Loudspeaker Complex Directional Response Characterization

William R. Hoy and Charles McGregor

Eastern Acoustic Works, Inc.
1 Main St., Whitinsville, MA 01588
PH: 508-234-6158 FAX: 508-234-6479
e-mail: bhoy@eaw.com, chuck.mcgregor@eaw.com

ABSTRACT

For complex directional response data to be useful, it must be gathered and deployed in a much more disciplined manner than has typically been applied to magnitude-only data. The loudspeaker under test and measurement microphone must be precisely positioned; geometrical errors must be corrected; and, temperature variations must be accounted for. An object oriented data structure is described which facilitates solutions to each of these challenges. Practical applications employing the new data structure are also presented.

0 INTRODUCTION

0.1 Standard Practice

The standard techniques of measuring and reporting loudspeaker polar data are not accurate enough to predict array interaction. Following standard procedure, most loudspeakers are measured by placing them on a turntable as close to the center of rotation as possible [1]. A microphone is then positioned in the far field, as far away as possible, so as to reduce apparent apex error and near field response anomalies [2]. As the speaker rotates, the frequency response is measured at fixed angular increments. The data level is normalized and stored in a data table: frequency and turntable angle vs. response magnitude.

In the past, modeling programs have used these data tables to predict how the loudspeaker will behave in various rooms. Given a location in the room, the modeling program calculates the vector from the speaker's location to the observation point in the room. The program then calculates the point at which this ray intersects the balloon and looks up the response in the data tables. Then, if needed, it interpolates the magnitude from the data table and adjusts it according to the length of the vector using the Inverse Square Law.

While the data table technique can be used with minimally at least two bands of measured polar data, typically horizontal and the vertical, sometimes full balloon data is also employed. However, it is cumbersome and time-consuming to gather full balloon polar data. Even with this data, most modeling programs still need to perform interpolation, using the magnitudes of the frequency responses found in the polar data. Such modeling programs are not able to predict accurate array interactions using the data tables, especially for sources that are large relative to the wavelengths of interest. This is because time domain (phase) information is neither part of the data nor taken into account by the modeling program algorithms.

0.2 A New Technique: Loudspeaker Directional Response Model

In order to obtain more accuracy in loudspeaker modeling and array modeling, it is necessary to collect complex frequency response data. This includes the electro-mechanical and propagation delays that occur before the wavefront emanates from the sound source. Complex data must be gathered in a much more disciplined manner than magnitude-only data. This means that the loudspeaker under test and the microphone must be positioned with high precision, all geometrical errors must be calculated and removed, and the air temperature during the measurement process must be accounted for. In this paper, we propose a new method of constructing loudspeaker directional models that will accurately provide the complex transfer function for any point in space around the loudspeaker.

The loudspeaker directional models that we construct are object-oriented and hierarchal in nature. Object-oriented refers to the fact that our models contain both raw data and data processing algorithms. Fig. 1 shows the layout and contents of each model.

Loudspeaker Directional Response Model
- ***Spherical Response Object 1 (Balloon Objects)***
    - **Polar Response Object – 1**
        - Complex Frequency Responses
    - **Polar Response Object – 2**
        - Complex Frequency Responses
    - **Polar Response Object - n**
        - Complex Frequency Responses
- ***Spherical Response Object 2 (Balloon Objects)***
    - **Polar Response Object – 1**
        - Complex Frequency Responses
    - **Polar Response Object – 2**
        - Complex Frequency Responses
    - **Polar Response Object - n**
        - Complex Frequency Responses

Fig. 1. Loudspeaker Directional Response Model Hierarchy

A loudspeaker model consists of several **Spherical Response Objects** called **Balloon Objects**. Each component in the loudspeaker is measured separately because superpositioning will be applied. Thus, each has its own balloon object. Inside each balloon object are **Polar Response Objects**, with at least two polar responses or slices taken at 90 degrees relative to each other. Additional polar slices can be included in the model. In turn, each polar response object contains the measured complex frequency response data that has all geometric errors accounted for and removed. These errors are corrected by implementation; data is "as measured." In addition to this, the temperature for each frequency response measurement is also recorded. Using all of this data, a much more accurate loudspeaker model can be constructed.

1 The Data Acquisition Process

1.1 The Measurement Setup

At the very heart of the loudspeaker directional models are the complex frequency responses that are collected in the laboratory. This data is used for interpolation and prediction of balloon data in the models. The setup used to collect this data consists of permanently fixed components: a turntable with a Genie lift bolted to it, and a microphone mounted at a fixed point in the far field. The loudspeaker to be modeled is divided into its separate components, each transducer comprising a "cell." The loudspeaker is placed on the turntable with each of the transducers wired and measured separately and independently, to avoid acoustical interaction between them. Each transducer is referred to as a "cell". This method also provides greater flexibility for any post processing that might be applied to the data.

Data is collected at various turntable angles using a custom Polar program. The Polar program automates rotating the turntable and collecting the complex frequency responses. At each rotation increment, a synchronous noise signal is sent out to the loudspeaker. An FFT with 32,768 points, sampled at 44.1 kHz, is employed. By using the longest dimensions of the room and a windowing function, all room reflections are removed from the data, leaving behind only the direct sound from the device under test.

Since the data will be used later in an interpolation process, the turntable angles at which data is collected is important. On one hand, more data allows more precision. On the other hand, more data also requires more time to be spent collecting data and performing calculations. More data is also subject to possible measurement artifacts resulting in bad data points. To improve the efficiency of the data collection, data is reduced in amount by eliminating less critical data. To this end, the data acquisition software allows the turntable angles to be broken up into three general categories: within the beamwidth, beyond the beamwidth, and the rear hemisphere. This allows data to be collected at close increments (for example: 2.5°) within the beamwidth where precision is most desirable. Where less precision is necessary, wider increments can be used outside the nominal beam and for the rear hemisphere, (for example: 5°, 10°, or 15° intervals). Polar slice data is thus collected for each cell in the loudspeaker.

1.2 Parameters for Each Measurement

To make the loudspeaker models more accurate, the phase of the frequency response is recorded along with the magnitude. The phase will allow for the required superpositioning accuracy when the data is convolved to model the complete loudspeaker system. The phase information also allows accurate prediction of acoustical interaction between complete loudspeaker systems. However, measuring phase requires precise and repeatable physical positioning of the loudspeaker and measurement microphone for the data collection process. For results to be valid, the phase error must be within 30°. For 8 kHz, according to equations (1) and (2), this translates to a maximum physical error of ±0.3583 cm. Fig. 2 shows *that* the magnitude error when adding two sine waves 30° out of phase is only 0.30 dB.

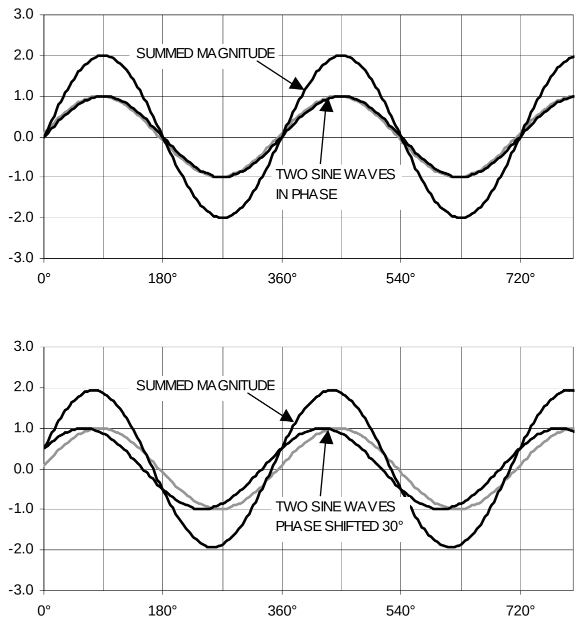


Fig. 2. Sine wave magnitudes added in phase (top) and phase-shifted 30° (bottom)

$$\lambda = \frac{c}{f} = \frac{344m/\sec}{8000Hz} = 0.043m \quad (1)$$

$$\frac{30^\circ}{360^\circ} \times 0.043m = 0.003583m = 0.3583cm \quad (2)$$

Hence, we must know the exact location of the sound source relative to the center of rotation for each measurement. The Polar program requires that this information be loaded before any measurements can be made.

