

# **Auralization of Microphone Placement**

## **for the Recording of a Fender Deluxe Amplifier**

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Submitted in partial fulfillment of the requirements for the  
Master of Music in Music Technology  
in the Department of Music and Performing Arts Professions  
in The Steinhardt School  
New York University  
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1/10/2012

## ACKNOWLEDGEMENTS

This thesis would not have been possible without the research that was organized and conducted by my mentor, Dr. Agnieszka Roginska, and by Jim Anderson and Alex Case. The rigors upon which they insisted provided a sound basis my work. I am also grateful to Justin Mathew, Joshua Guthals, Cheng-I Wang, Merter Oğuz Yıldırım and Braxton Boren for their help. I especially thank Areti Andreopoulou for her assistance in reviewing the draft and offering numerous helpful comments and suggestions, along with her ongoing encouragement to me. I would also like to thank Sennheiser for supplying the microphone array that we used to collect the radiation pattern data.

I am also grateful to Apple Inc., which came to my rescue when my hard drive died a couple of days before this paper was due.

Most of all, thanks to my loving parents for their tireless support.

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## PREFACE

I am interested in using different senses to experience the same phenomena. If there is a great visual effect, I want to sonify it. If there is a great auditory phenomenon, I want to feel it... or see it. Radiation patterns offer a unique opportunity to explore this interest. The auditory phenomena of sound radiation are captured through sampling locations around a source. This data can then be used to create informative visualizations of the sonic world. These visualizations give the eyes a chance to understand the ears. Beyond that, the data can be used to auralize. This thesis explores the practical way in which radiation patterns can be used to quickly and effectively guide a recording engineer to greater success in microphone placement. Although the auralization of specific radiation pattern measurements will be the focus here, my underlying motivation is in the longer term connection of the sonic and the visual. Both are communicative, but when experienced together, they complement each other... the sum is greater than the parts.

# I. INTRODUCTION

After his interviews with highly respected recording engineers, Bobby Owsinski, author of *The Recording Engineer's Handbook*, cautions against sight-based microphone placement, "...which is a mistake that is all too easy to make (especially after reading a book like this). The best mic position cannot be predicted, it must be found." (Owsinski 74). Here, Owsinski warns engineers against using the recommendations in his own book without also referencing what they hear. To be fair, books like *The Recording Engineer's Handbook* offer great insight into what seasoned recording engineers have found and learned, largely through in-studio experimentation. Likewise, Jürgen Meyer's work in radiation patterns, *Acoustics of the Performance of Music*, as it has been, should certainly be studied and used by the recording engineer to find the 'sweet spot' for microphone placement. However, visual representations of microphone placement and radiation patterns plots need not be separate. A photograph of a suggested microphone placement does not give the engineer an intuition for why it was placed there. Why not 1mm higher? Could that be better? What about pictures showing microphone placement and radiation patterns? Radiation pattern plots can inform an engineer about why the timbre is changing with microphone movements, but do not allow the engineer to hear the results of different positions. What if the engineer had not only a picture showing microphone placement and

radiation patterns, but also the ability to virtually audition alternate placement locations? That would provide a complete picture.

In the music and audio research field, we should be able to hear research results... that is, at least, when they can be heard. As would be expected by the complexity of the radiation patterns of musical instruments, the timbral differences between potential receiver locations is often audible/perceivable. Radiation pattern plots of a musical instrument certainly inform a recording engineer's microphone placement. Nevertheless, recording engineers still rely on their ability to mentally connect what they see to what they expect to hear. This gap is not always necessary. Microphone placement is part of the recording engineer's art. Other data might not always have such a direct audible corollary, but... radiation patterns do.

The author of *Auralization*, Michael Vorländer, writes, "Auralization is the technique of creating sound files from numerical (simulated, measured, or synthesized) data" (Vorländer 3). Sonification, by contrast, refers to an invention of sound from data. It is rather a "mapping of data to sound parameters" (Madhyastha & Reed). This data-to-sound invention may be random or well thought. Auralization in this paper will be a technique of creating sound that is grounded in the inherent acoustic meaning of the numerical data.

### **Radiation Pattern Basics:**

Sound is produced by waves of pressure variations. A sound source excites the surrounding medium, causing particles of the medium to compress and rarefact in reaction

to the source. These sound waves are transduced by our ears. Ultimately these waves become the experience of sound.

When a reproduction speaker is the sound source, a speaker cone vibrates back and forth in order to excite the air surrounding the speaker. The particle density throughout the entire field around the speaker is affected. The way in which the speaker has affected this field is different for every point in the field and for every moment of time as the wave progresses through the medium. The arrival time of a wave can easily be calculated using the speed of sound and the distance between the sound source and a point in the field. The spectral differences, however, are far more complicated. These differences are the result of many variables, such as the size and material of the speaker cones and the size and material of the cabinet enclosing the speaker cones. The speaker cones' size, for example, has a direct effect on the frequencies which the speaker can effectively reproduce. The cabinet resonates, reflects, etc., differently for all waves of different frequency. All of these factors lead to spectral differences in the field surrounding the speaker.

When considering the radiation patterns of a sound source, the audio scientist's goal is to objectively measure how a sound leaves and interacts with the source itself and the surrounding environment. A recording engineer, however, has a different goal. At the end of the day, a recording engineer needs a way to find out where a microphone should be placed around an instrument or speaker to get a "good sound". Often the engineer relies on his or her educated intuition, acquired over time primarily from the testing of many different microphone placements and mentally documenting the corresponding subjective experience. Therefore, radiation patterns are not of direct concern to the recording engineer; rather, the engineer's concern is a capturing pattern, i.e. the result of microphone

placement. The capturing pattern, in this example, refers to the subjective experience of timbre as a function of microphone placement.

### **Interpolation Basics:**

The intent of interpolating between two data points is to create an accurate third data point. Interpolation between data points serves an audio researcher by smoothing out a data set. Hesitation must be taken when doing an objective analysis using interpolated data, for obvious reasons. However, when making radiation pattern plots for an instrument, for example, a continuous plot is more congruent with the inherent meaning of the data. These measurements, after all, are the discretization of a continuous pattern. Interpolation for auralization allows for the auditioning of a discrete data set as though it were a continuous data set. For the recording engineer, this can be beneficial, by allowing the engineer to hear the capturing pattern of an instrument at various potential locations in the acoustic space.

The radiation pattern measurement methodology used by researchers at New York University and the University of Massachusetts at Lowell, upon which this thesis is based, will be described here and the results analyzed.

Although auralization tools for musical instruments offer benefits to the audio researcher, this thesis focuses on the perspective of the recording engineer. The importance of interpolation between data points will be shown to be vital for this application. The data sets that are used here were recorded using a Fender Deluxe Amplifier as the sound source. Two independent auralization tool implementations are

offered. It is the goal of this thesis work to provide a proof of concept auralizer, which could successfully accelerate microphone placement experience.

## **II. LITERATURE REVIEW**

In 2011, fellow researchers, along with the author, developed an impulse response navigation system called Interactive Room Response (IRR). In this case, the goal was to auralize the perceivable auditory affect of room size. For this system, impulse responses were precomputed, using acoustic modeling software to generate a range of incremental room sizes, in which a binaural listener received sound from several fixed source locations. Once this data set was obtained, convolution was used to produce the auralization. In this thesis work, a similar path was taken to auralize microphone placement. For this purpose, an acoustic data set needed to be collected and then effectively put to use.

### **Acoustic Radiation Patterns:**

The classic text on the radiation patterns of musical instruments is that of Jürgen Meyer: *Acoustics of the Performance of Music* (1978). In this work, Meyer offers a detailed description of the directions that waves of various frequencies leave an instrument. This is referred to several times in his work as the “directions of radiation”. Meyer couples the radiation directions of instruments with performance hall acoustics toward informing optimal musician placement in a performance group. His study can be seen from many perspectives. The orchestrator, for example, can more intelligently position his/her musicians to best accommodate a piece for live performance. For the recording engineer,

this kind of information can be used to position musicians for a live recording.

Microphone position, of course, can be thought of as similar to audience position, especially when placing microphones at a distance to capture the whole sound, including the room, instead of close microphone techniques.

Jürgen Meyer's work offers a radial/spherical interpretation of instrument radiation patterns. Sampling the radiation patterns of an instrument on a radial/spherical grid surrounding an instrument offers certain benefits. If an instrument were a point source, the 'acoustic center' of the instrument represents the specific physical location from which all the sound radiates. A spherical system takes advantage of the inherently spherical radiation of waves. But locating this 'acoustic center' is no simple task for many sound sources. Hagai et al. (2011) developed an algorithm to re-align spherically recorded radiation pattern measurements to take into account the ambiguity of some acoustic centers. Comments will later be made toward the inherent difficulty of interpolating between data points on a Cartesian coordinate system when the acoustic system is spherical in nature. It should be briefly noted that for the measurements used in this thesis work, a Cartesian coordinate system was chosen, in part, to eliminate the problem of locating an 'acoustic center'. The near-field radiation focus used for this thesis would have significantly amplified this problem.

Regardless of the coordinate system chosen, the radiation patterns of a musical instrument live in a three-dimensional space. That three-dimensional space will have its own acoustic characteristics. This space may be a free-field/anechoic space, hemi-anechoic space, or reverberant space. Sikora et al. (2010) chose not to remove the instrument from the room for their radiation pattern research in microphone placement.

They took a “recordist-centric approach” by viewing the room and the instrument as a “single acoustic system” (Sikora 1). A recording engineer does not record an instrument in an anechoic chamber, but rather records the instrument along with the room, intentionally or not. It was further determined by Leonard et al. (2011) that the room’s affect, in a room and piano coupled system, is significant; therefore, many more piano and room combinations would need to be tested before generalizations could be made.

In Sikora et al.’s case, the measurements were taken using a three dimensional Cartesian coordinate system with 25cm spacing between measurements. A three dimensional interpretation of the piano and room radiation spectral centroids was created to show locations in the room of “bright sound and dark sound” (Sikora 8). Expert recording engineers’ microphone placement suggestions proved to correlate with these locations. This is admitted to be anecdotal, but the direct relationship between the experts’ suggestions and the spectral centroids gives great credit to Sikora et al.’s methodology toward finding what is referred to in the author’s Introduction section as a capturing pattern.

All of the above-described radiation pattern research has been for purely acoustic instruments, i.e. non-electronic instruments. The electric amplifier is, after all, the instrument of choice here. Alex Case (2010) tested some classic microphone positions to scientifically show what happens at those locations. He compared the frequency responses of each position. These responses showed that more high frequency content is captured when the microphone is on-axis with the driver and less high frequency content is captured when off-axis. Increasing the distance from the driver results in a loss of lower frequencies, attributed to the proximity effect of the cardioid microphone that was used.

Small angle changes to the microphone position proved to have a relatively minimal effect. All of these scientifically gathered characteristics can begin to reveal to the recording engineer why different positions meet aesthetic goals.

### **Commercially Available Microphone Placement Auralizers:**

Microphone placement as a manipulable parameter is not new to guitar processing software. Several commercially available products include microphone placement as a parameter. At the time of this writing, this feature can be found in offerings from Red Wirez, Native-Instruments, Waves, and IK Multimedia.

Once an impulse response data set has been collected, simple convolution can be used to provide for auralization. Red Wirez, for example, offers a collection of impulse responses for a variety of amplifiers and microphones with typical microphone positions. This data set is comprehensive across amplifiers and microphones, but the available microphone positions are limited to those most commonly used. These placements rely upon the knowledge of experienced recording engineers. Native-Instruments' popular guitar amp and effect modeling software, Guitar Rig 5, makes use of the Red Wirez data set for its cabinet and microphone placement processing.

