CHALMERS

Identification of the effective directivity of the human voice for a person sitting in the car seat with head restraints

Master's Thesis in the Master's programme in Sound and Vibration

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Department of Civil and Environmental Engineering
Division of Division of Applied Acoustics
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Abstract

Today the comfort level inside the vehicle is drawing lots of attention. The optimal listening conditions inside a vehicle compartment are of paramount importance for the vehicle makers, as this is one of the most relevant points in assessing the “comfort” of the vehicle. An important parameter is the directivity of human voice. The requirements for vehicle interior acoustics have increased over the past years. Besides infotainment systems and telecommunication devices, communication between passengers plays the most important role for the specification of acoustic related interior parts.
The main aim of this thesis work is to establish the spatial dimensions of human speakers inside vehicles with head restraints. A three-dimensional model is created to show the directivity of the human voice. Instead of the artificial loudspeaker, volunteers are needed in this research. Head restraints are required. Related to vehicle-condition the free-field condition is necessary. For validation purpose another seat as vehicle-condition will be compared.

Key words: sound directivity, vehicle interior acoustics, human voice, head restraints

Acknowledgements

First of all I wish to thank GM for giving me the chance to do this thesis. I also want to thank the Department of Applied Acoustics for providing me with all the needed support for conducting this thesis work.
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Contents

1. Introduction

1.1. Thesis Background

The optimal listening conditions inside a vehicle compartment are of paramount importance for the vehicle makers, as this is one of the most relevant points in assessing the "comfort" of the vehicle. One of the intelligibility's parameters that have been reveled to be more sensitive inside vehicles is the directivity of voice. As a wave motion the human voices have obvious directivity, with the angles' changing of the directivity, the articulation index will be changed clearly.

The requirements for vehicle interior acoustics have increased over the past years. Besides infotainment systems and telecommunication devices, communication between passengers plays the most important role for the specification of acoustic related interior parts. For previous research, 2D model is always used; see Figure 1. Some researches used 3D model, however, the most popular method to research and measure the directivity of the human voices inside the vehicle is using the artificial speakers as the mouth simulator to imitate the real human voices. Though the artificial speakers have some advantages, like easy to use, simple to control the output level, the weak point is clearly, the loudspeakers are not as real as the human voices. However, the knowledge of the spatial dimensions of human speakers inside vehicles is still missing.

This article is to pay close attention to the directivity of human voice for a person sitting in a car seat with head restraints. It is well-known that the neck of human being is a kind of "multi degree of freedom system", which could keep the forward direction to the learners. However, for the most time of driving, the head of the person sitting in a car is restrained; the voice will be influenced by the environment, especially by the seat.

1.2. Objective

Because of the lack of 3D data, it was decided to use a "fullerene" model, shown as Figure 2. However, due to problems at SAAB, the fullerene was not provided in time to be used in this work. This made it necessary to use a simplified and less satisfactory approach on described.

For this article, using any artificial speakers is renounced. The person as the volunteer is considered as the sound source, which is much more close to the practical data. All the measurements are measured inside an anechoic chamber.

Primarily, directivity measurements of human voices under free-field conditions are required, which is used as the standard and reference value.

Two car seats are needed. One is used for the main measurements; another one is as the comparative analysis. The volunteers are required to hold their head close to the seat, which makes sure the forward direction to keep to the front of volunteers. The measured microphones are fixed at some regular angles to record the voice specimen.

The final project objective is to establish a statement about the directivity of human speakers sitting in car seats with head restraints.

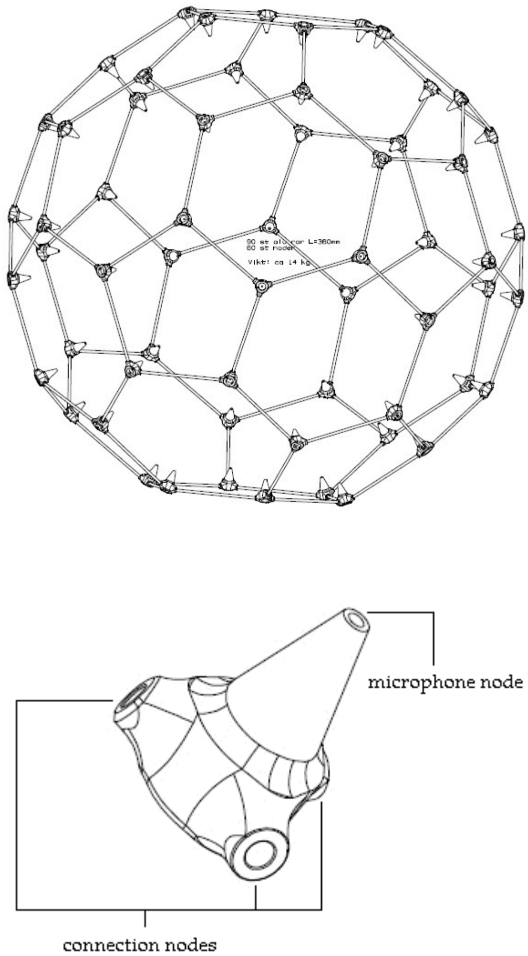


Figure 2: *Fullerene model and node.*

2. Measurement

2.1. Theory

2.1.1. Sound Pressure Level

The microphones were used for measuring the sound pressure in the air in this measurement. Sound pressure is the local pressure deviation from the ambient (average, or equilibrium) pressure caused by a sound wave. [1]

The sound pressure deviation (instantaneous acoustic pressure) P is:

$$P = \frac{F}{A} \qquad (2.1.1.1)$$

Where the F is force and the A is area.

In a sound wave there are extremely small periodic variations in atmospheric pressure to which our ears respond in a rather complex mechanism. The minimum pressure fluctuation to which the ear can respond is less than one-billionth of atmospheric pressure.

Because of the wide range of pressure stimuli, it is convenient to measure sound pressures on a logarithmic scale, called the decibel (dB) scale. Although a decibel scale is actually a mean for two sound pressures, we can define a decibel scale of sound level by comparing sounds to a reference sound with a pressure amplitude $P_0 = 2 *10^{-5}$ (N/m^2) assigned a sound pressure level of 0 dB. Thus, we define sound pressure level (SPL or L_p) as:

$$L_p = 10\log_{10}\left(\frac{P^2}{P_0^2}\right) = 20\log_{10}\left(\frac{P}{P_0}\right) \qquad (2.1.1.2)$$

Where:
The P is the root-mean-square sound pressure being measured.

2.1.2. Microphone sensitivity

The sensitivity of the microphones is an important parameter for the measurement with the microphone. The sensitivity of a microphone is usually expressed as the sound field strength in decibels (dB) relative to 1 V/Pa (Pa = N/m^2) or as the transfer factor in mille volts per Pascal (mV/Pa) into an open circuit or into a 1 kilo ohm load. [1]

2.1.3. Normalization

Since the measurement could only measure one direction at once, the normalization of recording data becomes very important. If not, the sound wave will be weakened in the anechoic chamber with the increase of distance between the mouth of volunteer and measurement microphones.

The normalization of the recording data includes two steps. The first one is called "Arc-self normalization", which normalizes the twelve measurement microphones on the arc with the reference microphone in one arc direction. (See Figure 7)

Formulating it as:

$$L_{p\,NorSelf\_Mic\_i} = L_{P\,Mic\_i} - L_{P\,Rf} \tag{2.1.3.1}$$

Where:
$L_{p\ NorSelf\_Mic\_i}$ is the normalized sound pressure level of measurement microphone No. *i*.
$L_{p\ Mic\_i}$ is the sound pressure level of measurement microphone No. *i*.
$L_{p\ RF}$ is the sound pressure level of reference microphone.

Because the reference microphone is very close to the mouth of volunteer and the measurement microphones are at least one meter far away from the volunteer, the normalized sound pressure level will be a negative value.

The second step is called "Arc-position normalization", which normalizes the twelve different arc positions. The arc positions are the relative positions related to the volunteers, because the arc is immovable while the seat is fixed on a revolving stage which is turned twelve times by 30 degrees during the measurement of each person. The reference is the face arc direction and is at the same height with the mouth of the volunteer, marked as (0,0) degree. In theory, the direct sound is the max value compared with other directions in the anechoic environment.

Formulating it as:

$$L_{p\,Nor\_Mic\_i} = L_{P\,NoSelf\_Mic\_i} - L_{P\,(0,0)\,\deg ree} \tag{2.1.3.2}$$

Where:
$L_{p\ (0,0)degree}$ is the sound pressure level of the microphone which measured the direct sound.

During the two steps, the recording data is turned into the normalized form relative to the mouth point and the unit is "one".

