EAW POLAR MEASUREMENTS

INTRODUCTION

This is a summary of several technical papers presented by EAW at Audio Engineering Society Conventions. These papers describe a new way to measure loudspeaker polar data and how the measured data can provide accurate computer models of loudspeakers and loudspeaker arrays. The mathematics are also applicable to other measurement and modeling tasks.

POLAR DATA

Typically, polar measurements are made by rotating a loudspeaker on a turntable and measuring its output at each turntable position. The data measured shows how much output a loudspeaker produces in various directions compared to the on-axis output. This data is most frequently used to design and specify a loudspeaker's dispersion angle (e.g. 90° Horz x 40° Vert) and to predict how evenly a loudspeaker will "cover" a given audience area. Several popular computer programs, such as EASE, use this kind of data to model a loudspeaker's coverage in simulated rooms.

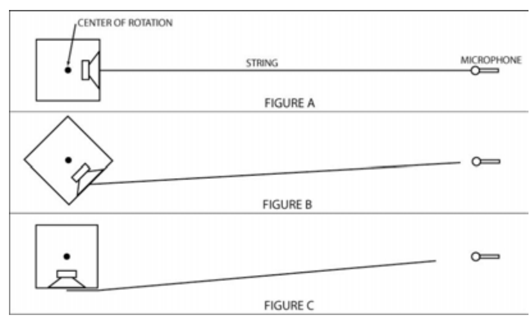
DATA ERRORS

Unfortunately, the typical polar measurement method has three significant flaws:

1. The assumed distance from the microphone to the loudspeaker is usually different from the actual distance between the microphone and loudspeaker. This causes errors in the measured output.
2. The assumed measurement angle (angle of the turntable for each rotation increment) is usually different from the actual angle between the microphone and loudspeaker. This causes errors in the plotted angles.
3. Because of #1 and #2, any phase data collected will be incorrect, eliminating any useful time domain information.

DISTANCE ERRORS

Distance errors can be easily seen in **FIGURES A** through **C**. A microphone is placed some distance from a loudspeaker on a turntable. A string is attached to the center of the loudspeaker's transducer. The other end just reaches the microphone. The string represents the direct path of the sound.



In **FIGURES B** and **C**, the loudspeaker is rotated 45° and 90°. In each case, the string no longer reaches the microphone. The string did not change length, the distance from the transducer to the microphone changed. This means that, even if the loudspeaker produced the exact same output in all directions, the sound level at the microphone would change with the turntable angle because the distance to the microphone changes. Clearly, this would be an incorrect result for this measurement.

ANGULAR ERRORS

Angular errors can also be seen in **FIGURES B** and **C** where the angle of the string from the microphone has changed. Ideally, the string would stay positioned on an imaginary straight line from the microphone to the center of the loudspeaker's rotation. Thus at 90° the string should be perpendicular to the direction the loudspeaker is facing. It is not, meaning the sound is not being measured at 90°. However, most measurement systems would record it this way.

CAUSE OF THE ERRORS

Both of these errors occur because of the physical geometry of the measurements. The larger the loudspeaker the greater the errors are for a given microphone distance. If a loudspeaker were reduced to a tiny point placed at the center of rotation, these geometrical errors would not occur. However, to date, no such loudspeaker has ever been designed.

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THE SOLUTION

Correcting these errors is relatively easy. The method is to provide the measurement program with accurate physical dimensions of the loudspeaker and its position relative to the microphone. Using straightforward geometry, the actual microphone to loudspeaker distances and angles can be computed for each measurement. The result is that the correct angle and level for each measurement can be determined.

IMPORTANCE OF PHASE

Phase concerns time. For example, in a 2-way loudspeaker with a HF horn and woofer, the HF transducer is further back in the enclosure than the woofer. This means the sound from the HF horn will arrive at the microphone later than the sound from the woofer. In order to "see" this effect in the data, the sound arrival times (which is translated to phase data) must be correctly measured. Without correct phase data, it is impossible to correctly model the interaction of two or more sound sources, such as for the 2-way loudspeaker or for two or more loudspeakers in an array.

Measuring phase requires comparing the time arrival difference between a reference time ("time zero") and the measured signal. If the signal arrives after time zero then it is phase-delayed or "out of phase". When measuring loudspeakers we only want to measure phase differences that are caused by the loudspeaker(s). These can occur for a number of reasons.

CAUSES OF PHASE ERRORS

Using the standard polar measurement method, even if the time arrivals are recorded, there will be two sources of errors that will make the data useless.