Several loudspeaker distance parameters are thus recorded so that geometrical errors that occur during the measurement process can be corrected.

The position of the each loudspeaker's cell on the turntable is recorded accurately. This involves using a CAD drawing, or physically measuring the enclosure, to get the locations of each cell in the loudspeaker. Fig. 3 shows the dimensions of a sample four cell loudspeaker consisting of four HF transducers each reproducing the same passband. All measurements are made with reference to the center of the face of the enclosure.

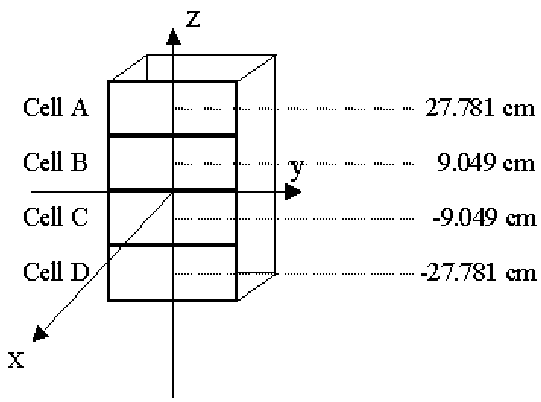


Fig. 3. Loudspeaker Cell Locations

In the figure, X is in the direction of the loudspeaker to the measurement microphone. Y represents the distance the cell is horizontally off center. For this loudspeaker, all cells are centered at Y = 0. Lastly, Z represents the distance the cell is vertically off center. Collectively, X, Y, and Z approximately locate the cell being measured in 3D space.

Since the microphone is placed in a fixed position, enough information is known to calculate an actual microphone distance and microphone (off-axis) angle, as shown in Fig. 4. The sound source is located at (x_c, y_c, z_c) and the microphone is located at: (x_m, y_m, z_m).

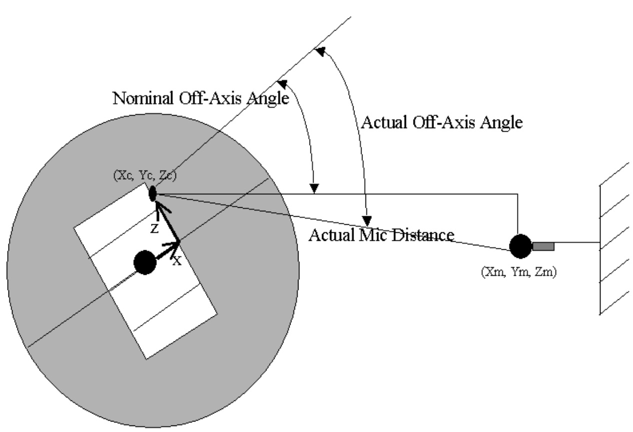


Fig. 4. Calculating the Actual Microphone Angle and Distance

The calculations are shown in Equations (3) and (4). For every turntable angle and each cell, these two parameters can be calculated.

$$MicDistance = \sqrt{(x_m - x_c)^2 + (y_m - y_c)^2} \qquad (3)$$

$$MicAngle = \tan^{-1}\left(\frac{y_m - y_c}{x_m - x_c}\right) \qquad (4)$$

The inclusion of these two parameters for each complex directional response measurement frees the model from the apparent apex error that occurs if the center of each cell is not placed at the center of rotation. Turntable angles are not actual microphone angles. Thus, a 45° turntable angle will not be an actual 45° measurement if the cell is positioned to the left or right of the center of rotation. Each measurement made is tagged with the actual microphone distance and the actual microphone angle. When the data is used for modeling, the turntable angles will not be used; the actual distance and microphone angles will be used.

2 The Spherical Response Object

2.1 FChart Basics

When the Polar program data collection process is complete, a hierarchal object has been created that can be loaded into FChart. FChart is an object oriented Windows based application written in C++. It is essentially a one-dimensional spreadsheet in which all values are a function of frequency. It can display transfer functions in the frequency and time domains. FChart contains a virtual 3D space in which sound sources, such as tessellae, can be positioned. FChart and its handling of sound sources have been the subject of two previous papers [3] [4].

The loudspeaker directional response objects are handled in FChart in the same manner as other sound sources. The objects are not simply sets of frequency response curves but serve as 3-dimensional models of sound sources. In addition to containing enough response data to fully define the 3-dimensional response of each source, the objects also contain the procedural code required to interpolate any requested data point.

2.2 Contents of the Spherical Response Object

Fig. 5 shows a tree view of the contents of an example loudspeaker directional response object file, simplified to only show the horizontal and vertical polars.

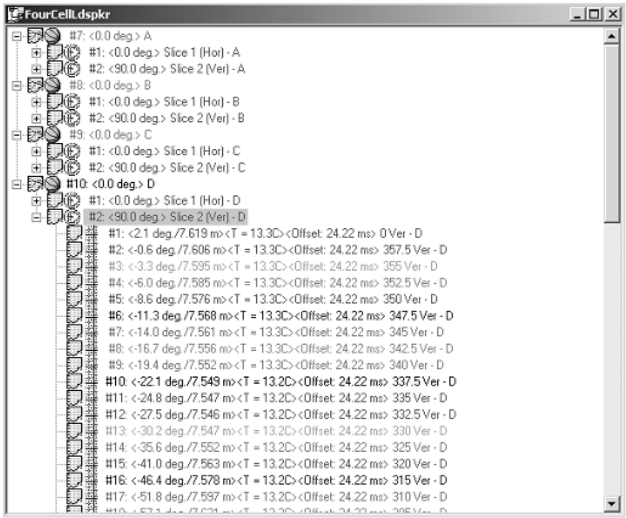


Fig. 5. Tree View of a Simplified Loudspeaker Directional Response Object

This object is a four cell high frequency loudspeaker, with its four cells labeled: A, B, C, and D. At the top of the tree is the name of the object. The next level contains the Spherical Response Objects, or balloons, denoted by . There is one balloon for each cell in the measurement. Inside of each Spherical Response Object are Polar Response Objects, or polars, which are denoted by . This example contains a minimal number of slices: the horizontal and vertical. Lastly, underneath each Polar Response Object is the collection of frequency response curves collected in the laboratory, as explained in Section 1, The Data Acquistion Process.

Note that each frequency response curve is tagged with a label containing certain information. For example, Curve #5 for cell D's vertical Polar Response has the following label:

<-8.6 deg/7.576 m><T = 13.3C><Offset: 24.22 ms> 350 Ver – D

The first two pieces of information are the actual microphone angle and the actual microphone distance. As explained in Section 1.2, the Polar program calculated the actual distance between the cell and the microphone to be 7.576 m, and the actual angle between the two to be -8.6°. "T" equals the room temperature and "Offset" equals the propagation delay between the onset of the test signal and its receipt by the measurement microphone.

The last information listed in the line above is "350 Ver-D." This is the turntable angle and a description of what cell was being measured. It can be seen that due to apparent apex error, this cell was not positioned at the center of rotation. Thus, the turntable angle does not equal the actual angle of measurement. The -8.6° (actually 341.4°) will be used in subsequent calculations, not the 350°.

Each spherical response object is programmatically considered to be one sound source and has the parameters listed below.

