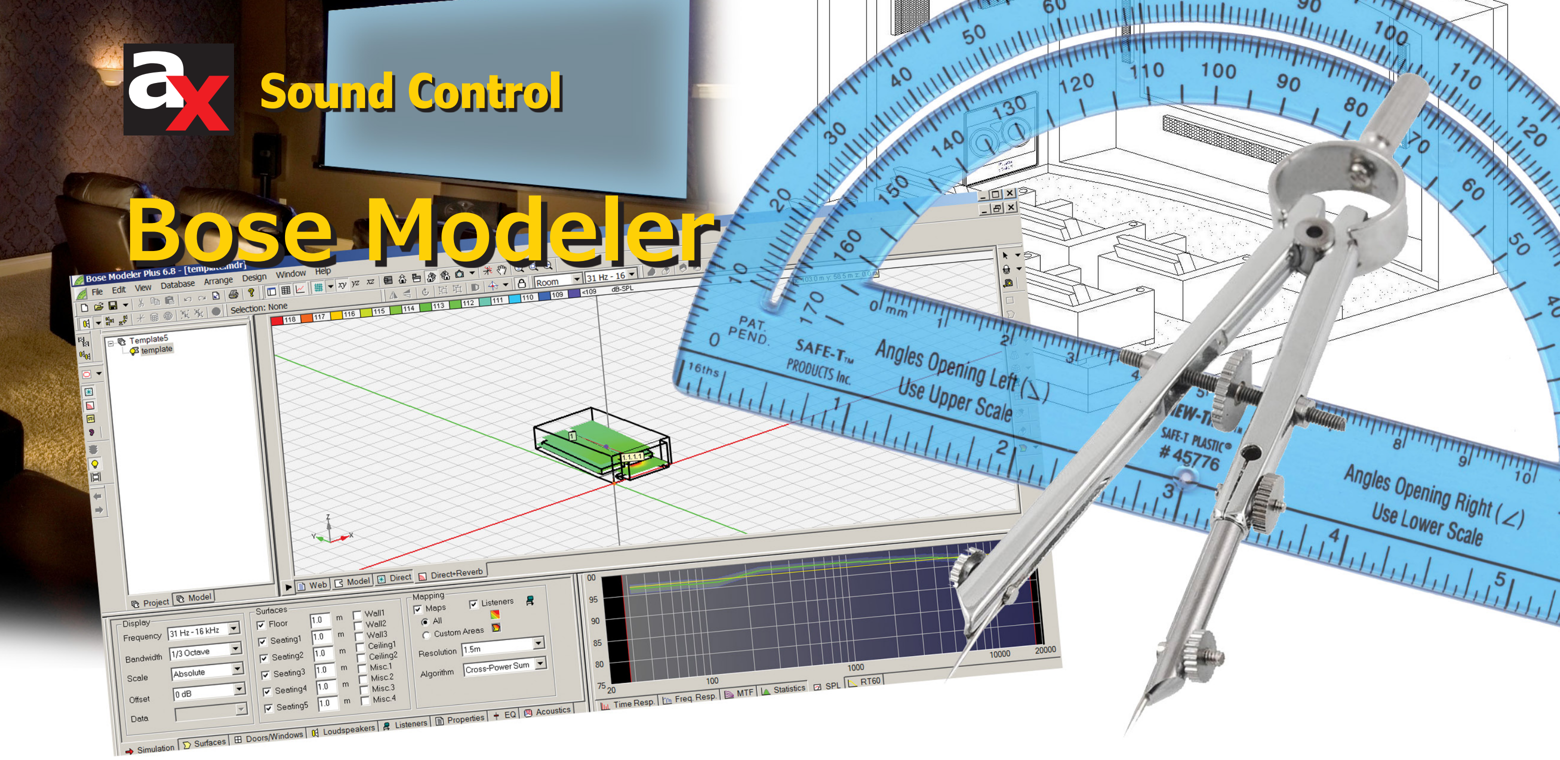
Bose Modeler



Bose Modeler electroacoustic prediction software is one of the earliest ray-tracing computer acoustical modeling programs. It was first released for MacIntosh in 1986. The current Windows version is Modeler 6.8. It is designed to be used as an aid to scientific sound-system design and as a sales tool for sound contractors (especially in regard to its auralization capabilities). It is not intended to be a complete room-acoustics modeling program.

