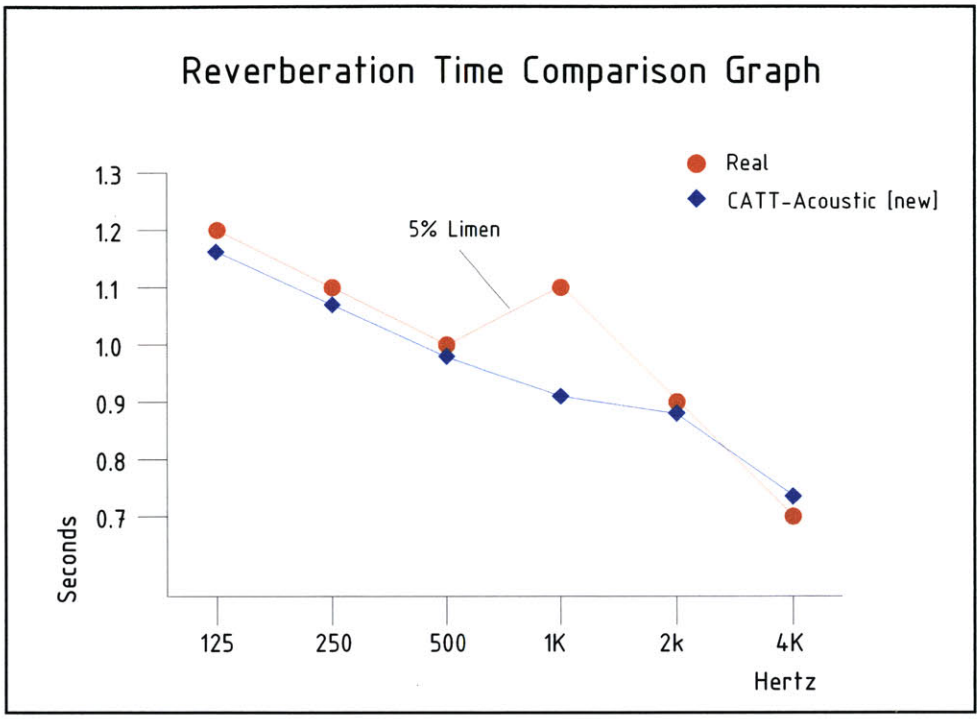
To explore if this explanation is valid and if the software can predict the RT more accurately, a new run with absorption coefficients for thicker plywood was done. Finding the coefficient for firmer plywood proved difficult; the published charts do not have the desired data. It was decided to use a mathematical mean of the absorption coefficients from 3/8" plywood and wood flooring [Cavanaugh and Wilkes 1999], which would closely resemble the characteristics of the ½" wood rigidly braced. Table 6 shows new "rigid wood" coefficients:

	125	250	500	1K	2k	4k
3/8" plywood	0.28	0.22	0.12	0.09	0.10	0.11
Wood floor	0.15	0.11	0.10	0.07	0.06	0.07
"Rigid wood"	**0.21**	**0.16**	**0.11**	**0.08**	**0.08**	**0.09**

Table 6, new wall absorption coefficient; Octave bands center frequencies, Hz.

The new results, as shown on graph 2, were even closer to the measured RT, further proving the usefulness of this software as a new analytical tool. Future development of extensive and more accurate absorption coefficient data is a logical step that must be taken to help us model the targeted spaces more accurately.

Frequency [Hz]	125	250	500	1K	2K	4K
Real	1.2	1.1	1	1.1	0.9	0.7
CATT-Acoustic [new]	1.16	1.07	0.98	0.91	0.88	0.74



Graph 2, new reverberation time comparison.

CATT-Acoustic, in addition to predicting the reverberation time much better than previous formulae, also calculates many other acoustical data that will be as useful as RT for designing a successful hall.

The computer predictions differ from the average measurement results by the same magnitude as any two individual field measurements. All are within the 5% subjective difference limen range [Gibbs & Oldham 2001] except in the 1KHz frequency band. Thus, we can deduce that the accuracy of the best computer calculations is roughly as good as two consecutive measurement results and can be considered acceptable. See graph 2.

Chapter 3
Speech and Music Objective Parameters - D_{50}, C_{50} and C_{80}

3.0 Introduction

This chapter presents how some objective parameters are used to evaluate the acoustical clarity of a room. Definitions, criteria and details on the calculations are offered to understand the highlights and shortcomings of these methods. Comparisons between predicted room acoustic indices (D_{50}, C_{50} and C_{80}) from a computer model and real values based on measurements carried out for the Ruth Shapiro Theater are explained and the results discussed.

3.1 D_{50}, C_{50} and C_{80}

D_{50}, C_{50} and C_{80} are mathematical manipulations of acoustical energy over time that are initiated by a sequence of events beginning with the arrival of the initial signal, followed by a succession of reflections that progressively dissipate because of losses caused by sound absorbing surfaces and the air. The early portion of the acoustical energy, consisting of the early reflections, is especially important and this portion can be further divided into early into early and late time periods, with the former being more significant. Sound quality is largely established in the early-reflection period. The early and late categories must be defined with the type of activity in mind, e.g. speech versus music or type of music. In simple terms reflections arriving within the first 50 msec will usually contribute beneficially to speech clarity; those within 50 to 100 msec may or may not be beneficial; and those arriving subsequently will probably harm clarity [Cavanaugh and Wilkes 1999].

These early-late ratios have proven to be as reliable as any other measure for clarity. A 50 msec dividing time is generally used for speech, while for music these useful/harmful times can be extended to 80 msec; this explains the early-late ratio difference in integration time used for music and speech. These measures are known as C_{50} and C_{80} respectively and expressed in dB. Similar to C_{50}, distinctness or D_{50} (Deutlichkeit) is a ratio of the sound energy in the first 50 msec after arrival of the direct sound to the total sound energy arriving. It is usually expressed as a percentage [Beranek 2004].

3.2 D_{50}, C_{50} and C_{80} calculations in real spaces

The physical measurement is the ratio of the early sound energy to that in the reverberant sound and is determined from an impulse response of the hall.

