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Directivity Analysis of an Open-Back Combo Electric Guitar Amplifier's Radiation Characteristics

Introduction:

Frequencies emitted from a sound source theoretically propogate throughout the surrounding three-dimensional space in every direction. With different auditing positions relative to the sound source, however, listeners notice changes in the timbre of this sound source. This frequency response shift occurs at least partially because of the particular directivity of the sound source. Directivity is a description of how the strengths of specific frequencies change with variations of listening angle. More accurately, directivity attempts to measure the level at which these frequencies leave the sound source in various directions with respect to the level of the direct, zero degree path. In a practical sense, directivity reports as to where particular frequencies radiate most strongly from an instrument.

The purpose of this experiment involves analyzing the directivity of an open-back combo electric guitar amplifier. Such speaker enclosures, compact and suited for medium volumes, are quite common in modern recording studios. The prevalant use of such combo amplifiers in recording situations argues for a flexible and informed microphone technique specifically appropriate to such unique cabinets. Directivity analysis data helps to advise this technique by enabling predictions as to how changes in microphone placement affect the frequency response. In general, therefore, this paper serves to endow the sound engineer with a more robust approach to microphone placement with open-back combo guitar amplifiers through the knowledge of such a speaker cabinet's particular directivity characteristics.

Derivation:

The inspiration for this project derived from the well-written text, Tonmeister Technology by Michael Dickreiter. In this book, frequency dependent radiation characteristics are illustrated for most of the common orchestral instruments (as shown on pages 28 [fig. E], 34 [fig. E], 46 [fig C.], etc.). These radiation characteristics, also simply known as directivity, show the relative strength at a particular angle of a certain frequency emanating from the instrument in three dimensions. By referring to such diagrams, a recording engineer can acquire a starting point in determining the microphone placement for a particular instrument. Dickreiter's illustrations allow an engineer to knowledgeably select an area in which to place the microphone that will best capture the frequencies for the desired tonality of the instrument to be recorded. One current limitation of these fine diagrams is their omission of radiation patterns for popular instruments such as the bass and guitar amplifier. Popular microphone techniques for guitar amplifiers make no mention of how speaker directivity affects microphone placement (Pedersen 80+). While basic radiation characteristics for speakers are well known, I became interested in the specific radiation patterns of guitar amplifiers. Guitar amplifiers, especially small combo units preferred in most recording situations, have unique qualities differentiating their construction from typical speaker cabinets. For example, an open-back design, the nearness of speaker housing to the amplifier, the overly heavy-duty cabinet wood for durability, etc. are all qualities found almost solely in guitar combo amps. The question arises, therefore, as to how these different factors affect speaker directivity. While this question, the "why" of directivity, must be left unanswered by this project, I wish to address the "what" of directivity for small combo guitar amplifiers, i.e. what is the basic directivity response in the audible spectrum with these unique speaker cabinets. The results of such an experiment will be a polar plot for each frequency of analysis, plotting the directivity of that frequency for the amplifier under study.

In an effort to produce spectrum radiation patterns of similar quality to those in Dickreiter's text, I consulted the source from which these illustrations were derived, Jurgen Meyer's Acoustics and the Performance of Music (translated from the German). Starting on page 75, Meyer's chapter is devoted to "Directional Characteristics of Musical Instruments." Since no mention of the means of experimentation is made in Dickreiter, I was happy to find some in Meyer's publication. In describing his methodology, Meyer says, "the measurements required for this purpose have been carried out in an anechoic room, with a microphone distance of [11.5 feet]." (Meyer 75). After completing the experiments, sound pressure data was organized into shaded lines-of-equal-amplitude polar plots, with demarcations for less than 3 decibels of loss (half the power), less than 10 decibels of loss (half the perceived loudness), and greater than 10 decibels of sound attenuation. As it turns out, the diagrams in the Tonmeister Technology text were compiled from the data for less than 3 decibels of loss. Further information on Meyerís scientific process was unavailable and most references unattainable (as the books were German publications).

Other experiments in sound source directivity show similar methods to those of Meyer, with some slight deviations. Smaller microphone distances have been used in similar studies (Nakashima 2502), although still at relatively large distance of 6.5 feet. For angles of investigation, most sources recommend increments of just 10 degrees (Nakashima 2501). Some successful directivity analyses, however, have been conducted using much larger differences between microphone position, on the order of 22.5 degrees (Otcenasek 1330). All published directivity experiments were, of course, conducted inside an anechoic chamber.