2.2. Method

2.2.1. Volunteers' Questionnaire Survey

In this measurement, some volunteers joined in as the sound source. In previous researches, an artificial torso with a mouth simulator is always used as the sound source. Although the sound of volunteers is uncontrollable and the level will not be as stable as the output of artificial mouth loudspeaker, it will be closer to the practical situation. Also long recording time, enough volunteers and multiple metering will decrease the error probability.

Seventeen people joined in this measurement, seven of them are female and ten are male. All the volunteers were twenties and used English as the test language.

The table below is the information of volunteers.

Male			Female		
No.	Age	Native Language	No.	Age	Native Language
1	25	Swedish	1	27	Swedish
2	24	Chinese	2	22	Chinese
3	27	Icelandic	3	25	Chinese
4	26	Chinese	4	24	Chinese
5	27	Spanish	5	24	Swedish
6	24	Chinese	6	25	Chinese
7	24	Chinese	7	29	Chinese
8	29	Spanish			
9	27	Chinese			
10	23	Chinese			

Table 1: *Volunteer Information.*

2.2.2. Anechoic Chamber

The anechoic chamber at Applied Acoustics department, Chalmers, is used as the measurement environment in this research. The anechoic chamber was built in 1969; the size is 10*10*8 m^3. Sillan wedges which are 1 meter long were glued directly to the concrete inner surfaces. Some other parameters of the anechoic chamber are at below: [4]

Background level: Less than 17 dBA.

Useful Frequency Range: 75 Hz - 10 kHz with mean reflection coefficient less than 0.1 giving a sound absorption higher than 99 %.

During this measurement, the speech of volunteers was recorded. Speech has a wide frequency range, having frequency components from approximately 100 Hz to about 10 kHz. So this anechoic chamber is fit for experimental requirements. [1] (The photo of anechoic chamber looks Figure 7)

2.3. Measurement Set-up and Equipment

2.3.1. Vehicle Seats

Three different chairs were used as test seats in this research, two of them were vehicle seats which came from SAAB, and the third one was a box.

The two vehicle seats were of same size but used distinct surface materials. One vehicle seat was a leather trim seat, which was used for the main measurement of vehicle-related condition. The second one was a fabric upholstered seat, which was used for the contradistinctive test of vehicle-related condition. The vehicle seats are shown in Figure 3 and Figure 4.

Figure 3: *The vehicle seat one, which was the leather trim seat and used for the main measurement of vehicle-related condition.*

Figure 4: *The vehicle seat two, which was the fabric upholstered seat and used for the contrast test as vehicle-related condition.*

The third chair was a wooden box. The box was used as a test seat in the contradistinctive test to measure the free-field condition and the height was nearly the same as the vehicle seats. The wooden box is shown in Figure 5.

Figure 5: *The wooden box, which was used as the other seat in the contrast test to measure the free-field condition.*

2.3.2. Microphones and Bracket Arc

Thirteen Panasonic WM60 electret microphones were used for data acquisition. Twelve of microphones were fixed on the bracket arc to record the direct sound at different angles. The last one, as the reference microphone, was put in front of volunteer's mouth very closely. The microphone positions are shown in Figure 7 (Real Object) and Figure 8 (Sketch).

Microphone's sensitivity is very important for the measurement. Although the model number of microphones is same, the sensitivity of microphones is different. The table below shows the microphones sensitivity:

No.	Sensitivity	No.	Sensitivity
Mic 1:	47	Mic 7:	53
Mic 2:	37	Mic 8:	47
Mic 3:	48	Mic 9:	44
Mic 4:	46	Mic 10:	47
Mic 5:	43	Mic 11:	48
Mic 6:	49	Mic 12:	47
Mic Reference:	49		

Table 2: *Microphone Sensitivity (mv/Pa).*

**The microphones from top to bottom are arranged as number from 1 to 12.*

Microphones at different positions were soldered on the cables with different length for distinct positions. A plug was soldered on the end of each cable. There was a 2.2 kΩ resistance is inside the plug. An example of microphone and plug is shown in Figure 6.

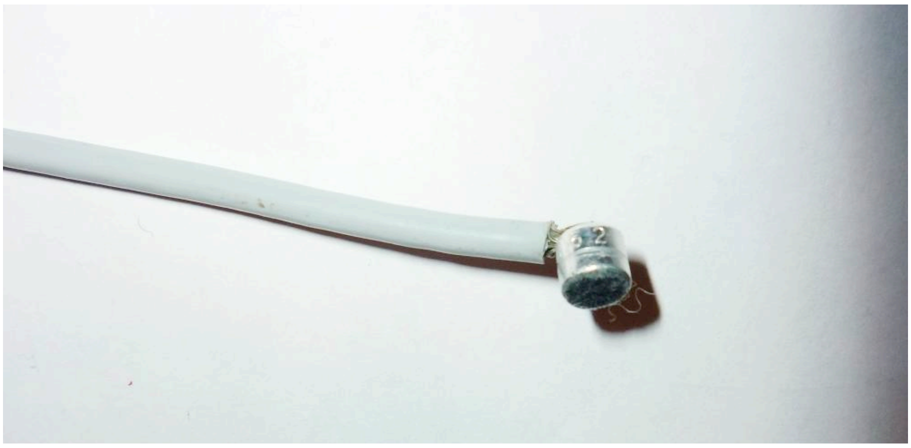


Figure 6: *An example of the Panasonic WM-63 microphones used for the measurements.*

The bracket arc was home-made. A long aluminum alloy bracket was bent as an arc, and then twelve microphones were taped on it. The vehicle seat was at the middle of anechoic chamber, on a mobile revolving stage, which was rotated anticlockwise by 30 degrees.

A reference microphone was taped on the side of seat back very closely. An A4 paper with the reading text was put at more than one meter away from the vehicle seat. The photo is shown in Figure 7 (Real Object).

The angles of microphones had some disparities with preliminary plan. The preliminary plan was put the microphones at every ten degrees from 90° to -10°or -20°. However, in the range of every ten degrees, there was at least one microphone, so it could get enough sound samples for the follow-up test. A sketch which marks the angle of each microphone position is shown in Figure 8.

The reason for angular deviation is because the bracket arc had to be hung on the ceiling. In this way the area of sound reflection could be decreased as much as possible, also hanging the arc could prevent the electromagnetic interference from the wire netting floor.

Figure 7: *Bracket arc, microphone positions and vehicle seat one at the anechoic chamber.*

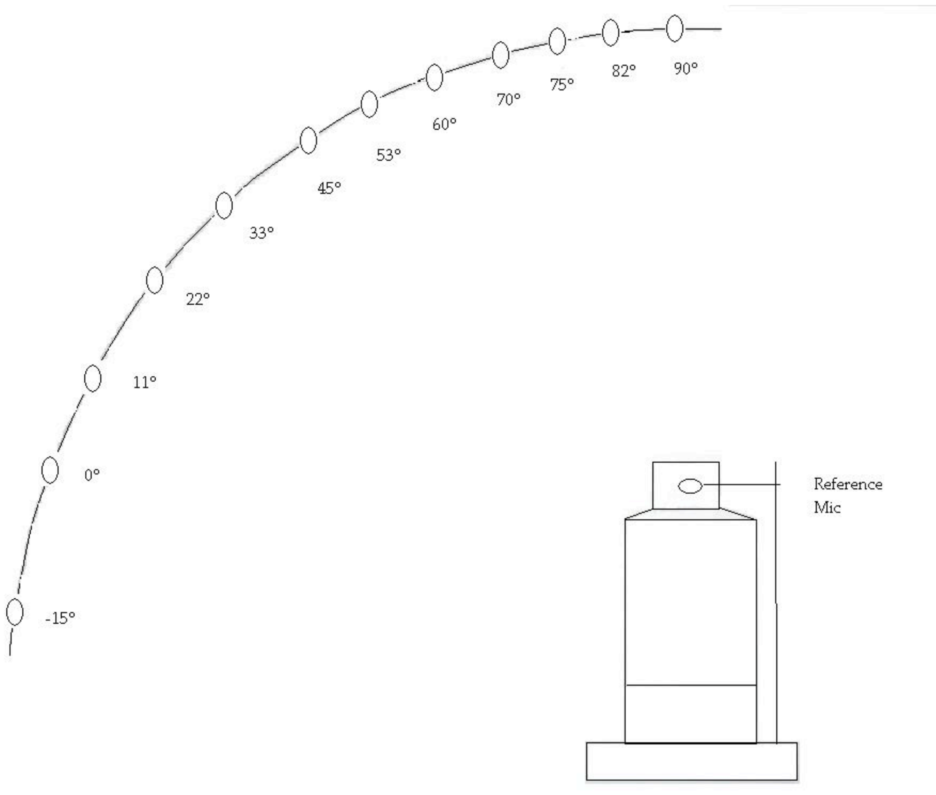


Figure 8: *A sketch of microphone positions.*

The distance between the measurement microphones and sound source (mouth position of volunteers) is shown in Table 3.