To measure D_{50}, C_{50} and C_{80} two pieces of equipment are required: one produces an impulse sound, such as a bursting balloon, a pistol shot or a short intense beep from a loudspeaker; and the other is a sound level meter or a tape recorder that records the impulses. From the recorded data two quantities are obtained. First is the energy of the sound that arrives directly from the source plus all the reflections from surfaces in the space that occur within the initial integration time (50 msec for D_{50} and C_{50} or 80 msec for C_{80}). Second is the energy of the sound that arrives after the initial integration time (up to 1 or 2 seconds). The ratio of the first divided by the second, expressed in decibels (percentage for D_{50}) is C_{50} or C_{80} [Beranek 2004]. The equations for D_{50}, C_{50} and C_{80} are:

$$C_{50} = 10\log\left(\frac{\int_0^{50} p^2}{\int_{50}^{\infty} p^2}\right)$$

$$C_{80} = 10\log\left(\frac{\int_0^{80} p^2}{\int_{80}^{\infty} p^2}\right)$$

$$D_{50} = \frac{\int_0^{50} p^2}{\int_0^{\infty} p^2}$$

Equations 2, 3 and 4; D_{50}, C_{50} and C_{80}.

If in a hall there is no reverberation, the room will be very "dead", C_{50} and C_{80} will render a large positive value. If the reverberation is very long, such as exists in a large gymnasium, C_{50} and C_{80} will take a large negative value and the sound will be unclear. C_{50} and C_{80} equal 0 when the early energy is equal to the reverberant energy [Beranek 2004]. Figure 14 equates D_{50}, C_{50} and C_{80} objective measures to a subjective rating scale [Cavanaugh and Wilkes 1999].

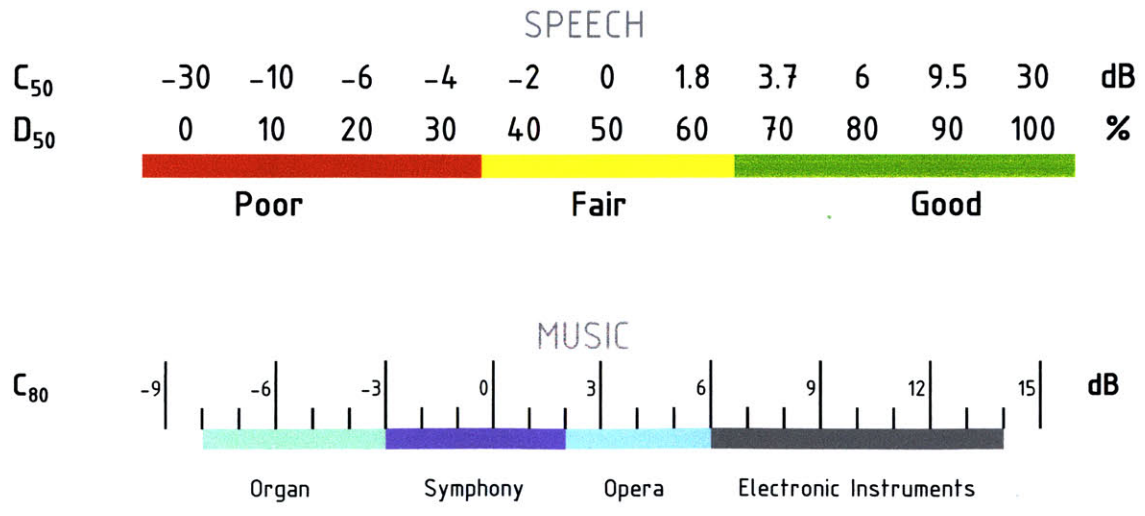


Figure 14 D_{50}, C_{50} and C_{80} subjective rating scale.

A good thing about D_{50}, C_{50} and C_{80} is that they are rather simple to understand. Since it is easy to hear whether the music or speech is clear or sounds "muddy" or the balance between the reverberant and the early sound is not satisfactory, these early-late ratios can be judged qualitatively by a careful listener [Beranek 2004].

3.3 D_{50}, C_{50} and C_{80} computer based calculations

For the calculation of these indices the computer uses "The Ray Tracing Method", where a large number of particles, which are emitted in various directions from a source point, are traced around the room while they loose energy according to the absorption coefficient of the surfaces they hit. In the same way as for reverberation time, a sound decay curve is generated, and from it the software calculates the relevant ratios to render the D_{50}, C_{50} and C_{80} indices. For a more in depth explanation on how the software generates a sound decay curve see *1.4 "How the software works"*.

3.4 Case study

The Carl and Ruth Shapiro Campus Center Theater
Brandeis University, MA

Using the model with the absorption coefficient corrections from "***Reverberation Time Case Study"*** on Chapter 2, a set of new computer calculations were developed with the CATT-Acoustic software to obtain predicted D_{50}, C_{50} and C_{80} at a given location, see figure 15.

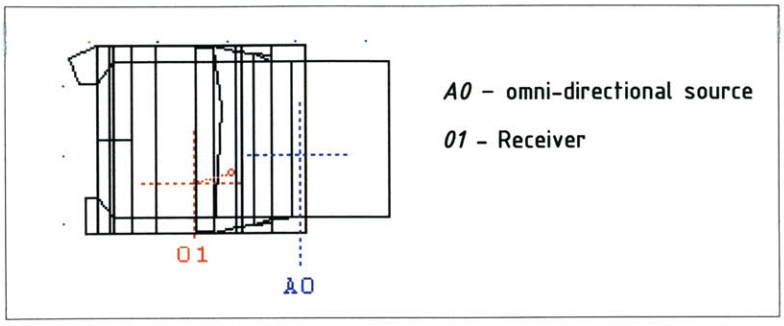


Figure 15, locations of receiver and source.

The results of the computer calculations are based on the predicted reflection sequence for each octave band. For the purpose of this paper 500 Hz was chosen for the comparison, the results are as follow:

COMPUTER RESULTS

D_{50}	C_{50}	C_{80}
63%	2.5 dB	5.7 dB

For the reality based values, measurements were taken with a Rion NA-27 sound level meter, which was programmed to take a decibel reading every two milliseconds from a pink noise sound decay. This renders a matrix of 2,000 decibel readings on each of the six octave bands that can be visualized as a sound decay as shown on figure 16.