According to a demo video on the product's website, Waves' GTR software offers different aesthetically chosen microphone placements. These locations were picked by experienced recording engineers. IK Multimedia's AmpliTube 3 also offers virtual microphone placement, likewise in traditional locations around an amplifier. AmpliTube Fender provides models, officially endorsed by Fender, of many classic Fender amplifiers, including the Fender Deluxe.

## **Microphone Placement Interpolation:**

A portion of the thesis work offered here will be toward improving upon a basic convolution auralizer's flexibility with the use of interpolation. The Red Wirez data set, for example, is, as mentioned, comprehensive for traditional locations, but not for nontraditional locations. Without hearing all locations, the reasons for the traditional locations cannot be analyzed.

Interpolation between data points can take advantage of as much or as little information as the interpolation method dictates. Radke et al. (2002), for example, developed an algorithm for interpolating between two microphones in an anechoic situation with multiple sources. In addition to the two real recordings, they took advantage of knowing the distance between the two microphones. Masterson et al. (2009), instead, used inter-response information to inform the interpolation. They developed an interpolation method entitled IMPRESSION, which stands for “Impulse REsponse Synthesis and InterpolatiON”. The IMPRESSION system breaks an impulse response into the “direct sound with early reflections and the diffuse decay” (Masterson 2). The direct and early reflection section of the impulse responses must be aligned as closely as possible in order to perform linear interpolation. If direct and early reflections are not aligned, the resulting interpolation will include a “significant smearing of reflections” (Masterson 2). This is an inherent problem in acoustic impulse response interpolation. By using Dynamic Time Warping (DTW), the two impulse responses are stretched and warped to be aligned - this alignment problem will be discussed further in the Analysis section. Masterson et al. used Pseudo-Random phase shifts to decorrelate the interpolated decay, but maintained the

timbre of the recorded decay. In this case, the DTW was used to line up direct impulses and their reflections, vis-à-vis the differences in arrival time. The decay interpolation was created to match the timbre, whereas the early reflection interpolation was created to match the timing. Subjective testing was completed to verify the success of the interpolation.

Although a more simple interpolation method is used in this thesis work than those offered by the mentioned researchers, the solutions offered in this literature informed the analysis here.

## **III. METHOD**

### **Overview:**

Using a multi-microphone array, impulse responses were recorded of the field surrounding a Fender Deluxe amplifier. The array was sequentially repositioned to gather a data set that sampled the radiation pattern of the amplifier. This process was repeated, with modifications, at two separate research institutions, namely New York University (NYU) and the University of Massachusetts at Lowell (UML). This research was headed by Dr. Agnieszka Roginska (NYU), Jim Anderson (NYU), and Alex Case (UML), with assistants, including the author.

### **The Microphone Array:**

The microphone array contained 31 omni-directional Sennheiser microphone capsules, centered on a 31.8mm grid. The capsules were aligned along the enclosure of the array.

Although it cannot be said that that array itself was acoustically transparent, the array's bulk was minimal and for the frequency range considered here (primarily 100Hz through 8kHz), the acoustic impact of the array can be ignored. High frequencies



**Figure III-1: 31 Microphone Array.**

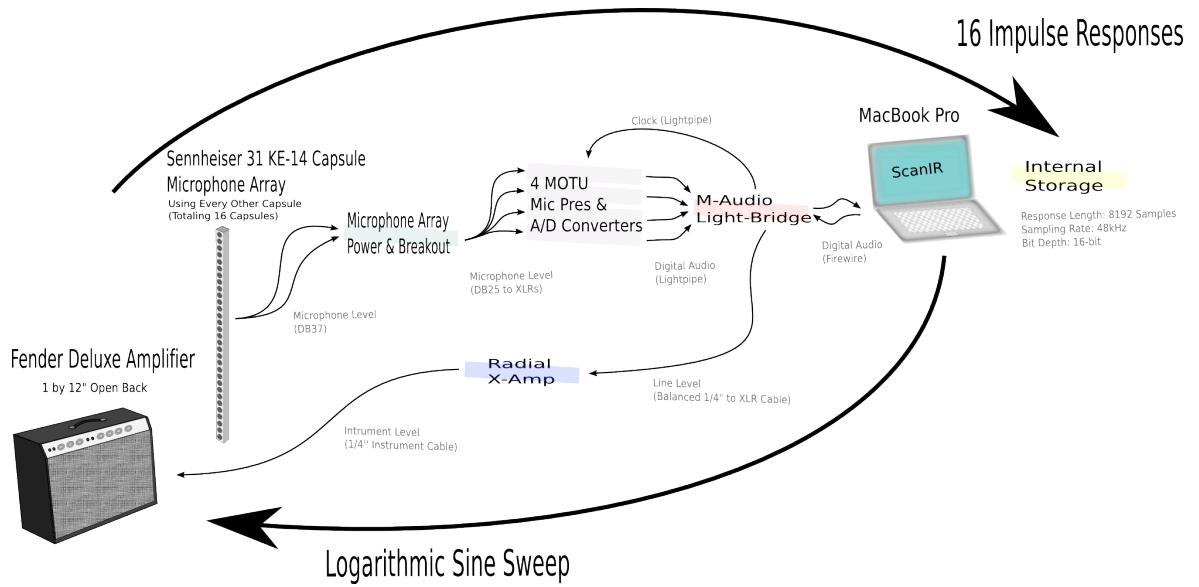
that might have been affected by such a barrier, due to their wavelengths, fall outside that range.

### **Measurement Density:**

A set number of measurement locations must be selected. Since, in theory an infinite number of points lie between any two positions in continuous space, it is physically impossible to make a measurement from every location. Although arbitrarily chosen, the microphone array's capsule spacing of 31.8mm was more dense than any field radiation pattern measurements that the author found in the Literature Review. The effectiveness of this spacing is explored in the Analysis section in detail.

### **Other Equipment:**

All measurements used the MOTU 8Pre analog-to-digital converters and the M-Audio Light-Bridge, using a MacBook Pro. The MOTU units were all connected to Light-Bridge over light-pipe cabling. All units were clocked by the M-Audio Light-Bridge. The line-level output signal of the Light-Bridge was converted to instrument level signal. For details of the signal flow used at UML, see Figure III-2.



**Figure III-2: Detailed Signal Flow (larger version attached after Works Cited).**

The Matlab impulse response utility ScanIR was used for all measurements. ScanIR generates the logarithmic sine-sweep, records the response, performs deconvolution, and stores the data (Boren & Roginska 2011).

A sampling rate of 48kHz was chosen primarily due to file size cautions. It was also known that the amplifier would not reproduce content-containing frequencies that would require a higher sampling rate.

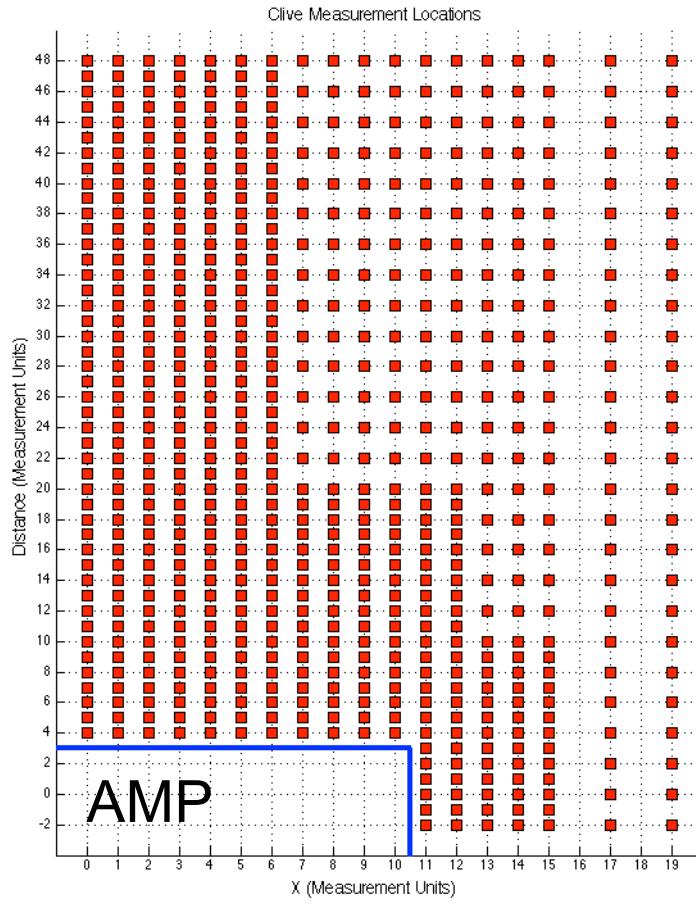
### Grid:

A rectangular/Cartesian grid was placed on the floor under the amplifier, to mark the locations of each measurement set. The microphone array was placed lengthwise vertically and as close to the floor as possible, without touching the floor. The microphone array was suspended in position with a large boom stand. Using laser levels, the array was

checked to be perfectly vertical in relation to the floor. A logarithmic sine sweep was then sent through the amplifier and the responses at the location of each of the capsules were recorded simultaneously. The array was then moved to the next location and the process continued.

### **Clive Davis Measurements (NYU)**

At the Clive Davis Institute of Recorded Music studio at NYU (Clive), impulse responses were collected using a Fender Deluxe guitar amplifier. At this location, all 31 microphones capsules were used for every *x/y* location on the grid. The grid used spacing of 31.8mm, to match the distance between each capsule. 31.8mm will be referred to as a ‘Measurement Unit’. Due to time constraints, the density of measurements was reduced as the microphone array-to-source distance was increased.



**Figure III-3: Clive Measurement Locations.**

Figure III-3 shows a bird's eye view of the locations of the Clive measurements. For each  $x/y$  location marked in Figure III-3, 31 impulse responses were collected. This resulted in 20,491 impulse responses.

In this and later figures,  $x$  refers to motion left and right and  $y$  refers to distance, forward and back from the amplifier, both as viewed from behind the amplifier.

The box in the lower left-hand corner of Figure III-3 represents half of the amplifier. The other half of the amplifier would reflect over the  $y$ -axis. It was assumed that the amplifier would radiate similarly for each side of the amplifier.

The amplifier was placed in the relative center of the live room. It was hoped that it would be possible to remove or at least diminish room reflections from the responses via time-domain windowing, using this source and measurement position. Nevertheless, during the data analysis, it became apparent that the walls and ceiling were too close to the measurement locations to be windowed away. These reflections will be discussed later, with regard to the errors that occur when interpolating between measurement points.