No.	Distance	No.	Distance
Mic 1:	1.83	Mic 7:	2.07
Mic 2:	1.85	Mic 8:	2.16
Mic 3:	1.89	Mic 9:	2.26
Mic 4:	1.92	Mic 10:	2.34
Mic 5:	1.93	Mic 11:	2.42
Mic 6:	1.99	Mic 12:	2.49
Mic Reference:	0.44		

Table 3: *The distance between the measurement microphones and sound source (meter).*

**To the reference microphone, the distance is an average value of 17 volunteers.*

2.3.3. Measuring Instrument and Software Introduction

Sixteen channels amplifier box was used (see Figure 9). Two batteries as the direct current electric source gave the power supply, which was quiet and impossible to influence the measurement. The box had enlarging function and supplied the electric powered for the microphones, then transported signals to data-processing system by cables.

An Agilent VXI system with two sixteen channels high speed digitizer cards (E1432A) was used, see Figure 9. Due to the size of the memory card, two cards were used for sharing. Two high speed digitizer cards were chose because the RAM inside the card was too small to supply for so many channels with long time recording. Although two cards were used, the recording time for once was approximately half a minute. To collect enough time signals, repeated measurements are needed.

Figure 9: *Measuring instrument.*

Left one is the sixteen channels amplifier box, home-made.

Right one is the Agilent VXI system (E8408A), with two sixteen channels high speed digitizer cards (E1432A). [10]

HP DAC Express (version 2.01) is used as main software. HP DAC Express is combined time-based measurements for static and dynamic signals. The working drawing looks Figure 10. Setup information is at below:

Recording time (s): *30*
Sample rate (Sa/s): *51200*
Sample period (s): *19.53E-06*
Blocksize: *1024*

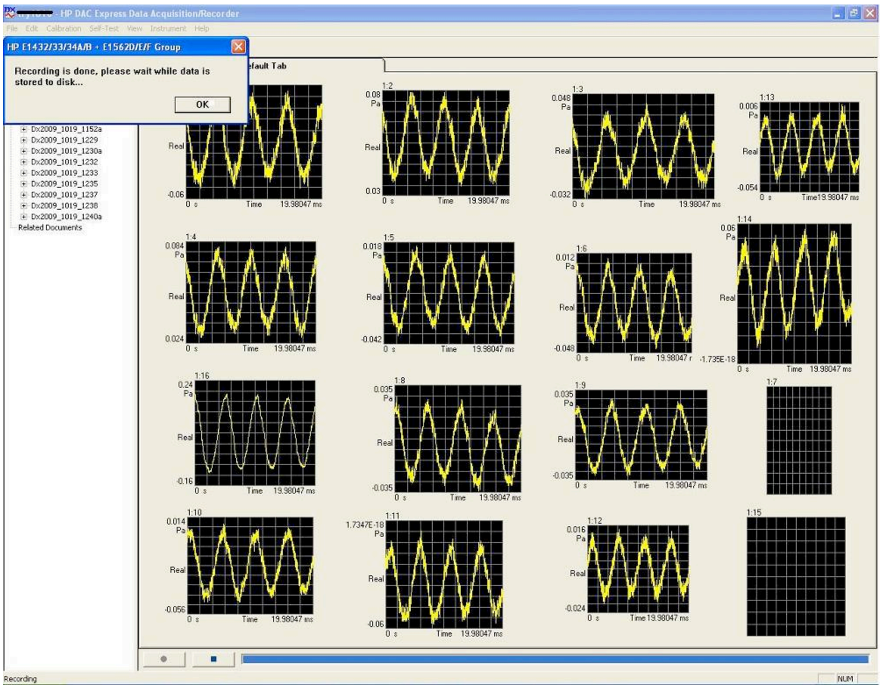


Figure 10: *A sketch of microphone positions.*

A sketch for setup is shown in Figure 11.

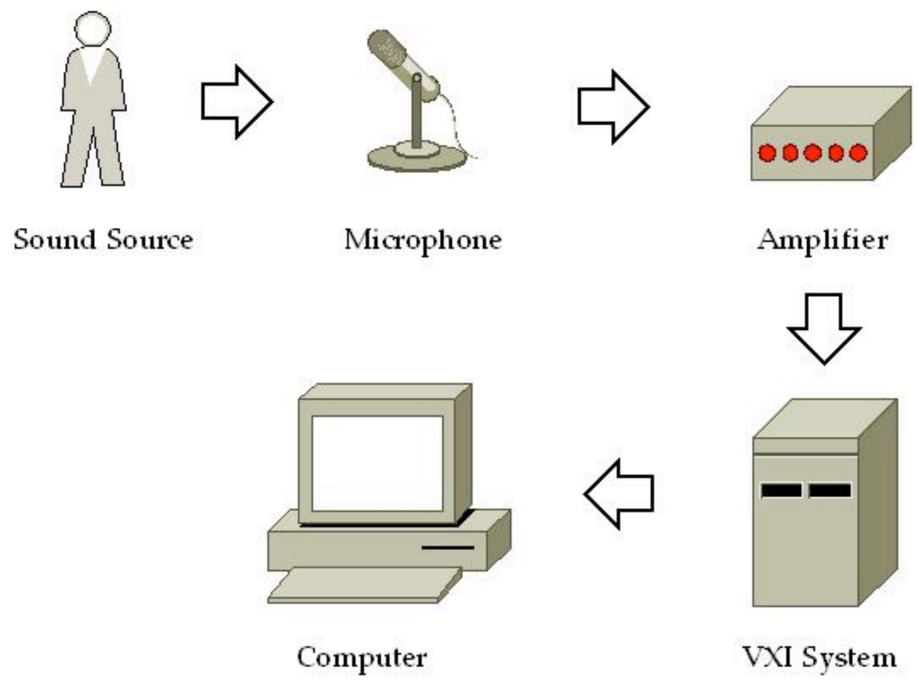


Figure 11: *A sketch for flow chart of setup.*

3. Results

3.1. Background Noise

Before the measurement, a simple test to calculate the background noise inside the anechoic chamber is essential.

Figure 12 is shown the background noise inside the anechoic chamber. Horizontal axis is 1/3 octave band frequency plot as log; vertical axis is 1/3 octave band sound press level in dB. The black line is the average value of each microphone but without the reference microphone, because the reference microphone shakes too much. The inverted triangle is the maximum value of microphones and the triangle is the minimum value.

This test is measured by measurement microphones but not the perfect microphones so there is different noise bandwidth with the reference value of anechoic chamber. A hanning window is used to calculate the FFT.

Obviously, there is a peak around 50 Hz, which is the electronic signal noise. However, the value is decreased very quickly. There is approximately -10 dB around 100 Hz. Normally the average SPL of speech for men and women is around 60 dB at a distance of 1 m in front of the talker [1]. 60 dB is much higher than the background noise. So this acoustics environment is fine for measurement.

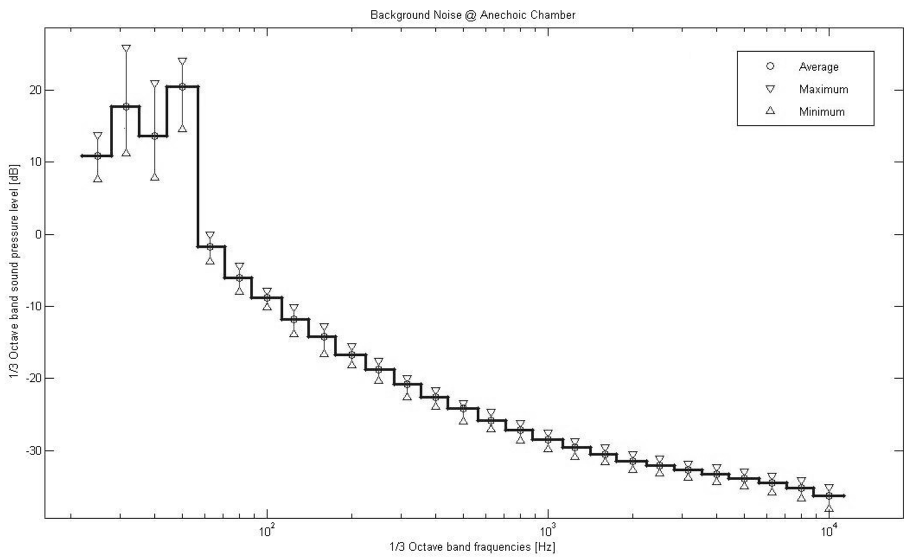


Figure 12: *Background noise in anechoic chamber.*

3.2. Main Vehicle-Related Condition

A 3D model is built to show the sound field around the volunteers, see Figure 13. Each node is as the same angle as microphone position except the bottommost circuit. The nodes on the bottommost circuit are added as -25 degree in the vertical position. Every three or four nodes make up a mesh; each mesh will be filled in some color to express the related sound level.