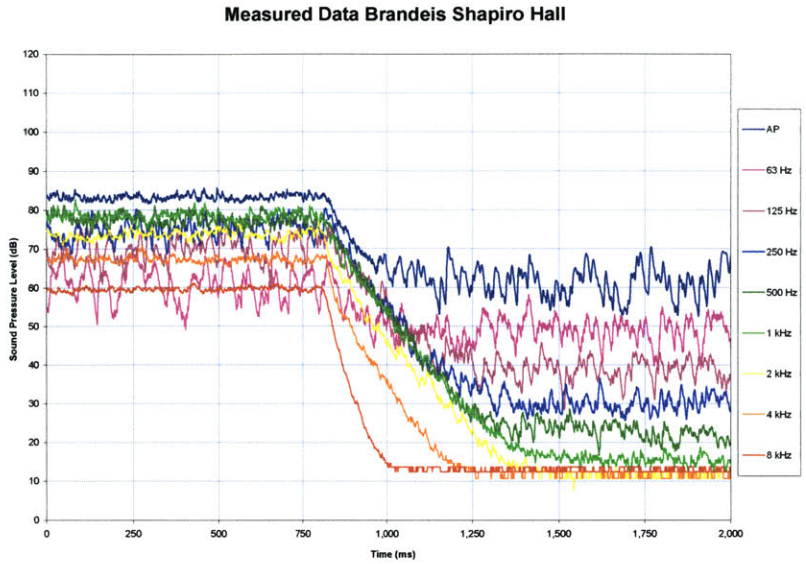


Figure 16, sound decay from Ruth Shapiro Theater, Brandeis University.

Each reading on the 500 Hz octave band was converted from **Intensity Level (dB)** to **Pressure Square (p^2)** in order to add the readings in the time intervals required for each parameter; see equation 5. Table 7 shows the results for D_{50}, C_{50} and C_{80} on 500 Hz for the hall in question.

$$p^2 = 10^{\frac{dB}{10}}$$

Equation 5, conversion from intensity level to pressure square.

C50		D50		C80	
	sums		sums		sums
8.94E+08	0 to 50	8.9E+08	0 to 50	1.1E+09	0 to 80
6.83E+08	50 to end	1.6E+09	0 to end	4.5E+08	80 to end
1.2	C50	**57%**	D50	**4.0**	C80

T

Table 7, D_{50}, C_{50} and C_{80} results on 500 Hz.

3.5 Conclusions [Chapter 3]

When comparing the real measured values of D_{50}, C_{50} and C_{80} against those predicted by the computer a significant difference in the results is perceived. For C_{50} the difference is more than double the subjective limen of 0.5dB. For D_{50} The difference is a bit more that the 5% limen of "just noticeable difference" [Akustikon 2004]. In the case of D_{50} and C_{50} it is just enough to push the results from "fair" to "good", so caution must be exercised when discussing these parameters; see figure 17.

	D_{50}	C_{50}	C_{80}
Real	57%	1.2 dB	4 dB
CATT-Acoustic	63%	2.5 dB	5.7 dB

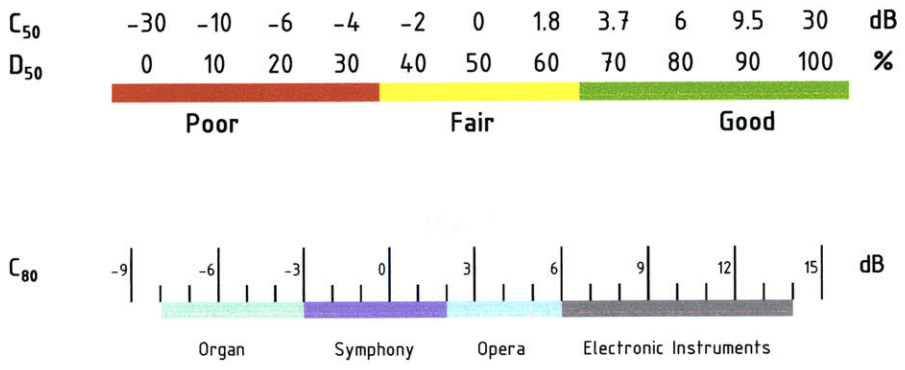


Figure 17, D_{50}, C_{50} and C_{80} comparison chart.

The reason for this discrepancy could be that because of geometrical simplifications and distortions in the computer model, reflection interference that can occur at a slightly

different location in the room or in the reflection sequence is not accounted for in the computer model. Adding to this, there is often a strong local positional variation in the values of D_{50}, C_{50} and C_{80}, and one is not quite certain of the absolute position of the source or receiver when transferring between a real hall and its computer generated model description. On a paper written by X. Perlson, dealing with variability of room acoustical parameters, he states that a positional accuracy of +/- 30cm in transferring from a real hall to a geometrical model is probably only rarely achieved [Naylor and Rindel 1992]. For instance, all this can cause a reflection to occur just before the specified interval of either 50 or 80 msec. instead of just after, thus disturbing the values.

Further problems can arise because of the oversimplification in the computer model of the audience plane. The software does not take into account the attenuation of sound waves at grazing incidences over the seat rows of the audience. Sound passing over such a complex structure is very difficult at present for a computer to model in a simple way, and this particularly affects the modeling of the direct sound and the first reflections [Vorländer 1995].

With these uncertainties in mind we know that although this parameters agreement is not what we desire, they can still serve as a useful tool in comparisons of different options to be applied into a space.

Chapter 4
Binaural Auralization

4.0 Introduction

The subject of this chapter is to discuss and evaluate the binaural auralization process as a tool for subjectively rating the acoustics of a space. Explanations of how binaural recordings are made in real space as well as how computer models can produce binaural signals are presented.

To validate the accuracy of the auralization process, soundtracks from binaural recordings from a real space and computer generated auralizations will be compared and rated on their ability to convey the source location, the reverberation time of the space and the tone/timbre of the source.

4.1 Binaural auralization

Listening to sound in a room before its construction will make it easier to understand the acoustical implications of a given project, thus becoming one of the most employed tools of the acoustician in the future. The technique that uses a computer generated room impulse response to make a room audible has been called auralization in analogy to visualization. Auralization techniques offer the possibility to use the ears to listen and judge the acoustics of a room during the design process [Kleiner, Dalenbäck and Svensson 1993]. Several acoustical problems that are difficult to understand and express with a numerical parameter can be detected by the ears and qualified subjectively.

One technique to recreate the acoustical ambience of a room is to feed the ears independent signals (through a pair of headphones) that would correspond to the sound

perceived in the real listening environment. The aim is to produce in the ears and the brain of the listener the illusion of a pattern of direct and reflected sound.

4.2 Software techniques for binaural auralization

The simplest auralization option available on CATT-Acoustic is based on binaural technology allowing the creation of an acoustic simulation over a pair of headphones. In the computer model, at the receiver location, a pair of impulse responses are calculated [one for each ear] taking into account all previous calculations for the room. The impulse response contains information on received reflections, location of the source and receiver and all of the source characteristics. A set of these impulse responses are called "*Binaural Room Impulse Response*" (BRIR) and can be though of as the sonic fingerprint of the space. See figure 18 for an example [Rindel 2000].