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Modeler's current version provides an implementation of what Bose calls the D^{2}R∆STIc procedure for designing a sound system:

- Calculate **D**irect coverage
- **C**alculate **D**irect + **R**everberant coverage
- Predict time arrivals—direct, first-order reflections, and second-order reflections (∆)
- Predict speech intelligibility using the speech transmission index (**STI**)
- **C**onfirm the design by auralizing, using Bose Auditioner

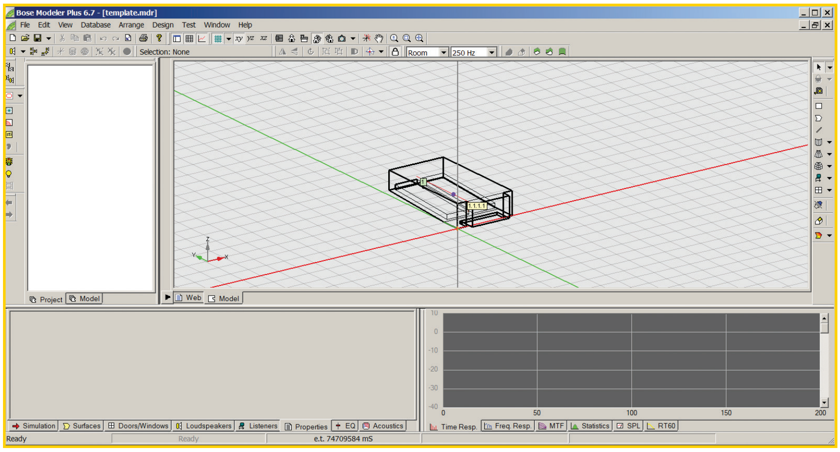


Photo 1: A screen shot similar to this one appears once you build or import a model.

Upon opening Modeler, a screen pops up containing a grid on which you can build your model by clicking on corners (with the aid of X, Y, and Z coordinates) in the XY, YZ, and XZ planes. You can build the floor and extrude the walls. You can add material characteristics such as absorption and scattering in octave bands from 31 Hz to 16 kHz. Alternatively, you can import an EASE or DXF model if you already have one or prefer modeling in EASE or CAD. Successfully importing a DXF model requires that there be a materials layer in the DXF file.

Once you have built or imported the model, you will see the screen shown in **Photo 1**. The model shown in the photo is a small shoebox auditorium with a raked floor and a small stage. The box containing the numeral 1 represents a listener position; the wider box containing 1.1.1.1 represents the omnidirectional loudspeaker. Modeler also offers the ability to view the model in plan or to reverse the isometric view.

You can click on the Place Loudspeaker icon, and then click on the model to place the loudspeaker. A dialog box will open for you to input the speaker's coordinates and select the speaker model. Clicking Database|Loudspeaker Database brings up the box shown in **Photo 2**.

Material Selection

Materials can be selected from the software's material database. You can also enter data for new materials. Next, you can click the RT60 button in the lower right. A graph similar to **Figure 1** will appear in the associated space on the screen.

Modeler includes a process to automatically make the predicted reverberation time (RT) match measured RT values by modifying material properties. This process can be useful, but must be intelligently used. It makes sense to modify absorption properties for which the values are not well-known (e.g., audiences and draperies), but the absorption of some other materials, such as painted concrete floors, are well known and should not be modified in an attempt to force predictions to match measurements.

In any real room, the effects of a material's increasing absorption in a certain octave band will depend on the material's location in the room. The Modeler procedure enables the user to choose which material to adjust and even which frequency for that material. Materials can be "locked" so their absorption will not be changed. The maximum permitted change in reflection strength is 3 dB every time the matching cycle is run.

If you have included speakers and seating areas, you can right click on the model and select "Direct (1 to 4 kHz)" from the D²RΔSTIc pop-up list. The speaker's direct-field coverage will then be displayed on the model (see **Figure 2**). Since this model uses an omnidirectional source, the direct coverage is symmetrical from side to side, and of course the sound pressure level (SPL) decreases with distance from the source.

In this model, the room has hard walls and ceiling so sound levels experienced by the audience will be significantly affected by the reverberant field also. Modeler also includes a "Direct + Reverberant (31 Hz to 16 kHz)" option in the D²RΔSTIc pop-out list. For this model, the results are shown in **Photo 3**.

Notice that there is a frequency response curve shown in **Photo 3**'s lower right corner. A similar curve is also included in the direct response display, although it was not included in **Figure 2**. Fans of omnidirectional speakers may notice that the direct:reverberant coverage is uniform within about 1 dB over most of the audience area. However, the omnidirectional speaker puts a hot spot near the front of the stage, which would make feedback squeal a problem. If SPL were the only consideration, a single loudspeaker with a wide horizontal coverage could be used in this room.