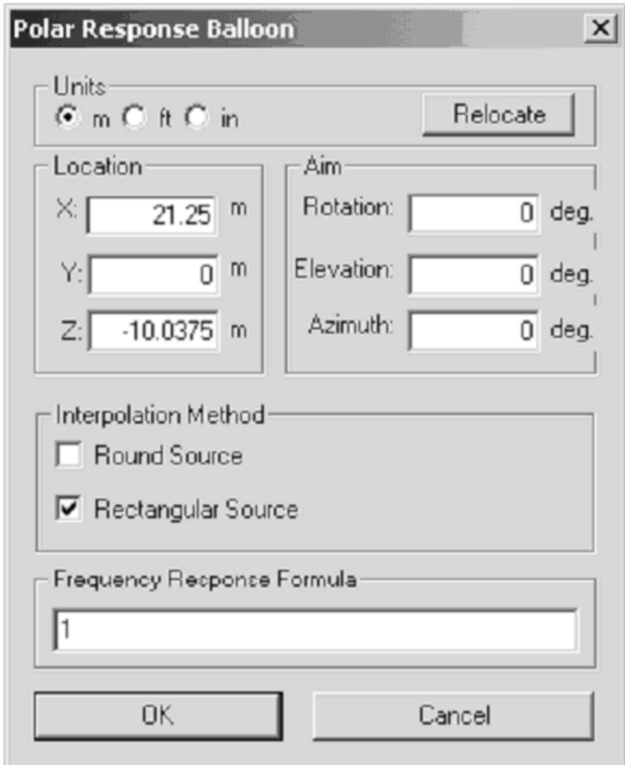


Fig. 6. Spherical Response Object Parameters

These are similar in fashion to the parameters used for a tessella. Each balloon object can be located in space. Thus, for our models, we set the location of the balloon to be the same place that it was during the measurement.

2.3 Using a Spherical Response Object

Once complete, the loudspeaker directional response object, with its spherical response object for each cell, represents a full model of the loudspeaker, with each cell located at the same position it was during the measurement. As it stands, the object does not display any curves. The information contained in the object is displayed in FChart through the use of virtual microphones. In FChart, a virtual microphone is placed into the 3D space with the spherical response objects. The virtual microphone, with its X,Y,Z coordinates then queries each spherical response object (balloon) in the model for its frequency response in relation to the direction and distance to the virtual microphone. The spherical response objects thus provides the appropriate frequency responses in response to the query. Superpositioning is then used to sum these responses as shown in Fig. 7.

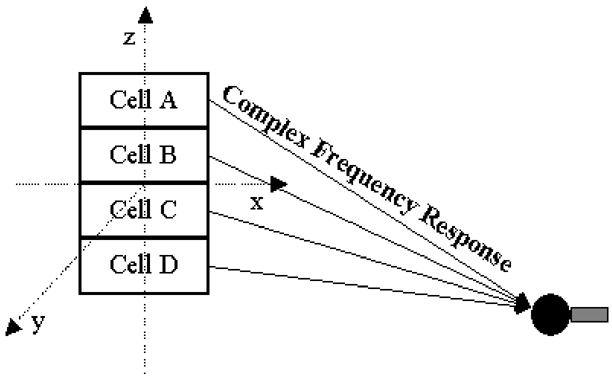


Fig. 7. Using Superpositioning to Find the Response at Any Location

To provide the response at any location, the spherical response object performs complex interpolations between the polar response objects, which in turn interpolate between measured transfer functions. The interpolation methodology is the subject of another paper entitled “Transfer Function Averaging and Interpolation,” by David Gunness [5]. Briefly, when a virtual microphone is placed in FChart’s 3D space, a vector is calculated from the microphone to each spherical response object located in the loudspeaker directional response object. The cross-correlation of these two transfer functions is then calculated. The peak of the cross-correlation will occur at the time corresponding to the difference in arrival times. In a polar, adjacent frequency response magnitudes should have a high degree of similarity, so the arrival time differences should be quite distinct. After calculating the analytic impulse of the cross-correlation, the arrival time difference is read as the time corresponding to the peak value. By removing the propagation delay from the later arriving transfer function consequently, the phase responses of each transfer function will also be similar. The primary impulses occur at the same time. Then, the complex transfer functions are interpolated.

To use the loudspeaker directional response models, virtual microphones are inserted in the locations they are needed. The frequency response for each will be displayed on the screen, as shown in Fig. 8. Since the magnitude measurements in the polar response objects are calibrated, all SPL levels are accurate for the actual distance of the microphone and the input signal magnitude. In this figure, the uppermost curve is the on-axis frequency response. Below this curve, in order, are the frequency response curves for microphone positions 10°, 20°, and 30° vertically below the enclosure.

At this point, it is also possible to insert DSP Filters and apply them to each balloon (spherical response object) by means of the Frequency Response Formula entry box shown in the bottom of Fig. 6. It is then possible to adjust filter parameters to provide optimum summation at various listening positions.