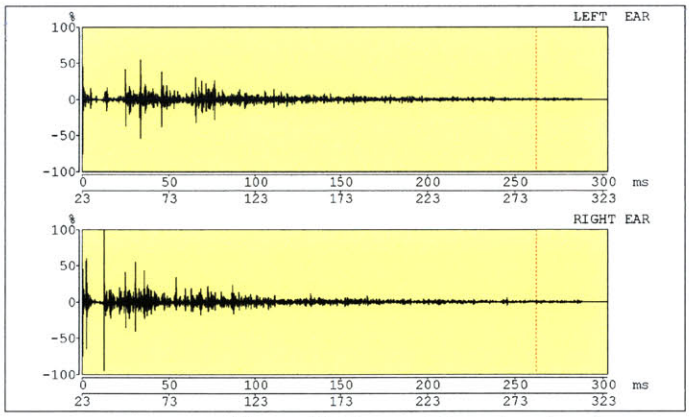


Figure 18, Binaural Room Impulse Response.

CATT-Acoustic then convolves the "*Binaural Room Impulse Response*", a built-in "Head Related Transfer Function", representing how sound acts around an average human head, and the sound signal to be played back. This sound signal can be speech, music, bursts of sound or whatever is relevant for the listening test; the only requirement is that it must be anechoically recorded to avoid the introduction of acoustical characters from the recording room into the room model impulse response. An anechoic soundtrack is one

that has been recorded in an anechoic chamber: all surfaces of the chamber are 100% absorptive and only the direct sound is registered by the microphone. Figure 19 shows an anechoic chamber.

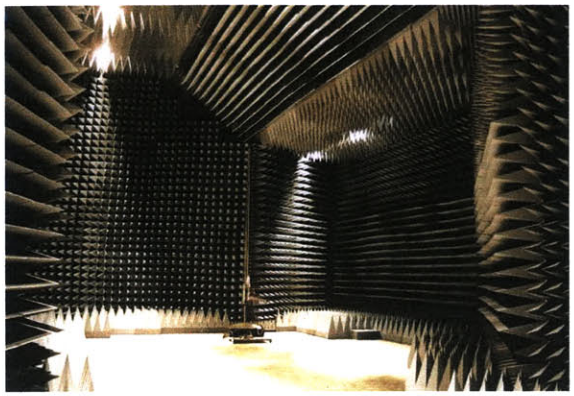


Figure 19, Anechoic Chamber.

4.3 Recording techniques using binaural head recording hardware (dummy head).

Due to the shape of the human head and the ears, sound recorded through a standard microphone will not sound natural and it will not deliver a sense of localization of the source. Because of this, recordings aimed at recreating a 360° sound environment are done with an artificial head (dummy Head) that has a pair of microphones on each ear canal. The artificial head replicates the average properties of a human head and captures all the frequency adjustments that happen naturally as sound wraps around the head and is altered by the form of the outer and inner ear.

This technique provides the brain with changing spectral characteristics depending on the direction of particular sounds. The ridges of the ear, as well as the other features of the head act as multiple "frequency-selective band-pass" filters tuned to the azimuth and elevation of every sound in our environment. The minuscule differences in frequency response, phase and sound level enables the brain to localize the sounds [The binaural

source, 2004] Recordings done through it will sound analogous to reality. Figure 20 shows a dummy head for binaural recording.

Figure 20, binaural recording dummy head [Neumann].

4.4 Case Study

Binaural head recording vs. binaural auralization.

The Carl and Ruth Shapiro Campus Center Theater

Brandeis University, MA

To evaluate the naturalness and validity of a computer binaural auralization a comparison between dummy head recordings and an auralized room simulation were developed. The dummy head recordings were used as reference signals. Figure 21 shows the computer model and the location of the source and the receiver.

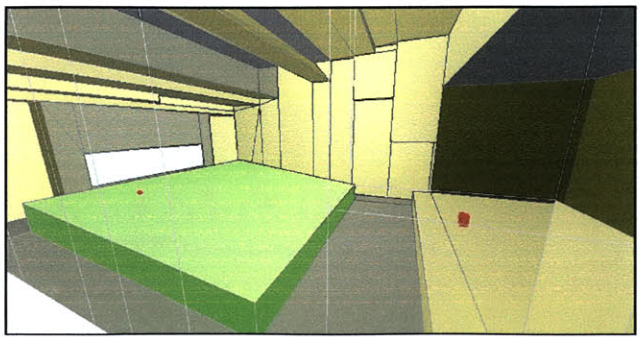


Figure 21, computer model, source and receiver shown in red.

The first goal for the room acoustic simulation was to create a totally artificial environment with the computer model previously developed in the CATT-Acoustic software. The sound source qualities, sound propagation and locations of source (see figure 22) and the receiver were carefully integrated through a post-processing module to the room characteristics. Three different anechoic soundtracks were employed for the tests: a female voice, classical music and a sample of percussions.

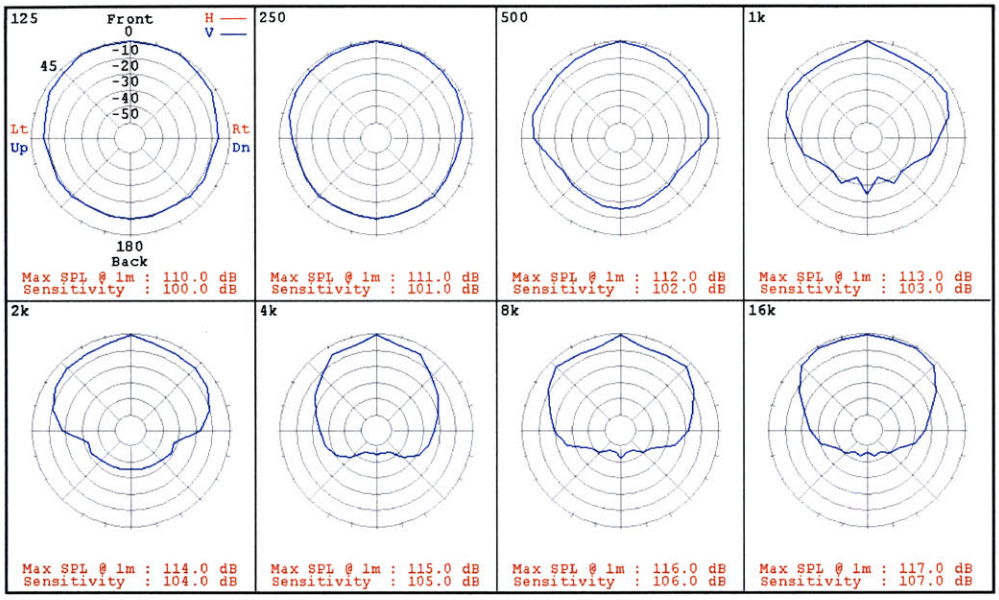


Figure 22, Directional characteristics of the sound source, EV SxA100+.