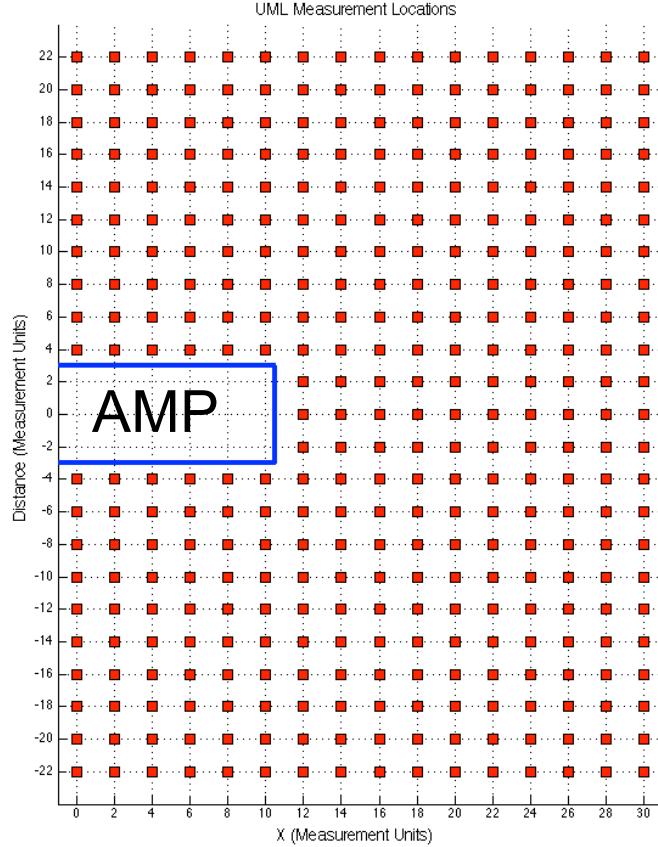


**Figure III-4: Clive Measurement Sessions.**  
The top half of the Microphone array is shown peering  
over the Fender Deluxe amplifier.  
(Photo by Jim Anderson)

### UML Measurements

At the Durgin Theater of the University of Massachusetts at Lowell (UML), impulse responses were collected from the same Fender Deluxe amplifier used for the Clive measurements. At this location, 16 capsules were used for every  $x/y$  location on the

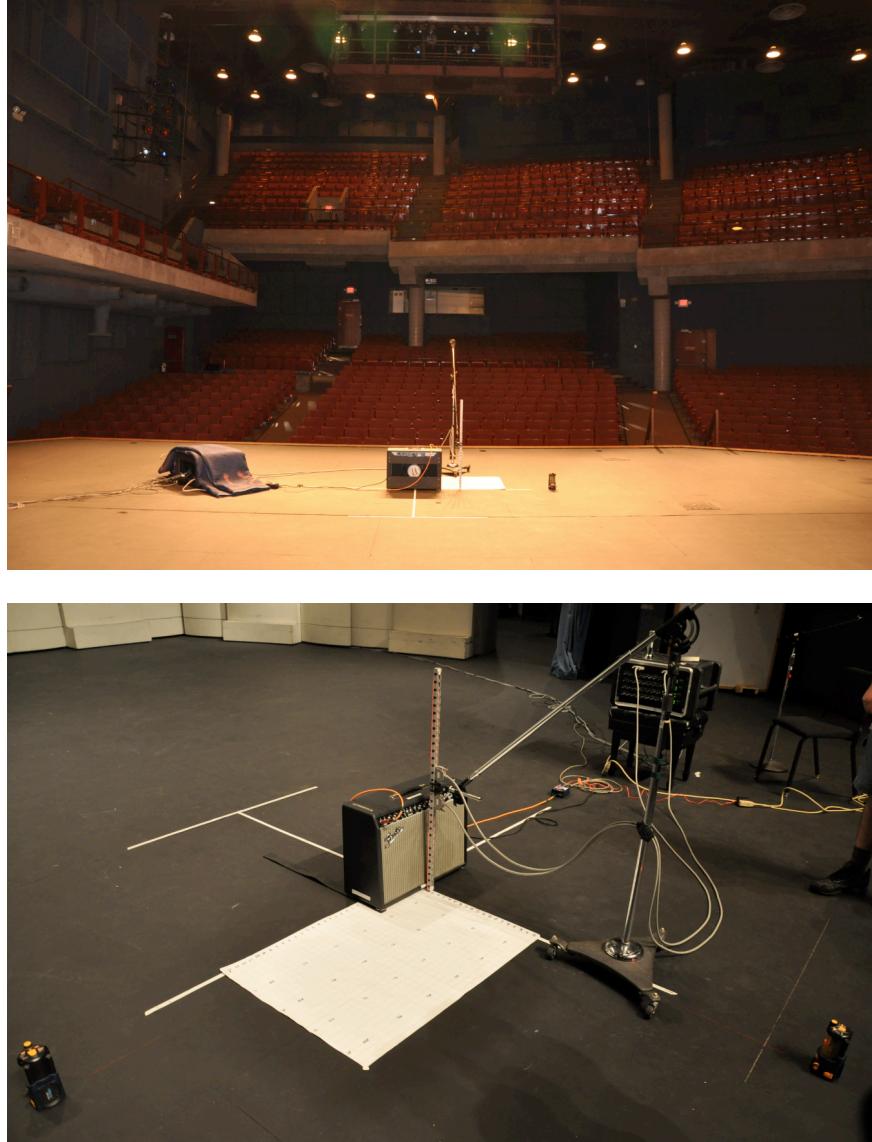
grid. The grid used spacing of 63.6mm (2 Measurement Units), to match the distance between each capsule in use.



**Figure III-5: UML Measurement Locations.**

Figure III-5 shows a bird's eye view of the locations of the UML measurements. For each  $x/y$  location marked on this plot, 16 impulse responses were collected. This resulted in 5,600 impulse responses.

The amplifier was placed at relative center-stage. In this venue, the walls of the theater are much farther from the amplifier in comparison to the Clive studio setup. This allows for time-domain windowing, as will be discussed in the Analysis section.



**Figure III-6: UML Measurement Sessions.**  
**Top:** View from behind Fender Deluxe looking out into the theatre.  
**Bottom:** View of microphone array being positioned for the first locations.  
Laser levels were used to align the array vertically.  
(Photos by Jim Anderson)

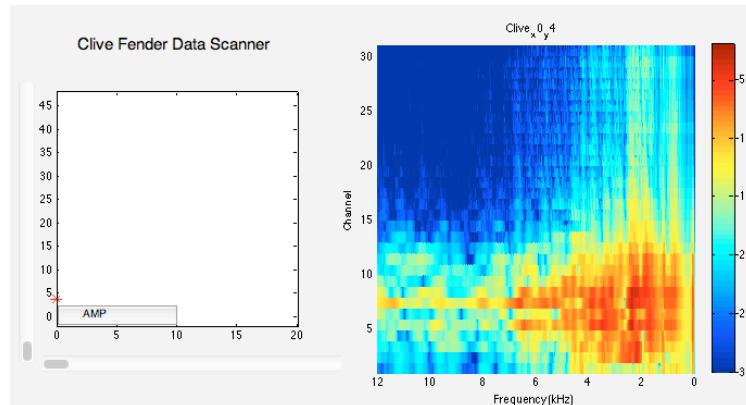
## **IV. PRELIMINARY RESULTS**

### **Overview:**

A preliminary analysis of the data sets was undertaken, using several frequency domain representations. The complexity of the radiation pattern of the amplifier was revealed with the use of utilities created by the author. These tools are described, to explain the methodology. The initial results found using these tools are also described.

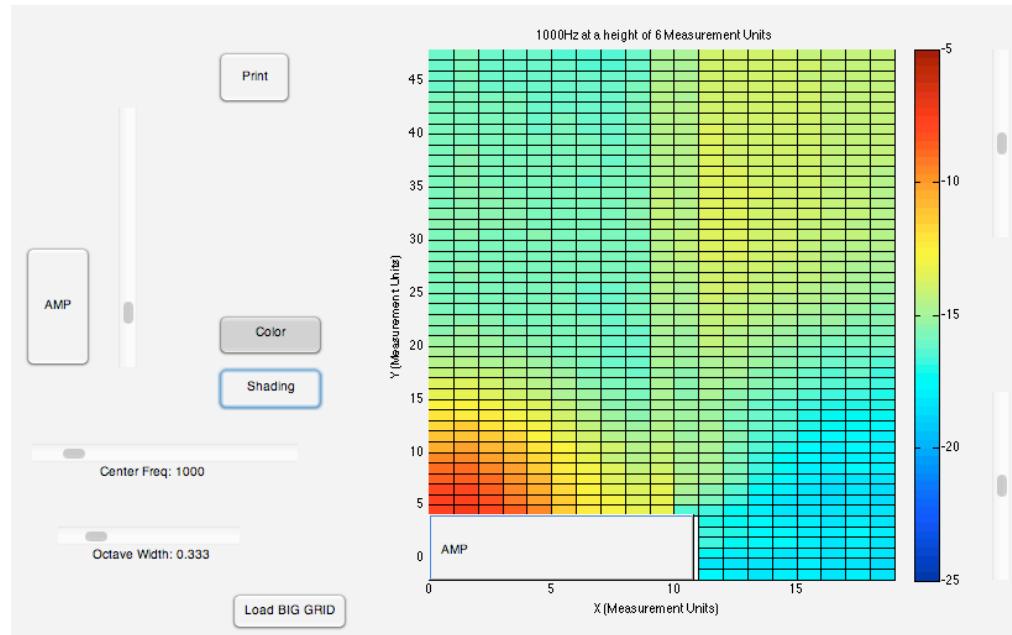
### **Navigation Tools:**

The Clive and UML data sets can be used in a variety of ways to represent the radiation patterns of the Fender Deluxe amplifier. As part of this research, several navigation tools were developed in Matlab for both data sets.



**Figure IV-1: Clive Data Set Utility - displays log-magnitude spectrum of specific locations for all capsules in the array – location (0, 4) shown.**

The tool shown in Figure IV-1, shows the log spectrum magnitude of a vertical column of measurements at a given  $x/y$  location. Inter-capsule differences were not taken into account. This tool assumes that the capsules have the same transfer function. Using this tool, the directionality of frequencies as a function of height is revealed. The intensity of higher frequencies seems to beam out, while the lower frequency energy is more diffuse. This is the only vertical plane representation that was used. Using this tool, it is possible to quickly move between locations. This ability to navigate within the data provided the opportunity to see centroids of energy move between locations.