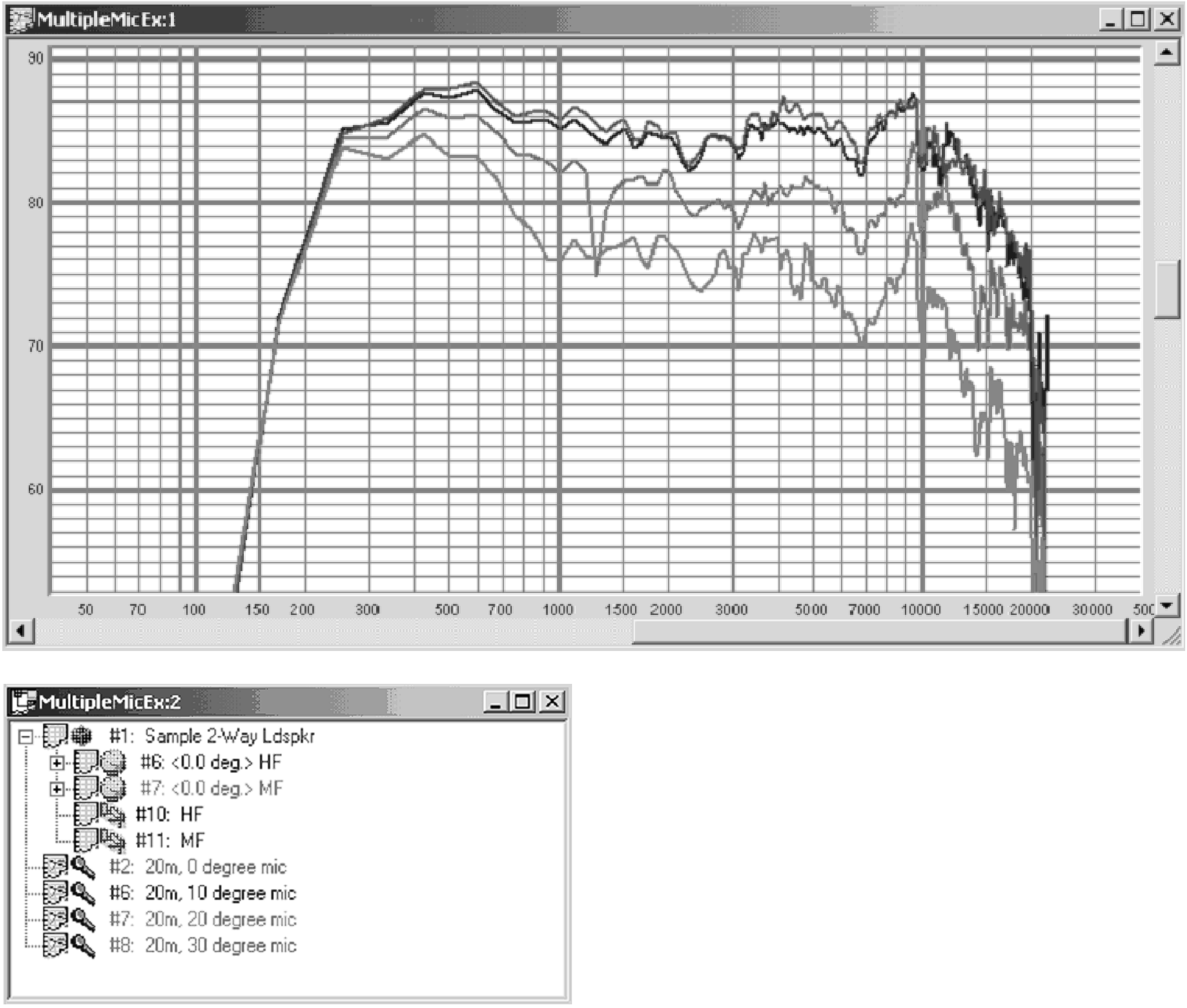


Fig. 8. A Sample Loudspeaker Directional Response Display

3 Temperature

3.1 The Necessity of Accounting for Temperature

In order to apply the actual microphone distance and angle that is calculated for every measurement, each distance must be converted into a propagation delay by using the speed of sound. Further, in order for complex data to be applicable, approximately ±0.3583 cm of accuracy is required when positioning the enclosure, as discussed in Section 1.2. Consequently, care must be taken when converting distance into propagation delay. From work with Phased Point Source Technology (PPST) [6], it was discovered that physically placing the enclosures as carefully as possible is not sufficient to guarantee the accuracy required. The speed of sound is extremely sensitive to temperature, and the temperature will vary from one measurement to the next. To attain sufficient accuracy, the absolute temperature at the time of measurement must be recorded.

From Beranek [7], the speed of sound can be written as a function of temperature.

$$c = 331.4 \times \sqrt{\frac{T}{273}}\ m/s \qquad (5)$$

T is the temperature in Kelvin. Near room temperature, the approximate speed of sound can also be calculated using:

$$c = 320.6486 + 0.3371 \times F \quad m/\sec \qquad (6)$$

where F is the temperature in degrees Fahrenheit. It can then be shown that speed of sound varies roughly by 1.106/1135 or 0.1% per degree F.

Our measurements are made at roughly 6m in our fixed setup. This means:

$$\frac{0.3371/345.948}{1^\circ F} \times 6m \times \frac{100cm}{1m} = \frac{0.585cm}{1^\circ F} \qquad (7)$$

0.585 cm for every 1° change in Fahrenheit. During a half hour measurement procedure, it is not unreasonable to assume the temperature could change by 5° F. This would result in a change of 2.92 cm in apparent distance. Clearly an intolerable change when phase accuracy is required. This change would prevent being able to combine loudspeakers measured on different days.

3.2 Applying Temperature to the Models

Each measurement inside our polar response object is assigned a temperature that is the temperature in the room at the specific time of measurement. This is an example label for a measurement:

<-8.6 deg/7.576 m><T = 13.3C><Offset: 24.22 ms> 350 Ver – D

The temperature at the time of this measurement was 13.3° C.

The temperature information, along with the actual microphone distance and angle, provides the accuracy necessary to use complex frequency responses. When the Spherical Response Object is asked for its data by a virtual microphone, all of this information is used. The object requests the response from its objects. The microphone asks the balloon, which asks its polars, which interpolates the measurements and returns the result to the balloon which interpolates between the polars.

The temperature is used to calculate the speed of sound that existed in the laboratory during the measurement. Using this speed of sound, a corrected arrival delay can be calculated for each measurement. Thus, each measurement is correctly aligned in the time domain. If there are several Spherical Response Object objects in the loudspeaker Directional Response Object, then each of these interpolated complex frequency responses will also be aligned correctly, allowing for accurate superpositioning of several sources.

4 Conclusions

4.1 Results

As an example, two high frequency transducers from the four cell loudspeaker shown in Fig. 3 were modeled. Cell A is 27.781 cm from the center and cell C is 9.049 cm away from the center, in the –z direction. Fig. 9 shows the independent measurement for each cell (transducer) when the enclosure is positioned on the turntable, on its side, being measured vertically, at the 10° rotator position. Hence, these are vertical comparisons. The upper curve is the transducer closest to the microphone (cell A) and the lower curve is for the transducer further away (cell C). The upper curve is actually at 8.6°/7.576m and the lower curve is actually at 11.4°/7.640m.

The modeled result at 2.5° using the virtual microphone and an actual measurement at 2.5° is shown in Fig. 10. The measured curve was obtained by turning on both HF transducers and making a measurement at 2.5°. The modeled curve is interpolated from measured data of the two individual transducers. Note that the measured data used for interpolation does not contain any data actually measured at a 2.5°. Instead, the program used data from measurements made at other angles. The coincidence of the both the magnitude and phase curves reflects the model’s accuracy.

In Fig. 11, the modeled result, using a virtual microphone, and an actual measurement at 10.0°, are shown for the two high frequency transducers.

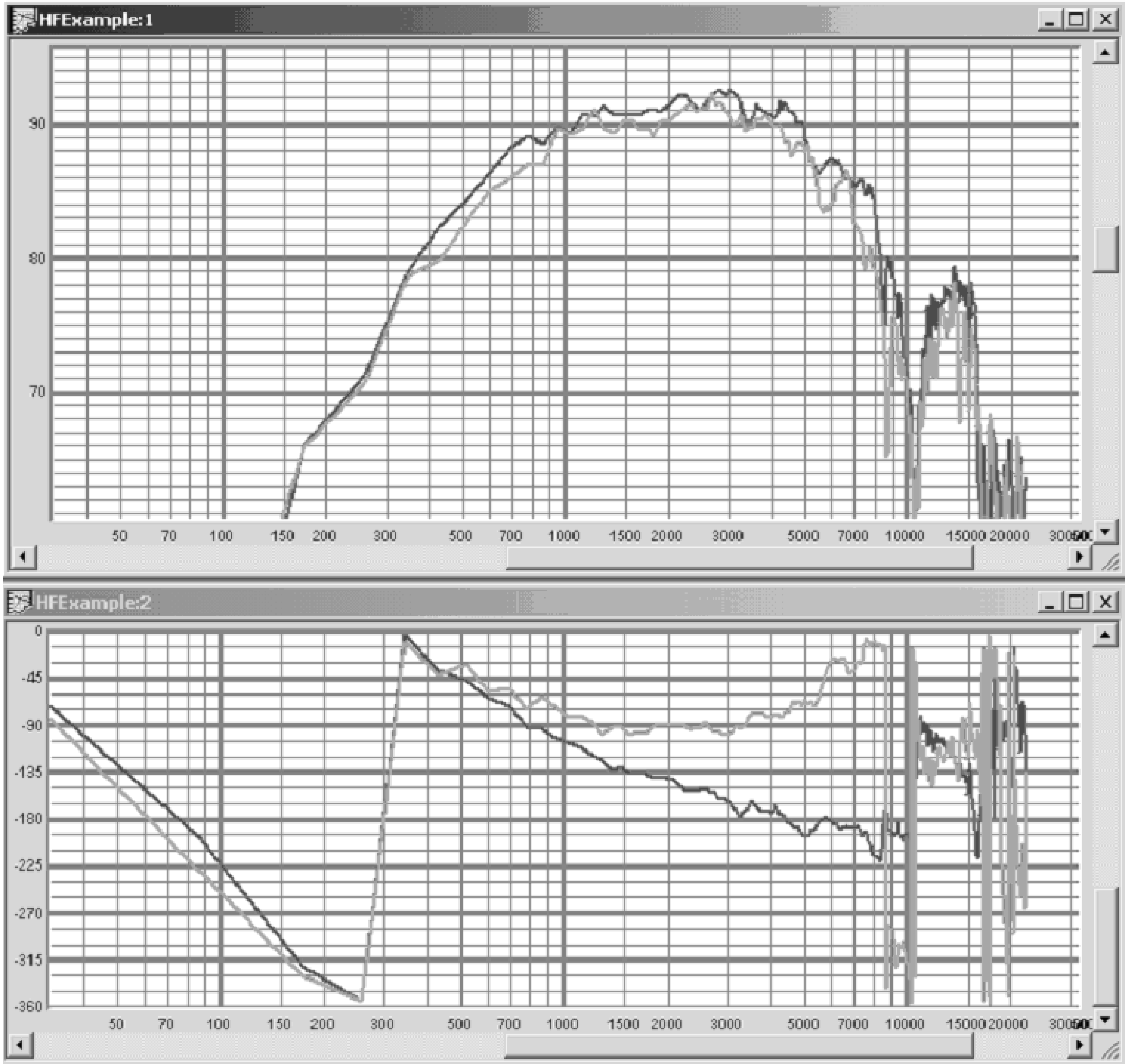


Fig. 9. Two HF Transducers Measured Independently. Top: Magnitude. Bottom: Phase.