The same three anechoic soundtracks were played back in the Ruth Shapiro Theater through an MP3 player and an active loudspeaker, EV SxA100+. A dummy head with condenser microphones in each ear canal was hooked up to a DAT recorder and was used to capture the sound in the hall. The recordings were then transmitted to the computer and prepared for headphone listening.

A listening test was conducted to judge the accuracy of the computer model auralizations with regards to: the location of the source, the reverberation time, and the tone/timbre of the samples. Both sets of samples (recorded and auralized) were played to several listeners who could switch between them.

All 12 listeners were acoustic consultants who are familiar and well versed in the parameters to be judged. The analysis consisted of three pairs of samples (classical music, female voice and percussions) to be rated on a five point scale from very similar to very dissimilar. A CD with binaural samples of these simulations is attached to the appendix.

Results

The overall results were promising; the auralizations showed good representation of reverberation time, especially with percussion and music. Most of the times the listeners judge the samples as "similar" to "very similar". Although not as good as reverberation time, sound directionality still got close to "similar" with the exception of the female voice. The test for tone/timber was not as successful as the two previous tests; listeners said that even though the samples had different tone/timbre characteristics they could still get a somewhat similar feeling of the sound of classical music and percussion in the space. Out of the three sets of sound samples, the percussion was the most favorable while the female voice was the least helpful to listeners. Figure 23 shows results graphically in the results table used, table 8 shows the result values.

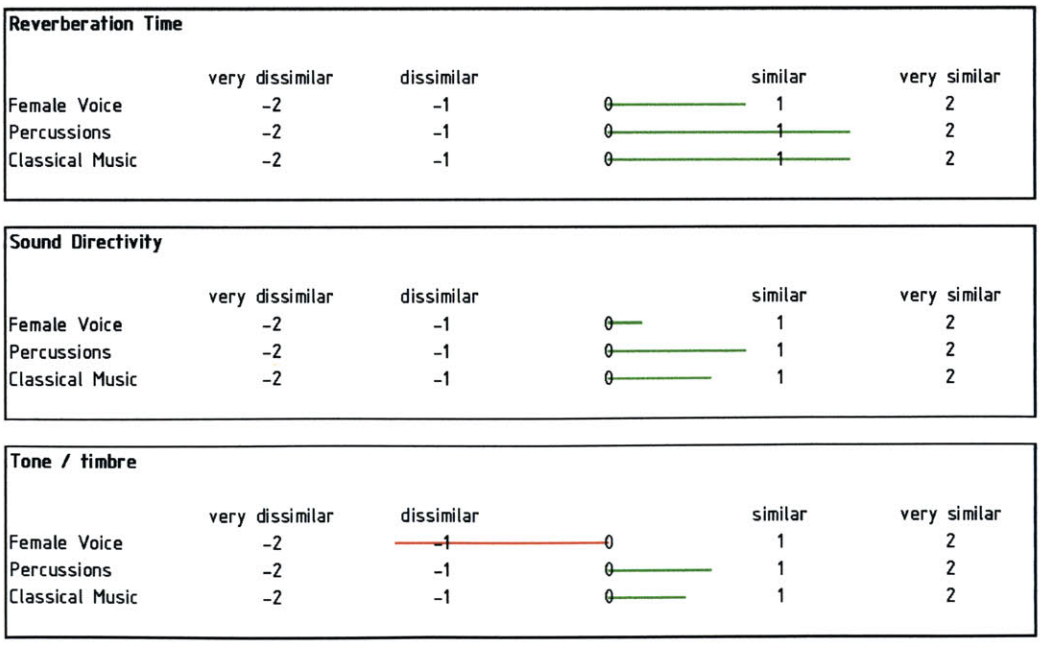


Figure 23, binaural auralization subjective rating table with graphical results.

Female Voice			Percussions			Classical Music		
RT	Directivity	Tone/Timbre	RT	Directivity	Tone/Timbre	RT	Directivity	Tone/Timbre
0.8	0.2	-1.25	1.4	0.8	0.6	1.4	0.6	-0.45

Table 8, binaural auralization subjective results.

4.5 Conclusions [chapter 4]

With the listening test, it was shown that auralizations developed by CATT-Acoustic are capable of reproducing the acoustical properties of a room. Many of the test subjects could not tell which sound sample was generated by the computer simulation and which was recorded on the physical space. Binaural recordings can be convincing in the reproduction of localization of a sound and reverberation time, however some differences between the recorded and auralized soundtracks were found as expected.

Some of the differences, mainly in the tone/timbre listening test, can be caused by the imperfections of the loudspeaker used as the sound source. CATT-Acoustic allows input of source characteristics only on central octave band frequencies (125, 250, 500, 1k, 2k, 4k, 8k, 16k) which leaves a large portion of the frequency response of the speaker out of the simulation process, primarily the lower frequencies and the events that occur in between such octave bands. The source data were acquired from the speaker manufacturer's website and some discrepancy can be expected, thus rendering to some extent different sound. In the future, work should be done to incorporate more of the source characteristics to better depict the tone/timber characteristics in an auralization.

However, for the purpose of an acoustical preview of the sound distinctiveness of a space and as a tool for identifying desired and undesired acoustical characteristics, auralization has proven to be successful in recreating an acoustical virtual environment with good resemblance to the real world.

Chapter 5
Ambisonics:
Virtual space, real listening.

5.0 Introduction

This chapter explains the basics Ambisonic computer auralization. It includes details on how human hearing uses several techniques to localize sound sources, how we can simulate factors that influence human auditory perception with computer software, and how we can reproduce the listening experience for a space that has not been built. It also includes a case study that creates a sound simulation of a space and allows the listener to subjectively "grade" the acoustical qualities.

5.1 Ambisonic, how it work

As stated before, devices capable of realistically simulating typical auditorium sound fields will prove to be a useful tool for understanding the acoustical qualities of spaces. They will help the acoustician as well non-acoustically versed people make decisions on projects that address acoustical issues.

By being able to control the significant acoustical parameters we can preview a proposed acoustical solution. Through the auralization it is possible to identify the objective parameters that correspond to certain subjective reactions experienced by listeners.