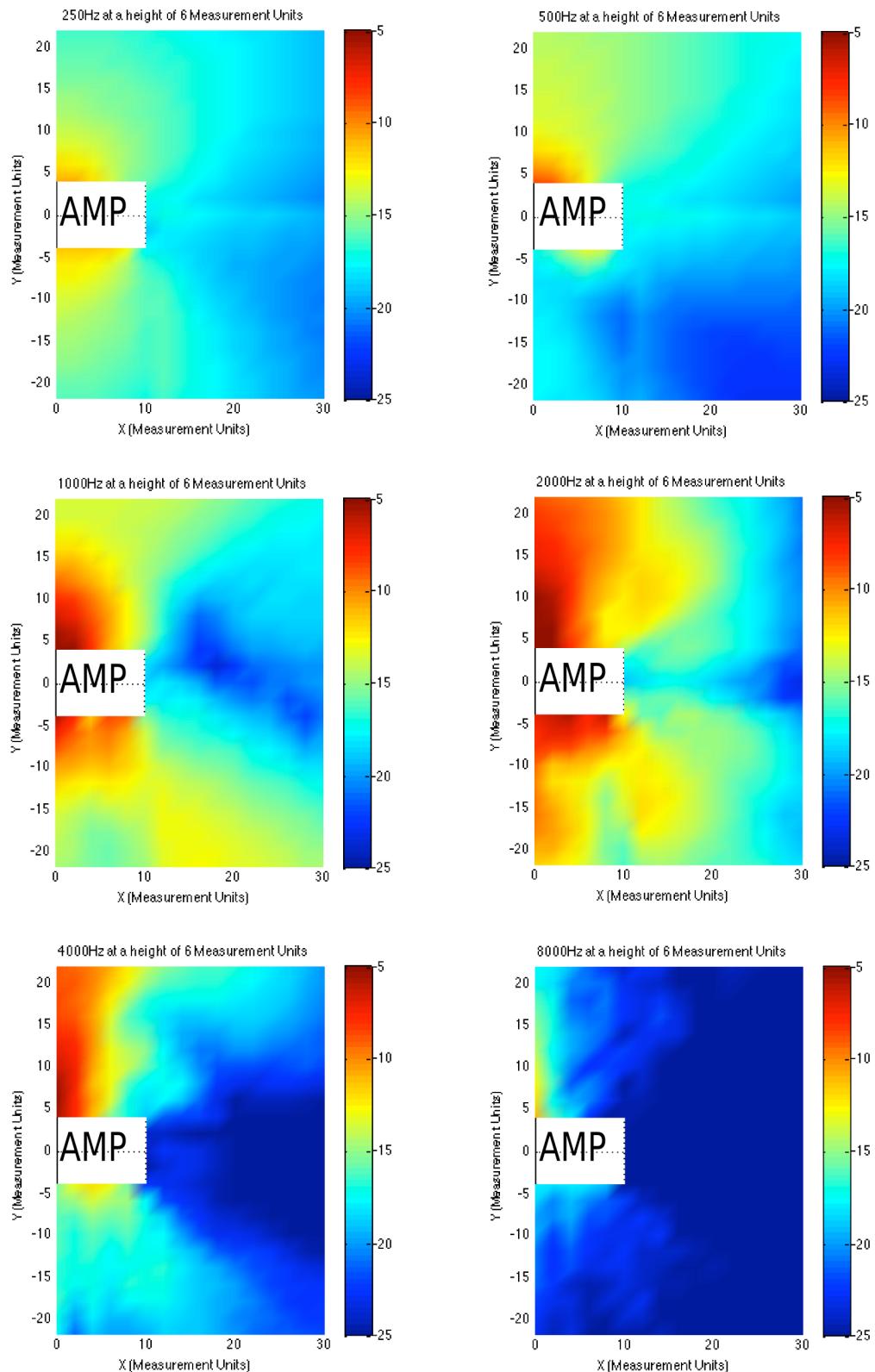


**Figure IV-2: Horizontal Plane Spectral Intensity Utility – intensity for Clive data set shown.**

The tool shown in Figure IV-2 shows the magnitude of an  $n^{\text{th}}$ -octave frequency band of a horizontal plane, relative to the floor. Each horizontal plane consists of measurements of a single capsule. As in Figures III-3 and III-5, the  $x$ -axis represents measurement locations left and right, by Measurement Units, relative to the amp. The  $y$ -

axis represents measurement locations farther from and closer to the amp. The intensity, shown in color in Figure IV-2, of a 1/3<sup>rd</sup> octave frequency band is shown in dB. This color scale is shown on the right. As an example for interpreting the plot, using an (x, y) notation, (0, 8) on the plot represents the intensity found directly on-axis with the center of the speaker cone, at a distance of 8 Measurement Units away from the center of the amp.

Looking at a variety of ascending center frequencies for 1/3<sup>rd</sup> octave frequency bands, it can be observed that the intensity of higher frequencies is more directional than that of the lower frequencies. The relative directionality of different frequencies is, of course, an audio characteristic that can be found in virtually any text that covers audio basics. Figure IV-3, on the following page, shows this characteristic for six frequency bands.



**Figure IV-3: Spectral Intensity (dB) Surrounding a Fender Deluxe Amplifier - showing ascending center frequency of 1/3<sup>rd</sup> octave frequency bands.**

Using Figure IV-3, it is clear that to attain more high frequency content, the microphone should be placed on-axis. These displays serve as a good visualization of the radiation pattern of the amplifier. The following three tables include several analyses of the quantities shown in Figure IV-3.

Frequency (Hz)	125	250	500	1000	2000	4000	8000
Intensity (dB)	15.46	10.01	14.54	18.97	20.42	22.29	22.52

**Figure IV-4: Range of Intensity (dB) of 1/3<sup>rd</sup> Octave Bands – for the 6 Measurement Units horizontal plane – between maximum intensity and minimum intensity.**

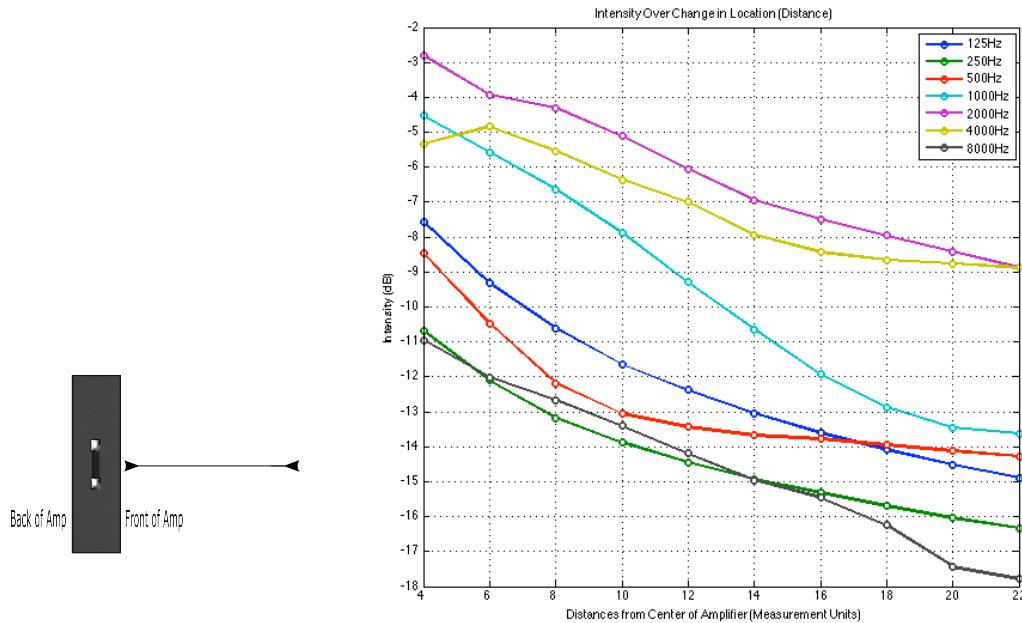
Frequency (Hz)	125	250	500	1000	2000	4000	8000
Intensity (dB)	4.12	4.59	3.99	5.74	6.37	7.10	6.31

**Figure IV-5: Maximum Change of Intensity (dB) of 1/3<sup>rd</sup> Octave Bands – for the 6 Measurement Units horizontal plane – between adjacently located capsules along the x axis.**

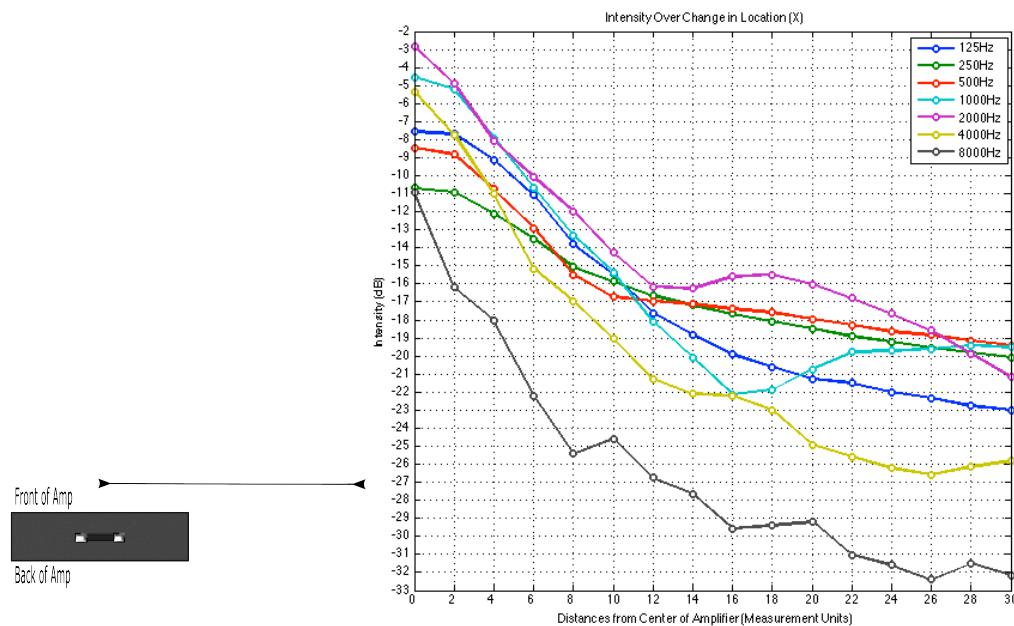
Frequency (Hz)	125	250	500	1000	2000	4000	8000
Intensity (dB)	4.06	1.51	2.72	2.79	3.56	4.98	6.67

**Figure IV-6: Maximum Change of Intensity (dB) of 1/3<sup>rd</sup> Octave Bands – for the 6 Measurement Units horizontal plane – between adjacently located capsules along the y axis.**

Figure IV-3 and consequently Figures IV-4, IV-5, and IV-6 consider the entire horizontal plane of capsules moving through the center of the speaker cone. This includes two spatial dimensions. Figures IV-7 and IV-8 display intensity of a single spatial dimension for the same 1/3<sup>rd</sup> octave bands. Figure IV-7 shows the spectral changes for capsules on-axis with the speaker cone in front of the amplifier. Figure IV-8 shows the spectral changes for capsules from the center of the speaker cone to the side of the amplifier.



**Figure IV-7: Spectral Intensity (dB) – for on-axis capsules for the 6 Measurement Units horizontal plane – shown at selected 1/3<sup>rd</sup> octave bands.**



**Figure IV-8: Spectral Intensity (dB) – for capsules moving from the center to the side of amplifier, for the 6 Measurement Units horizontal plane – shown at selected 1/3<sup>rd</sup> octave bands.**

Figures IV-7 and IV-8 provide alternate visualizations of spectral change as a spatial dimension changes. The change for two specific vectors of capsules is shown. For example, a 1/3<sup>rd</sup> octave band centered at 2000Hz shows a drop of 2dB from the nearest on-axis capsule to the capsule two Measurement Units to the side and a 5dB drop to the nearest on-axis capsule four Measurement Units to the side.

Considering the nearest and farthest measured positions for on-axis capsules, Figure IV-9 shows the drop in intensity (dB) for each of the selected 1/3<sup>rd</sup> octave bands. The first position, using (x, y) notations oriented as seen in Figure IV-3, is (0, 4) in Measurement Units. The last position, using the same (x, y) notation, is (0, 22) in Measurement Units.

Frequency (Hz)	125	250	500	1000	2000	4000	8000
Intensity (dB)	7.32	5.64	5.82	9.10	6.07	3.55	6.84

**Figure IV-9: Drop in Intensity (dB) of 1/3<sup>rd</sup> Octave Bands – for the 6 Measurement Units horizontal plane – between position (0, 4) and (0, 22).**

Considering the closest on-axis capsule and the farthest off to the side of the amplifier, Figure IV-10 shows the drop in intensity (dB) for each of the selected 1/3<sup>rd</sup> octave bands. The first position, using (x, y) notations oriented as seen in Figure IV-3, is (0, 4) in Measurement Units. The last position, using the same (x, y) notation, is (30, 4) in Measurement Units.