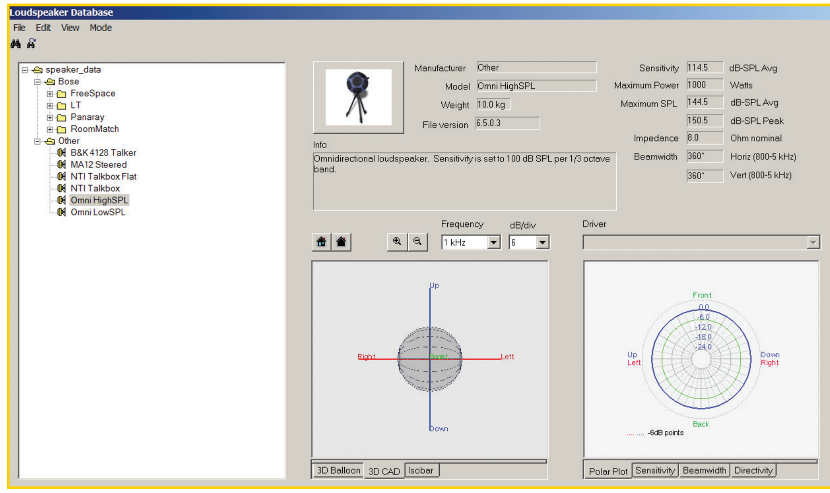


Photo 2: Modeler's loudspeaker database contains technical details for several loudspeakers.

Sound-System Design Factors

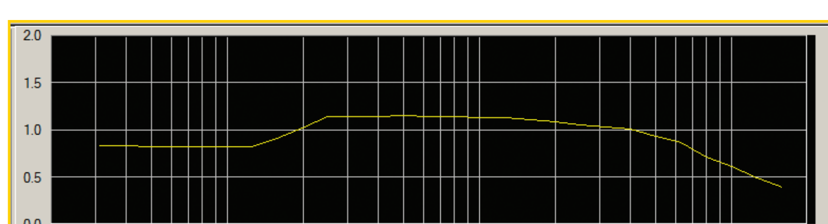
Late-arriving sound (greater than 50 ms after the direct sound), noise, some types of frequency-response anomalies, and distortion created in the sound system are also important considerations in sound-system design. These can be quantified by speech intelligibility measurements.

Most modeling programs, Modeler included, assume that the sound system is correctly set up so distortion is not a factor. Modeler includes a method of incorporating EQ to eliminate frequency-response anomalies. Modeler and most other modeling programs permit the user to enter specific noise signatures (noise levels in various octave or one-third-octave bands) so the noise can be included in speech-intelligibility calculations. STI is the most nearly standard metric for speech intelligibility. Modeler includes its most comprehensive tool for examining late-arriving sound under the STI rubric.

To examine the late arrivals at a specific listening point, you can click the STI button on the left-hand side of the screen, click the Time Response tab near the bottom center of the screen, then choose a single frequency in the drop-down menu. The menu can be accessed via the Simulation tab at the lower left of the screen. **Photo 4** shows the resulting image.

In the graph at the lower right of the illustration, the red vertical line represents direct sound from the loudspeaker. If more than one loudspeaker were used, there would be multiple red lines, unless all

Figure 1: Modeler shows the model room's predicted reverberation time (RT).



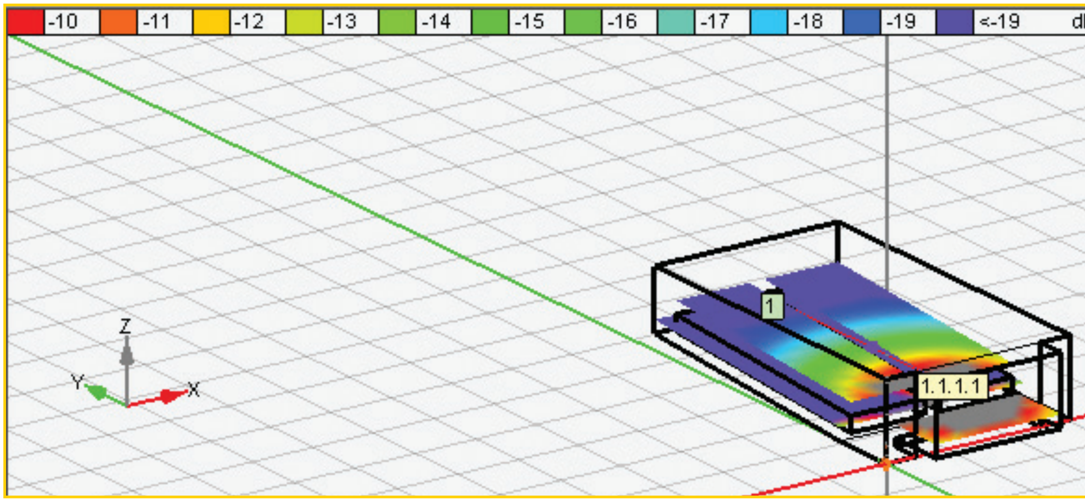


Figure 2: This is the direct-field coverage from an omnidirectional source in the small auditorium.