The first source of error concerns the time it takes of the sound for each measured signal to travel from the loudspeaker to the microphone. This delay is called the propagation time. This delay has nothing to do with any delays caused by the loudspeaker(s). Therefore, the propagation times must be accurately determined, then factored out of the data. Without correct distance information for each measurement, this cannot be done.

The second source of error concerns temperature. The speed of sound is affected by air temperature. If the temperature for one measurement is different from another, the propagation times will be different, even if the distances are identical. Temperature changes that can affect the data can easily occur within minutes, even in an acoustics laboratory.

THE SOLUTION

Correct propagation times can be determined using the corrected distances as described above. These are then divided by the temperature-corrected speed of sound for each measurement. When these results are factored out of the data, only the phase differences caused by the loudspeaker(s) are left.

HOW THE NEW METHOD IS USED

Each transducer of a loudspeaker system is measured individually. A post-processing program then combines the data into a complete loudspeaker as if all transducers were turned on and measured at the same time. It does this using the corrected distance, level, angular, and phase data to recreate the position of each sound source in a virtual space. The program can then model the combined outputs at all frequencies and directions, complete with any acoustical interactions between them.

MODELING EXAMPLE

FIGURE D shows the off-axis frequency response and phase measurement for each of two identical HF transducers mounted in one enclosure. The microphone is on-axis of one but off-axis from the other, and thus at a slightly longer distance. The lower curve is the one further away from the microphone. As expected, the level is lower. **FIGURE E** compares an actual measurement of both transducers operating at once with a computer model made by combining the individual measurements of **FIGURE D**. Both the frequency and phase response curves are virtually identical. The expected deep notch is caused by differences in time arrival, and thus phase, between the two transducers at the microphone position. The model accurately shows this pronounced phase-related interaction between the two transducers.

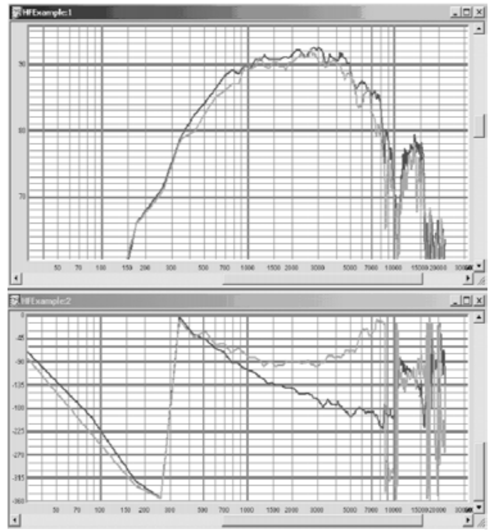

FIGURE D: Two HF Transducers in One Enclosure

FIGURE F shows a similar comparison between the actual measurement of both transducers operating at once and a model of both made from the individual measurements. However, in this case, the model has no phase information. It is obvious that the model does not show the measured performance, particularly the 30 dB “hole” near 4 kHz. This is actually how most modeling programs combine the outputs of multiple sources.

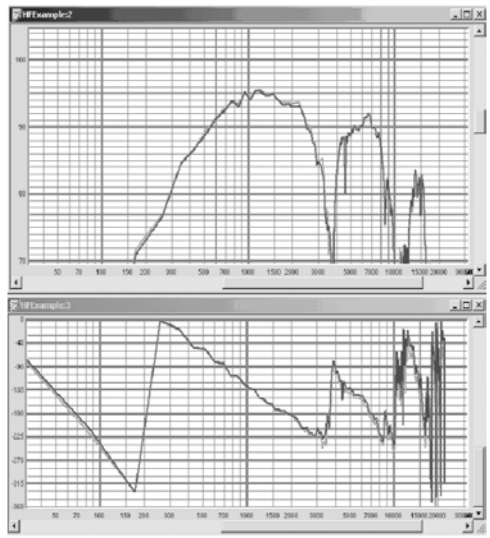

FIGURE E: Model and Measurement of Both Transducers On

ADDITIONAL MODELING

Using corrected data as described, the interaction between transducers in adjacent pass bands (e.g. LF, MF, and HF transducers) and in multiple loudspeakers can be accurately modeled from the individual transducer measurements. Thus, not only complete loudspeakers, but entire loudspeaker arrays, can be accurately “assembled” within a modeling program.