To determine the location of the sound source, our brain uses a number of different cues, among them are [Malham 1998]:

a. Sound reaches an individual's two ears at different times. As long as the source of the sound is not directly behind or in front of a listener, the sound will arrive at one ear before it arrives at the other. The time difference is known as "Interaural Time Delay" (ITD). This effect occurs only at frequencies where the wavelength is less than twice the distance between the ears (aprox. 1KHz); by this technique alone, humans are unable to determine the location of sound with longer wavelengths.
b. Sound reaches an individual's two ears at different sound levels. When a sound source is located to the side of a receiver, the sound reaches one ear directly. The other ear receives sound only after it has diffracted around the head. Sound arriving at the occluded ear will therefore be quieter.
c. The human brain can distinguish sound position relative to a phenomenon called the Head Related Transfer Function (HRTF). HRTF is a frequency dependent response that varies with source location, and is based on the shape of the head and the external part of the ears. "When the source gives an ambiguous ITD, this is the brain's main position-sensing mechanism" [Malham 1998].
d. By moving one's head, a listener can vary the ITD and adjust the HRTF between the ears, giving the brain more information to determine the sound source location.

Familiar systems like "Quadraphonic" and "5.1" surround sound are simply an extension of "intensity" or "amplitude" stereo. Sound is positioned between loudspeakers only with the use of relative level of two channels, just panning the sound to generate a phantom image of localization of sound. The problem is that, as stated above, the human ear and brain rely on numerous localization techniques at different, overlapping frequencies and is not satisfied by the use of level only as means of localization [Ambisonics.net 2004].

One of the best systems to recreate the acoustic qualities of a space is a surround sound alternative developed by a British research group (notably M. Gerzon and P. Fellgett) known as Ambisonics. It offers features impossible to realize through other methods. With this system it is possible to capture a sound event (such as a musical performance) and replay it such that, as far as possible, the original sound and

acoustical environment of the original performance is faithfully recreated. Ambisonics satisfies simultaneously as many as possible of the mechanisms used by the brain/ear to localize sound, reproducing them over practical loudspeaker systems in such a way as to fool the ears of the listeners into thinking that are hearing the original sounds correctly located. This can take place over a 360-degree horizontal soundstage (pantophonic system). An additional practical benefit is that the realistic listening area for Ambisonic Surround Sound is larger than that of conventional stereo [Elen 2001].

Using the so-called "B-Format" signals to carry the recorded information, an Ambisonics system generates a 4-channel signal that contains all the information in the soundfield (direction, delay, sound intensity, etc). These four channels record the event into left-right, front-back and up-down information plus a mono reference signal [Elen 2001]. Figure 24 shows the B-Format signal, channels X, Y and Z relate to the coordinates in a 3D space; W is an omni-directional signal used as reference [Malham 1998].

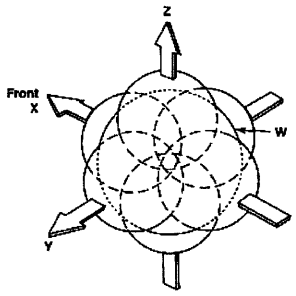


Figure 24 Graphical representation of B-Format [Ambisonics.net].

When replayed, the Ambisonic signal is processed and fed from a decoder to each speaker independently to create a horizontal surround. (A minimum of four speakers is required.) Each signal contains all the elements of the original recording but in different ratios, collaborating together to recreate the acoustical atmosphere of the original space.

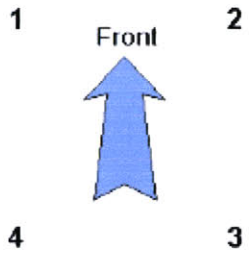


Figure 25, loudspeaker placement [Ambisonics.net].

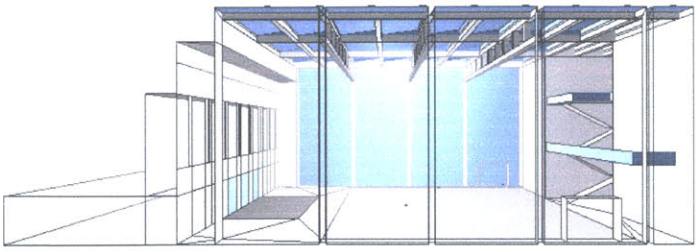
Ambisonic uses many of the methods of localization employed by the ear/brain combination to localize sound sources. There is a wavefront reconstruction to a degree. The results have noticeable benefits. First of all, you can put the speakers more or less where you want them (you tell the decoder where they are). Second, the surround effect is pronounced and stable over a very wide listening area. You can even stand outside the speaker array and experience a kind of "sonic image" emanating from within the array [Ambisonic.net 2004].

The important thing to note is that there is no need to consider the actual details of encoding and decoding when doing B-Format recording, synthesis or reproduction: if B-Format specifications are followed and suitable loudspeaker/decoder setups are used, "all will be well" [Malham 1998].

CATT-Acoustic [Dalenbäck 2002], an acoustical prediction software, is capable of generating a B-Format room response of a room which has not been yet built. Multivolver, a software application that works in conjunction with CATT-Acoustic, processes the room impulse response from the computer model, the source material (e.g. music or speech) and the loudspeaker layout of the test room where the auralization will be reproduced. The final result is a set of audible music files that are fed independently to each loudspeaker to recreate the desired environment.

5.2 Case Study

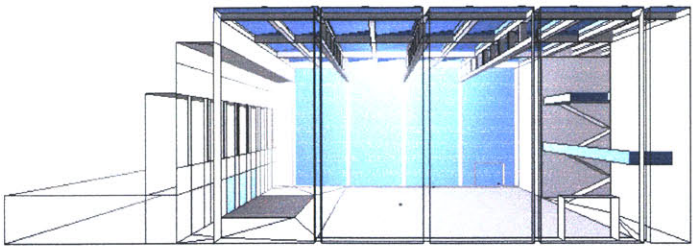
Atrim



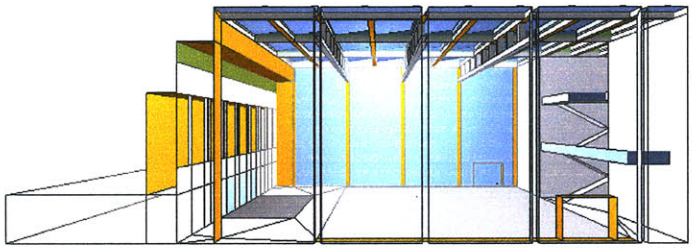
The need for an "acoustic test drive" arose in the development of a large atrium space. This enormous glass structure, of 800,000 cubic feet, is planned as a multi-purpose space where large dining events with music can be held. The lack of absorptive materials in the original concept, resulting in an unacceptably long reverberation time, was a concern shared by the owner, architect and acoustic consultant from the early stages of the design process.

