BIOMECHANICAL SOUND SOURCE POLAR PATTERN MEASUREMENT: WIND INSTRUMENT

SEBASTIÁN IEZZI [1], DANIELA PEPPE [2] AND DENISE ROMARIS[3]

[1] Universidad Nacional de Tres de Febrero, Caseros, Buenos Aires, Argentina.
sebastian.a.iezzi@gmail.com
[2] Universidad Nacional de Tres de Febrero, Caseros, Buenos Aires, Argentina.
daniela.peppe@gmail.com
[3] Universidad Nacional de Tres de Febrero, Caseros, Buenos Aires, Argentina.
dnic.91@gmail.com

Abstract – *The present paper explains the measurement procedure in order to obtain the polar pattern of a saxophone. Measurements took place at Universidad Nacional de Tres de Febrero (UNTREF). Although previous research indicates that this kind of measurement should be carried out in anechoic conditions, this procedure was made at University's Auditorium.*

1. INTRODUCTION

The directivity pattern of a sound source is a useful measure that allows us to know how the radiation of the energy in the space is. This data is necessary for many applications, for example, the design of concert halls, among others.

The polar pattern of musical instruments has been studied by several authors, mainly by Jurgen Meyer, who calculated the average directivity of musical instruments at the whole frequency range. Actually, this is the most used information in auralizations and room simulation, however, in many cases, the directivity of instruments changes abruptly with frequency. In the measurements made decades ago Meyer, in order to obtain the polar patterns of several instruments, the location used was an anechoic chamber, and the distance between the sound source and the microphone was 3,5 meters [1]. Pätynen and Lokki studied the directional characteristics of the instruments in an orchestra, using an anechoic chamber and a distance of 2,13 meters [2]. Different dynamics were used in its investigation. Otondo and Rindel [3] [4] also made several measurements of instruments directivities, and they used anechoic conditions as well as a distance from the source of 1,5 meters.

The current information about directivity of instruments for each frequency is limited. On the other hand, most of the published papers are focused on the physical properties of the instrument measured, so they use mechanical devices to play it, although this way the measurements are accurate, it does not allow to take in account the contribution of the musician.

This paper presents the polar directivity measurements by third octave bands of a high saxophone, measured every 10º at the horizontal plane, the frontal plane, and the sagittal plane. The saxophone was measured at UNTREF's Auditorium (Universidad Nacional de Tres de Febrero). This paper is organized as follows: first, a theoretical background is presented, then the measurement setup and procedure is detailed. The results are presented and analyzed, and then the conclusions are obtained.

2. THEORETICAL BACKGROUND

Most instruments radiate sound in different directions with different intensities. This connection between the direction and the radiated sound pressure can be seen as the directional characteristic.

Although there are not enough details about instruments directivity measurement methods, most of them are made in anechoic conditions, using the same distance from the source to the microphone, in different positions.

The directivity index expresses how much the sound level is higher, in a specific direction, than it would be with an omnidirectional sound source. It is expressed in dB.

Woodwind instruments present many differences in their characteristics with respect to other groups of instruments. The sound production mechanism differs on one woodwind instrument to another. The directionality of these family of instruments is more complex than others, due to its radiation resulting from the open mouth of the bell, which is supplemented by the radiation from the open finger holes.

The saxophone is a member of the woodwind instruments family.

3. MEASUREMENT

3.1. Equipment and software

The following equipment was used for the measurement:

- Sound Level Meter Svantek SVAN 959
- Earthworks Microphones M50
- Electronic Turntable Outline ET230-3D
- Calibrator Svantek SV-30-A
- Tascam US-4x4
- Laptop ASUS i7

Also, the following software was used:

- Cubase 5
- SVANPC++
- Smaart 6
- Adobe Audition 3.0
- Matlab 2011

3.1.1. Calibration

Before and after the measurement, the sound level meters were calibrated. A Svantek calibrator was used, with a frequency of 1 KHz and a level of 94 dB. The values corresponding to "Cal Factor" were obtained for each sound level meter. These values are used in the uncertainty section.

In order to obtain the same levels with all measurement microphones, a calibration was made, also using the Svantek calibrator. The levels of the four microphones were paired at 1 KHz, regarding the results of the calibration in the software Cubase.

3.1.2 Measurements compensation

After the recording, a level compensation was necessary. The combination microphones-preamplifier-audio interface, and the wav record of the sound level meters, had level differences.

First of all, a compensation to convert from Volt to SPL units was necessary. With the digital audio recording of the sound level meters, located in the angle 0° when the music was reproduced in Forte mode (reference recording), and the Svantek files of three octave band SPL average (in excel type file), the first compensation was made. Those files were uploaded in the MATLAB software, where an average on three octave band with a filter design was applied on the digital sound recording, like is explained in the Processing section. The results of the difference between the excel file and the digital sound file on three octave bands were saved for future compensations.

All the future audio files were compensated for doing the sum of the three octave band and the compensation obtained.

The second part of the compensation implies the compensation of the microphones and the sound level meter differences. Once made the first compensation of all the sound level meter and the microphones files, an average of the two signals in the same angle was made, as shown in the section "Processing". Thus any reflection deficiency impact, like the comb filter, was reduced.

Finally, an average of all the reference files was made. All the angles measures were in different sound pressure levels, because of the human errors on the instruments played. So all the measured signals were compared with the average obtained, and compensated.

3.2 Frequency range and harmonics of the instrument

The instrument chosen was the high saxophone. This instrument is tuned up at Mib, that means that the scale is 1 tone and a half, or three semitones beneath the conventional tune of other instruments, for example the piano (Do). This tuning at a Do tonality, leads to consider that when it's been talked about a note interpreted in the saxophone, this refers to a frequency of three semitones above the frequency that's usual.

The musical record in which a high saxophone works covers from a Sib2 (in Do scale: a Reb3, which implies a frequency of 138,59 Hz), and ends in a Fa6 (Lab6 for Do 1661,21 Hz). The truth is that to extend this musical record at high frequencies, it is necessary to use techniques that not every saxophonist achieves with efficacy. Because of that the range of frequencies had to be reduced from Sib to Fa4 (Lab4 for Do, 349,23 Hz), that matches with the musical record of the easiest technique.

In order to choose the melody, first it was necessary to consider the post processing of the signal, taking into account the largest frequency range that is possible for the instrument to reproduce, even its harmonics.

In previous works, it was observed that the high saxophone has at least 4 harmonics referring to the fundamental frequency. That's why it was tried to make a melody with notes that doesn't have coincident harmonics between them.

In many musical styles, saxophonists use some distortion methods. For the measurement, it was asked the musician not to play the saxophone with distortion, in order to achieve an analysis of the harmonics from the instrument itself, independently of the musical style.

The melody interpreted is shown in Figure 1, and the frequencies at the pentagram is shown in Figure 2.

Figure 1: Saxophone sheet music tube on Mib.

Figure 2: Real notes on piano tune on Do.

3.3 **Location**

The location used for the measurement was UNTREF's Auditorium, with a capacity of 350 seats, and a volume of 2500 m3 [5]. Although it is not an anechoic chamber, is the location with the most similar characteristics available. The chosen area for the measurement procedure was the center of the stage. In Figure 3 it can be seen from the Auditorium's stage.

To determine the critical distance (CD) the following equation was used:

$$CD = 0{,}057\sqrt{\frac{QV}{RT60}} \qquad (1)$$

Where Q is the directivity (1), V the room volume, and RT60 the reverberation time.

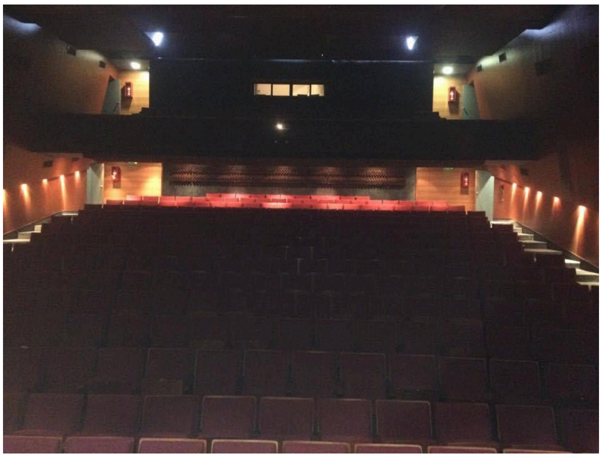

Figure 3: UNTREF's Auditorium. View from the stage.

It's important to consider that the Auditorium has water pumps. These pumps are functioning constantly. Since they never stop working, the sound levels produced by the water on the pipes contributes to the background noise in the room.

3.4 Procedure

The measurement took place at UNTREF's Auditorium. Four measurement microphones and four sound level meters were used. Another sound level meter for the reference was used.

The sound source (the musician) was placed on the electronic turntable on the floor, at the center of the stage. The floor was covered with absorbing material in order to minimize the effects of the first reflections.

The reference sound level meter was located in every measurement at 0° and 2 meters from the source. The objective of this reference measurement is to compensate by the sum or the subtraction of sound pressure level by every third octave band, to reduce the error generated by the musician variations.

For the horizontal plane the four microphones and sound level meters were placed at 2 meters from the source at 0°, 90°, 180° and 270° at 1 meter from the floor, taking 0° as the front of the stage. A microphone and a sound level meter were located at the same place, in order to obtain two results from different measurement instruments. To measure the angles on each plane, a bevel square, a conveyor, a rigid meter and a laser measurer were used. The measurement began at 0º with the musician standing on the electronic turntable, and then it was rotated every 10° from 0 to 80. In Figures 4, 5 and 6 the position of the musician and the microphones can be seen.

Figure 4: Positions of the microphones for the horizontal plane.

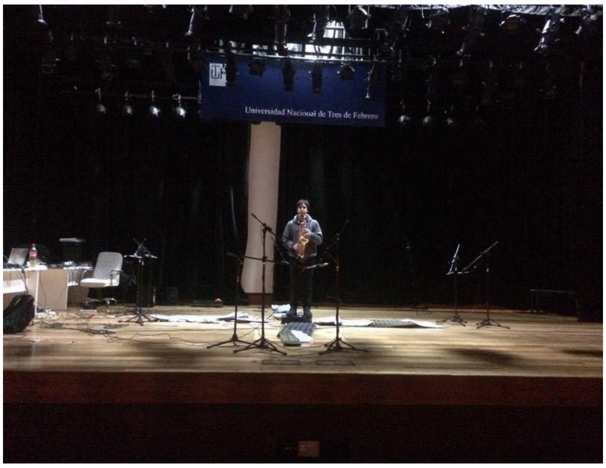

Figure 5: Position of the musician, on the electronic turntable, at the center of the configuration.

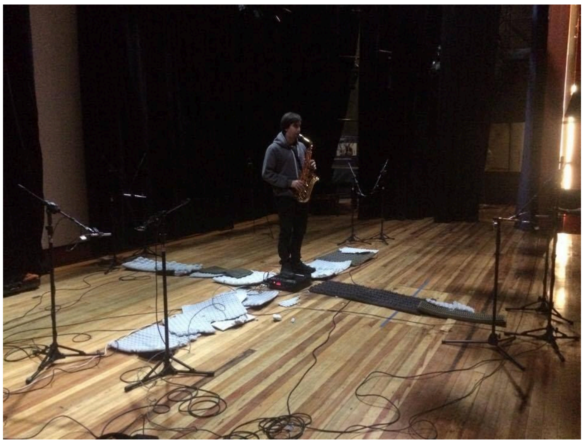

Figure 6: Position of the musician.