Frequency (Hz)	125	250	500	1000	2000	4000	8000
Intensity (dB)	15.41	9.39	10.94	14.99	18.35	20.47	21.21

**Figure IV-10: Drop in Intensity (dB) of 1/3<sup>rd</sup> Octave Bands – for the 6 Measurement Units horizontal plane – between position (0, 4) and (30, 4).**

Using Figure IV-3 alone, a recording engineer can gain a high level understanding of what kind of responses should be expected at numerous different microphone locations when considering a single frequency band. The plots reveal the complexity of the radiation pattern of the Fender Deluxe amplifier, including the dramatic differences between locations, which were discretely sampled. The plots in Figure IV-3 were graphically smoothed to suggest what happened in continuous space. This smoothing will be indirectly validated in the Analysis section. Even with such smoothing, however, a visualization only offers a sight-based approach to an understanding of the sound source. A recording engineer, however, is after a certain ‘sound’, as opposed to a visualization. Therefore, the recording engineer could make more use of this data when coupled with an auralization.

## **V. ANALYSIS**

### **Overview:**

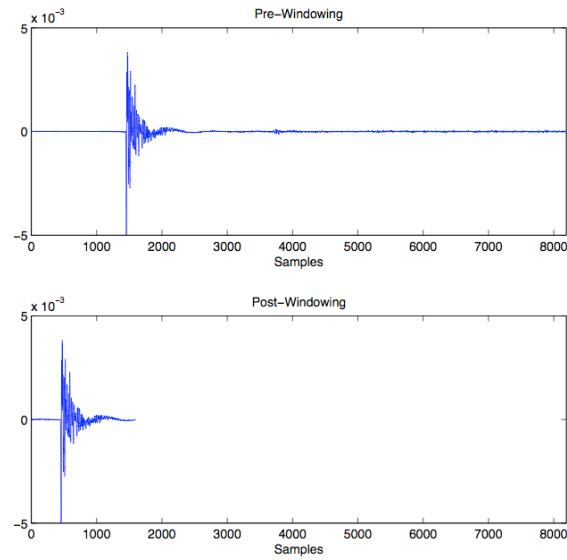
The preliminary analysis indicated that linear interpolation may be valid. To prepare the data for interpolation, time domain windowing was implemented. Each filter was then aligned in various ways. Converting the data to minimum phase filters was ultimately chosen to provide the necessary alignment. Interpolation between capsules, on a vertical and horizontal basis, was completed for comparison with a recorded ground truth filter. This was used to provide a basis for interpolating between adjacent capsules.

### **Linear Interpolation for Auralization:**

Can two recorded positions serve as an auralization of microphone position? As will be described in the Discussion section... not in a sufficient way. Therefore, toward the implementation of an effective auralization tool, the problem of measurement density is quickly apparent. One of the primary take-aways from the aforementioned data navigation tools is that the spectral radiation seems to have some linear behavior, upon casual visual inspection. When flipping through the plots using the first navigation tool, centroids of energy seem to move linearly for changing microphone locations along the  $y$ -axis, especially when  $x$  is 0, i.e. on-axis with the speaker cone. This apparent relative linearity offered hope for a simple linear interpolation, to virtually increase measurement density. The following is an account of the steps taken during the analysis of interpolation as a potential solution for auralization.

### Time Windowing:

Widnowing in the time domain was attempted for both the UML data set and the Clive data set. For the Clive data set, windowing in the time domain was not possible, because the early reflections arrived too soon. The room reflections were found to clash in time with the reflections corresponding to the amplifier's radiation. For the UML data set, however, time domain windowing was possible, because of the larger size of the room in which the measurements were taken. All of the UML impulse responses were windowed to 1,600 samples. This limits the data to quasi- "anechoic" responses, even though the floor reflection could not be removed.



**Figure V-1: Pre- and Post- Time Domain Windowing of UML Fender Deluxe Impulse Response.**

### **Time Domain Alignment:**

To interpolate between two filters, the filters must have an appropriate time relationship. Timing wise, they must be as though they were recorded as a coincident pair. For example, the sample index in a response that corresponds to the onset of the impulse, i.e. the direct sound, must be identical for both filters. They need to “line up”, so to speak. Without post-processing, this is, of course, impossible, since the goal of the measurements is to see what happens at different locations. The initial onsets will not “line up” unless the impulse responses in question are taken from equidistant locations to the sound source. Several options were considered for the purpose of “lining up” the responses, as detailed below.

The onset of the direct signal in an acoustic impulse response often has the peak absolute amplitude. This is certainly not always the case, but is common. At this stage, alignment was visually judged by the superimposed plotting of both responses. Lining up the maximum absolute values provided reasonable results for most impulse responses. For both the UML and Clive data sets, several of what is believed to have been the floor reflections had a higher peak absolute amplitude than the initial onset. For this reason, response alignment was sometimes off by as much as 1,000 samples. Yet, when the maximum values did correspond to the direct onset, the responses did visually appear to be in sync.

The cross-correlation of two impulse responses determines their best matching overlap. This method provided results that were more consistent, when compared to the maximum method, in general alignment, meaning the alignment was never as far off as it

was in the extreme cases of the maximum method, but consistently the onsets visually seemed misaligned by several samples.

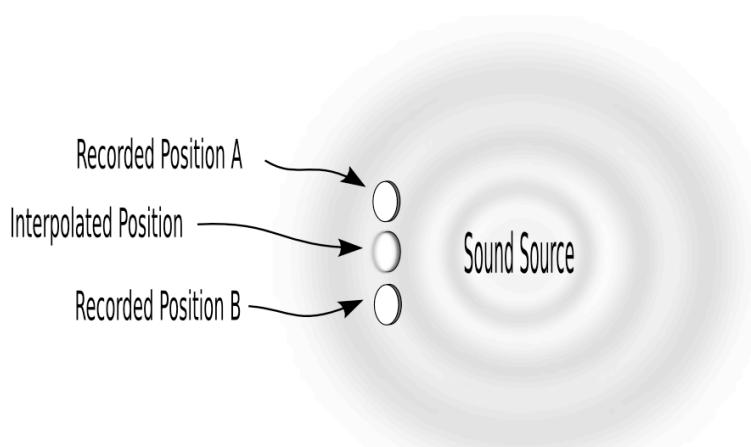
This trouble in achieving alignment was to be expected, because true alignment of the recorded impulse responses taken from different receiver positions is actually impossible. The onsets of the direct sound, the floor reflections, and other reflections from the cabinet and room all have different time relationships for each receiver location. Even with the best possible direct onset alignment, for example, these other onsets would be misaligned.

As mentioned in the Literature Review section, Masterson et al. (2009) use dynamic time warping as a solution to this problem. Also, as mentioned by Masterson et al., if this problem is not addressed, there will be a smearing of reflections in the time-domain of the interpolation.

For this research, the decision was made early on to focus exclusively on radiation patterns as seen through spectral changes. As a result, all reflection time differences can be removed, as long as the spectral magnitude stays consistent. This is exactly the point of using minimum-phase filters (Smith). By removing the phase, all the filters are as perfectly aligned as possible. The phase relationships cannot be reconstructed from the minimum phase filter, but since the spectral magnitude is being considered here, this is not a problem.

For each method of alignment tested, spectral interpolation error, as discussed in detail later, paralleled the casual visual inspection of alignment. The minimum phase filters resulted in less spectral interpolation error than the cross-correlation method, which in turn resulted in less spectral interpolation error compared with the maximum method.

## **Interpolation Methodology and Resulting Error:**



**Figure V-2: Interpolation Position.**

As mentioned earlier, the lack of infinitely dense measurement locations is an inescapable problem. The duration of the measurement process, which is a function of the density of the measurements, was also found to be an important factor. A compromise must be made between the density of measurement locations and the time needed to take such measurements. At the end of the day, the question is how sparsely can the measurements be taken to still give an acceptably precise picture of the radiation patterns.

Looking at this problem retroactively, we cannot concretely determine if the measurement density was too sparse, with interpolation error as the only metric. As will be explained, we can only concretely conclude if it was more than twice as dense as ‘needed’. The density ‘needed’ depends directly on the intended use of the data set.

The spectral differences between two adjacent locations cannot be used to conclude whether or not the measurements were dense enough for interpolation. To that end, the

differences may even be relatively extreme and yet the density still adequate. The more accurate the interpolation between measurements is found to be, the fewer measurements are needed. For example, if every desired point could be accurately interpolated from just three recorded points... then only three points would be necessary.

In order to characterize the accuracy of interpolation, a method of determining the interpolation error is required. By using every other measurement, we can see how accurately interpolated responses can be generated with the ground truth of the recorded response. For illustration, using three capsules in a series A-B-C, an interpolation is computed between A and C. This interpolated B is compared with the recorded B. If this interpolation is found to be accurate enough for the application, then the measurements could possibly have been more sparse. If the interpolation is not found to be accurate enough for the application, it is possible that the measurements should have been more dense.

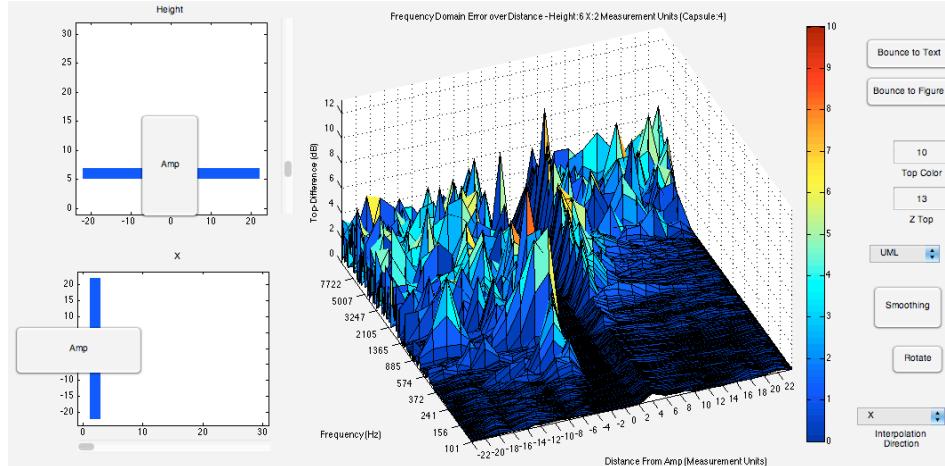
If the spectral change between points is in fact perfectly linear, interpolation is then both simple and accurate. Therefore, a basic frequency domain linear interpolation was chosen for this analysis, to explore the spectral linearity between points. Even though linear interpolation is, of course, not the only interpolation method that could have been used, it was the only method tested here.

The following process was completed separately for all horizontal planes of data. Only one capsule was used to measure a specific height. Therefore, each horizontal plane corresponds to data collected using a single capsule:

- All of the filters are converted into minimum phase filters.
- The matrix of time domain responses is converted to the frequency domain via the Fast Fourier Transform.

- Two copies of the matrix are offset, scaled, and summed, so that the filters of a specific ( $x$ ) location are averaged with the filters of ( $x+2$ ). This results in a matrix of interpolated filters.
- The Euclidean distance is found between the log magnitude of the ( $x+1$ ) filters and the log magnitude of the interpolated filters. This generates a matrix of interpolation error for each location and frequency bin.
- The error matrix is smoothed to 1/24<sup>th</sup> octave bands for visualization.