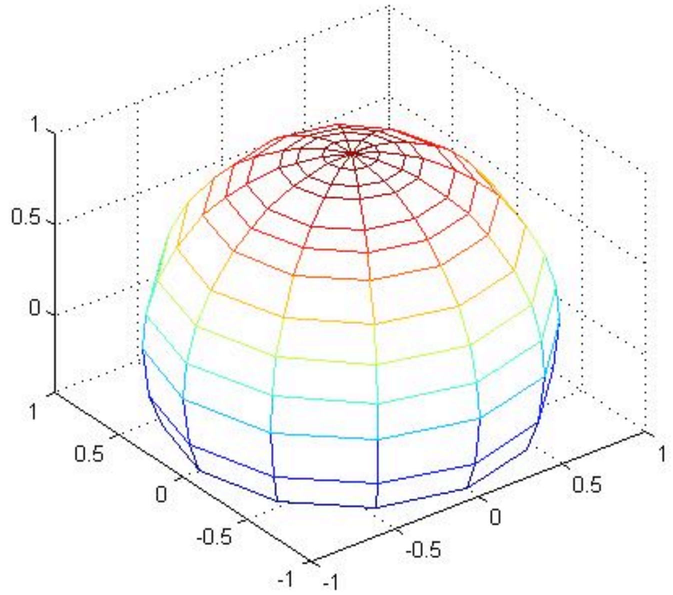


Figure 13: *A 3D model built as the same angle as microphone positions.*

The relative direction between the volunteers and the 3D model is shown in Figure 14. A female frequency model is used as an example.

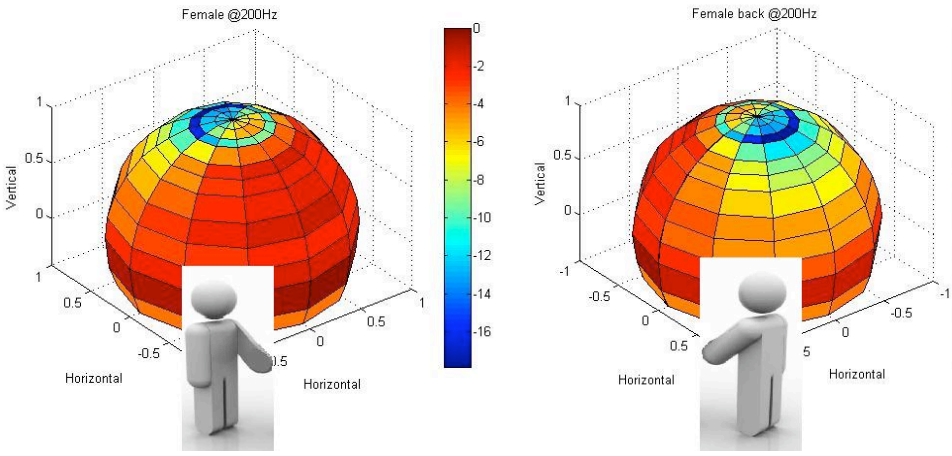


Figure 14: *An example of Volunteer's direction.*

3.2.1. Male

Figure 15 shows the directivity of the male voices similar to that shown in Figure 1. However, it is not as obvious as 3D model.

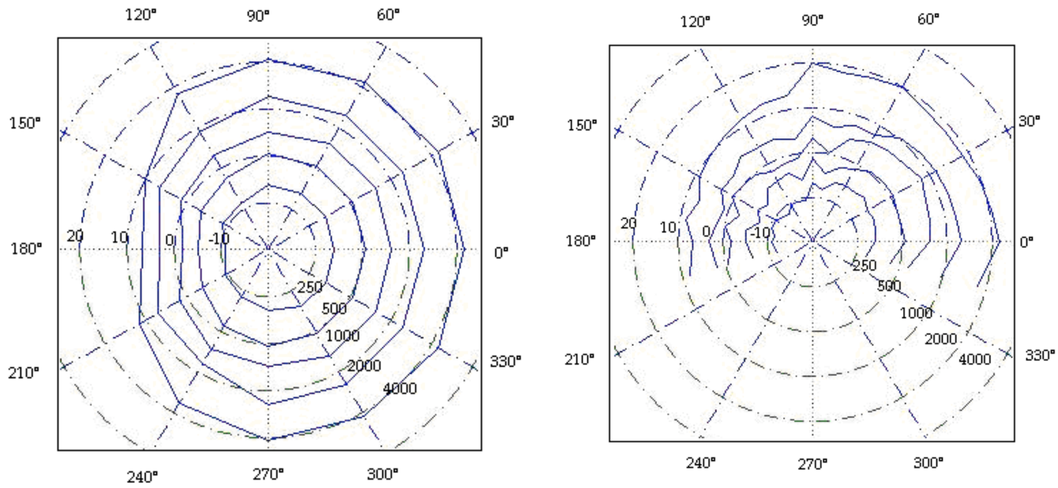


Figure 15: *Polar plot for speech directivity of male voices in the horizontal plane and the median plane.*

Figure 17 shows the figures of male directivity at different 1/3 octave band in 3D interpolated from related microphone positions. The frequency range is from 100 Hz to 10 kHz in 1/3 octave band. The left side is the front view and right side is back view. The relative direction is as Figure 14. The colorbar in the middle shows the relative level. The colorbar shows the relative level related to the right ahead direction of sound source. All the colorbars start at 0 and end with negative value.

It's clear that the closer the position is to the front or the horizontal level of mouth (the second line close to the bottom), the warmer the color is (close to the color at the top of the colorbar). Conversely, the color is colder (close to the color at the bottom of the colorbar). With the increase of the frequency, the contrast between the front and back is more obvious. especially, the mesh which shows the direct sound direction of the mouth becomes increasingly prominent with the growth of frequency.

However, there are some particular cases. One is at 400 Hz. Form 45°to 75°the level in the front side is lower than the back side. That case also could be found at 315 Hz and 500 Hz, around 60°. There is a reason to believe that the directivity of men voice around 400 Hz is not as strong as other frequencies in the front of the upper. Another particular case is in 75°at all frequencies. Obviously, the lowest level of almost all the figures is around 75° not in 90°.

The Figure 1t 6300 Hz should be paid more attention. Compared with nearby figures, the color at 6300 Hz is too "cold", even on the front side of sound source. It shows the sound at that frequency is much better directivity and less radiation.

3.2.2. Female

Figure 16 shows the directivity of the female voices similar to that shown in Figure 1.

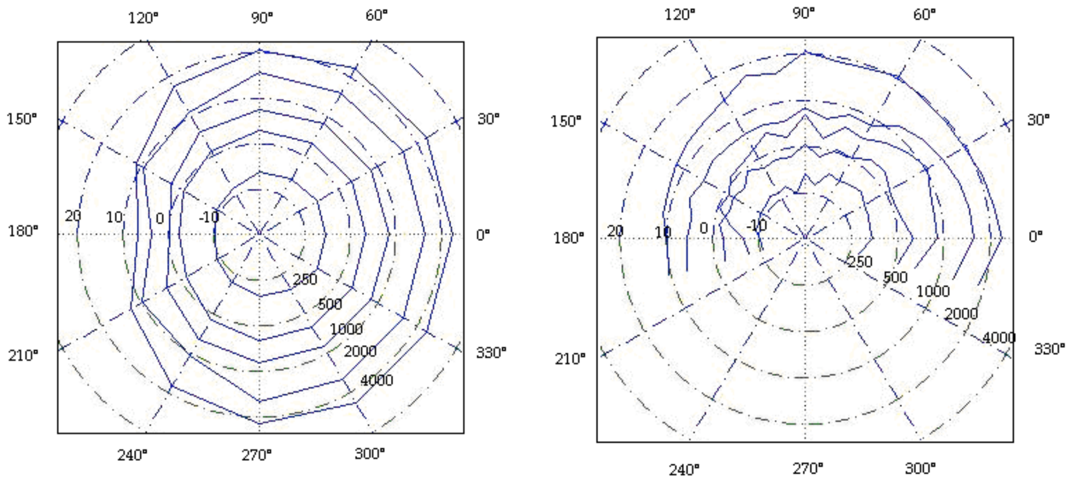


Figure 16: *Polar plot for speech directivity of female voices in the horizontal plane and the median plane.*

The female directivity is in Figure 18. Since the frequency of female voice is much higher than the male, the 3D model is without the low frequencies and the range is from 200 Hz to 10 kHz.