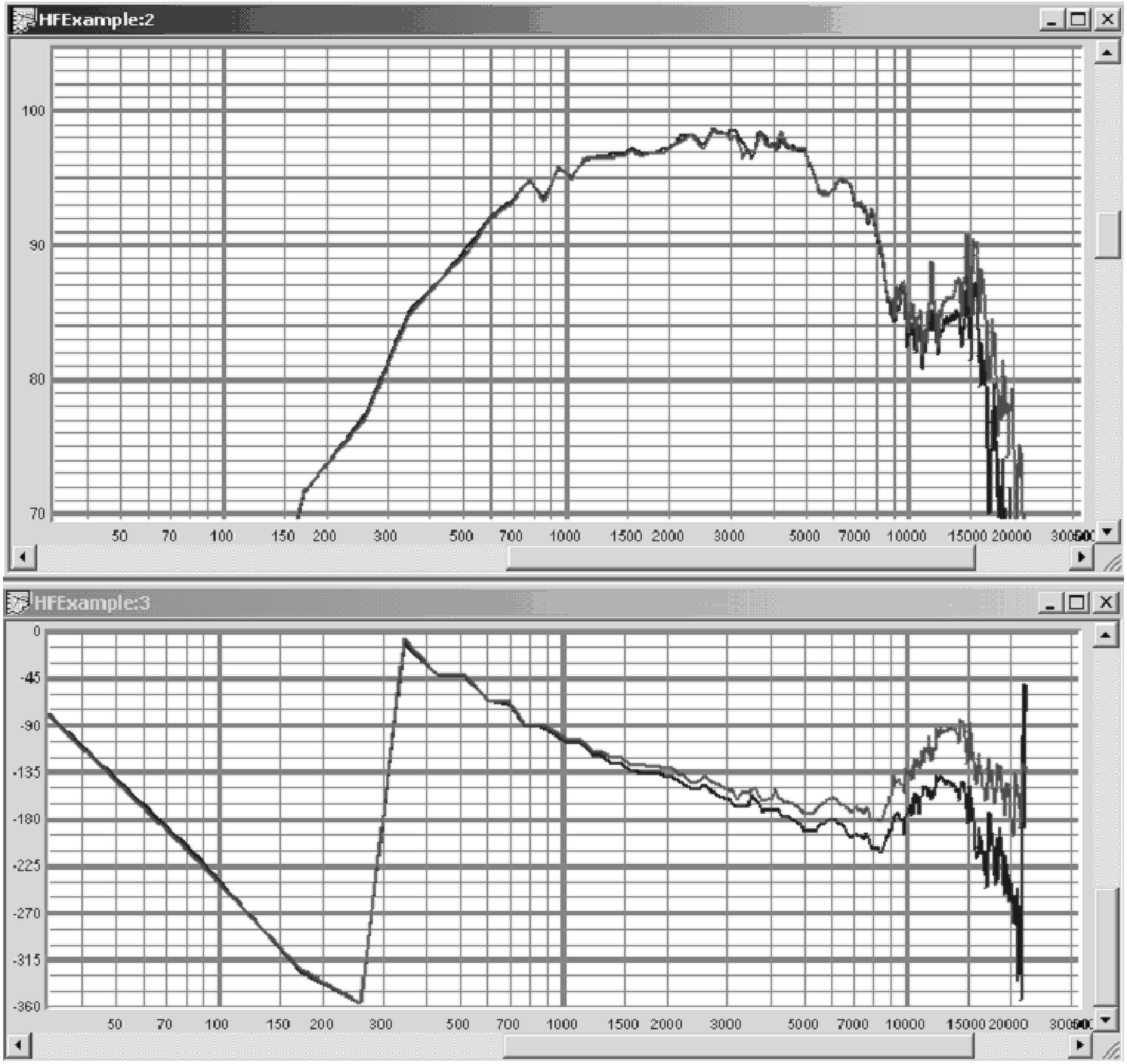


Fig. 10. Modeled Vs. Measured Curve at 2.5°

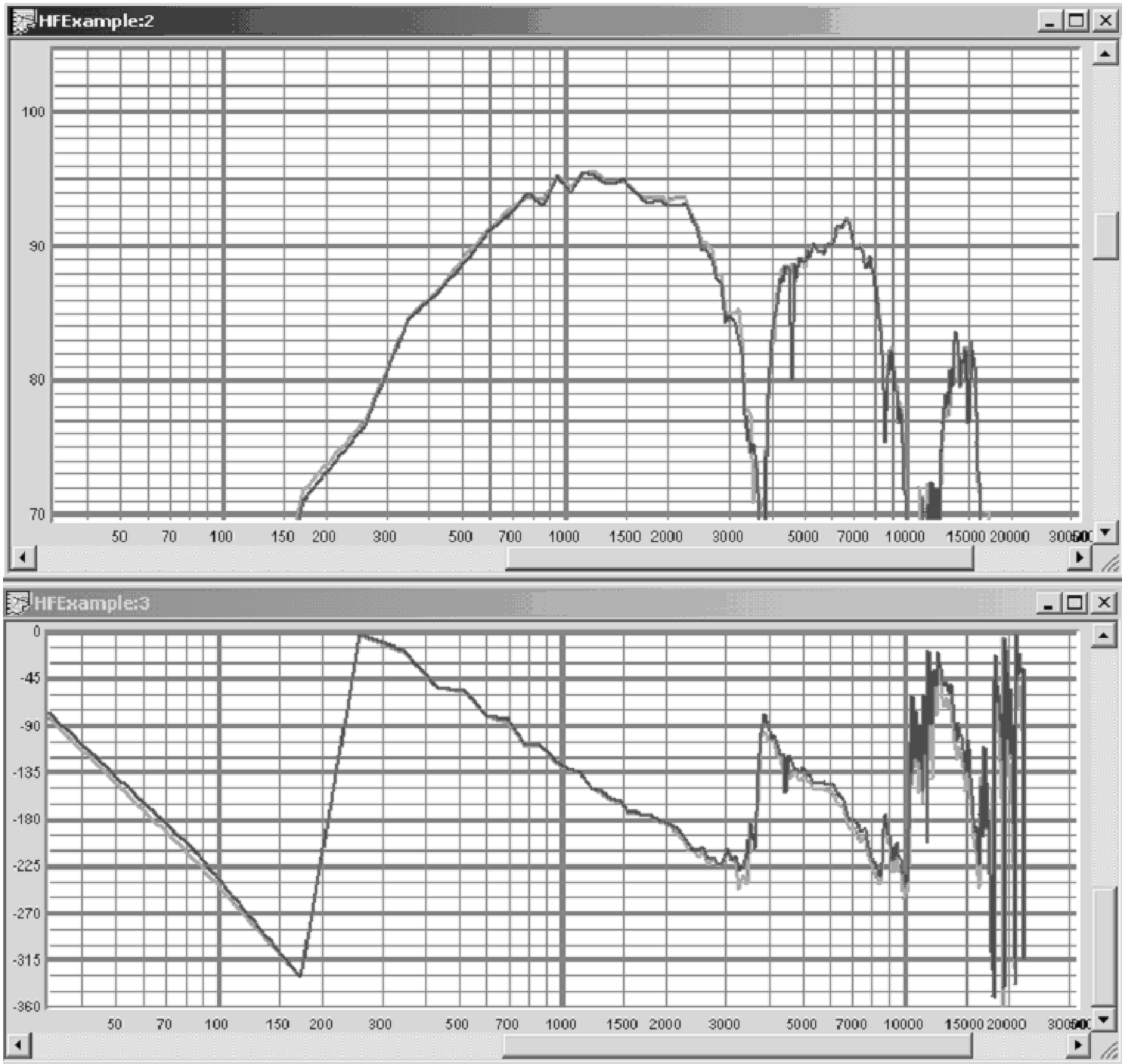


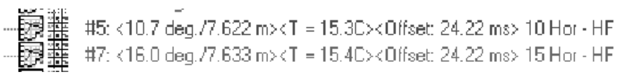
Fig. 11. Modeled Vs. Measured Curve at 10.0°

4.2 Benefits

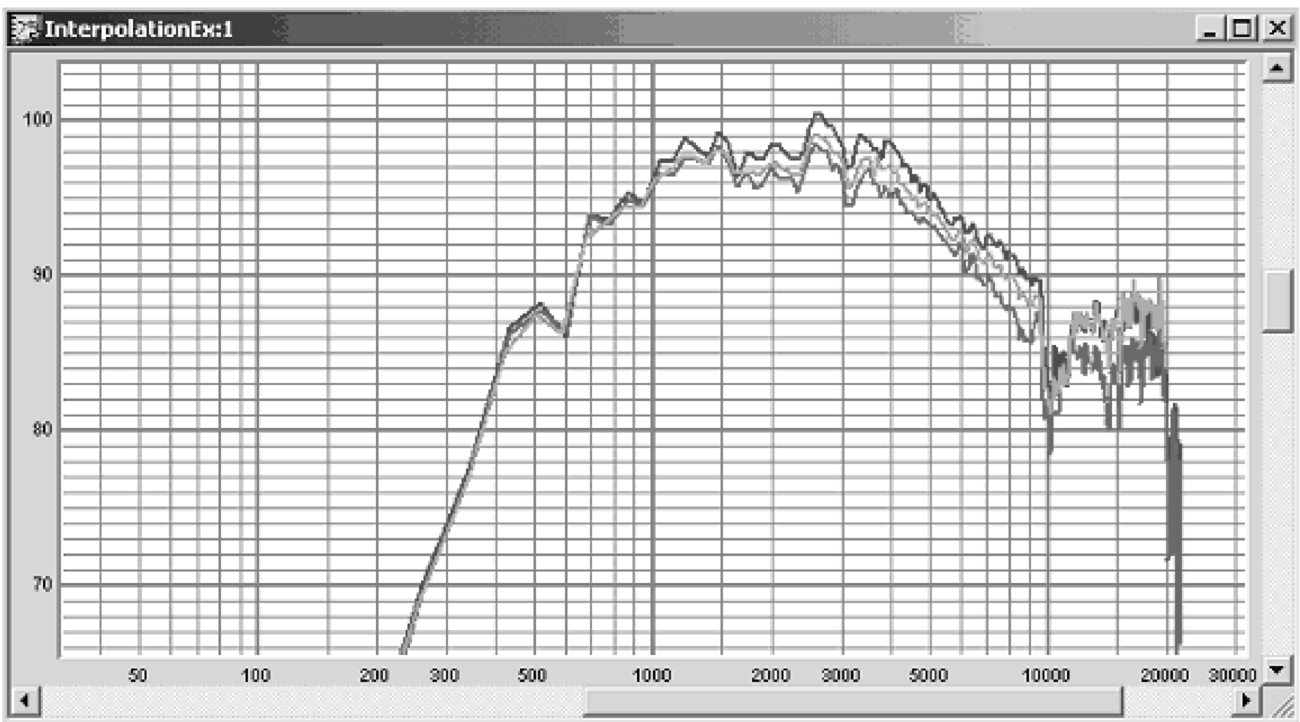
4.2.1 Frequency and Angle Resolution

The loudspeaker directional response objects offer several benefits. First, they are fully interpretable. Hence, although data is collected at a finite set of angles, the use of the complex interpolation technique described in David Gunness's paper "Transfer Function Averaging and Interpolation," allows the frequency response to be calculated at any location with phase response accurate enough for interference calculations.

For example, in Fig. 12 (top), the upper frequency response curve is the measurement recorded at 7.622 m, 10.7° horizontally from an HF horn. The lowermost curve is the measurement recorded at 7.633 m, 16.0° from the same horn. The center curve is the virtual microphone's response at a distance of 7.627 m and an