By using the microphone placed at 0° as a reference, the frontal and sagittal plane measurements were made. For this measurement the room is supposed to be perfectly symmetric. Four microphones with four sound level meters were placed at 10° from each other and 2 meters from the source. The frontal plane was measured by placing the musician at 90° and 270°; and the sagittal plane by placing him at 0° and 180°. The four measuring points remain steady. This way, it obtained four results for every microphone that's been moved. For example, the measurement started with a microphone at 70, 80, 100 and 110 degrees (the 90° measurement of the frontal and sagittal plane matches with the 0 from the horizontal plane, so it had already been measured). When the musician was placed at 90° anticlockwise from the front of the stage, the results of 70°, 80°, 100° and 110° from the frontal plane were obtained; when he was placed at 270°, those from 290°, 280°, 260° and 250° were obtained (respectively of the frontal plane). In the same way, when he was placed at 0º, the results of 70, 80º, 100º and 110º of the sagittal plane were obtained. And when he is at 180°, the results of 290°, 280°, 260° and 250° were obtained.

In Figure 7, a diagram of the angles used for the frontal (and sagittal plane) can be observed,

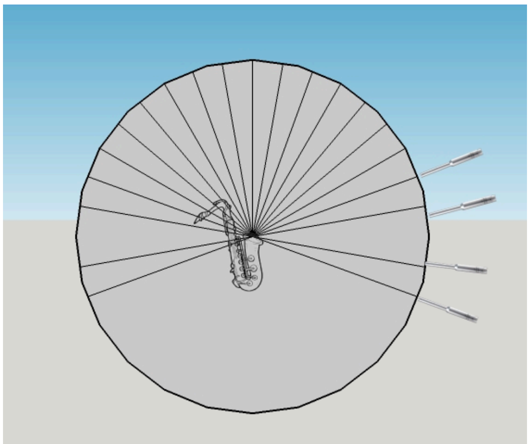

Figure 7: Frontal plane.

4. PROCESSING

4.1. Frequency analysis and octave band sound level

A MATLAB code was written in order to make a correct interpretation of the data obtained in the audio files.

The processing of these files consists of importing them to the software. The audio files were previously saved with a specific name. There are four groups for each measurement: A, the reference level at the axis of the mouth of the instrument; B, the sound register of each microphone position (with an specific angle respect of the axis); and C the sound register of another microphone in the same angle of "B", but 20 cm further from the center.

The data for each measurement angle is entered on the MATLAB code. In order to obtain its frequency information, a Fast Fourier Transform is used. Then, a third octave filtering is made and the equivalent sound level is calculated for every third octave, according to the ANSI S1.11-2004, as it can be seen on Equation 2 [6].

$$Leq_{third} = 10log\left(\frac{\left[\left(\frac{1}{T}\right)\int_0^T V_{in}(t)dt^2\right]}{V_o^{\,2}}\right)dB \tag{2}$$

Where T is the elapsed time of integration, V-in is the instantaneous input signal as a function of time t, Vo is an appropriate reference quantity such as 20uV and Leq-third is the average value for each third octave.

A MATLAB's precalculated third octave filter design was used for filtering bands. This filter design is also according to the ANSI S1.11-2004, and frecuencia response for all band, is shown in the Figure 8 [7].

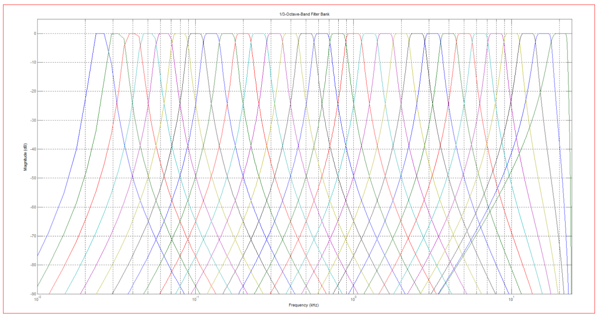

Figure 8: Third band filter by MATLAB.

Then, it is necessary to use the values obtained from those that correspond to an angle of the polar pattern: on one side, the B measurements, and on the other the C measurements. Using Equation 3, an average between every third octave is made, where N is the number of the measurements.

$$Lter = 10log\left(\frac{1}{N}\sum_{i=1}^{N} 10^{L_i/10}\right) \tag{3}$$

In the same way, the averages by third octave of all measurements A are taken. The 0° value is fixed as a reference of all measurements. For each angle, a difference between the third octave is considered (between the reference and every position). This can be seen in Equation 4.

$$Dif_i = 10log\left(10^{Lter_{0°}/10} - 10^{Lter_i/10}\right) \tag{4}$$

Where i correspond to a determined angle. This is made to avoid the interpretation intensities differences.

Finally, in order to obtain the definitive value of every third octave on each angle, the values of the averages between measurements B and C are used. It's important to do a normalization on these values. This normalization is detailed in Equation 5.

$$Nor_i = 10log\left(10^{Dif_i/10} + 10^{Lter_i/10}\right) \tag{5}$$

4.2 POLAR PATTERN BY FREQUENCY

Once the normalization of each measurement was made, the polar pattern related to the frequency was obtained, using the "contourf" Matlab function. The graphic consists of an axis corresponding to the radiation angles, and another axis corresponding to the frequency. The colors show the intensity on each point.

With the normalized values "Nori", that corresponds to the different angles, a matrix named "PPol" is created, to make the graphics with the already mentioned function.

On the other hand, in order to obtain the polar pattern of curtains frequencies, it's necessary to take the values that correspond to them from the matrix "PPol". With "polar" function the graphic for a specific frequency can be obtained.

4.3 Characteristics pitched

For the study of the characteristics pitched in every measurement, an frequency component analysis was made. The values obtained with the FFT, previous to the third octave calculus, are the ones that's been used.

The notes that's been executed by the musician were selected in order to avoid a superposition by the harmonics, achieving less processing than with superposition.

Through the FFT analysis, it was easy to establish the harmonics values and their respective fundamental relation. The influential frequencies and the amplitude relation had been determined, comparing the recordings of Piano and Forte, and observing the difference between them.

Finally, the harmonics values were used to calculate the polar pattern of those frequencies.

For a better interpretation of the harmonics involved, two files were chosen: one in Forte mode, and one in Piano mode. With the software Audacity, the separation of each played notes was performed, and then a FFT transform in MATLAB software was made. The Figure 9 shows the FFT of the lower note

(184.9 Hz) in Forte and Piano mode, and the Figure 10 shows the FFT of the highest note (783.9 Hz).

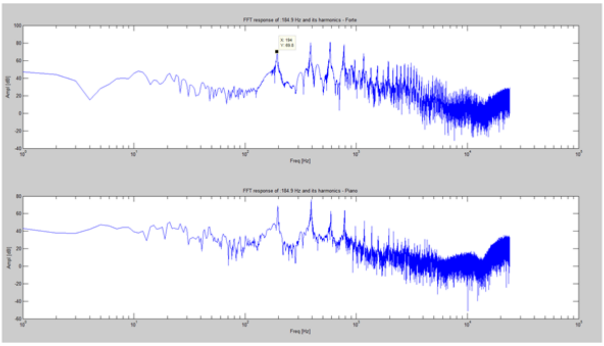

Figure 9: FFT of the lower note (184.9 Hz) in Forte and Piano mode, and its harmonics: (a) Forte (b) Piano

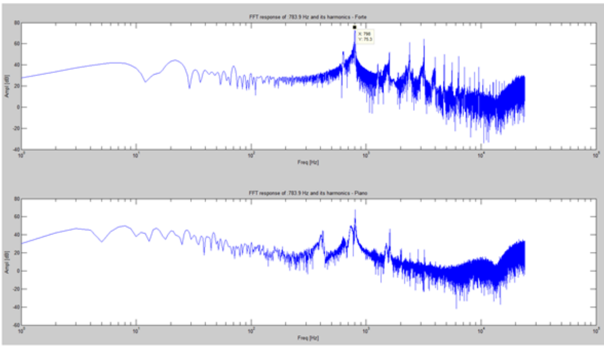

Figure 10: FFT of the highest note (783.9 Hz) and its harmonics: (a) Forte (b) Piano

Finally, the level of each harmonic was estimated. The Table shows the difference of the levels between Forte mode and Piano mode, of the fundamental and first three harmonics. The rows show each played note. All values are normalized.

Table 1: relation of the Forte and Piano harmonics

Played Note [Hz]	Fundamental Freq [dB]	1rs Harmonic [dB]	2nd Harmonic [dB]	3rd Harmonic [dB]
185,00	0,00	15,45	19,52	25,53
155,56	0,00	-5,30	3,32	0,44
196,00	0,00	-0,43	9,16	10,56
233,08	0,00	-1,45	8,37	-3,91
311,13	0,00	4,22	-1,61	16,93
783,99	0,00	14,40	9,09	4,51
466,16	0,00	-10,58	10,77	32,42
622,25	0,00	14,57	13,02	16,13
783,99	0,00	11,97	27,99	29,25
622,25	0,00	15,04	17,54	25,60
466,16	0,00	-4,95	0,37	16,48
392,00	0,00	0,71	11,23	7,12
311,13	0,00	5,41	-0,95	17,22
233,08	0,00	-2,97	-0,39	-2,90
185,00	0,00	4,33	13,58	7,97

5. RESULTS

Once finished the programming of the calculation on MATLAB, all the recorded files were uploaded to the software. Then, the graph of the three axes (horizontal, frontal and sagittal), for the two musical interpretation intensity (Forte and Piano) was obtained. Also, the polar patterns of a few frequencies in the same cases were plotted.

The selected frequencies for the graphics were selected according to: a. the lower note played; b. the higher note played; c. the 2nd and 3rd harmonics of them, d. 3 specific frequencies: 1 KHz, 3 KHz and 5 KHz.

5.1 Horizontal plane

5.1.1 "Forte" results

Figure 11 shows the contour color MATLAB function, and represents the frequency response vs. the angles of the horizontal plane. The color corresponds to the sound pressure level (SPL). This graph gives an idea of the polar pattern in all frequencies.

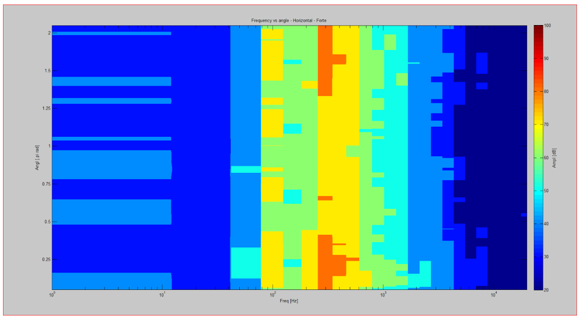

Figure 11: Contour plot for horizontal axis, "Forte" intensity.

Figure 12 shows the polar pattern of the lower note played by the musician, with a frequency of 185,9 Hz.