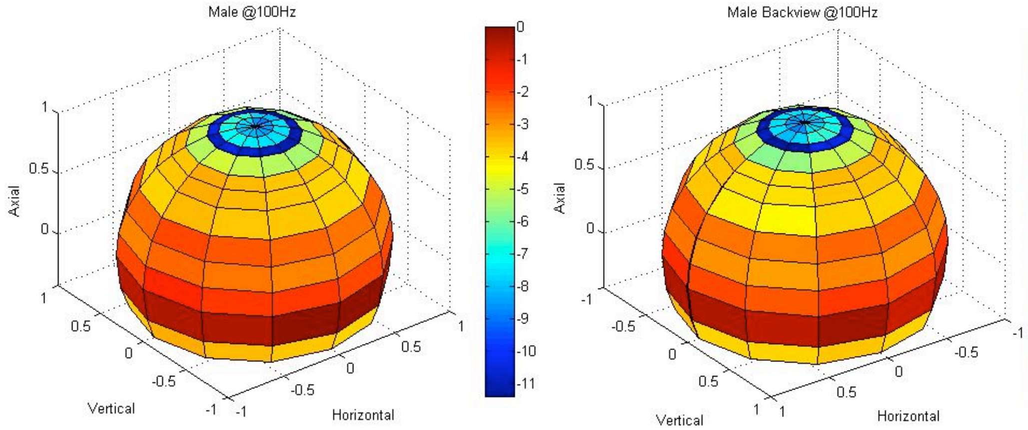
Similar as the figures of male, the closer the position is to the front or the horizontal level of mouth, the warmer the color is. With the increase of the frequency, the contrast between the front and back and the mesh of the direct sound from mouth direction is more obvious.

Apparently, particular cases are similar. On one hand, the phenomenon that the level on the front view is lower than the back is also around 400 Hz, but more obvious than male’s. On the other hand, the lowest level is still in 75°, which means maybe the vehicle-related condition is really like that, or probably due to the equipment problem.

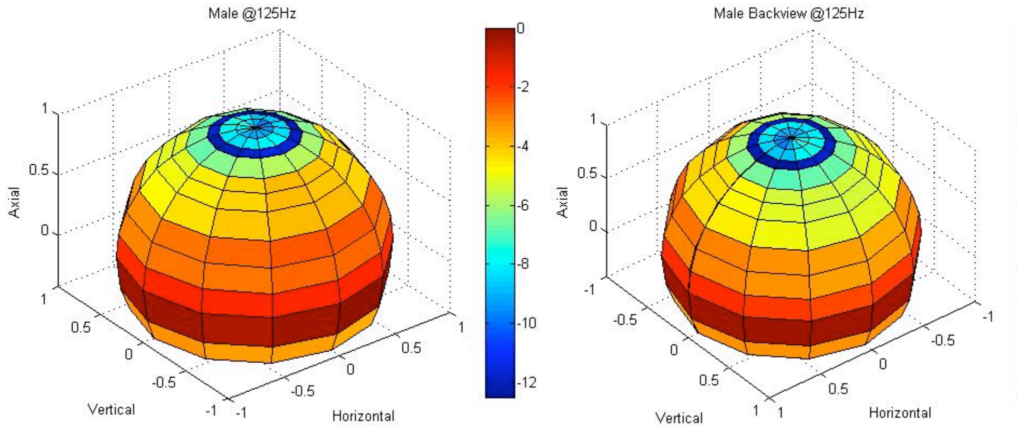
Compared with the figures of male, it is clear that the female voice is much better directivity and less radiation. This may be because the woman's voice is more acute. Compared with low male voice, the acute female voice shows better directivity.

This kind of phenomenon becomes increasingly prominent with the growth of frequency. At high frequencies, especially for higher than 1 kHz, the back view of female has more cold color area than male, which means related to the right ahead direction of sound source, fewer sound waves are radiated to the back side. At the horizontal level of mouth (the second line close to the bottom), the decrease of female level is much faster than the male.

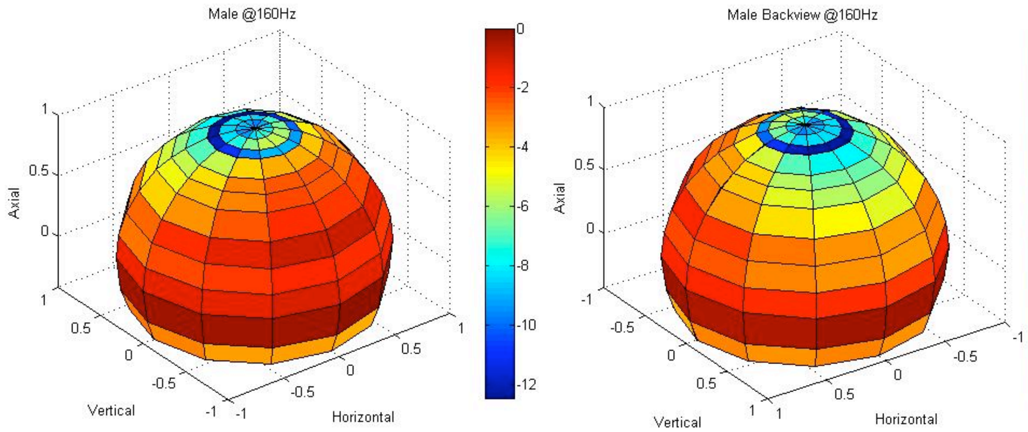
Figure 17: *Male directivity model. The range is from 100 Hz to 10 kHz in 1/3 octave band. Left one is the front view and right one is back view. The colorbar in the middle shows the relative level.*



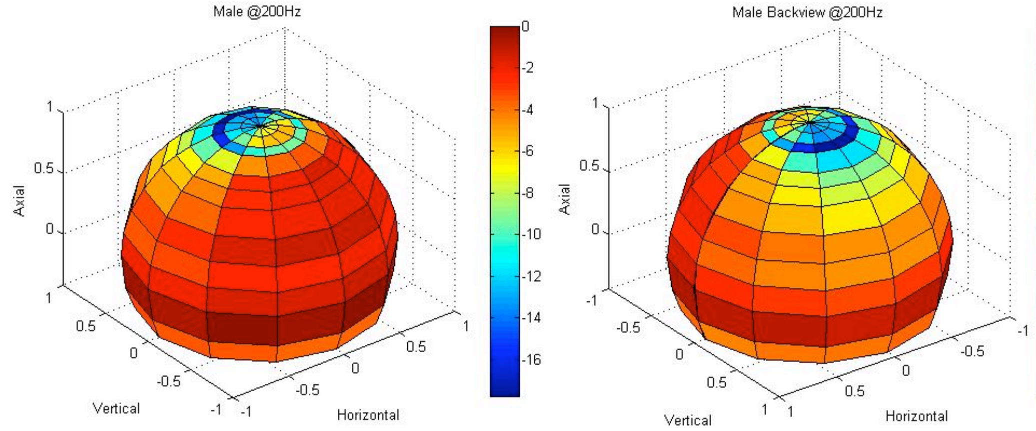
Male, at 100 Hz.



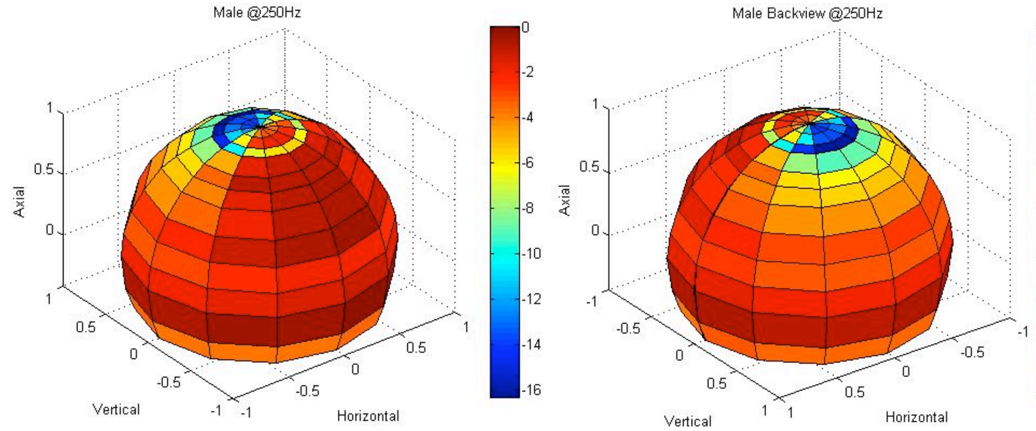
Male, at 125 Hz.