angle of 13.4°. As expected, the interpolated frequency response fits precisely between the two measured curves. This is the file label for the two measurements:



An actual measurement was taken at the same location as the virtual microphone. This and the interpolated curve should be coincident if the interpolation is correct. As shown in Fig. 12 (bottom), they are within fractions of a dB. This is the label for the actual measurement file at 13.4°:



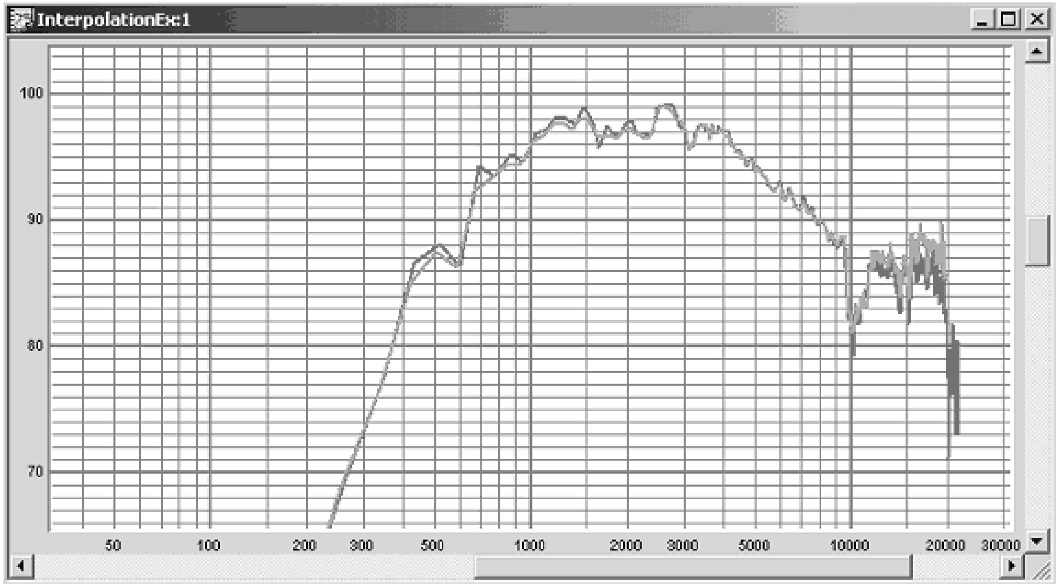


Fig. 12. Interpolation Example: Magnitude. Top: Data measured at 10.7° and 16.0° with interpolated result at 13.4°. Bottom: Data measured and interpolated at 13.4°.

Fig. 13 (top) shows the same results as Fig. 12 but in the time domain. The 10.7° curve arrives first, and the 16.0° curve arrives last. As expected, the interpolation puts the 13.4° curve between the surrounding curves. Fig. 13 (bottom) shows data measured at 13.4° along with the interpolated results for 13.4°. The coincidence of the two curves shows the precision of the interpolation.