The single most important variable that influences the acoustics in the atrium is the number of sound absorbing units, or sabins, in the space. A sabin can be defined as a totally absorptive area of 1 sqft. The absorption coefficient ranges from 0.01 to 1; fuzzy porous materials (velvet, glass fiber insulation) are on the higher region of the range while hard and heavy materials such as marble have a low coefficient. This study provided the tools to establish objective goals for the acoustical treatment in the proposed design.

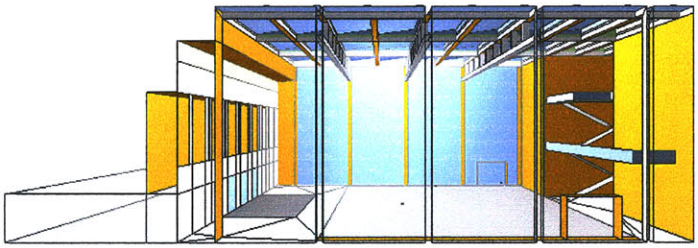
CATT-Acoustic was used to calculate the room response and acoustic signature of the modeled space. Three different versions of the modeled space were considered. Figure 26 shows the acoustical treatment in a darker shade (orange if in color) on each version.



T-0



T-1



T-2

Figure 26, Treatment schemes

T-0: all hard surfaces, no acoustical treatment, average absorption coefficient (total number of sabins evenly distributed over all surfaces) of 0.03 and a reverberation time of ~14 seconds (never actually considered, but useful for comparison).

T-1: Inclusion of sound absorptive treatments on one of the walls and some structural elements, average absorption coefficient of 0.18 and a reverberation time of 3.2 seconds.

T-2: treatments as proposed in Scheme T-1, plus an additional area of sound absorption, around 4,300 sabins, to represent the amount of treatment that is believed will achieve the goals of the owner. This version has an average absorption coefficient of 0.25 and a reverberation time of 2.7 seconds. The actual location of this extra material is on the wall located directly in front of the previously treated wall.

The particular auralization technique developed for this project allows the listener to hear the various conditions in the laboratory test room without having to wear headphones. The sound surrounds the listener as the sound would in the real space. An Ambisonic reproduction of the complete sound field one would experience during an actual event was generated and allowed reasonable comparisons of different acoustical conditions, replicating what a listener would hear in the projected space for each of the given conditions.

As source material, recordings of the sound of groups in a dining environment are used. The recording was done in a small, rather non-reverberant restaurant environment so that the room acoustics of the recording space did not influence the final auralization. The recordings were processed into the acoustic computer model so that it simulates the environment of 500 diners, situated at tables around the main volume.

At the acoustic consultant's laboratory, seven different types of events in the three differently treated spaces were developed. The laboratory is a semi-anechoic room treated with absorptive ceiling and wall panels as well as carpeted floor; the room characteristics prevent any reflection from the real space to hamper the virtual simulation. Figure 27 show an image of the semi-anechoic laboratory.

Figure 27, Acentech's laboratory and dummy head.

For the simulation, four B-Format decoded signals were fed to four loudspeakers situated in a horizontal square array. This created a Pantophonic (360°) system that recreated the sound stage of the future atrium. This helped key persons participating in this project make their own judgment of the relative effect of different amounts of sabins in the space. Table 9 is a table given to the participants to select any of the 21 auralization available.

	T-0	T-1	T-2
Estimated Reverberation Time (sec)	~ 14.0	3.2	2.7
Estimated Average Absorption	0.03	0.18	0.25
Auralization Choices:			
Banquet, 500 diners only			
Banquet, 500 diners, plus classical			
Banquet, 500 diners, plus jazz			
Banquet, 500 diners, plus rock			
Banquet, 500 diners, plus solo			
Banquet, 500 diners, amplified speech			
Typical use, just passers-by			

Table 9, Auralization table.

Figure 28 is intended to orient the listener to the actual position of the different sources as well as the receiver in the "Atrium". When listening to the auralization simulation in laboratory test room, one experiences the atrium (virtually) as if one were

at the location marked with the arrow. The direction of the arrow indicates the direction of the front of the simulation room (toward a rear-projection screen that showed the image of the version being heard). Dots indicate example locations used for creating the impression of diners, speakers, and music within the space during a typical banquet.

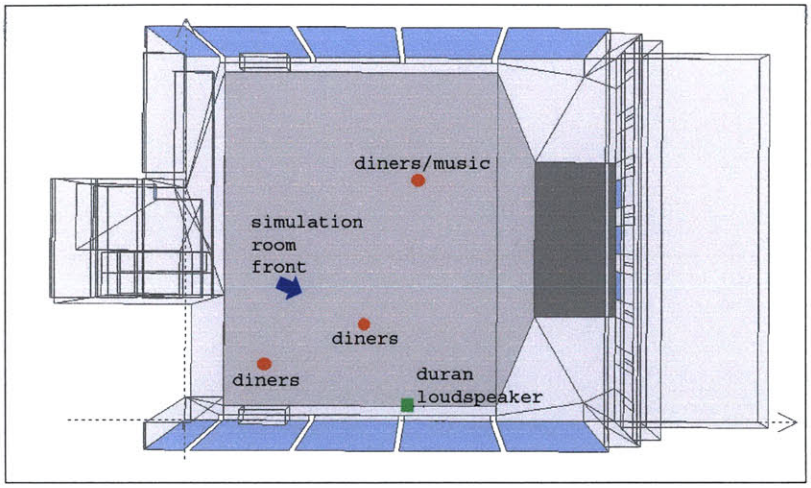


Figure 28, Simulation layout of source and receiver locations, Atrium.

The result was a fair and accurate representation of the various acoustical conditions and ambience. A clear difference was perceived by the listeners; some even experimented by having conversations while in the laboratory simulation such as they would have at a cocktail party. With versions T-0 and T-1 it was difficult to perceive the music playing in the background, and it was very uncomfortable to carry out a conversation. Although T-2 was by no means quiet (remember it is 500 diners plus music) it was much more comfortable to speak to individuals seated on the opposite side of the table, and the music could be clearly distinguished. The extra absorption also had the added advantage of noticeably reducing the intensity of the ambient noise levels.