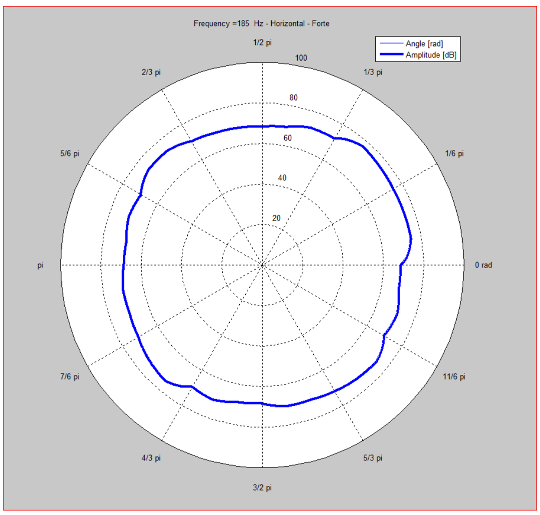

Figure 12: Polar Pattern of the horizontal axis: 185,9 Hz.

Figure 13 and Figure 14 corresponds respectively to the 2nd and 3rd harmonics, 370 Hz and 554,9 Hz.

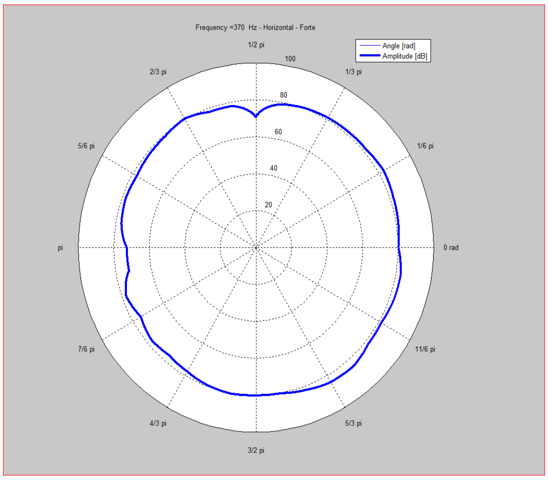

Figure 13: Polar pattern, 370 Hz.

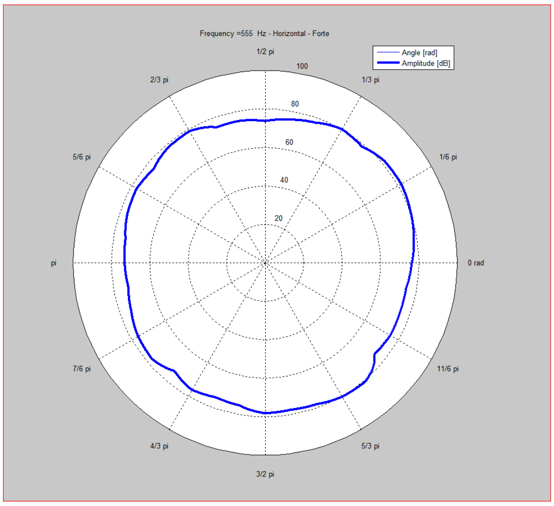

Figure 14: Polar pattern, 554,9 Hz.

The highest note played corresponds to a frequency of 784 Hz. Its polar pattern can be observed in Figure 15.

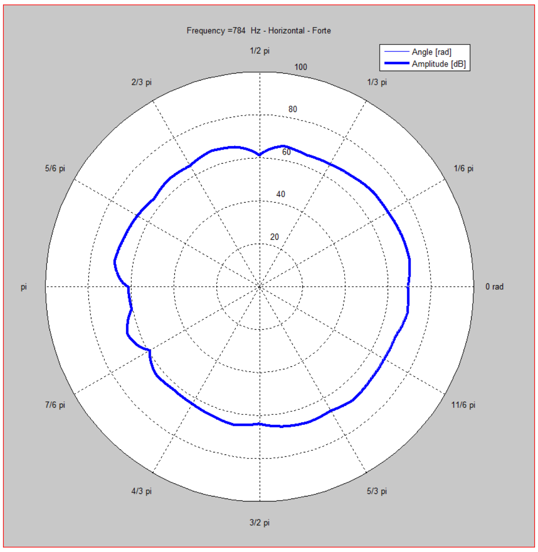

Figure 15: Polar pattern, 784 Hz.

Figures 16 and 17 show the polar patterns for the 2nd and 3rd harmonics of the highest note played: 1568 Hz and 2352 Hz.

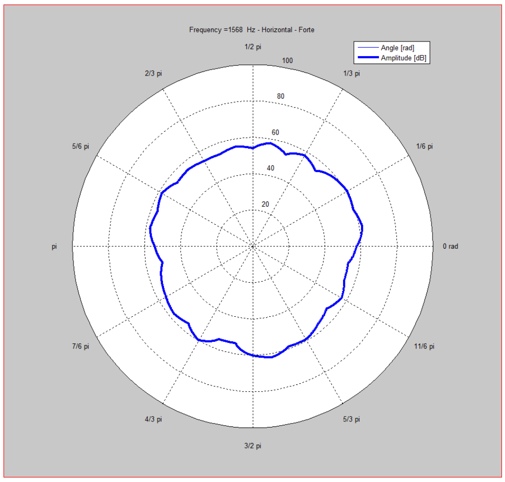

Figure 16: 1568 Hz.

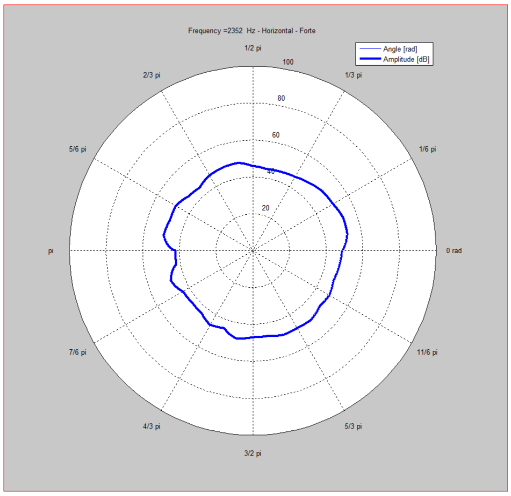

Figure 17: 2352 Hz.

On Figures 18, 19 and 20, the polar patterns of the selected frequencies 1 KHz, 3 KHz and 5 KHz can be observed.

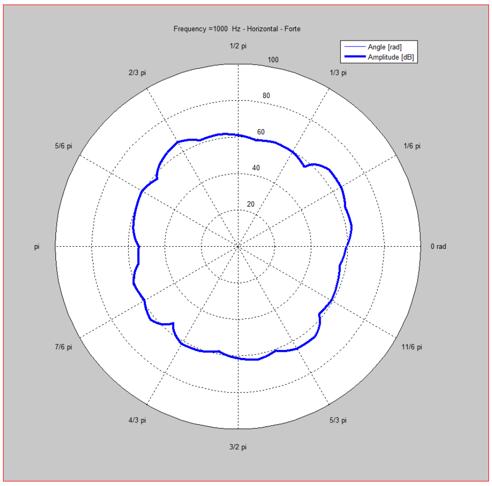

Figure 18: 1 KHz.

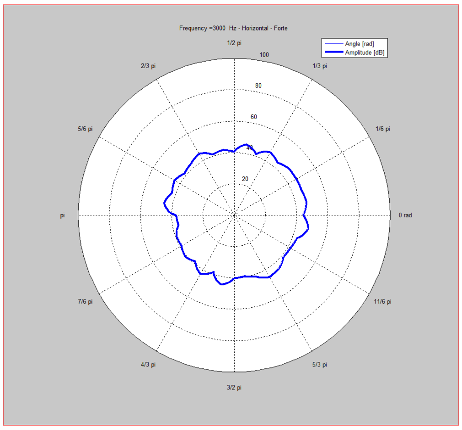

Figure 19: 3 KHz.

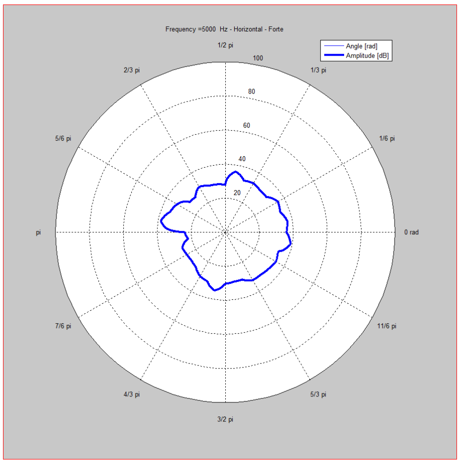

Figure 20: 5 KHz.

5.1.2 "Piano" results

Figure 21 shows the contour color MATLAB function, which is the frequency response vs. the angle of the horizontal axes. The color represents the sound pressure level (SPL). This graphic gives an idea of the polar pattern in all frequencies.

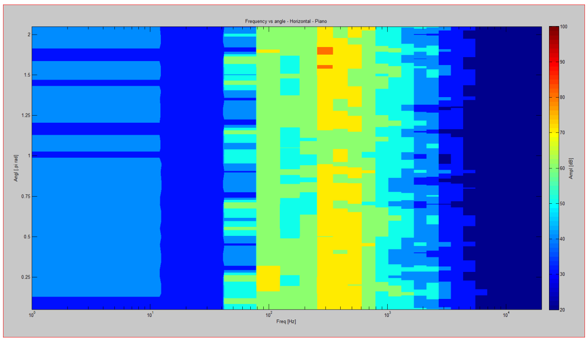

Figure 21: Contour plot for horizontal axis, "Piano" intensity.

Figure 22 shows the polar pattern for the lowest note played, whose frequency is 185,99Hz.

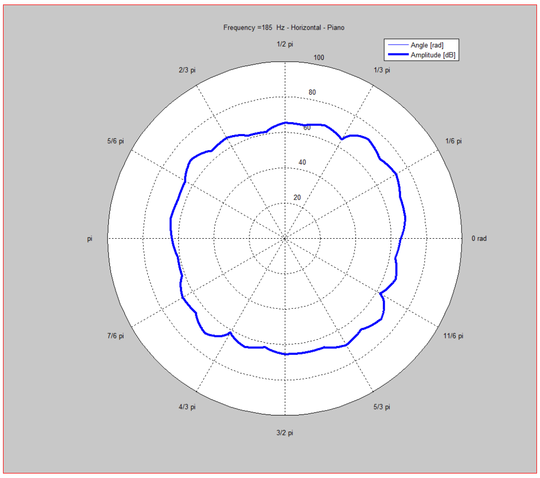

Figure 22: Polar pattern, 185,9 Hz.

The 2nd and 3rd harmonics (370 Hz and 554,9 Hz) are shown in Figures 23 and 24.

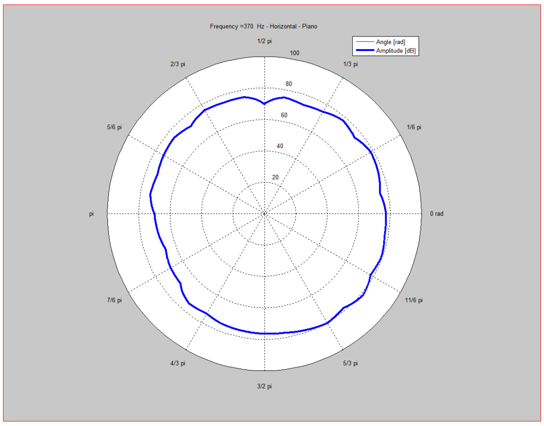

Figure 23: 370 Hz.