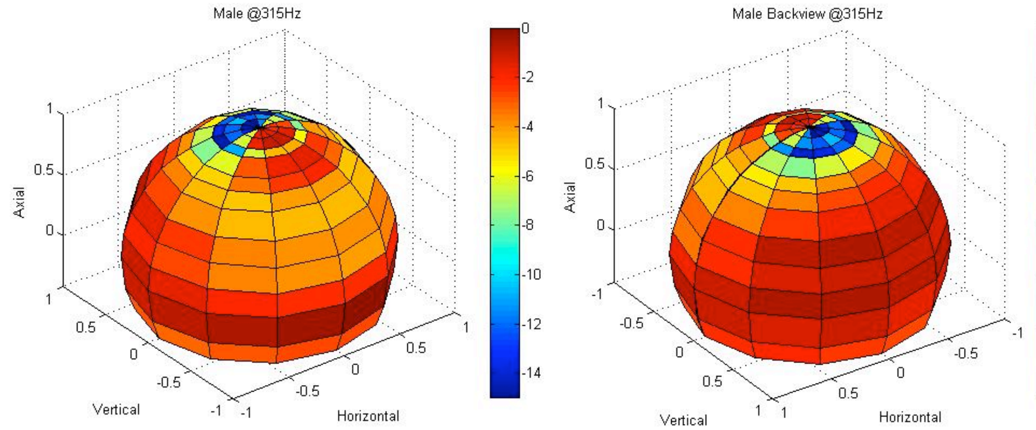
Male, at 160 Hz.



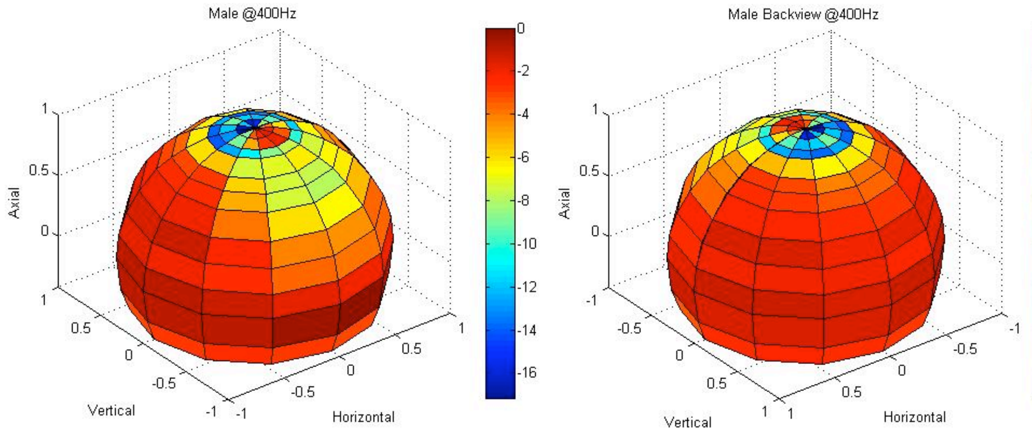
Male, at 200 Hz.



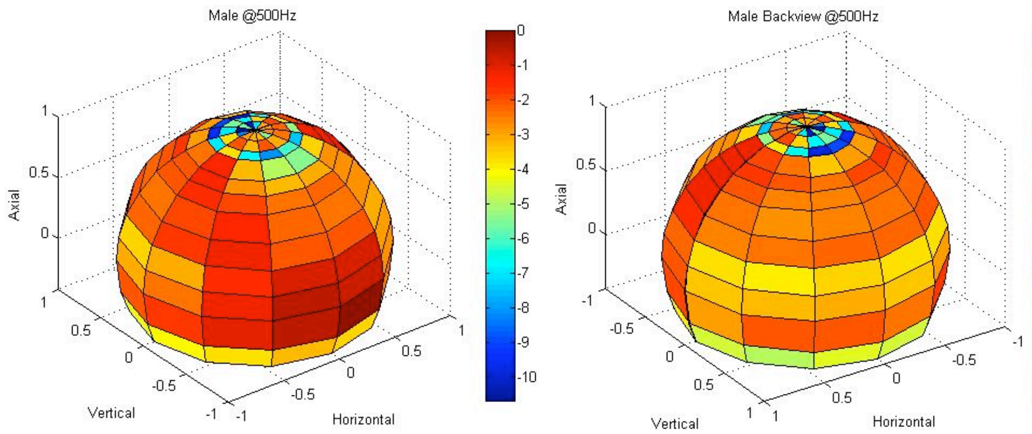
Male, at 250 Hz.



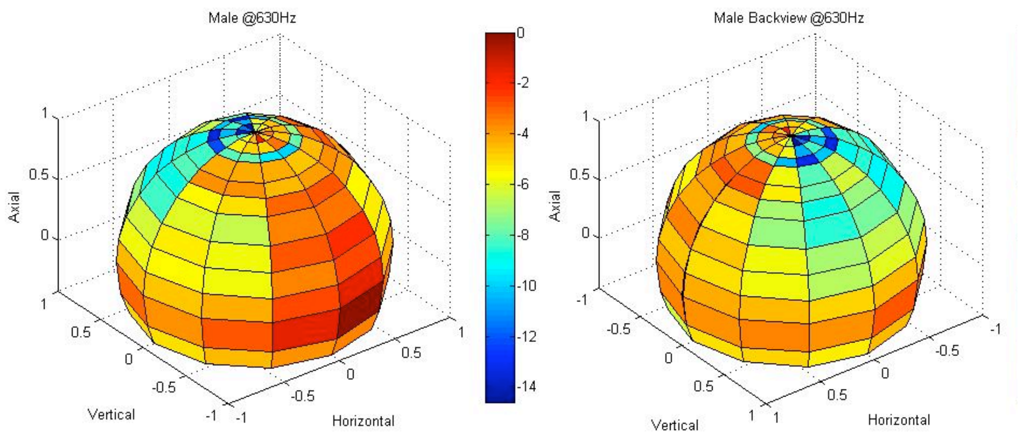
Male, at 315 Hz.



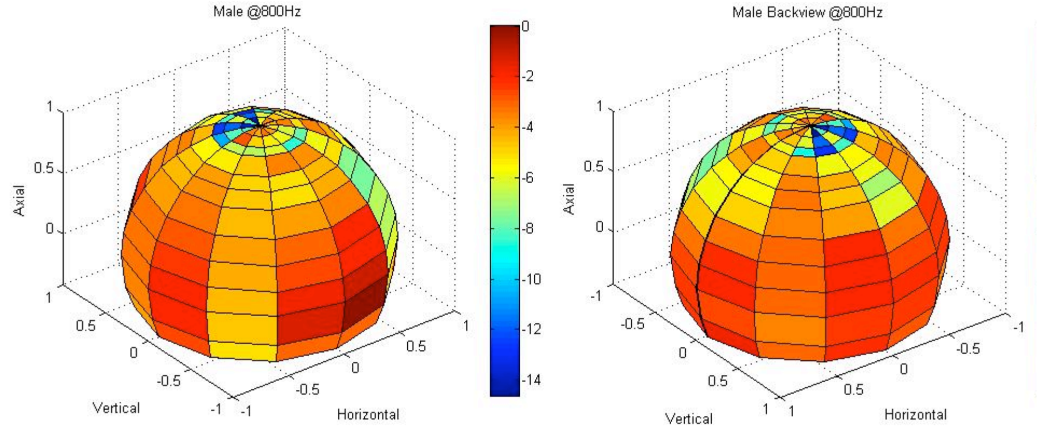
Male, at 400 Hz.



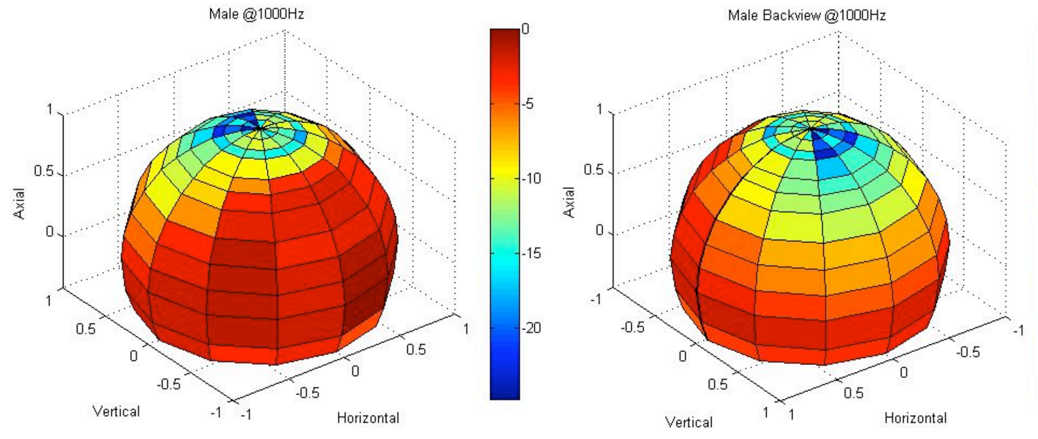
Male, at 500 Hz.