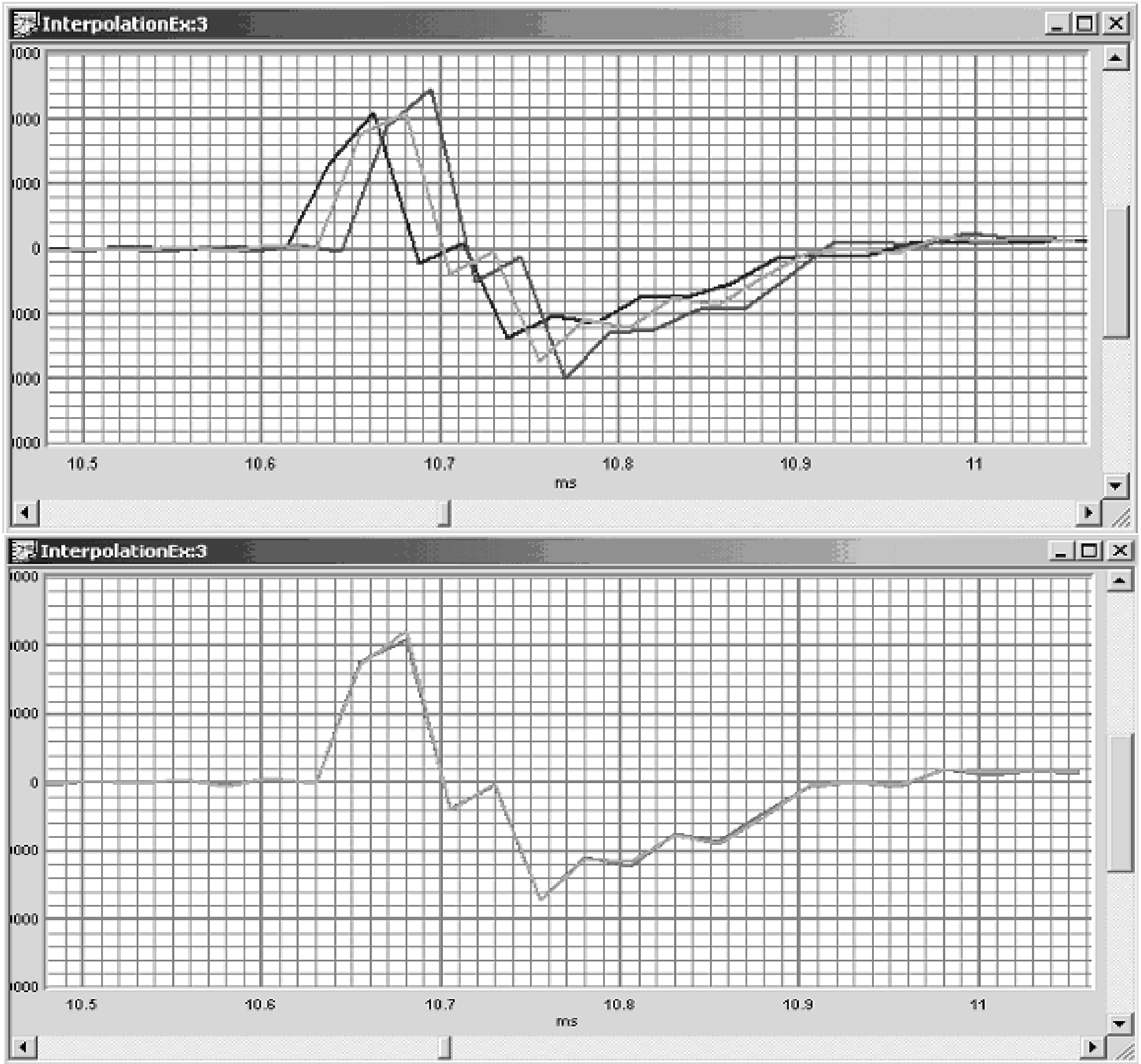


Fig. 13. Interpolation Example: Time Domain. Top: Data measured at 10.7° and 16.0° with interpolated result at 13.4°. Bottom: Data measured and interpolated at 13.4°.

4.2.2 Apparent Apex Error

The Loudspeaker Directional Response Objects are largely free from apparent apex error. Apparent apex error is simply defined as the error in the computed beamwidth that occurs if the apex of the beam is not at the center of rotation. Fig. 14 illustrates this error.

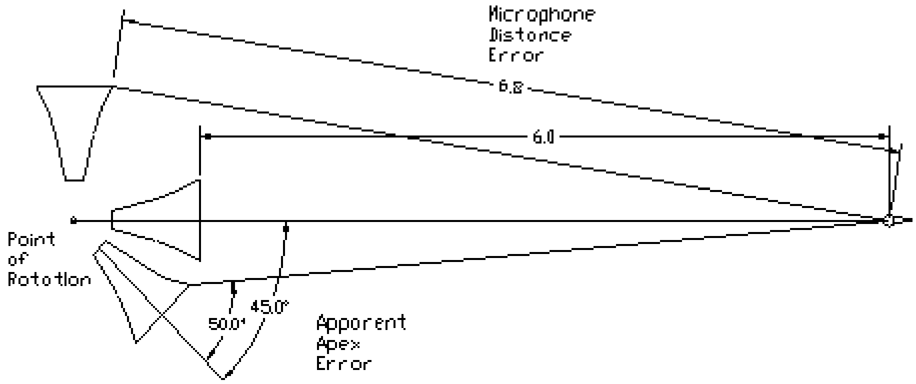


Fig. 14. Apparent Apex Error

For nearly all loudspeakers, it is impossible to place the center of the loudspeaker exactly at the center of rotation. The error occurs when data recorded at a turntable position of 45° is labeled and used as data recorded 45° off-axis of the loudspeaker. The turntable angle is not equal to actual off-axis angle because the loudspeaker was not on the axis of rotation. As shown in Section 1.2, since the locations of the cells are recorded, a more accurate distance and angle to the microphone can be calculated for our models.

When the loudspeaker directional response object is asked for its data at a certain point, it uses the corrected microphone angle as its lookup. For example, if we ask the Cell A Spherical Response Object for its Horizontal 0° measurement, it will not return the line of data that was collected at the turntable 0° position. It will return an interpolated result using the surrounding measurements. Fig. 15 shows this result. The upper curve is what the virtual microphone returns as the frequency response for Cell A at 0° on axis. The lower curve is the measurement made at a turntable angle of 0°. However, since cell A is at the top of the box, it was not at the center of rotation during the measurement process. In fact, the lower curve was actually measured at 2.1° below Cell A's axis and is therefore the incorrect frequency response for a 0° measurement.