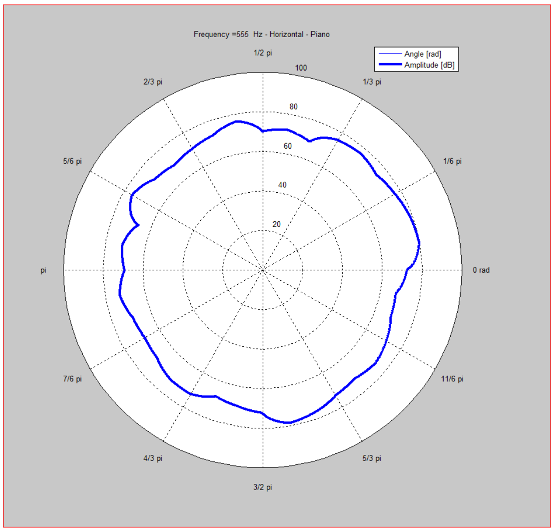

Figure 24: 554,9 Hz.

The polar pattern for the highest note played, which corresponds to 784 Hz, is shown on Figure 25.

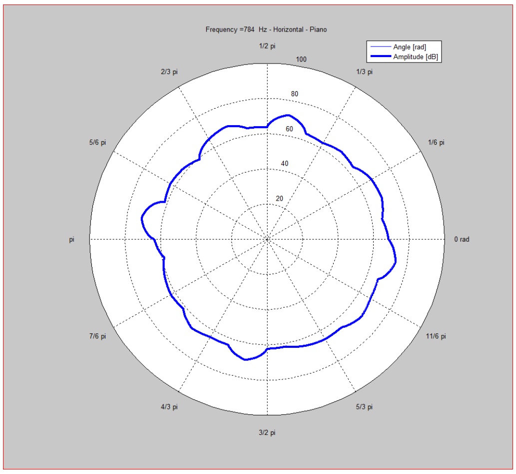

Figure 25: Polar pattern, 784 Hz.

Figures 26 and 27 show the graphics for the frequencies 1568 Hz and 2352 Hz, 2^{nd} and 3^{rd} harmonics, respectively.

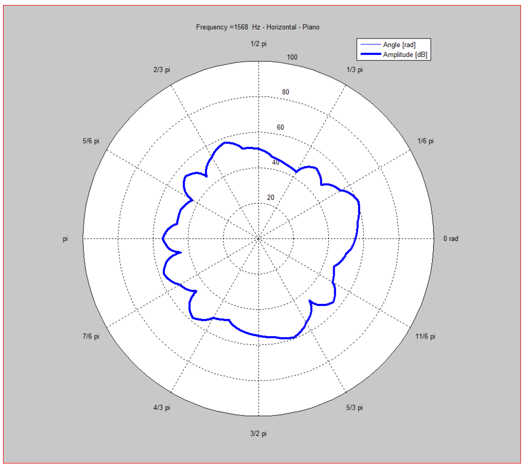

Figure 26: 1568 Hz.

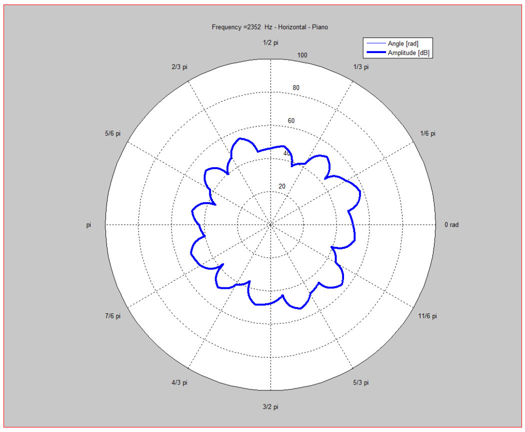

Figure 27: 2352 Hz.

The polar patterns for "Piano" mode of 1 KHz, 3 KHz and 5 KHz are shown on Figures 28, 29 and 30.

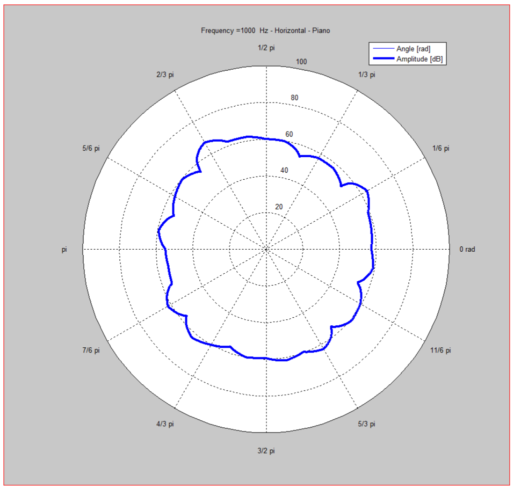

Figure 28: 1 KHz.

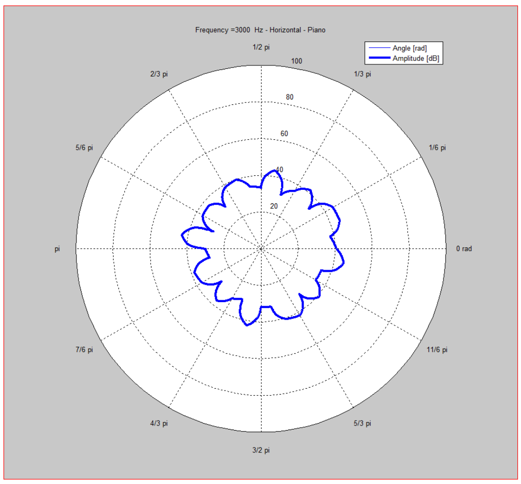

Figure 29: 3 KHz.

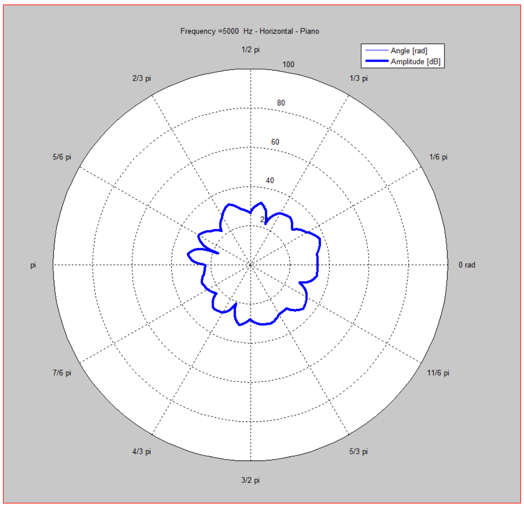

Figure 30: 5 KHz.

5.2 Frontal plane

5.2.1 "Forte" results

Figure 31 shows the contour color MATLAB function, which is the frequency response vs. the angle of the horizontal axes. The color represents the sound pressure level (SPL). This graphic gives an idea of the polar pattern in all frequencies.

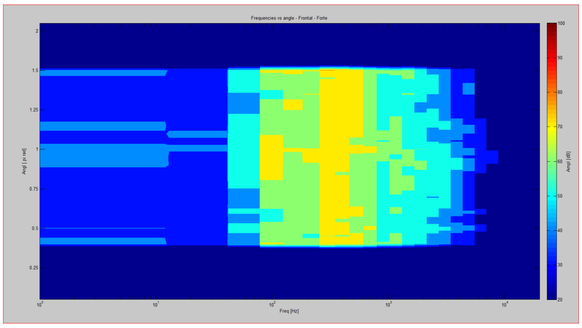

Figure 31: Contour plot for frontal axis, "Forte" intensity.

Figure32, shows the polar pattern for the lowest note played, whose frequency is 185,9Hz. Figure 33 and Figure 34, shows the 2^{nd} and the 3^{rd} harmonics respectively, 370 Hz and 554,9 Hz.

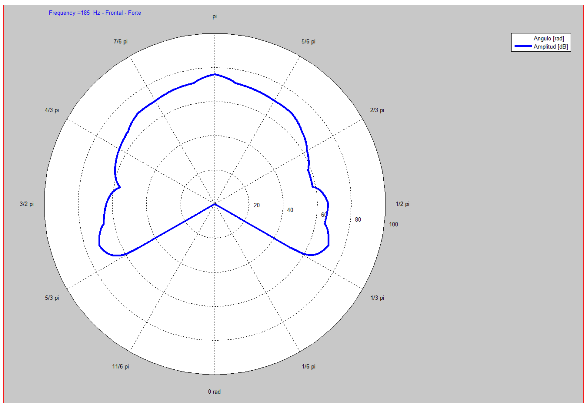

Figure 32: Polar Pattern of the frontal axis: 185,9 Hz.

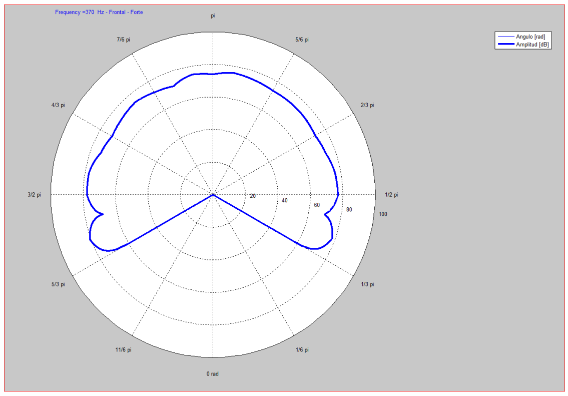

Figure 33: 370 Hz.

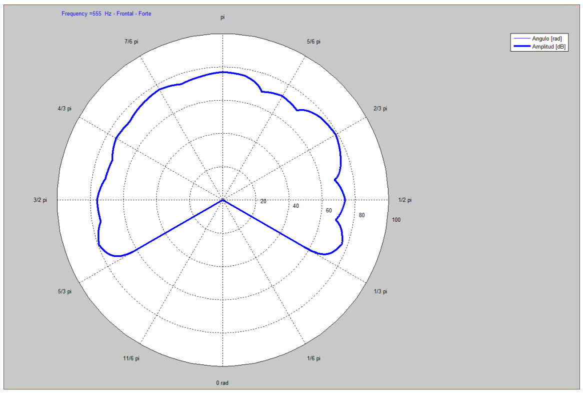

Figure 34: 554,9 Hz.

Figure 35 shows the polar pattern for the highest note that's been played, whose frequency is 784 Hz. Figures 36 and 37, shows the 2nd and the 3rd harmonics respectively (1568 Hz and 2352 Hz).

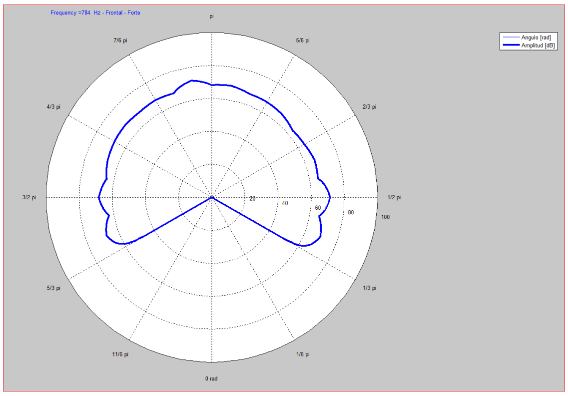

Figure 35: 784 Hz.