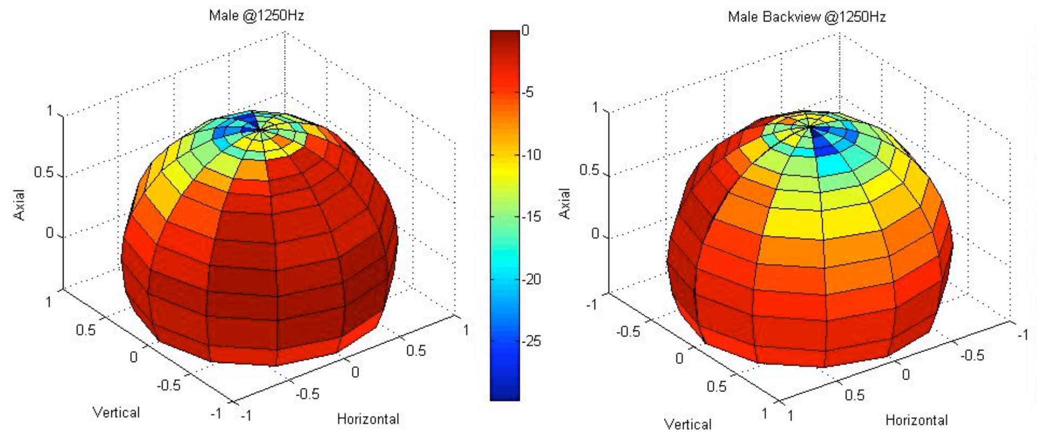
Male, at 630 Hz.



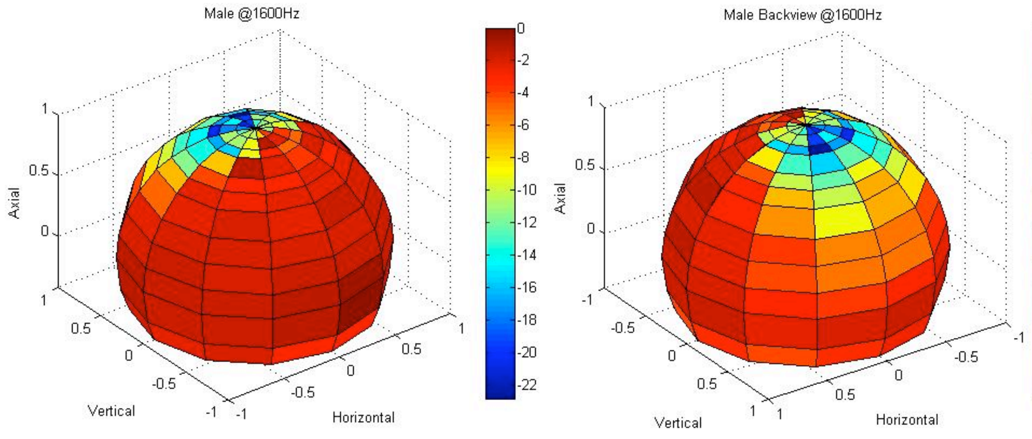
Male, at 800 Hz.



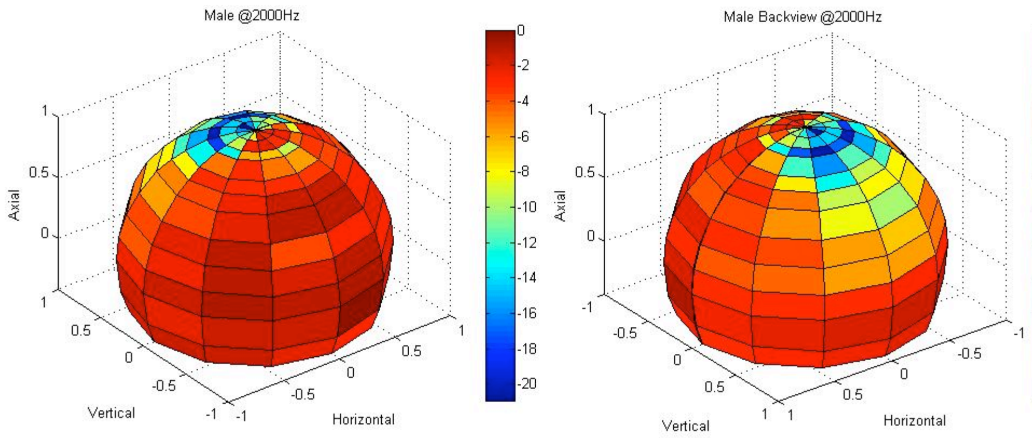
Male, at 1 kHz.



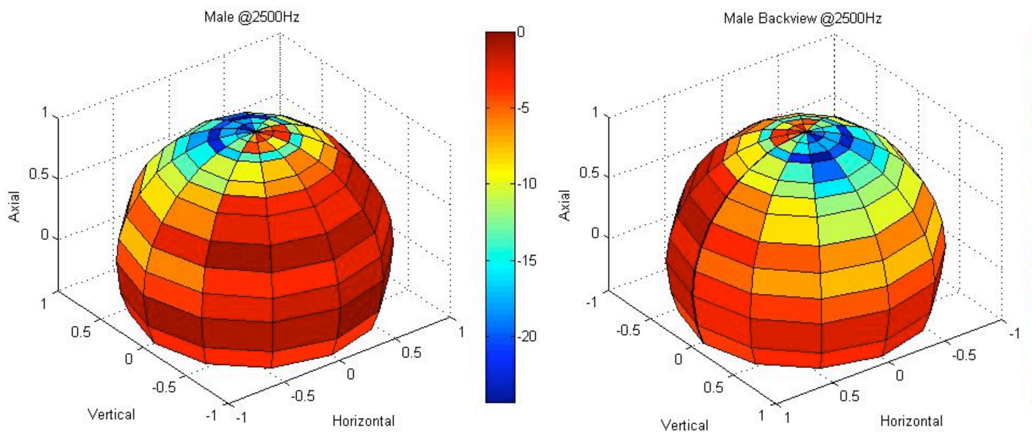
Male, at 1.25 kHz.



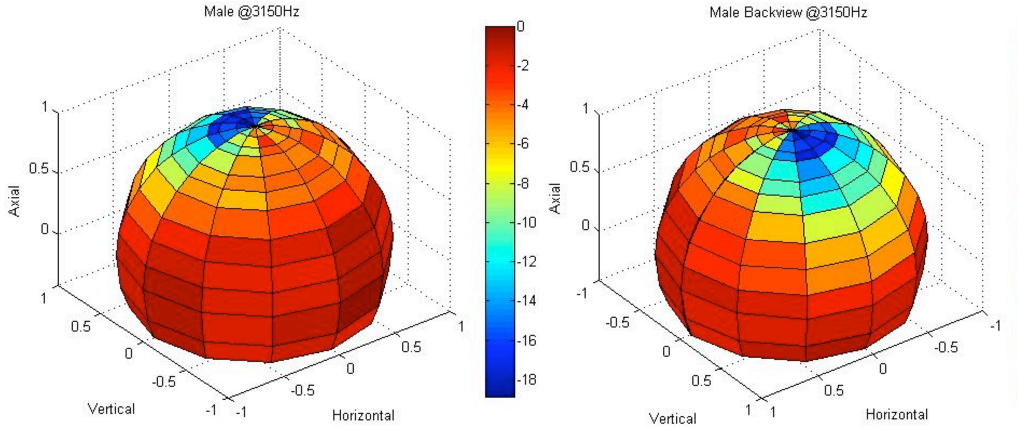
Male, at 1.6 kHz.