speakers were time-aligned. Green lines are first-order reflections (i.e., sound that has reflected off one surface). Blue lines are second-order reflections, which is as far as Modeler tracks reflections. The yellow line shows how reverberant energy builds and decays in the room, and is called the reverberant field envelope function (RFEF).

In this model, the latest second-order reflections are almost exactly 50 ms later than the direct sound, so they would not be heard as echoes. Higher-order reflections could be perceived as echoes, if they arrived more than 50 ms after the direct sound and had sufficient level. However, Modeler does not examine these late reflections, nor does it include an algorithm to predict the listener disturbance resulting from an echo at a given delay time and SPL.

Acoustic Imaging

Correct timing and SPL of late arrivals is also important for proper acoustic imaging: making the sound seem to come from the actual acoustic source (e.g., the talker, singer, musician, etc.) rather than from one side or the other. The earliest-arriving sound's arrival direction is considered by the ear and brain to define the source's location. False directional cues are caused by strong reflections or secondary loudspeakers with incorrect delays.

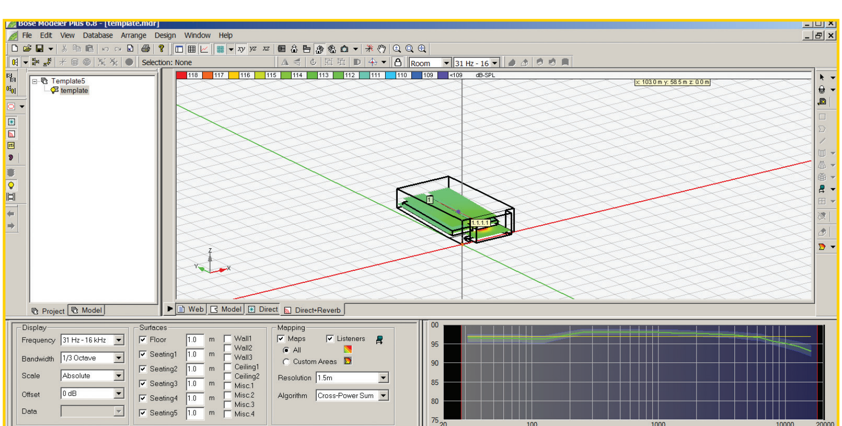


Photo 3: The direct + reverberant coverage is usually much more even than the direct coverage.

Clicking F8 causes Modeler to present the STI map and data, per the 1988 standard. It also predicts per the 1998, the 2003, and the 2011 standards for both male and female voices.

For most of the room, STI is about 0.55, which is considered fair speech intelligibility. It is in the good range (STI = 0.6 to 0.75) in the front one-quarter of the auditorium. This distribution of STI values is likely caused by the use of an omnidirectional speaker, which provides a poor direct-to-reverberant sound ratio for most of the room.

The graph at the lower right of the screen shows the modulation transfer function (MTF) in the various octave bands. The bands are color-coded: red = 125 Hz, orange = 250 Hz, yellow = 500 Hz, green = 1 kHz, blue = 2 kHz, indigo = 4 kHz, and violet = 8 kHz.

The MTF is a representation of how much the signal is degraded by the sound system and room. It is plotted as the decimal fraction of modulation (information in the signal) that is preserved (vertical axis) vs the modulating frequency. Perfect conditions would be indicated by an MTF of 1 at all frequencies. (It would appear as a horizontal line across the top of the graph.) Reverberation causes a downward-sloping MTF (see **Photo 5**).

In this model, the room is not particularly reverberant, but the speaker does not have the directivity needed to put enough direct sound into the rear of the audience area. Background noise causes a decrease in MTF at all frequencies. Strong echoes cause a notch in the MTF. Thus by examining the

MTF, we can identify the cause(s) of poor speech intelligibility.

Auralization

The last step of the D^{2}RΔSTIc procedure is to confirm the design by auralization. Auralization involves convolving some anechoically recorded sound with the room's impulse response. With music performed live in a venue, the music itself is free of echoes, and the process of playing it in the room acoustically convolves it with the room's acoustics. Auralization software performs the convolution mathematically, using the impulse response derived from ray tracing.