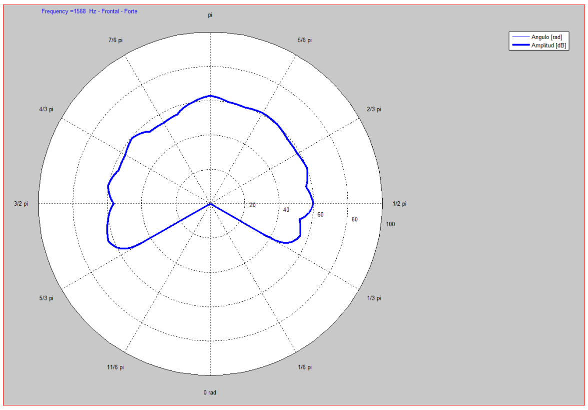

Figure 36: 1568 Hz.

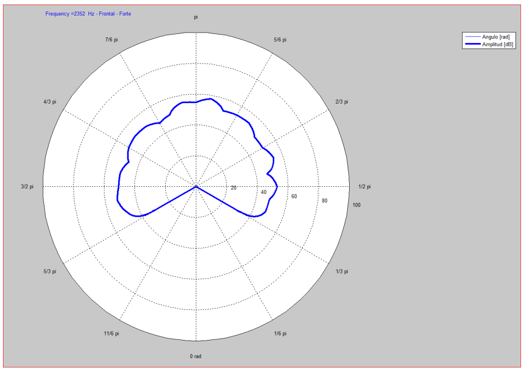

Figure 37: 2352 Hz.

Figure 38, 39 and 40 shows the polar patterns for the frequencies of 1 KHz, 3 KHz and 5 KHz for the frontal plane, "Forte" intensity.

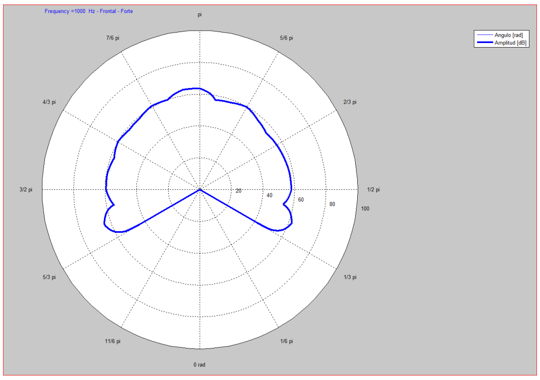

Figure 38: 1 KHz.

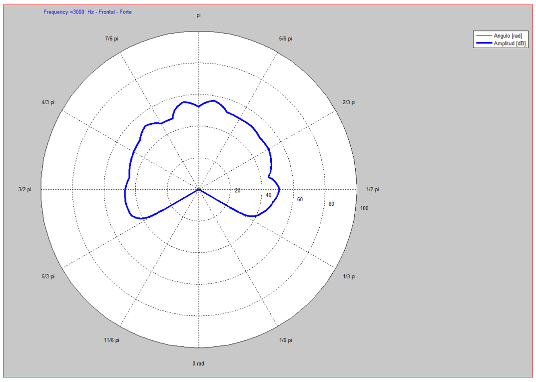

Figure 39: 3 KHz.

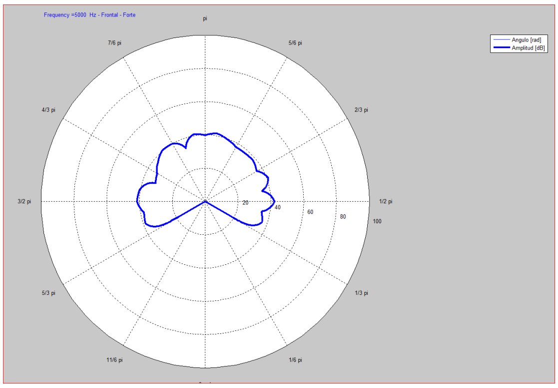

Figure 40: 5 KHz.

5.2.2 "Piano" results

Figure 41 shows the contour color MATLAB function, which is the frequency response vs. the angle of the horizontal axes. The color represents the sound pressure level (SPL). This graph gives an idea of the polar pattern in all frequencies.

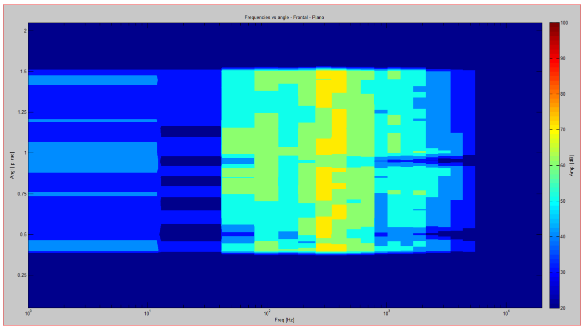

Figure 41: Contour plot for frontal axis, “Piano” intensity.

Figure 42 shows the polar pattern for the lowest note played, whose frequency is 185,9Hz. Figures 43 and 44 show the 2nd and the 3rd harmonics respectively (370 Hz and 554,9 Hz).

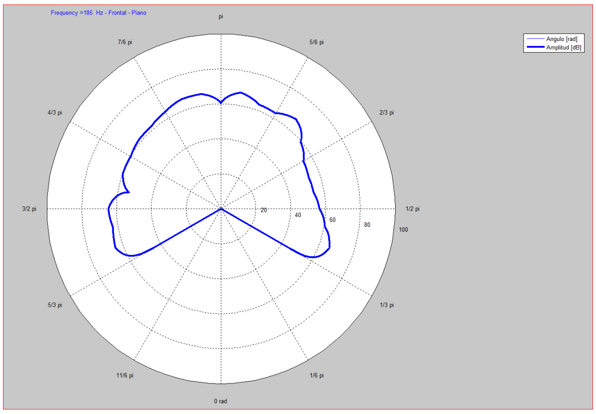

Figure 42: 185,9 Hz.

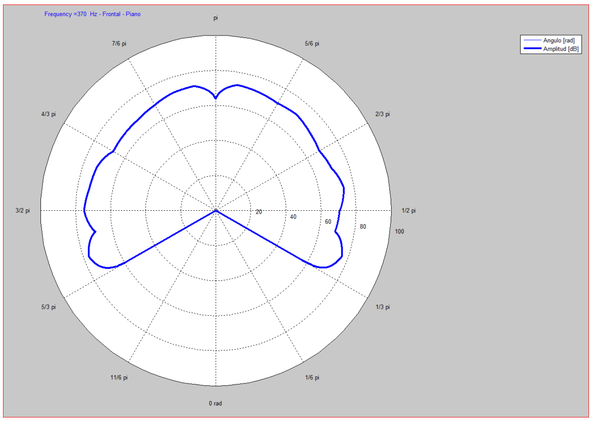

Figure 43: 370 Hz.

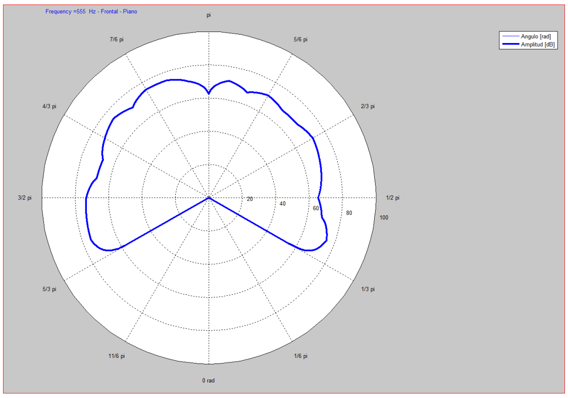

Figure 44: 554,9 Hz.

Figure 45 shows the polar pattern for the highest note played, whose frequency is 784 Hz. Figures 46 and 47, shows the 2nd and the 3rd harmonics respectively (1568 Hz and 2352 Hz).

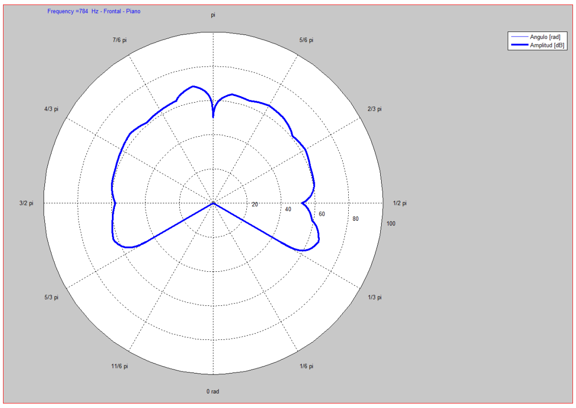

Figure 45: 784 Hz.

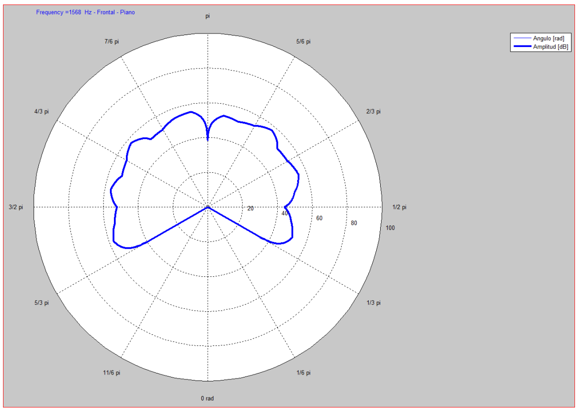

Figure 46: 1568 Hz.

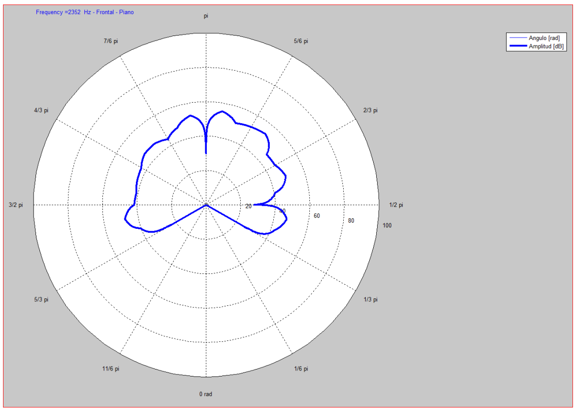

Figure 47: 2352 Hz.

Figures 48, 49 and 50 show the polar pattern for the frequencies of 1 KHz, 3 KHz and 5 Khz, for the frontal plane, “Piano” intensity.

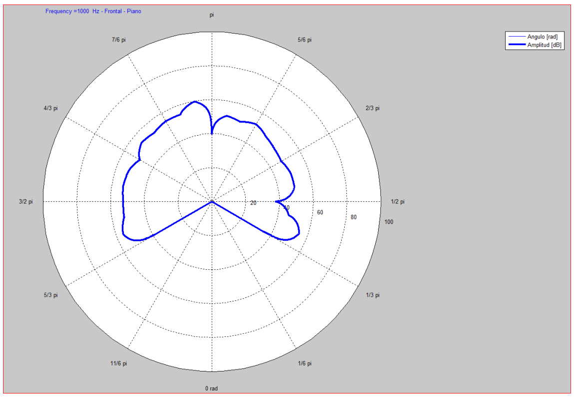

Figure 48: 1 KHz.

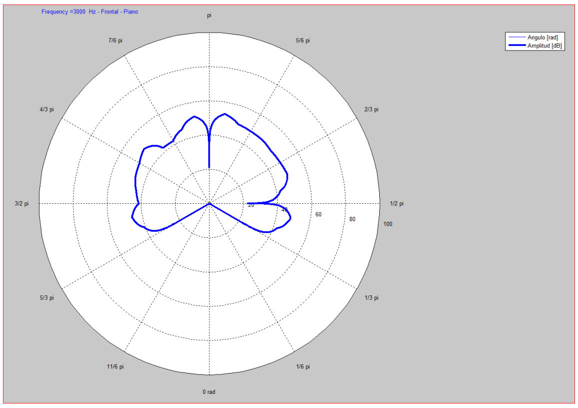

Figure 49: 3 KHz.