The feedback from the listeners was very positive and they commented favorably on the help this simulation rendered for the decision-making process. Version T-2 with the extra 4,300 sabins of treatment was selected as the most desirable option and is now incorporated in the design. A CD with binaural samples of these simulations is attached to the appendix.

5.3 Conclusions [Chapter 5]

Previously acoustics could be measured, quantified and charted, but with the help of auralization the subjective character of sound can now be simulated, experienced and judged. In the past, acoustical consultants could only try to convince the client/architect with calculations and geometrical plots that they could create an acoustically exceptional space.

In the simulation of the unbuilt space, architects and acousticians collaborated to create iterations of the material treatment until the acoustic goals were achieved. The client was able to identify the influence of sound reflections and different material treatment schemes and relate them to a subjective impression of the sound in the proposed space; to do this you have to be able to listen to what the space will sound like.

The auralization techniques offer the possibility to use the ears and listen to the acoustics of a room during the design process. Several acoustical problems can be detected by the ears, whereas they may be difficult to express with a parameter that can be calculated. Using these tools the acoustician can communicate the acoustic consequences of a design to the client/architect effectively. This technique can be used very early in the project to achieve the desired results.

Nevertheless, it is important to state that this is still only a model. With listeners focusing on the acoustical tests, and in a visually artificial environment, the experience can obviously never be totally representative. We must always remind the participants of these physical and psycho-acoustic limitations of the model.

Chapter 6
General Conclusions

Although there are independent conclusion on every chapter, this section will present a few closing comments.

It was shown that a computer model is extremely flexible and much more accurate than previous methods of predicting the acoustic qualities of a space. It was also observed that computer model results can generate several iterations very fast once the model is set up and can render several numerical parameters as well as auralizations making it possible to subjectively judge the sound of a space. It has been demonstrated that a quite good agreement between calculated and measured results can be achieved and that CATT-Acoustic provides stable and reliable predictions over a wide range of acoustical parameters.

It is clear that the CATT-Acoustic software is an important step in the prediction and understanding of the acoustics of a room and that it comes very close to the real measured characteristics of a room. In the case of reverberation time is roughly as good as any two consecutive physically based measurements. The prediction of many parameters and the accuracy of the echograms generated by computer models make them the tool of choice for the acoustician.

While it is clear now that computer modeling technologies have reached a state where acoustical consultants can model a hall to search for positive characteristics and eliminate bad ones, a reliable value will only be obtained if the absorption coefficients of the surfaces are well approximated. So, as future work in the area, more extensive and more accurate absorption coefficient should be developed. Also, modeling software must allow the input of more reliable and accurate information on the directivity and frequency response of sound sources that must be obtainable from the manufacturers.

In the case of a parameter with strong local positioning sensitivity such as the clarity indexes, we now comprehend that accurately modeling how sound acts in the presence of very complex structures, such as a coffered ceiling or an audience plane, will hinder the reliability of the results. Knowing this enables us to determine that results for these values should be taken as a comparison between proposed solutions rather than literally as a numeric value.

As for binaural auralizations, there is now an answer to the age-old question "How good will the sound of a room be?". With the aid of this document, it was revealed that room acoustic qualities are well presented and the techniques have matured to a degree to which a normal human listener will have difficulty telling whether it is a simulation or not. However, some differences between a recorded and an auralized soundtrack can be encountered, as expected.

By certifying the auralizations with listeners' reactions it was apparent that reproduction of the acoustical feel of a space can be successfully reproduced. Either with binaural or with Ambisonic technology, accurate recreation of a virtual environment with resemblance to the real world can be achieved. As the ultimate judge of the sound quality within a space will be a human listener, tests directed to the ears will prove useful in the evaluation and detection of desired/undesired sound behaviors.

This thesis will help acousticians to feel comfortable with this new technology, helping them understand the benefits and limitations of the computer models. It also shows that computer models can deliver new ways to communicate the acoustical implications in a project. Unlike in the past, the acoustician no longer has to try to convince the client to blindly, or better said, deafly, trust him to generate an acoustically exceptional space.

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Appendix

Attached to this document is a CD containing sound files of the auralizations. The files are encoded as MP3s and can be played back through any computer. The first 8 soundtracks are the comparisons done on chapter 4. This set of 4 pairs of auralizations will help the listener understand the difference that exists between a dummy head recording done in a physical hall and the simulation created through a computer model of that same space. The soundtracks included are:

1 Voice Real.mp3
2 Voice Model.mp3
3 Percussions Real.mp3
4 Percussions Model.mp3
5 Classical Real.mp3
6 Classical Model.mp3

Also included on the CD are dummy head recordings of the Ambisonics simulation done for the case study on chapter 5. These simulations were recorded through a dummy head so that anyone can listen to the sound samples through headphones and get a sense of the 3D virtual environment. Figure 29 shows an image of the dummy head recording the sound simulation at Acentech's laboratory.

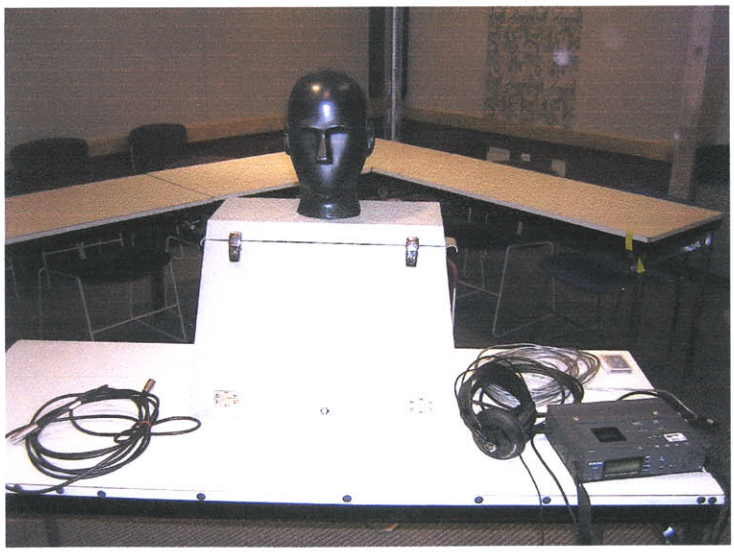

Figure 29, Dummy head recording of Ambisonics simulation.

A pause in the sound simulations will indicate the switch from one computer model to the next, T-0 to T-1 to T-2. The simulation soundtracks are:

7 500 diners+Jazz.mp3
8 500 diners+Classical.mp3
9 500 diners+Rock.mp3
10 100 passers-by.mp3