For the auralization experience to be valid, the listener must be in a controlled acoustical environment; otherwise, the listening room's acoustics will color the results to an unacceptable degree. One way to control the listening environment is to incorporate binaural processing into the wave file produced during auralization, and ensure that the listener uses excellent headphones with good rejection of external sound. Here's the rub: Few clients (with the exception of recording and broadcast studio owners) have—or even know how to identify and where to get—excellent headphones, even if they were willing to spend the necessary money for them.

Bose created a portable set of speakers with on-board electronics to provide an acoustically controlled environment for listening to auralizations. This system is called the Bose Auditioner (see **Photo 6**).

Modeler is designed so that it will only permit auralization if an Auditioner system is connected to the computer, thus providing a measure of quality control for the auralization experience.

Overall Function

As stated earlier, Bose's Modeler is a sophisticated software tool designed to facilitate scientific design of sound systems. It performs this function well, with a friendly user interface. The calculation time for each function is extremely short. Model input is straightforward once users become familiar with the method. The other option is to import a DXF or EASE model.

Some non-Bose speaker data in the CLF or EASE formats can be imported into Modeler. GLL and DLL data cannot be imported. A special key is required to permit importing non-Bose speaker data.

Modeler is not designed as a general-purpose acoustical modeling program and does not include functions such a program would have including the ability to view echograms and ray traces of reflections beyond second-order, EK grad echo analysis, calculation of strength (G), clarity (C80), definition (D50), mean free path analysis for analyzing reverberation, walkthrough auralization, or auralization using headphones or external speakers.

A prospective user should carefully evaluate his/her needs before choosing Modeler or any other modeling program written by a loudspeaker manufacturer over a general-purpose modeling program. Modeler and software by other manufacturers such as Yamaha and Danley can be cost- and time-effective solutions for sound system designers who do not need the extended capabilities provided by the programs designed for more general applications.

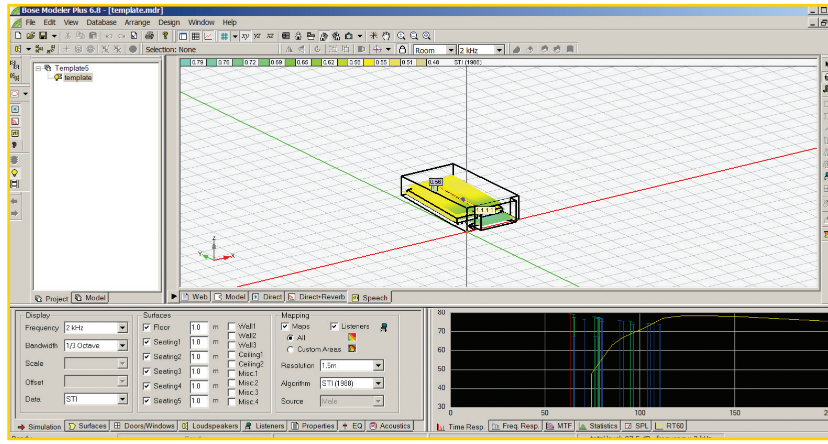


Photo 4: Modeler can enable users to examine late sound arrivals.

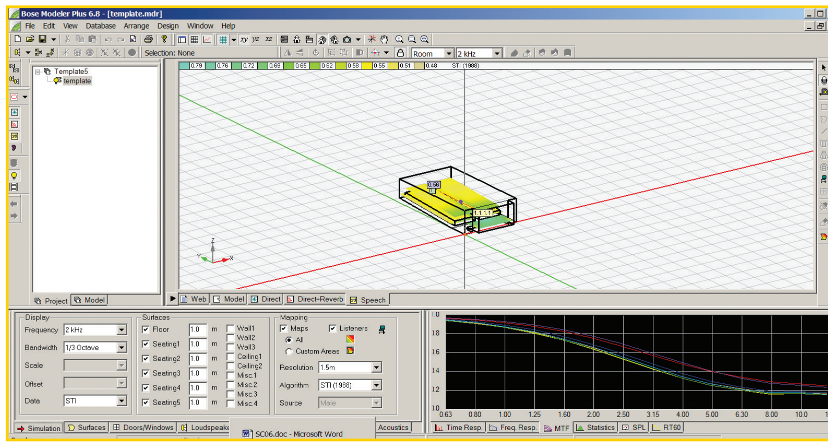


Photo 5: Modeler includes a significant amount of STI data in a single screen.

Photo 6: The Bose Auditioner is designed to provide a controlled environment for listening to auralizations. (Photo courtesy of Bose Corp.)