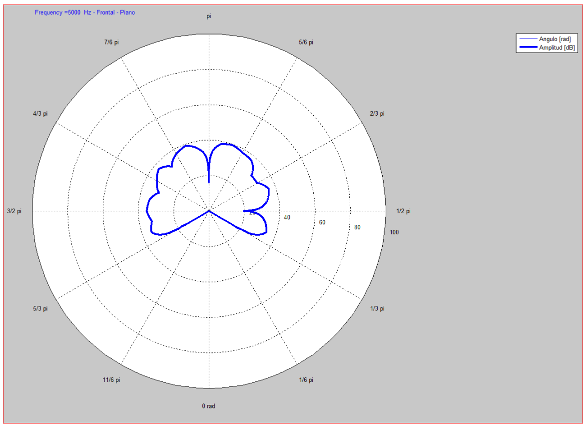

Figure 50: 5 KHz.

5.3 Sagittal plane

5.3.1 “Forte” results

Figure 51 shows the contour color MATLAB function, which is the frequency response vs. the angle of the horizontal axes. The color represents the sound pressure level (SPL). This graphic gives an idea of the polar pattern in all frequencies.

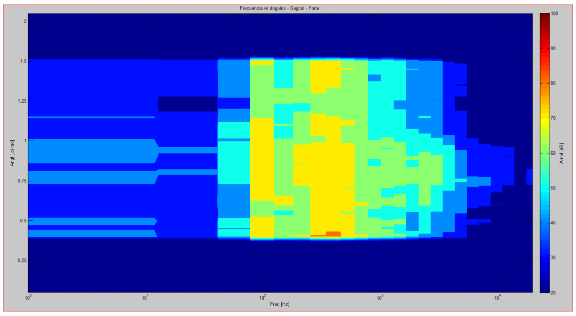

Figure 51: Contour plot for sagittal axis, “Forte” intensity.

Figure 52 shows the polar pattern for the lowest note played, whose frequency is 185,9Hz. Figures 53 and 54 show the 2nd and the 3rd harmonics respectively (370Hz and 554.9Hz).

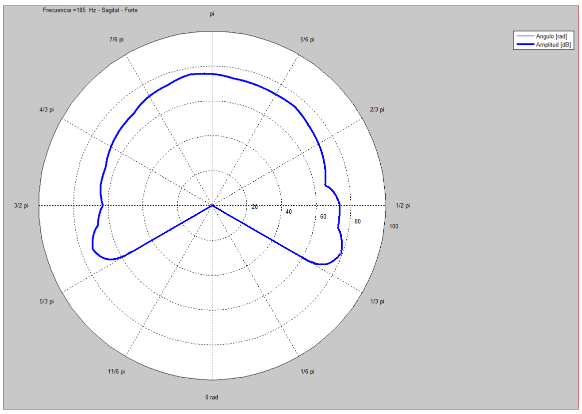

Figure 52: 185,9 Hz.

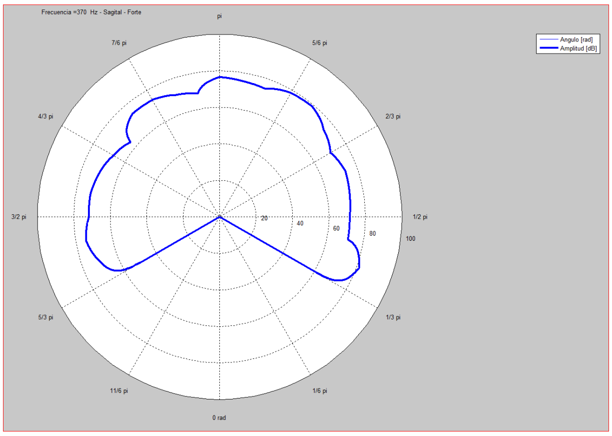

Figure 53: 370 Hz.

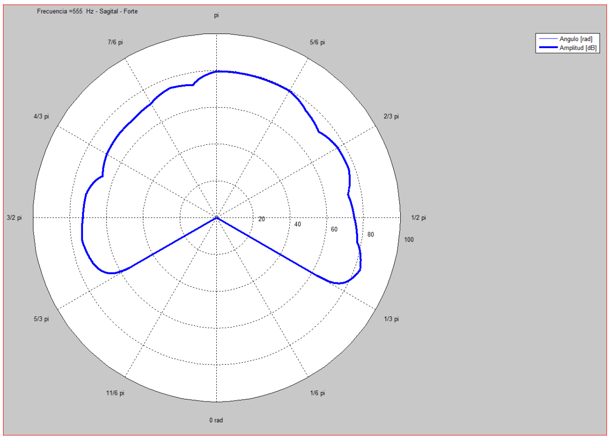

Figure 54: 554,9 Hz.

Figure 55 shows the polar pattern for the highest note played, whose frequency is 784 Hz. Figures 56 and 57 show the 2nd and the 3rd harmonics respectively (1568 Hz and 2352 Hz).

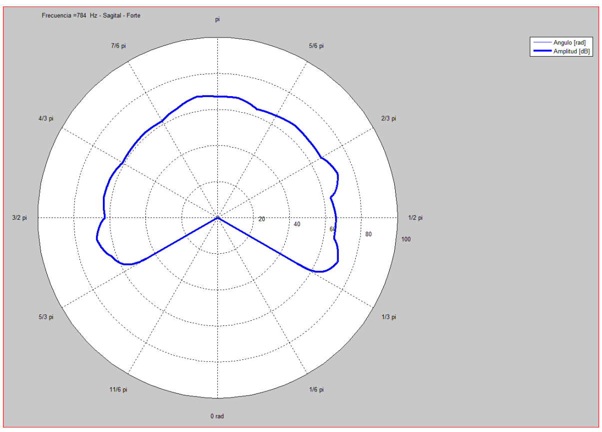

Figure 55: 784 Hz.

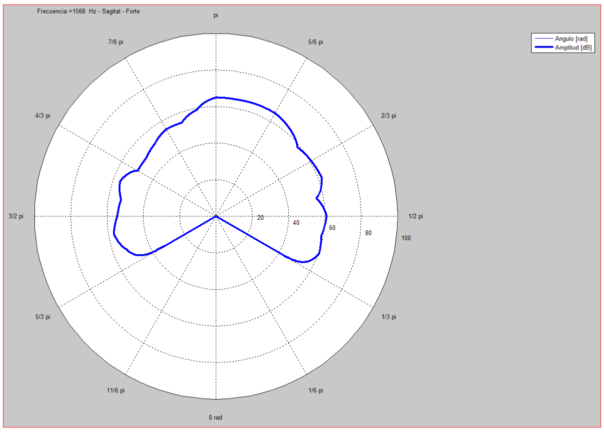

Figure 56: 1568 Hz.

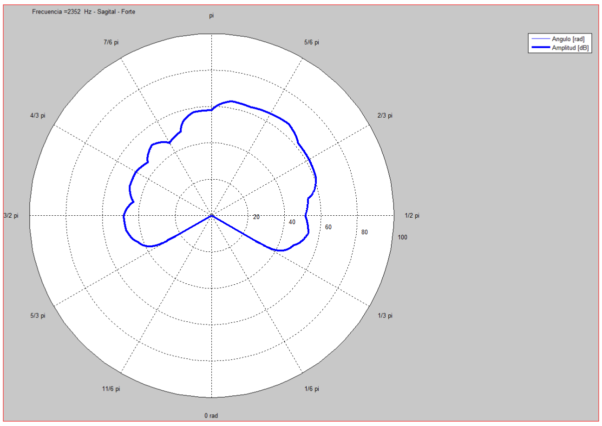

Figure 57: 2352 Hz.

Figures 58, 59 and 60 show the polar pattern for 1 KHz, 3 KHz and 5 KHz respectively.

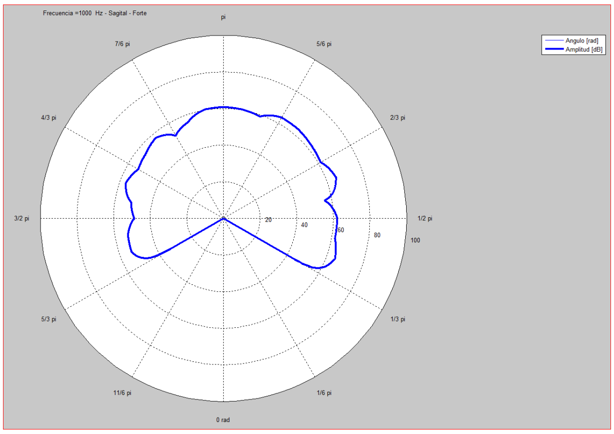

Figure 58: 1 KHz.

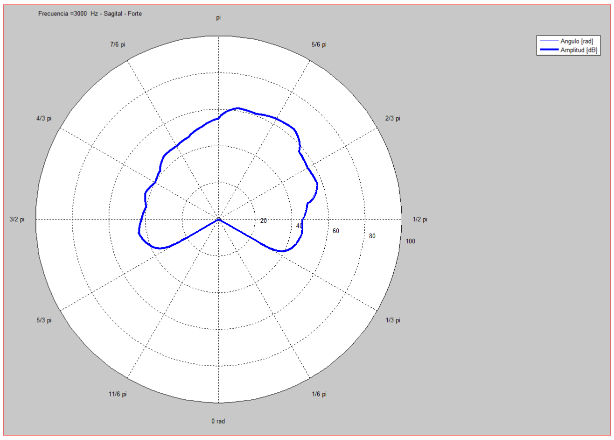

Figure 59: 3 KHz.

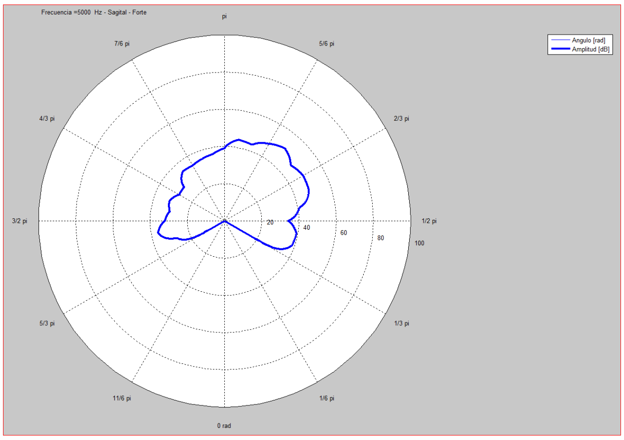

Figure 60: 5 KHz.

5.3.2 "Piano" results

Figure 61 shows the contour color MATLAB function, which is the frequency response vs. the angle of the horizontal axes. The color represents the sound pressure level (SPL). This graphic gives an idea of the polar pattern in all frequencies.