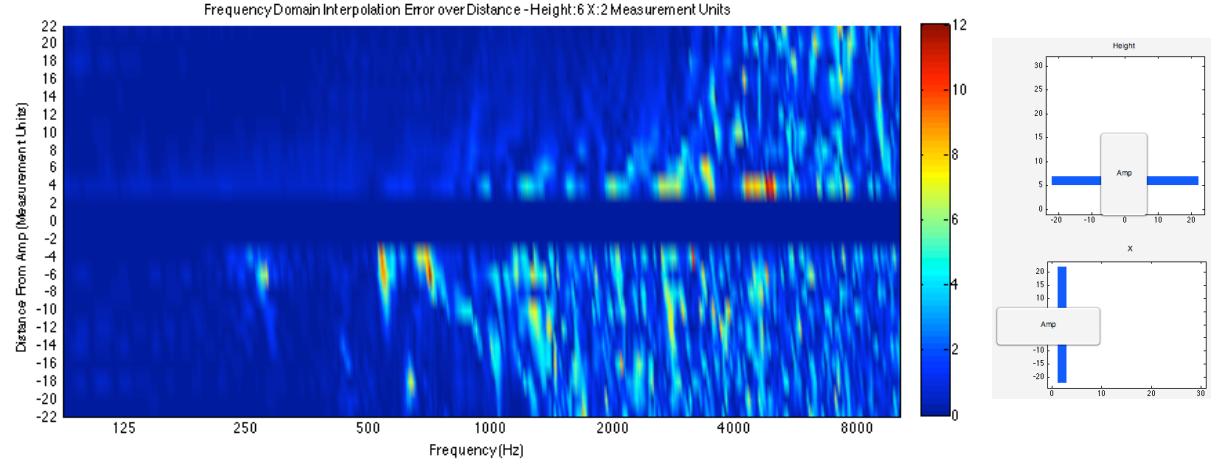
A Matlab GUI, shown in Figure V-3, was built to navigate the error plots. In the GUI, the user can surf through visualizations of the error at each capsule height and each  $x$  position. Each plot shows the dB difference between the recorded and interpolated results for all distances, across 1/24<sup>th</sup> octave bands.



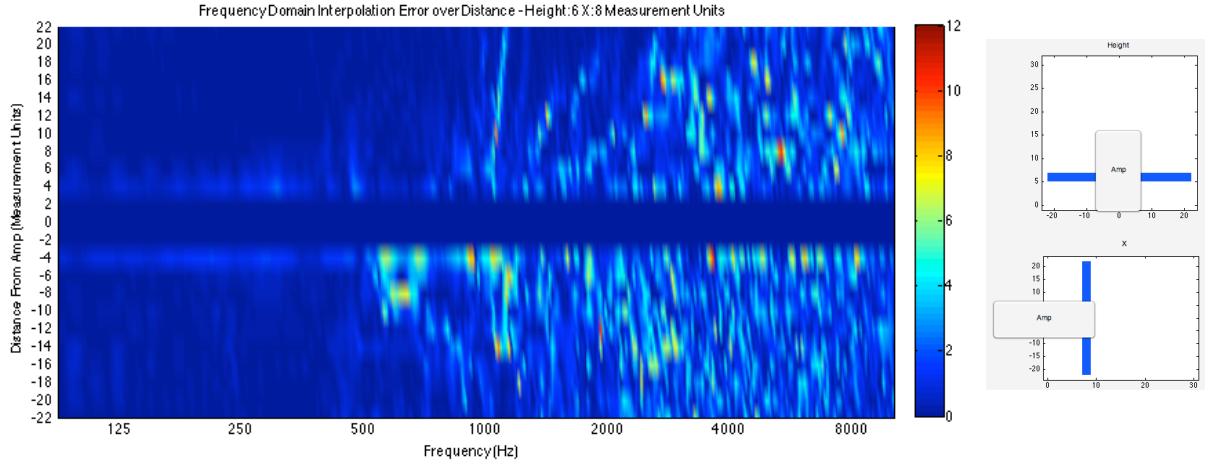
**Figure V-3: Interpolation Error Utility.**

The frequency domain was windowed to 100Hz -12,000Hz. For higher frequencies, there is a significant drop in intensity when approaching 6-8kHz. The response for frequencies above this threshold are thought to be primarily noise and relatively irrelevant. Below 100Hz is not shown because of the resulting overuse of

frequency bin data points with limited resolution, when smoothing to 1/24<sup>th</sup> octave bands for lower frequencies.



**Figure V-4: Frequency Domain Interpolation – over distance for capsules 63.6mm off the center of the speaker driver.**



**Figure V-5: Frequency Domain Interpolation – over distance for capsules 254.4mm off the center of the speaker driver.**

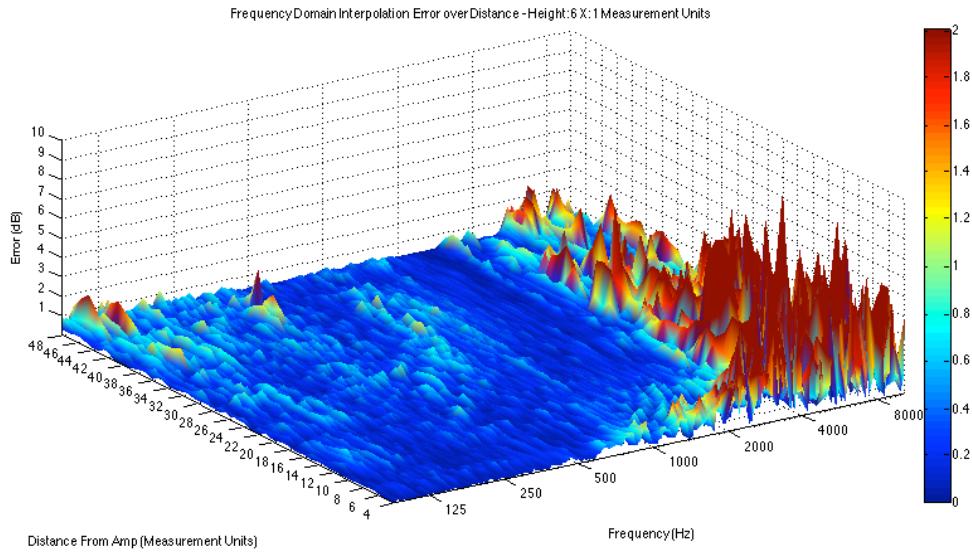
The interpolation error plots, as seen in Figures V-4 and V-5, confirm the thesis that spectral linearity between Cartesian spaced radiation pattern measurements is a function of distance from the sound source and is frequency dependent. When the microphone array is close to the amplifier, interpolation is less accurate than when the

microphone array is farther away from the amplifier. This is especially true with regard to the higher frequencies.

		Peak Interpolation Error					
		Frequency Band (Hz)					
		125 - 250	250 - 500	500 - 1000	1000 - 2000	2000 - 4000	4000 - 8000
Distance from Center of Amp (Measurement Units)	22	0.317	0.176	0.490	0.496	3.060	8.023
	20	0.166	0.398	0.405	0.481	2.006	7.979
	18	0.358	0.496	0.635	0.818	3.150	5.743
	16	0.151	0.270	0.694	0.771	3.801	7.801
	14	0.194	0.519	0.998	1.204	5.705	3.480
	12	0.152	0.382	1.365	1.175	3.769	3.764
	10	0.274	0.530	1.638	1.340	6.160	5.586
	8	0.407	0.735	2.054	2.697	4.344	5.083
	6	0.550	0.826	1.889	3.587	7.801	4.622
	4	0.810	0.861	3.746	6.010	8.195	11.355
	2	AMP					
	0						
	-2						
	-4	2.822	2.925	9.507	5.548	10.123	6.116
	-6	1.700	6.567	12.298	7.699	7.093	8.590
	-8	0.509	2.851	5.663	4.069	6.526	6.682
	-10	1.199	1.747	4.659	7.373	8.097	4.166
	-12	0.710	1.433	4.881	7.921	7.849	6.463
	-14	0.647	2.583	5.603	5.502	5.876	5.676
	-16	0.270	2.513	4.856	6.803	8.588	4.716
	-18	0.397	0.899	6.742	5.378	6.142	5.758
	-20	0.229	0.976	3.080	5.965	4.954	6.252
	-22	0.167	0.903	1.611	8.276	4.566	5.901

**Figure V-6: Peak Frequency Domain Interpolation Error over Distance – showing peak errors corresponding to Figure V-4.**

The time domain windowing of the UML data set increased the linearity of the interpolation. The room effect of the Clive data set is evident in the interpolation error plots, where the error is reintroduced as the array was moved away from the amplifier. As the microphone array is moved away from the amplifier, the sound has more time to interact with the room before reaching the microphone array.



**Figure V-7: Frequency Domain Interpolation Error - over distance for the Clive data set - showing a reintroduction of error in frequencies below 1kHz, as distance is increased.**

This error does not return in the UML data set. For this reason, the conclusion can be made that, for the Clive measurements, the room contributed non-linearities to the radiation pattern measurements.

### Interpolation Direction:

For a basic frequency domain interpolation between two points, the desired point must lie on a line between the two points.

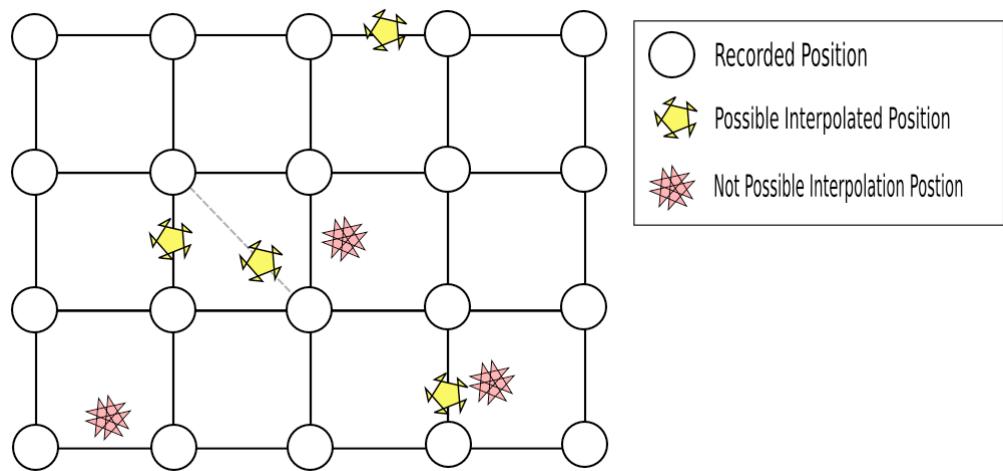
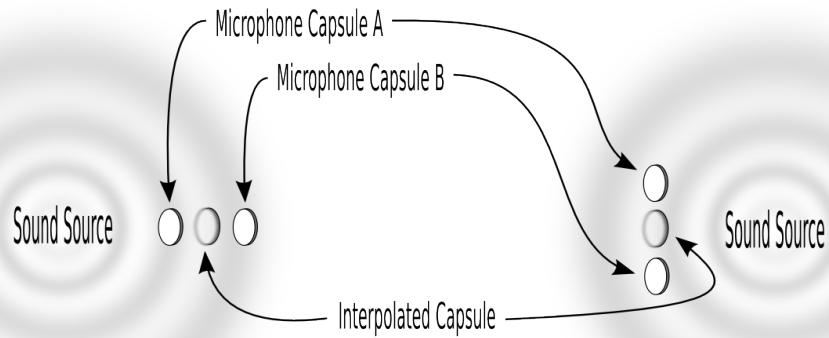


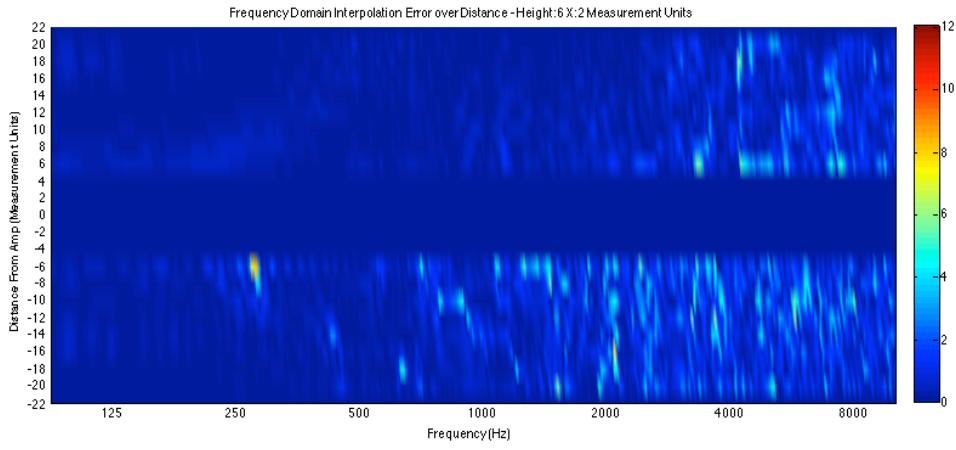
Figure V-8: Potential 2-Point Interpolation Examples.