To elucidate how other New York University students have conducted similar research, Weon-Bae Kim Master's thesis from Spring 1998 (investigating the directivity of Korean instruments) should have proved a reference starting point. Such a paper should have addressed such problems as: the lack of access to an anechoic environment, how to best analyze results without high-powered frequency analysis engines, how to ensure sounds for each microphone position were identical. Unfortunately, (besides being fraught with extremely unintelligible English) Kim's thesis includes only a brief description of an apparently loosely rigorous scientific method. Merely seven microphone positions are used for each instrument, and these microphone positions are analyzed on the basis of seven different takes by the performer (apparently no access to a multi-track or seven "Noeman" (sic) KM184 microphones). The importance of having each tone exactly the same for a different microphone position is paramount. Differences in the frequency response of separate performances cannot otherwise be isolated from differences in the frequency response of separate microphone locations. When testing the directivity of a French Horn, previous acousticians have gone so far as to play the instrument with an amplified electric oscillator connected to a Western Electric 555 brass conical adapter (Martin 310). Fortunately, because it is investigating speaker directivity, this project was not dependent on real-time performances and could be conducted with previously recorded sine tones.

Design:

The limited availability of proper tools, environment, and time necessarily had a problematic impact upon this research. Only one open-back guitar amplifier was used for this experiment, an Ampeg Jet II (Model J-12T). The entire enclosure measures 18.5" wide, 8.5" deep, and 15" tall. The cabinet was loaded with one stock, unknown maker 8-ohm 12" speaker. For the experiment, the amplifier was run with the volume knob on 3, the tone knob on 5, and all other knobs set to zero. For measurement purposes, a Rode NT2 microphone was used, set to cardiod pattern. This signal was fed to a Sytek Audio-Systems microphone preamplifier (Model MPX-4A) with the gain set to 4 (gain knob unfortunately not calibrated). Voltage levels were measured with a standard Radio Shack digital volt-meter. The experiments were conducted in a small room measuring 10' wide, 16' deep, and 9' high. The room was completely acoustically untreated as evidenced by the hardwood floors, multiple doorways and windows, and various pieces of furniture typical of a bedroom.

The lack of access to an anechoic environment necessitated specific design differences of this project from traditional directivity measurements. Foremost, a closer microphone distance was employed to reject room sound and capture as high a level of direct sound as possible. While extremely close (6 inches to 1 foot) microphone distances would have rejected the most room sound, a greater distance was chosen for three reasons: 1) to lessen the frequency response coloration of proximity effect; 2) as the amplifier was rectangular and had a width twice as great as its depth, to ensure a generally similar distance of microphone to amplifier when the cabinet was rotated on axis; and 3) to reconcile the greater distances used in traditional experiments with the close positioning of this experiment. After preliminary tests, a distance of 2 feet 4.25 inches from the center of the cabinet to microphone was chosen (half the depth of the cabinet plus 2 feet). Such near-field microphone placement is by far the most common method used in recording studios and concert stages. Moreover, no recording studio or concert stage can be considered an anechoic environment. In a sense, therefore, the practicality of conducting these experiements in a reverberant space with a close microphone distance better mirrors the real world application of microphone technique than do traditional scientific measurements.

Although performing these tests in a reverberant space represents the more common recording environment, differences in the frequency response of non-anechoic spaces, particuarly differences dependent on specific placement withing these spaces, demand that the experiment uses more than one amplifier/microphone position to seperate directivity of the speaker from remnant directivity of the room. In consideration of time, two positions were used. The first test location was made parallel to the shorter dimension of the room (see fig. A; microphone at point "A", speaker at point "B"). The second test location was made parallel to the longer dimension of the room (see fig. A; microphone at point "B", speaker at point "C"). From preliminary tests, these locations seemed to best give balanced results (where decibel loss was equivalent on one side of the cabinet to the opposite side).

Also being affected by time restraints were the angle of rotation and selected frequencies of analysis. As the largest difference in angle found in published works was 22.5 degrees, I settled on these areas of study. With a rotation of 22.5 degrees through a full circle, 16 angles of investigation existed for each frequency. In some sort of desire for conformity, 16 frequencies were also chosen. The full set of 23 frequencies analyzed in Meyer's publication for the directivity of a cello seemed appropriate for a guitar amplifier since cello and guitar have roughly the same playing range. The discarded analysis frequencies from Meyer were disused to also make the rise of frequencies more closely resemble a logarithmic progression and thus more closely resemble a balanced rise in pitch.