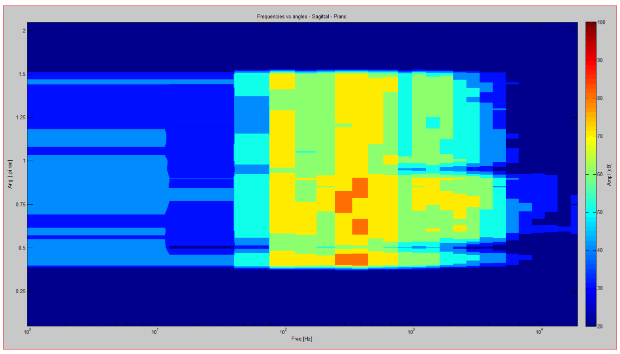

Figure 61: Contour plot for sagittal axis, "Piano" intensity.

Figure 62 shows the polar pattern for the lower note played, whose frequency is 185,9Hz. Figures 63 and 64 correspond to the frequencies 370 Hz and 554,9Hz, the 2nd and the 3rd harmonics respectively.

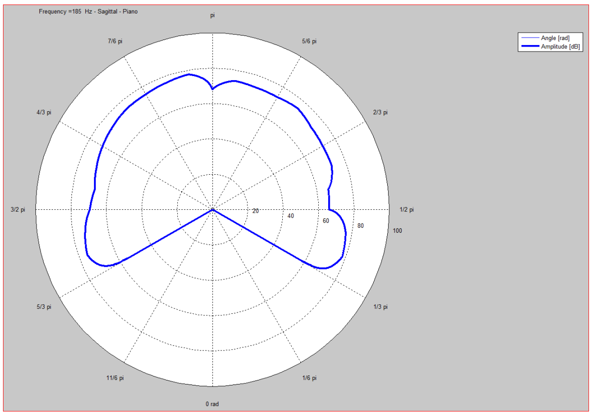

Figure 62: 185,9 Hz.

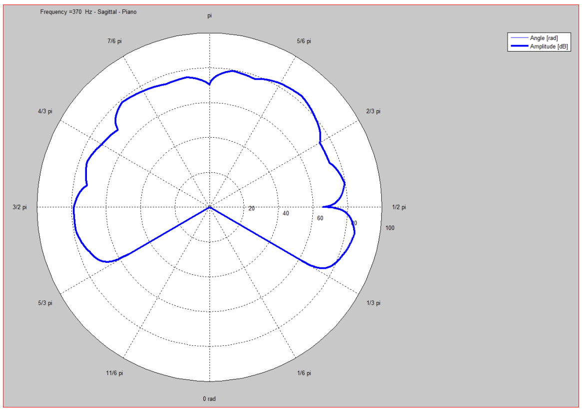

Figure 63: 370 Hz.

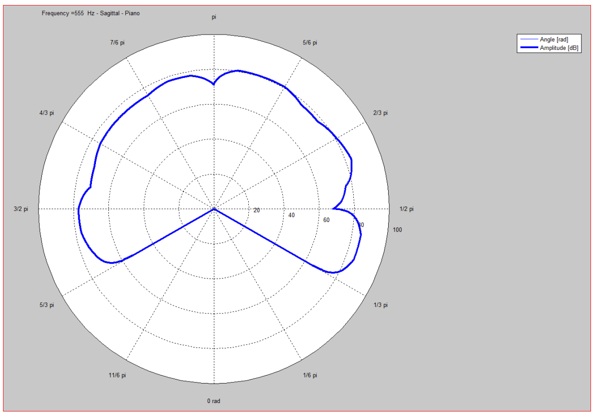

Figure 64: 554,9 Hz.

Figure 65 shows the polar pattern for the highest note played, whose frequency is 784 Hz. Figures 66 and 67 show the 2^{nd} and the 3^{rd} harmonics respectively (1568 Hz and 2352 Hz).

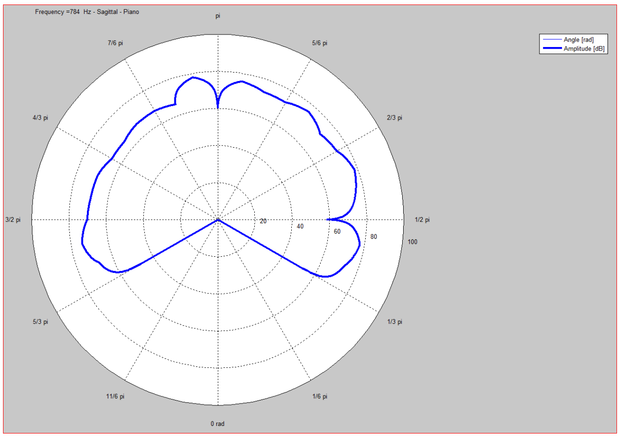

Figure 65: 784 Hz.

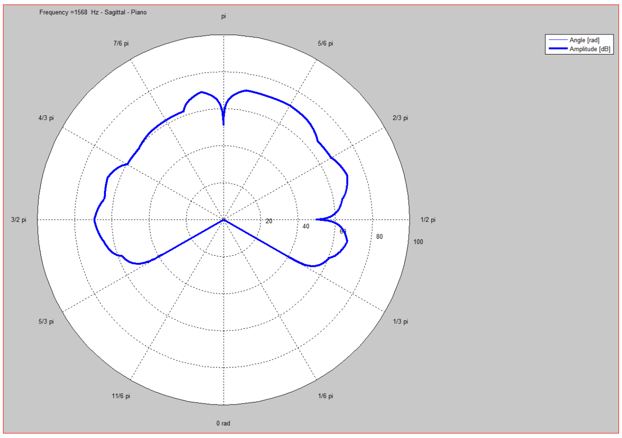

Figure 66: 1568 Hz.

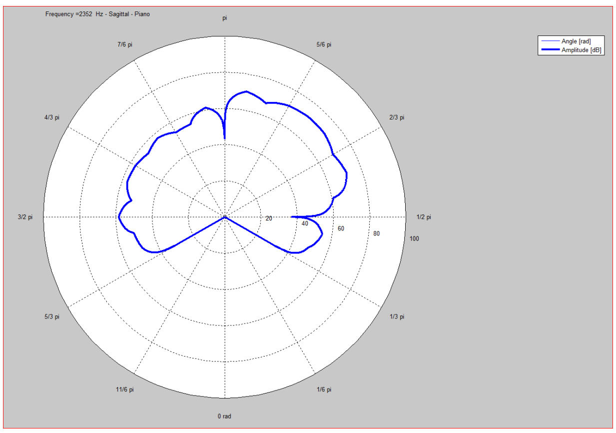

Figure 67: 2352 Hz.

Figures 68, 69 and 70 show the polar pattern for 1 KHz, 3 KHz and 5 KHz respectively.

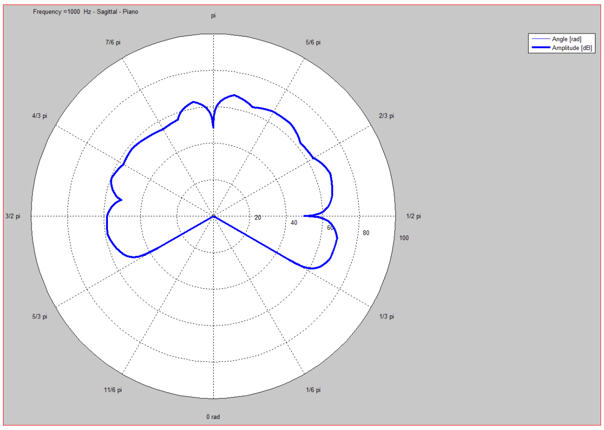

Figure 68: 1 KHz.

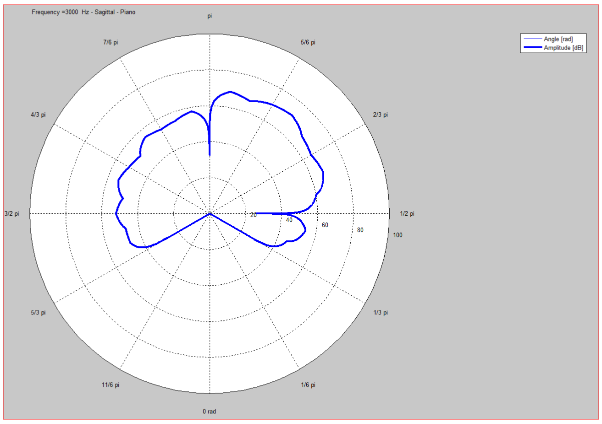

Figure 69: 3 KHz.

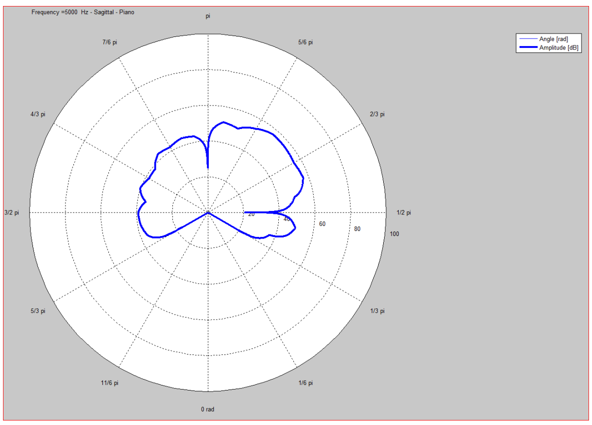

Figure 70: 5 KHz.

6. UNCERTAINTY

To obtain reliable results, it was necessary to determine the uncertainty of the measurement procedure. Different causes of errors can be the following:

- Measurement instrument tolerance. All the instruments, as the microphones, the sound level meters or the electronic instruments, have a deviation, which is specified by the manufacturer. Many of these deviations were offset in the Post-processing section, however, a deviation from the calibration was given, in the Calibration section.
- Differences between the positions of the microphones. The microphone or sound level meter location in different points of the theater, carries diverse acoustic features, therefore, they never capture the same sound field. With a few results obtained from the background noise measures, a deviation was obtained.
- Error in the distance of the measurement point.
    - The deviation in the angle of the microphone location introduces an error in the measurement.
    - In the same way, the human mistake produces an error in the microphone location, also the instrumentalist's natural movements. This type of deviation was in the three axes of the distance between the microphone location and the instrument (saxophone), and an estimated value was obtained with a laser distance meter.
- Musical interpretation intensity. The replay of the musical passage carries a deviation of the intensity of each note reproduction. With the "reference" measurement in front of the musician, a deviation was made.
- Changes of the weather condition. The temperature change on the day causes a deviation of the acoustical ear impedance, thus in the sound pressure measurement.

There are two types of uncertainty: the type "B" which is based on the certifications of the instruments, and the type "A" which is obtained from the measurement.

The group on item 1 corresponds to the type B, thus for each sound level meter a "Cal Factor" from the Calibration Section was obtained, which are detailed in Table 2. In the Calibration Section a microphone compensation was explained. This values of "Cal Factor" are the uncertainty (ΔX) of each sound level meter.