Capsules which can be diagonally grouped were not analyzed in this study. The distances between the capsules on the diagonal would differ from the distances on the non-diagonal, so comparisons between the directions would not be valid. Therefore, interpolation error was tested in only two directions, in the form of a Cartesian grid.

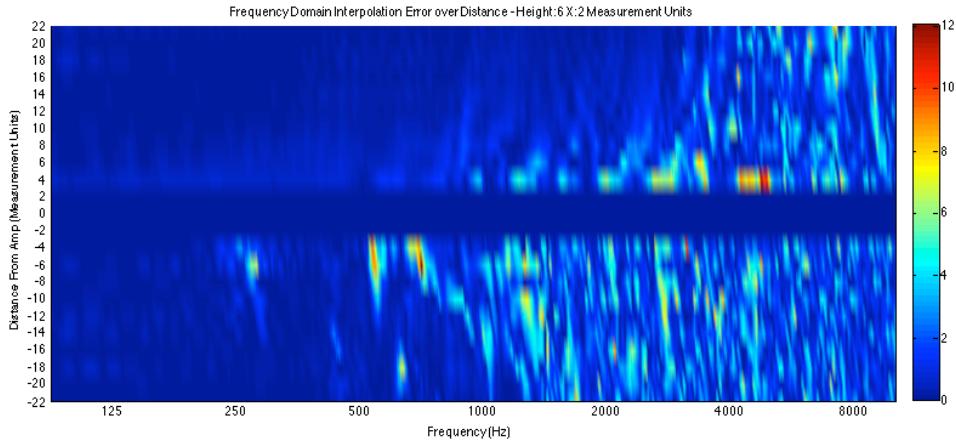


**Figure V-9: Two Interpolation Directions Considered.**

When interpolating between capsules in line with the sound source -where a straight line can be drawn through the sound source, the first and the second capsules (Figure V-9 left)- interpolation is more accurate. When interpolating between capsules not in line with the sound source -where a triangle must be drawn between the sound source and the first and second capsules (Figure V-9 right)- interpolation is less accurate. This interpolation error effect is shown in Figures V-10 and V-11.



**Figure V-10: Frequency Domain Interpolation Error – over distance using interpolation by distance.**



**Figure V-11: Frequency Domain Interpolation Error – over distance using interpolation by  $x$ .**

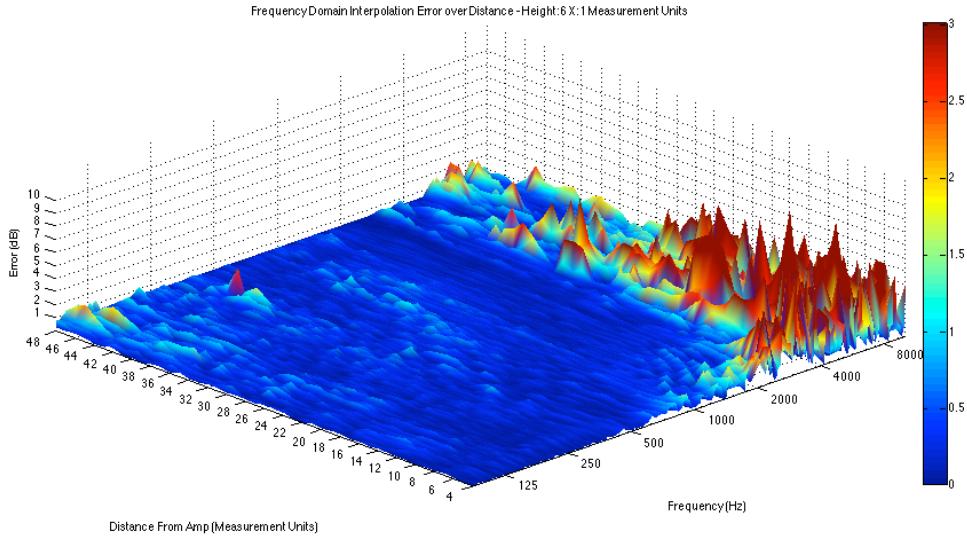
These error differences were to be expected, since sound radiates outward. These results indicate that the accuracy of linear interpolation in the frequency domain is also a function of the angle relationship between the source and the two recorded points. It also suggests that interpolation error may be reduced using a radial data set, but such a claim was not tested.

The Matlab ‘Sweet Spot’ auralization tool, to be described, makes use of both directions of interpolation and has the option to also use a four-point interpolation. The four-point interpolation allows for any location within the measurement grid to be interpolated.

### **Interpolation Between Adjacent Locations:**

The ultimate goal of interpolation in this research is to virtually increase the density of measurements, rather than to retroactively check the actual density measured. As mentioned earlier, the ‘needed’ accuracy of the interpolation should be determined by the task at hand. For the UML data set, for example, it was the decision of the author that this interpolation provided results accurate enough to attempt to further interpolate between the discrete points.

For some justification, comparisons can be made between the two data sets. Although the Clive data set cannot be considered a one-to-one comparison with the UML data set, because of the aforementioned room effect, this effect should only make the linear interpolation somewhat less accurate. The interpolation error for the Clive data set, on-axis and near on-axis capsules, provide less than 3dB error for all frequencies up to 2kHz (see Figure V-12).



**Figure V-12: Clive Frequency Domain Interpolation Error - showing less than 3dB error for frequencies below 2kHz.**

One would expect an even lesser error from the UML data set, because that set does not include the non-linear effects of the room reflections. How much greater cannot be determined, for lack of an intervening capsule for comparison.

The auralization implementations here can expect the above mentioned level of accuracy and it is up to each user to determine if it is acceptable. Ultimately, subjective testing comparing interpolated positions versus recorded positions would be necessary to determine the interpolation accuracy needed to remain perceptually identical. These results would no doubt be different for sine tones and a guitar signal. Again the specific applications of the interpolation ultimately drives the accuracy ‘needed’ decision.

Interpolation error analysis can also be used to visually inform the recording engineer of how the rate of change in timbre is affected by the distance from the sound source (Roginska et al. 2012). The interpolation error discussion here is presented to give a metric for the accuracy of the auralization tool.

## **VI. AURALIZATION**

Learning where exactly to place a microphone to record an amplifier takes years of experience. One needs the so-called “years-in-the-shed” to practice and, in relevant terms to this discussion, build up one’s own mental capture pattern data set. Engineers will often talk about moving the microphone outward, to gain a more ‘full’ effect, or inward, toward the center of the speaker cone, for a ‘brighter’ effect. Being able to accurately compare these different locations mentally is a skill of the trained recording engineer.

This process of testing out many microphone locations around an amplifier is a time consuming task. One must place a microphone in front of an amplifier in a live room, return to a control room and then audition the placement. Would .5 cm higher be better? Maybe? Move back into the live room... move the microphone... return to the control room... and audition again. This process takes time that a modern day recording engineer does not always have.

In the classroom, the effect of moving a microphone relative to an amplifier can be recognized by comparing several recorded microphone placements, but, especially to the ears of one just learning to appreciate detailed critiques of timbre, this mental data set is likely too sparse to fully understand the differences.

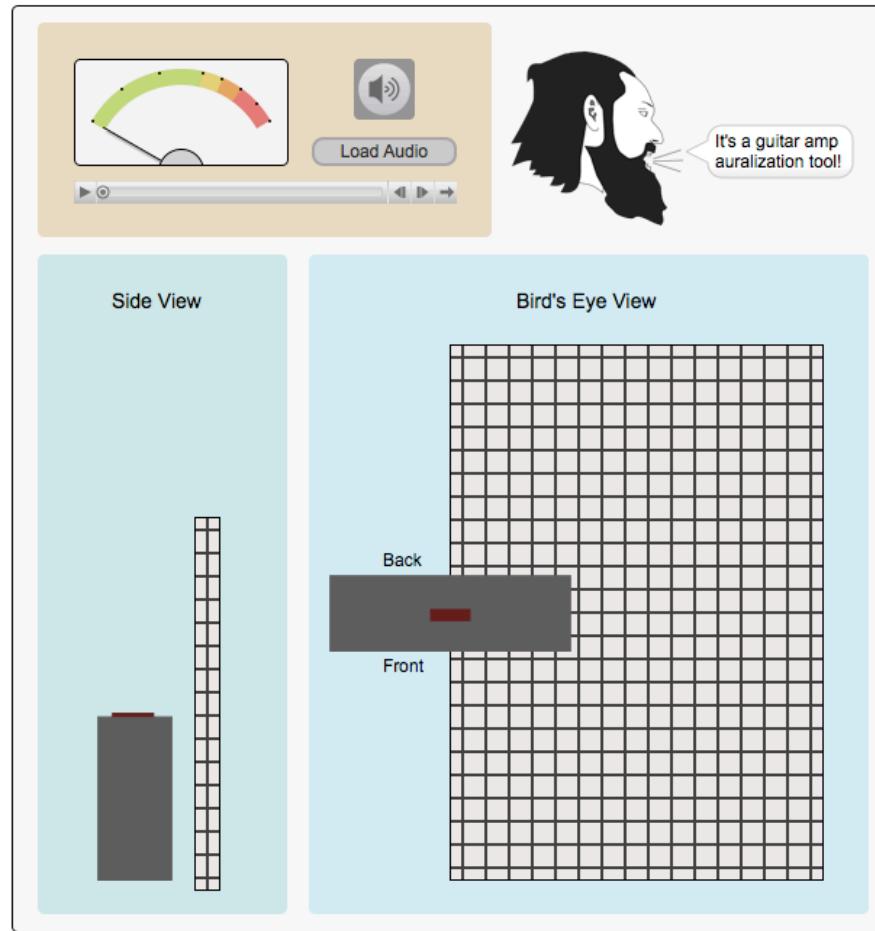
The Clive and UML data sets used omnidirectional microphones with a relatively flat response. Although learning the effect of different microphones is absolutely

necessary, even this data set with a single type of microphone can give a student an accurate appreciation of the differences that microphone placement makes. These studies were, after all, designed to record accurate radiation pattern measurements, so that the responses could inform microphone placement, regardless of other variables such as the effect of specific microphones.

Using an auralization tool by which auditioning microphone placement is instantaneous, as opposed to the physical trial and error method, would benefit the professional recording engineer and especially the student recording engineer. This can already be done to some extent with commercially available software, but not nearly with the high measurement density offered here.