The practical implementation of the experiment was as follows: Sine waves at the selected frequencies (shown in fig. D, E, and F) were recorded to digital audio tape at a level of -8 decibels from a Loftech generator. The output from the DAT was attenuated through a Mackie mixer (-40 decibels) and sent to the Ampeg combo amplifier. The output of the microphone was measured by a voltage meter connected to pin 1 and pin 2 of the microphone preamplifier's output. Each frequency was sent to the amplifier before the cabinet was rotated to the next angled position. The voltage of each frequency at each angle was recorded and then later converted to decibels using the zero degree position as the reference and employing the standard equation for volt/decibel conversion:

$$dB = 20 \log (V/Vref)$$

The results of these decibel conversions are included in the charts of figs. D and E. From these decibel levels, areas of less than 3 decibels of loss were organized into figure F. For frequencies that included angles with higher decibel levels than the reference zero degree level, the areas of less than 3 decibels of loss from the highest of these levels were only included. Often, the zero degree reference level could not be included in the results of fig. F since it fell more than 3 decibels from the highest reading at that frequency. From the areas of less than 3 decibels of loss in fig. F, the diagrams in figs. B and C were compiled. Fig. B shows general directivity results, while fig. C shows interesting specific directivities comparing both microphone locations. N.B. Since the angles under test were evenly spaced at 22.5 degrees, the assumption had to be made for graphing purposes that transitions occured somewhere between these points. For convenience's sake, this transition point was chosen midway, offset from the angle under test by 11.25 degrees. Thus, when comparing the graphs in figs. B and C to the areas in fig. F, the degrees will be slightly shifted to larger areas. Obviously, for example, at high frequencies, although radition was only measured with significant intensity at one point (zero degrees), a margin of error of 22.5 degrees (centered around the test frequency) accounts for the transition this frequency makes from high intensity to low intensity elsewhere.

Results:

Upon comparing the decibel levels for both microphone positions (figs. D and E), one notices a decent amount of variation in the results. Obviously, room acoustics have an effective role upon frequency response in all directions, even with microphone distances of only two feet or less. Accurate polar plots (similar to those of microphones) with such inprecise results seem ungrounded. General trends in directivity, however, are apparent even though specific comparisons are impossible. The possibility remains, of course, that even the general trends documented by this research have been affected by room acoustics. Such a possibility can only be proven or disproven by further experimentation in different surroundings, infeasible at this venture. Remarks on the general directivity for this specific open-back combo amplifier seem warranted, however, due to the reasonable and logical patterns in the results.

Figure B shows the basic organization of all the results. The shaded areas, again, are the areas indicating less than or equal to only 3 decibels of sound attenuation when compared to the highest level at any angle. The results from figure F inform some stark changes in directivity with changing frequency response. Firstly, the band of 100-200 Hz shows strongest intensity from the back, extending from around 123.75 degrees to 236.25 degrees. Although prior to the experimentation, bass frequencies were expected to exhibit omni-directional behavior, the open-back design must concentrate these frequencies more directionally. Both microphone positions show severe attenuation (greater than 10 decibels) at 90 and 270 degrees for these low frequencies. Somewhere between the 200 Hz and 300 Hz test frequencies, a shift to forward radiation becomes favored. From 300 Hz upwards, frequencies radiate out the front of the speaker in a cone of ever-decreasing width. Such increasing directionality with increasing frequency is to be expected. The greatest differences in directivity occured with the 1.0 kHz to 2.0 kHz range. This frequency band also exhibited some odd radiation characteristics, with high levels appearing somewhere between 90 and 180 degrees, as well as between 180 and 270 degrees. The reason for such unpredictable behavior in this range most probably relates to frequency as a function of speaker diameter. 1.0 kHz has a wavelength equivalent to the diameter of the speaker under test (12"), whereas 2.0 kHz has a wavelength half that of the speaker's diameter. Although such a phenomenon cannot be fully explained by this paper, some boundary zone of diffraction apparently occurs between frequecies whose wavelengths are between 1 and 0.5 times the diameter of the speaker cone. For the uniqueness of these frequency bands, full graphs have been included (fig. C) comparing the specific frequencies with each microphone position. Two specific frequency bands, 150 Hz and 1.5 kHz show unexpected characteristics: 150 Hz behaves almost as a transition from the 100 Hz-200 Hz band and the 300 Hz-800 Hz band by radiating strongly out both back and front; 1.5 kHz seems to belong more to the extremely directional group of high frequencies (2.5 kHz-5.0 kHz) than the group of scattered frequencies related to the speaker's diamter (1.0 kHz-2.0 kHz). Overall, while only giving simple descriptions of speaker directivity for a single open-back combo amplifier, the graphs in figs. B and C constitute the core results of this experiment; these graphs definitely inform sound engineers as to basic radiation characterisitics, as well as strongly suggesting interesting specific frequency bands for more exacting radiation study.

Conclusion:

Although the precision of this experiment was limited by available test and spatial resources, basic radiation characteristics of an open-back combo electric guitar amplifier have been discovered. These results were valiated through the use of two test positions and a logical organization to the experimental process. Interesting transition frequencies have been identified. Traditional concepts of increasing directivity with increasing frequecy have been comfirmed. The directional behavior of lower frequecies due to open-back design deserves possible further research. Also, the relationship of wavelength to speaker diameter warrants more detailed testing. In general, this paper should at least serve to acquaint the sound engineer with the basic directivity of speaker cabinets commonly found in the recording studio and on stage, thus allowing a more educated approach to microphone placement.

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