Table 2: "Cal Factor"

Sound level meter	Cal Factor (start)	Cal Factor (end)
1	0.23 dB	0.19 dB
2	0.2 dB	0.13 dB
3	0.4 dB	0.39 dB
4	-0.04 dB	-0.05 dB
5	0.4 dB	0.45 dB

Secondly, the groups on the Item 2 and henceforth correspond to the Type A of the uncertainty. Because the number of measurements exceeds a minimum value, a Normal (or Gaussian) approximation was made. The average of the measurement results can be estimated with the Equation 6, where X is the variable who was measure, $\bar{X}$ is the average of that variable, n is the number of measures, and Xi is the values of each measure.

$$\bar{X} = \frac{1}{n}\sum_{i=1}^{n} X_i \tag{6}$$

The standard deviation, which equals the uncertainty of the measurements results, can be estimated with the Equation 7, where σ_m is the standard deviation.

$$\Delta X = \sigma_m = \sqrt{\frac{\sum_{i=1}^{n}(\bar{X} - X_i)^2}{n(n-1)}} \tag{7}$$

The Equations 6 and 7, correspond to variation of the SPL in the measurement, only if the deviation error has direct involvement. But some errors, like the groups on item 3 and 5, have indirect involvement. In this cases, is necessary to do the partial derivatives, like shown the Equation 8, where ΔX is the final uncertainty, and ΔY_k is the indirect involvement standard deviation on X.

$$\Delta X = \left|\frac{dX}{dY_1}\right|\Delta Y_1 + \left|\frac{dX}{dY_1}\right|\Delta Y_2 + \cdots + \left|\frac{dX}{dY_k}\right|\Delta Y_k$$

(8)

An estimate of the average, the standard deviation and, consequently, the uncertainty were made with this Equation. For the groups in item 2 and 4, which their deviation have direct involvement, only the Equation 6 and 7 were used. Because the different values of SPL were obtained with the different microphones locations when the background noise was recorded for the group in item 2, and the different values of SPL were obtained with all reference measurements in each musical passage reproduction.

Figure 71 and 72 shows the average and the deviation for items 2 and 4. The uncertainty due to different SPL in each interpretation is 3.10 dB and the uncertainty due to the different positions of the microphones in the theater is 3.04 dB.

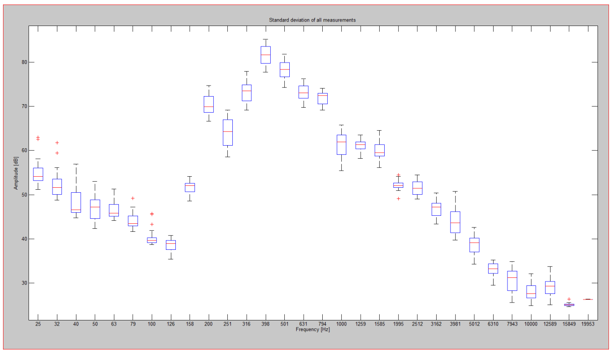

Figure 71: Boxplot of all reference measurements.

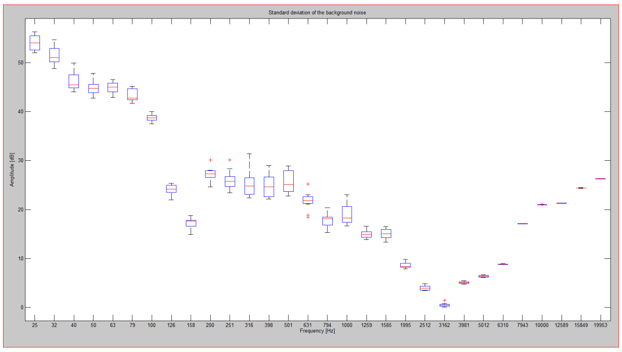

Figure 72: Boxplot of all background noise measurements.

The groups in the item 3 and 4, are indirect involvement deviations, thus the Equations 6, 7 and 8, was used. The partial derivatives arise from the SPL equation in the free field, where the acoustic source is considered with a spherical wavefront. So the SPL lost function of the distance and temperature (or the acoustical ear impedance), is shown in Equation 9.

$$SPL = 10\log(W) + 10\log\left(\frac{Q}{4\pi r^2} + \frac{4}{R}\right) + 10log\left(\frac{\rho_0 c_0}{P_{ref}^2} W_{ref}\right) \qquad (9)$$

Where:

Q= directivity factor of the source. In this case, is equal to 1.

r= distance between the microphone and the source.

R= reverberation constant of the theater.

P_0= ear density, whose value ranges between 1.2216 (for temperature of 15°C) y 1.1993 (for temperature of 20°C).

C_0= sound's speed, which depends on Equation 10.

W_{ref}= =reference acoustical power equal to 10 pW

P_{ref}= = reference sound pressure equal to 20μPa

SPL = sound pressure level

W = source acoustical power

$$c_0 = 332 + 0{,}608\,Temp \qquad (10)$$

To determine the uncertainty, the partial derivation of Equation 9 was made. The Equation 11 shows the derivative with respect to x, which is the horizontal distance between the musician and the microphones. This movement involves the forward and backward displacement.

$$\frac{dSPL}{dx} = \frac{d\left(10log\left(\frac{Q}{4\pi r^2}\right)\right)}{dx} = \frac{d\left(10log\left(\frac{Q}{4\pi}\right) - 10log(r^2)\right)}{dx} = -20\frac{1}{\ln(10)}\frac{1}{r} \qquad (11)$$

The Equation 12 shows the derivative with respect to y of the Equation 9, which is the horizontal movement of the musician or the microphones, perpendicular to the distance between. This movement involves the rightward and leftward displacement.

$$\frac{dSPL}{dy} = \frac{d\left(-10log(r^2)\right)}{dy} = -20\frac{1}{\ln(10)}\frac{y}{2(x^2+y^2)} \qquad (12)$$

Equation 13 shows the derivative with respect to z of the Equation 9, which is the vertical movement of the musician or the microphones, also perpendicular to the distance between. This movement involves the upward and downward displacement.

$$\frac{dSPL}{dz} = \frac{d\left(-10log(r^2)\right)}{dz} = -20\frac{1}{\ln(10)}\frac{y}{2(x^2+y^2)} \qquad (13)$$

Equation 14 shows the derivative with respect to the angle of the Equation 9, which is the angular movement of the microphones locations with respect to the horizontal or vertical axes between the musician.

$$\frac{dSPL}{d\alpha} = \frac{d\left(-10log(r^2)\right)}{d\alpha} = -20\frac{1}{\ln(10)}\frac{sen(\alpha)}{\cos(\alpha)} \quad (14)$$

The Equation 15 shows the derivative with respect to the temperature of the Equation 9, combinated to the Equation 10. The temperature modifies the acoustical ear impedance. Was necessary to consider the ear density as a constant, because its variation is minimal.

$$\frac{dSPL}{dT} = \frac{d\left(-10log\left(\frac{\rho_0 c_0}{P_{ref}^2}W_{ref}\right)\right)}{dT} = 10\frac{1}{\ln(10)}\frac{1}{\rho_0(332+0{,}608T)}0{,}608 \quad (15)$$

With a MATLAB code, the results of the uncertainty were obtained. The uncertainty due to the deviation of the positions or the music player is 0.38 dB; and the uncertainty due to the deviation of the temperature variation is 0.01 dB.

Finally, the maximum value of deviation Type A and the maximum Type B were chosen. The final result of the uncertainty of all measurements deviations corresponds to Equation 16.

$$devMAX = \sqrt{typeA^2 + typeB^2} \quad (16)$$

The final uncertainty was obtained. The definitive uncertainty due to different deviations of the measurement procedure is 3.13 dB.

7. DISCUSSION

The methodology of the present work tried to emulate those that's been studied and referenced on this paper. Since a great number of investigators made measurements in anechoic chambers, the Auditorium was selected to do the experiment. Also, a method consisting in measuring angles was carried out. These differences related to the conditions of the environment and the measurement techniques can represent a difference in the results that were obtained.

To get more precise results, it is important to make measurements with a circular structure for the vertical planes. Using a structure can guarantee that all angles are equal.

8. CONCLUSIONS

The measurement of the directivity pattern of a wind instrument was made. The study of it shows a few points of interest. The first observation is that the polar pattern of the saxophone is similar to an omnidirectional source. The graphics on Forte mode, in most of the frequencies analyzed, determine that the pressure level decreases in the same size of most angles. This decrease depends, evidently, on the decay of the harmonic's SPL. The frequencies range of the instrument with a simple melody and technique doesn't exceed 5 KHz, and between the lower note (approximately 180 Hz) and the highest, the decay is relatively linear in all angles.

But, on the other hand, in Piano mode lobes appear. In the three axis, when the frequency increases, the polar pattern is not constant. A few lobes appear in a few angles, and it gets worse. Evidently, that phenomenon depends on the intensity of the musical interpretation. Future works are necessary to determine the origin of these lobes. The specification of this phenomenon can only be done in an anechoic chamber, because it is uncertain if the causes are the first reflections of the floor, walls or cells. Another option is that the low intensity of the high frequencies and the higher harmonics of each note are masked with the background noise.

To complement this idea, an analysis of the harmonic relationship between the Forte mode and Piano mode was made. Is clearly that the relationship of each harmonic in the same angle, and different intensity, presents no constant relationship between them. So that means the polar pattern is modified by the intensity of the harmonics of the fundamental played note. In an anechoic chamber, maybe the situation will be different. But in a real situation in a theater with reverberation time unequal to zero and first reflections that influence the direct sound, the polar pattern of the instrument is not constant for each frequency and different play intensities.

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Annex 1

Curriculum Vitae

Axel Montes de Oca is a session musician, professional saxophonist and a Sound Engineering student from La Plata, Buenos Aires, Argentina.

He learned to play the saxophone and studied at the Gilardo Gilardi Conservatory from 2004 to 2008, obtaining the level FOBA, which corresponds to a Classical Musical academic training.

He also studied at the EMU Institute from 2007 to 2009 obtaining the Professional Musician degree.

He started his activities as a session musician in 2013, working with several musicians and musical groups: Luz Graziano, Lucía Giles, Quique Roca, Agathis Alba, DES, La Huella, Blanca Grande, among others. He had also been part of the soundtrack of the movie “Mis Margaritas”.

He also play the keyboard as a session musician.

Annex 2

This section presents questions to emphasize a few topics.

1. Given a constant room acoustics, what results do you think it has on the audience the difference found between the different intensities? For example changes in the direction of perception, in the direct sound, in the reverberant field, intelligibility, etc.

   Apparently, the perception of the direct sound doesn't suffer modification when the intensity of sound changes. Because in all cases the sound it's similar to an omnidirectional source. But, it's true that the decay intensity causes the origin of lobes in a few angles, for specific frequencies.

2. Has the relative amplitude of harmonic currents generated between the two reproductions? Results show signal processing.

   It's a fact that the relationship of the harmonic distribution in the two cases is different. The graph of that relationship was shown in the section Characteristic of the timbre. The difference is not constant, but the harmonic intensity doesn't decrease in the same proportion as the fundamental frequency.

3. If the acoustic measurements and designs provide for the use of omnidirectional sources, What changes (in metrology) would entail understanding the dynamics of polar patterns real sources for different levels of performance?

   When an omnidirectional source is used for measurements, in most cases, the reproduction of a real instrument is made in a simple way. The sound's character of that instrument is recorded in a specific intensity, and for its reproduction, only the level amplifier is modified to create the perception of loudness. But in this work, a new idea was present. Not only changing the level pressure of the source is enough: also the relationship of the harmonics and its effect on the polar pattern.

4. What would be the recommended distance and position of a microphone to the correct record of the source music recordings?

   Generally it's better to locate the microphone for recording a few meters from the instrument. But it is necessary to consider the acoustic properties of the chamber. As was shown in the work, it is better to locate the microphone into the free field. And that depends on the reverberation time, the chamber volume and the frequencies to be recorded.