INTERPOLATING AND AVERAGING DATA

Only a certain amount of measured data is usually collected due to time, hardware, and/or software realities. What if one needs to know the results at points not actually measured? Up until now, interpolating such data has been highly suspect. This is because attempts at interpolation and averaging have provided poor results because of both the mathematical methods used and the measurement conditions. While successful interpolation of magnitude information has been accomplished, what is needed is the accurate interpolation of complex data (magnitude and phase).

FIGURE F: Model Without Phase Data and Measurement

INTERPOLATION

It was empirically found that the geometric mean provides correct results for complex interpolations between data points. As an example, two measurements were made of a mid-high frequency horn at 15° and 20° off axis. The frequency response was interpolated for 17.5° and an actual measurement was made at 17.5°. These two curves are shown in **FIGURE G**, in between the measured upper 15° (upper) and 20° (lower) curves that were used for the interpolation. There are no significant differences between these two curves.

The lower portion of **FIGURE G** shows the impulse responses for the 15° and 20° measurements plus the measured and mathematically interpolated impulse responses at 17.5°. As expected, the 17.5° curves fall between the 15° and 20° impulses in time. This is because at 17.5° the distance to the microphone, and therefore the raw propagation time, is halfway between the distances for 15° and 20°. The shape and arrival time of the interpolated result is virtually identical to the actual measurement made at 17.5°. Using this interpolation method, the measurement process becomes quite tolerant of variable propagation times, normally a serious problem for interpolating acoustic measurements.

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SPHERICAL DATA

The far field radiation pattern of rectangular sound sources is actually well defined, but typical polar measurements rarely produce this result. Because of this, there is an assumption that full spherical measurements are required to accurately produce “full balloon” data. The reality is that such measurements are flawed due to problems already discussed and other, less obvious uncontrolled variables. In the specific case of rectangular shaped sources, the response in the corners can be accurately calculated by applying superposition of the horizontal and vertical off-axis characteristics. Extremely rigorous measurements have verified this result. However, maintaining this same rigor is not practical for everyday laboratory measurements. As such, superpositioning yields a more accurate result.

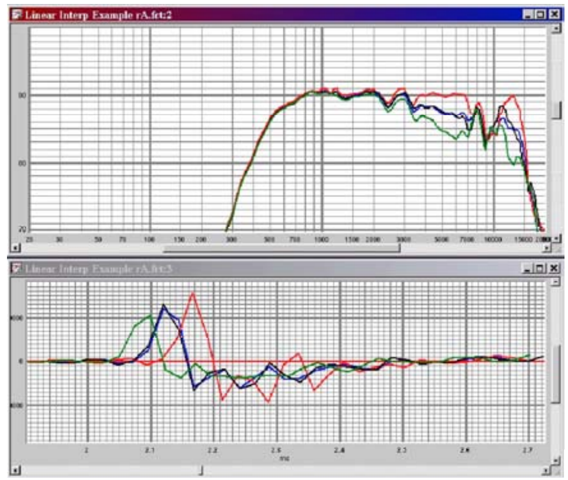

FIGURE G: HF Horn at 15° and 20°

AVERAGING

This new methodology can also be applied to averaging data, such as when a measurement system uses multiple measurements to increase signal to noise ratio. This is also true for averaging field measurements where measurement propagation times typically cannot be controlled. Complex averaging (i.e. averaging both phase and frequency data) can thus be used even outdoors and in large venues. Normally, averaging of complex data in such situations has merely shown the interference between successive measurements, resulting in a misleading high frequency response.

As another example of how well this method works, an interpolation was made between a second-order Butterworth filter and a fourth-order Butterworth filter. The result was an exact match in both magnitude and phase to a third-order Butterworth filter. Similar results are obtained between orders of Bessel filters. The most interesting aspect is that the essential behavior that characterizes a particular filter type survives the interpolation. For example, an interpolation between two maximally flat (Butterworth) filters is another maximally flat filter. It is also quite easy to synthesize “impossible” filters with arbitrary characteristics, such as one with an 8.6 dB per octave slope.

SUMMARY

These new methods represent significant breakthroughs in both measuring accurate polar data and in using this data to accurately model and predict the radiation patterns of loudspeakers and loudspeaker arrays. In conjunction with this, accurate interpolation and averaging of complex data is possible, particularly for field measurements where propagation times and other variables cannot be controlled.

The primary EAW applications for this new measurement method include:

- Engineering design of loudspeakers and loudspeaker arrays
- More accurate specifications
- Complex data for acoustical modeling programs such as EASE 4.x
- Analyzing field measurements

REFERENCES

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