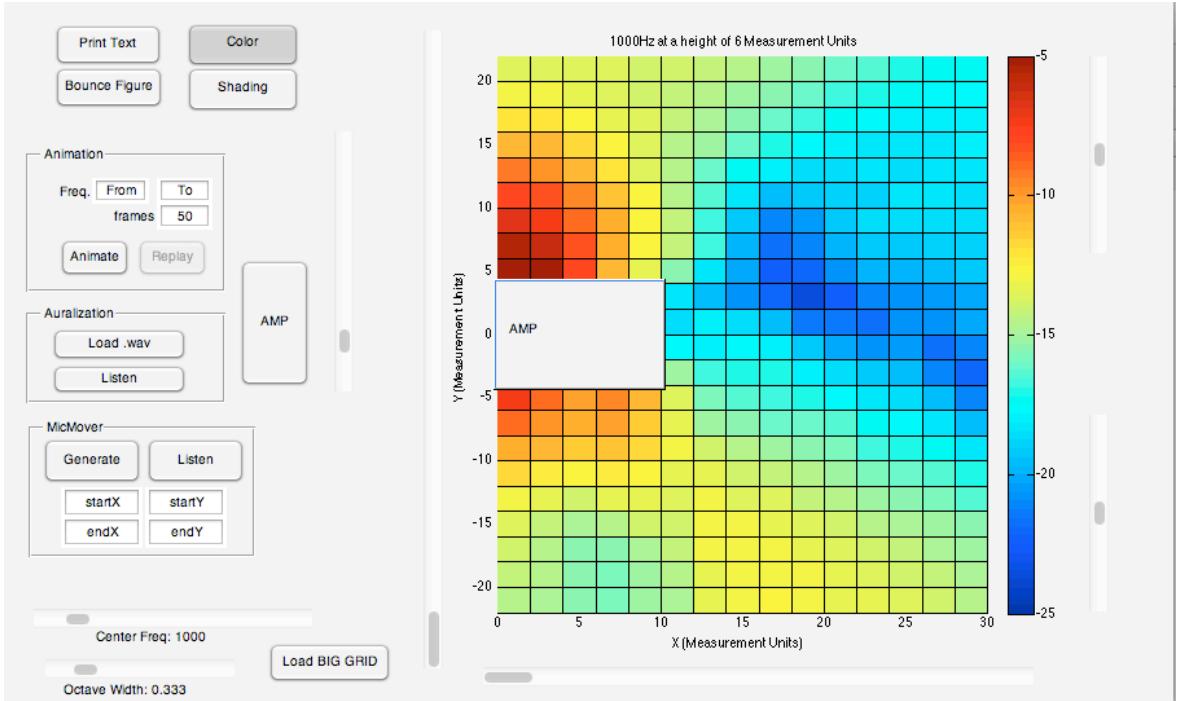
### **Auralization Implementations:**

The primary challenges in any impulse response-based auralization tool is navigating the impulse response library effectively and performing convolution. Using Max/MSP 6, the digital signal processing behind auralizing the amplifier is a relatively simple task. Built-in user interface objects like matrixctrl provide a graphical and interactive representation of the data set. This object allows the user to select the desired impulse responses from the library. Filter objects like buffir~ perform the convolution between the impulse response and the input signal.



**Figure VI-1: Presentation Mode of the Max/MSP ‘Sweet Spot’ Auralization Tool.**

The Max/MSP ‘Sweet Spot’ auralizer tool serves as an example interface for navigating the impulse response library. Using this tool, the user can navigate 5,600 impulse responses by requesting a position around the amplifier. When the user selects a position, the impulse response is read into the `buffir~` object. This object then performs real-time convolution between an input signal and the impulse response. The input signal can be any mono audio stream in Max/MSP. This can come from a wave file, for example, or a live direct guitar feed.



**Figure VI-2: Matlab ‘Sweet Spot’ Auralization Tool.**

An auralizer was also built in Matlab, to audition different locations while coupling this sonic data with visual data. In this case, the tool mentioned earlier, originally developed to only display  $1/n^{\text{th}}$  octave-bands magnitude for a horizontal plane, was also outfitted to include an auralization function. This auralization tool is equipped with linear frequency domain interpolation. Any point on each horizontal plane can be auditioned.

As mentioned earlier, when interpolating between points directly between two capsules, a two-point interpolation can be completed. For any point not between two capsules, a four-point interpolation is used. No testing was completed to verify the four-point interpolation’s accuracy. Once an impulse response is interpolated, it is convolved with the input wave file and played.

Using the gradient mode of the Matlab ‘Sweet Spot’ auralizer, a wave is generated to best depict a moving microphone. This allows for an aural sweep through the radiation patterns. The starting and ending points are selected. For every sample in an audio file, a complete impulse response is interpolated. Since both the impulse response and the audio file are discrete signals, this method allows for the most detailed simulation of a moving microphone allowed for by this data set and interpolation method.

## **VII. DISCUSSION**

To auralize a system with discrete settings, one can start by capturing the response of the system at each setting, forming a discrete data set. Some equalizers, for example, use discrete settings -turn the knob either 1,000Hz or 1,500Hz; settings in between are not available! For recall-ability of a mix, discrete settings can be a godsend, but for aesthetically driven mixing and recording, especially without time restraints, discrete settings are somewhat limiting.

With continuous settings, an engineer grabs the knob, turns it a little to the left and little to the right until it sounds spot-on. In this case, the engineer can theoretically hear the result of infinite possible settings, allowing for a level of precision that a discrete system cannot offer.

In the recording studio, microphone placement is often treated as if it were a discrete system, even though a microphone can be placed at an infinite number of locations between any two points. But to physically interact with microphone placement as one would interact with, say, a continuous control equalizer, the engineer would need either a precise and dexterous microphone placement robot or a super steady-handed, mind-reading, and possibly soon-to-be-hearing-impaired intern. In practice, each position must be auditioned separately, with separate recordings at each location, which takes precious time. For these reasons, auditioning many locations is often not practical. Rather, the

engineer must depend on his or her ability to mentally interpolate between the various placement locations, hopefully to arrive at the optimal one.

In the real world, an engineer often becomes comfortable with a few microphone locations in front of an amplifier, which become the 'go-to' locations. That engineer eschews experimenting with various placement locations and relies instead upon his or her ability to make adjustments later on, during the mixing phase, when settings can be manipulated more quickly.

Virtual auralization of microphone positions around an instrument allows a recording engineer to quickly audition many potential locations. It can be used either as an educational tool during training or as an integral tool within the recording process itself. Using a tool like the Max/MSP 'Sweet Spot' auralization tool, the recording engineer can promptly consider all the discrete locations measured in a data set.

In this case, however, the auralizer is a discrete system. Without some kind of interpolation, the benefits of auralization are primarily limited to saving the engineer some precious time, by zeroing in on a general location. This alone is enough to make the tool worthwhile. After all, a student recording engineer needs to learn the timbral differences between a recording captured by a microphone say around 1cm from the amplifier versus around 10cm from the amplifier. The density of the two data sets studied here serve this purpose well. The Max/MSP 'Sweet Spot' auralization tool provides many examples at the fingertips of a user.

If a single impulse response were taken of a room, it could be used to auralize the single source-to-listener relationship in that space. If two impulse responses were taken of a room, they could be used to auralize those two source-to-listener relationships in that

space. They each serve as discrete samples of the room's acoustics. Discrete samples do not serve to auralize microphone placement as well as they serve to auralize the system being captured, in this case the room. When auralizing microphone placement for recording a Fender Deluxe amplifier, as opposed to auralizing a Fender Deluxe amplifier, the focus on the density of the data set is greater. A single impulse response, for example, although capable of auralizing the amplifier, is not capable of auralizing the difference between microphone placements.

For this reason, interpolation is necessary for the complete auralization of microphone placements to record a Fender Deluxe amplifier. With the use of the basic auralization mode of the Matlab 'Sweet Spot' tool, an engineer can audition microphone placement locations both at and anywhere between the discrete locations measured in the data set. The continuous gradient mode of the Matlab 'Sweet Spot' tool allows a more precise understanding of the effect that small adjustments to microphone locations will have on the timbre. Such a tool removes the limitations of a discrete system.

As mentioned, measuring positions at infinite locations between two points is physically impossible. Recording at every perceptually different location is practically impossible. The measurable differences between minute receiver relocations is significant. The significance of these differences is suggested by the complexity of the radiation pattern of a musical instrument, as seen in visual representations, such as Figure IV-3. The perceptual significance of this difference is suggested by the practice of the recording engineer..., i.e. just ask a seasoned recording engineer how much microphone placement matters.

The interpolation error allowance that would be non-perceivable has not been studied here. As mentioned earlier, subjective testing would need to be carried out using musical signals, to determine the range of such an allowance. The perceivable difference between two sine tones, for example, may inform toward this minimal error, but likely does not tell much useful information for the recording engineer judging ‘tone’. Without subjective testing, the effectiveness of the auralization implementations discussed here cannot be concretely determined. If found to be too great of an error, the density of the measurements or the type of interpolation would need to be reconsidered. With that said, it is the author’s belief that the results shown here indicate the Matlab ‘Sweet Spot’ auralization tool is promising.

## **VIII. CONCLUSIONS & FUTURE WORK**

### **Conclusions:**

Radiation patterns of the instruments of musical creation are complex. In order to gain a complete knowledge of that complexity, real world measurements must be taken. In this thesis, a radiation pattern measurement methodology used by the team headed by Dr. Agnieszka Roginska (NYU), Jim Anderson (NYU), and Alex Case (UML), was relied upon and the resulting data sets were used to perform some preliminary analysis. Linear interpolation in the frequency domain was performed and analyzed to provide an increased density of data points. This interpolation proved to be a function of distance from the sound source and to be frequency dependent. The room effect found in the Clive data set added non-linearities, which were not present in the UML data set.

This technique of interpolation was then used for the application of auralization. Auralization of microphone placement is now possible for any location around a Fender Deluxe amplifier, within the bounds of the recorded data set. The flexibility offered by such interpolation is found to be necessary for a meaningful auralization of microphone placement for recording a Fender Deluxe amplifier.

## **Future Work:**

### **Interpolation Methods**

For this analysis, linear interpolation proved to be an effective method, given the high density of the measurements. Other interpolation methods should be explored that take advantage of other variables, such as the locations of the sound source and the microphone. An interpolation method that makes use of more information than just the two or four filters could also be more powerful.

### **Different Planes**

For this analysis, only horizontal plane microphones were considered. With a properly calibrated array, other planes, specifically the vertical plane, should be considered. Once this additional plane is made available, interpolation error can be studied between capsules in all directions. This would also allow for the possibility of an eight-point interpolation, where the available positions lie anywhere in three-dimensional space.

As mentioned earlier, this analysis included only two interpolation directions. The interpolation error of other directions, including diagonal (see Figure V-7), should be analyzed.

### **Different Amplifiers**

The Fender Deluxe amp data only tells us about one sound source's radiation patterns. Indeed, the difference between amplifiers may very well be greater than the difference between microphone positions. To make any kind of a generalizations for an auralization tool for microphone position, more amplifiers would need to be tested.

## **Microphone Comparisons**

Microphone placement is only one of the many variables that go into a recorded tone. In order to further develop this technique, different microphone patterns and capsules would need to be represented, since different microphone pickup patterns have different frequency responses. A cardioid microphone, for example, is subject to proximity effect. The spectral effects of the chosen microphone and chosen microphone angle should be studied and added to an auralization tool.

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# UML Measurement Session Signal Flow