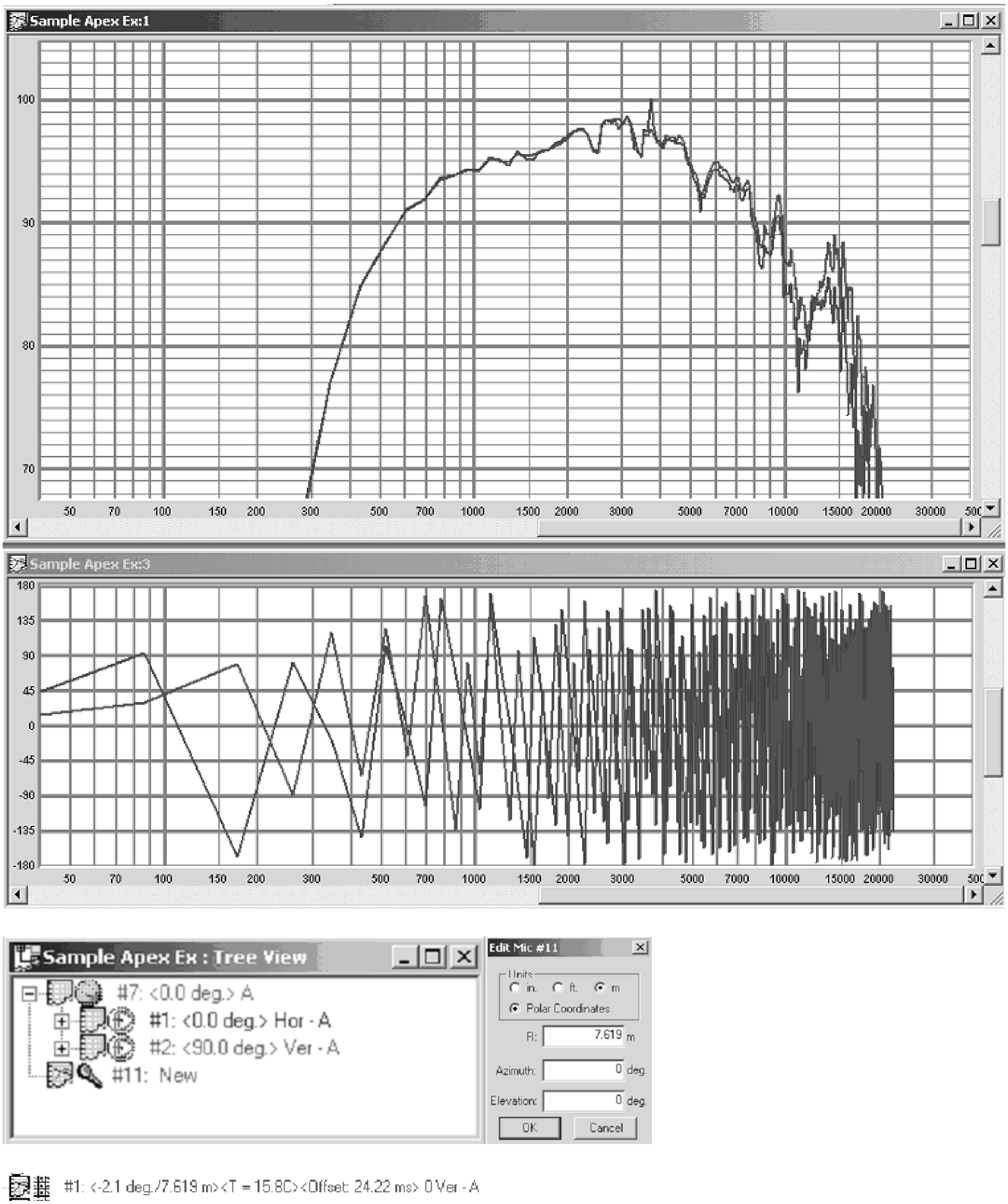


Fig. 15. Response at 0 Degrees

4.2.2 Accurate Array Interaction

The models are fully capable of predicting array interactions, simply because complex frequency responses are stored and used, as opposed to the magnitude-only data currently employed in typical modeling programs. The phase data is used to accurately determine and model the interaction between the components in the array.

Since each loudspeaker is broken down into individual cells, modeling an array of loudspeakers is no different than modeling one loudspeaker. Thus, the object-oriented approach applies perfectly to building an entire loudspeaker array. One loudspeaker contains several cells, each located at a different point in the loudspeaker's enclosure. Each cell contains its hierarchy of Spherical Response and Polar Response Objects. Thus, each loudspeaker becomes one object with these items contained within it.

Following the object-oriented approach, the Loudspeaker Directional Response Objects can be copied and added to a larger model to form columns and/or rows of loudspeakers. As a whole, we can position the loudspeaker object, which effectively positions all of the balloons and all of the measurements. Once a row or column is built, we can then group these and copy entire rows or columns. In this manner, a fully interpretable 3D loudspeaker array can be accurately modeled. When asked for its data, the overall model "asks" each one of the balloons it contains. All loudspeaker objects thus provide their data with both magnitude and phase information. From this information, the frequency response at any point and location around the array can be accurately determined.

4.3 Applications

4.3.1 Data Collection Engine

The loudspeaker directional response objects have several useful applications in the real world. The techniques described are currently being employed as the standard data engine for EAW's laboratory measurements. The models are particularly useful during the design phase of a project. Because all cells (transducers) are measured independently, they can be filtered and switched in and out of projects. This implies that cells can be combined virtually, and their combined frequency responses can be calculated without requiring a complete system to be built.

The complex data contained within each object can be used to calculate the interaction between loudspeakers and model the results. Since all distances are known, and since phase is recorded, we are able to use these models to generate the appropriate digital signal processor settings. For example, this makes the models perfectly suitable for Phased Point Source Technology (PPST) data. PPST refers to the technique of measuring each cell independently and then applying signal processing to it, so the entire array can be forged into a single acoustic unit, optimized to cover a certain area. In this way, the models allow the determination of the appropriate signal processor settings for various arrays used in various rooms.

4.2.2 DLL Usage

DLL's are "dynamic link libraries" and they are programming constructs that allow packages of code and data to be ported, without requiring any knowledge of what is inside. Because the models are set up in an object-oriented fashion, it is possible to package them as DLL's for use with predictive acoustic modeling programs. This involves designing a custom graphical user interface that wraps around the loudspeaker Directional Response Object. The user interface would allow adjustment of parameters, such as DSP settings, and array configurations, without having to manually interact with the data objects shown in this paper. Each product will be modeled by a DLL file. The software would use the data provided by these DLL's, instead of the currently employed data text files.

In addition to the independent measurement of each cell, the object-oriented approach to data collection is powerful because each array can be built on demand. In this sense, the model acts as a dynamic tool that allows the user to adjust array parameters. The DLL would simply build the desired array out of the available cell objects, insert the necessary virtual microphones, and then provide the frequency and phase responses at those microphone locations.

4.2.3 "Wizard" Software

In a similar fashion to the DLL usage of the models, the Loudspeaker Directional Response Objects lend themselves nicely to product deployment assistance programs (often referred to as Wizards). For this use, a user-friendly front end provides input dialog boxes where array information (such as loudspeaker type, loudspeaker count, and array configuration) can be entered. There is also an input dialog box for venue dimensions. Using this knowledge of the venue, arrays can be modeled and optimized for the venue.

5 REFERENCES

[1] M. S. Ureda, “Apparent Apex Theory,” presented at the 61st Convention of the Audio Engineering Society, 1978, Preprint #1403.
M. S. Ureda, “Apparent Apex Revisited”, 1991, Preprint #3040.
M. S. Ureda, “Apparent Apex, Part III: The Three-Dimensional Case”, 1991, Preprint #3166.
M. S. Ureda, “Apparent Apex, Part IV: Direct Radiators”, 1992, Preprint 3324.
M. S. Ureda, “Apparent Apex”, 1997, Preprint #4467.

[2] D. W. Gunness, “Loudspeaker Directional Response Measurement,” presented at the 89th Convention of the Audio Engineering Society, 1990, Preprint #2987.

[3] D. W. Gunness and W. R. Hoy, “Improved Loudspeaker Array Modeling, “ presented at the 107th Convention of the Audio Engineering Society, 1999, Preprint #5020.

[4] D. W. Gunness and W. R. Hoy, “Improved Loudspeaker Array Modeling Part 2, “ presented at the 109th Convention of the Audio Engineering Society, 2000, Preprint #.

[5] D.W. Gunness, “Transfer Function Averaging and Interpolation, “ to be presented at the 111th Convention of the Audio Engineering Society, 2001.

[6] D. W. Gunness and J.F.S. Speck, “Phased Point Source Technology and the Resultant KF900 Series, “ Eastern Acoustic Works, Inc., 1997)

[7] L. L. Beranek, *Acoustics* (McGraw-Hill, New York, 1954).