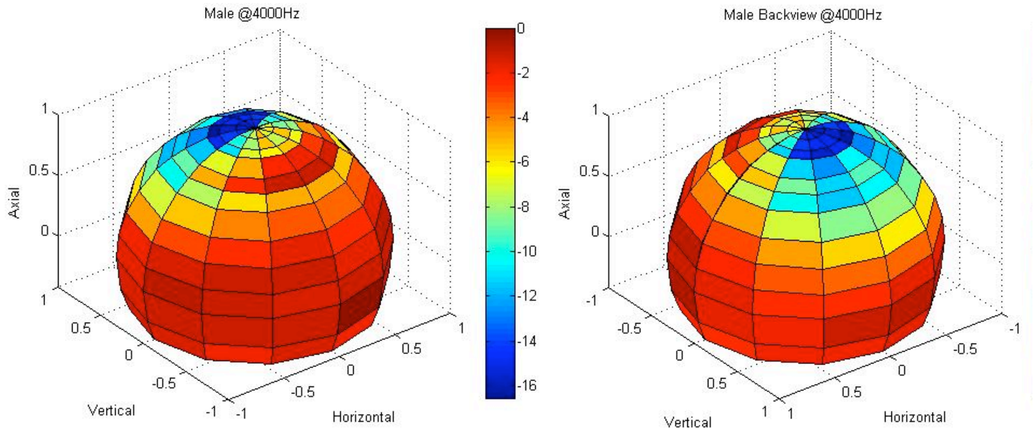
Male, at 2 kHz.



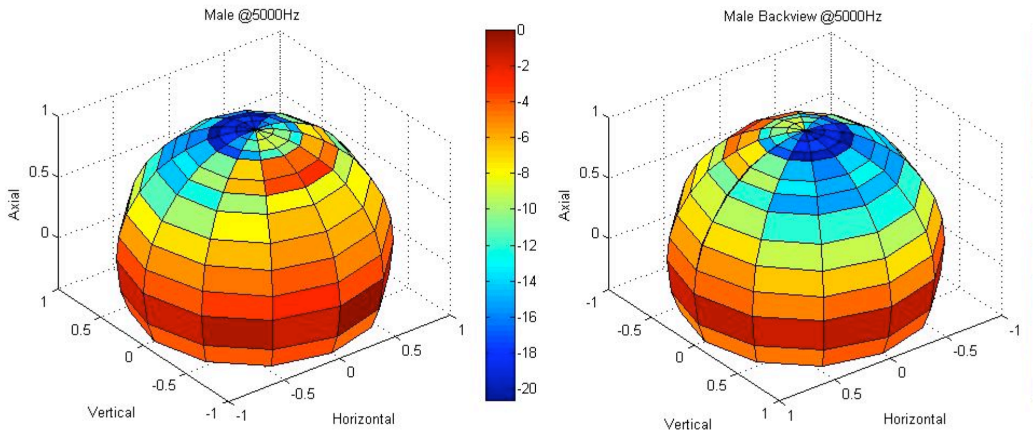
Male, at 2.5 kHz.



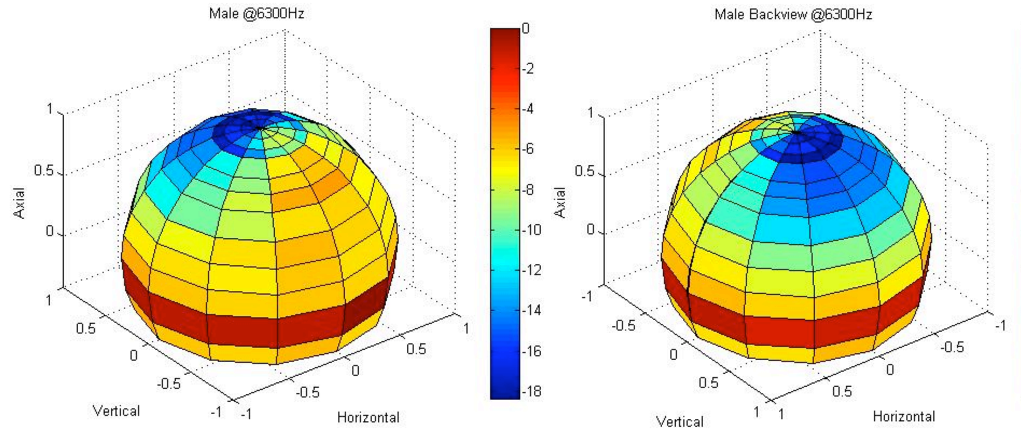
Male, at 3.15 kHz.



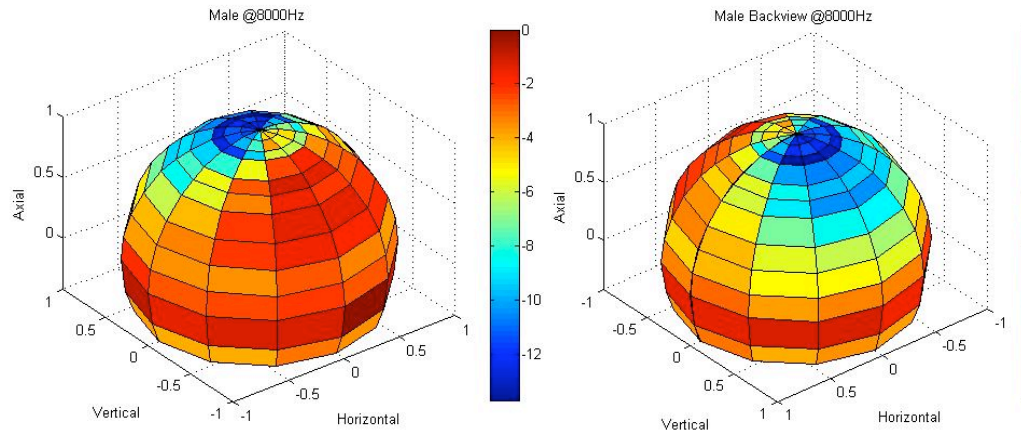
Male, at 4 kHz.



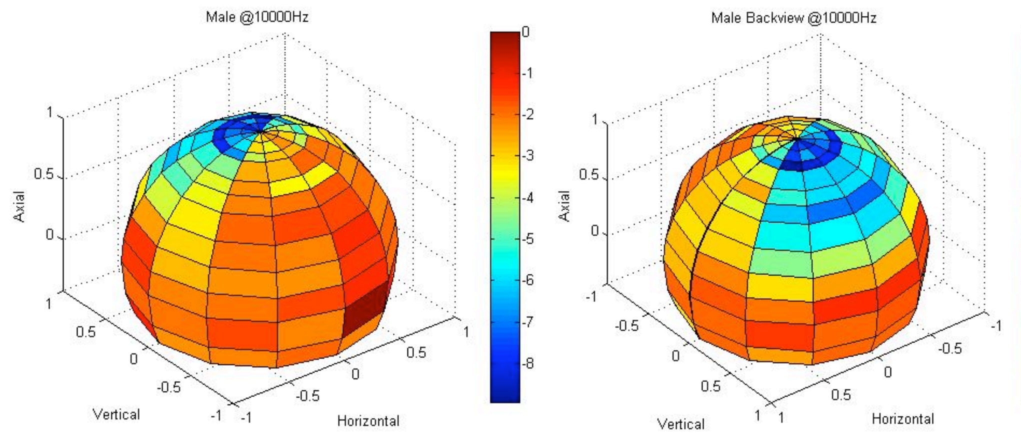
Male, at 5 kHz.



Male, at 6.3 kHz.

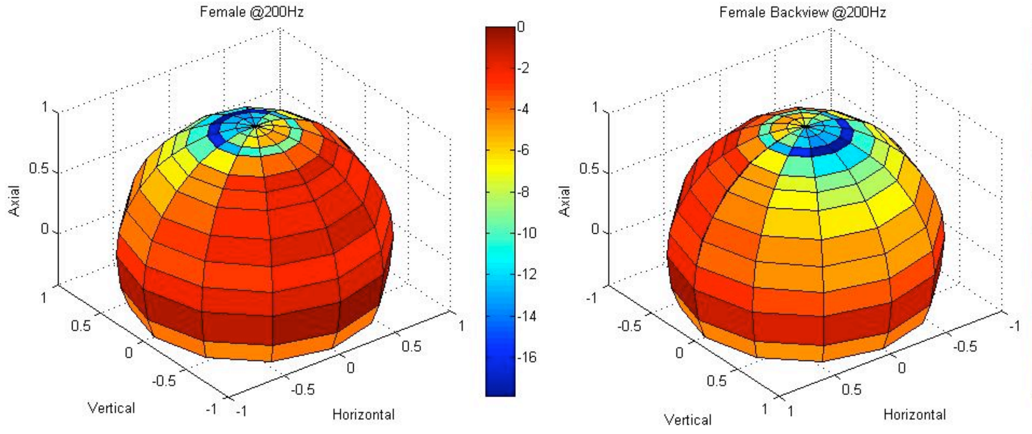


Male, at 8 kHz.

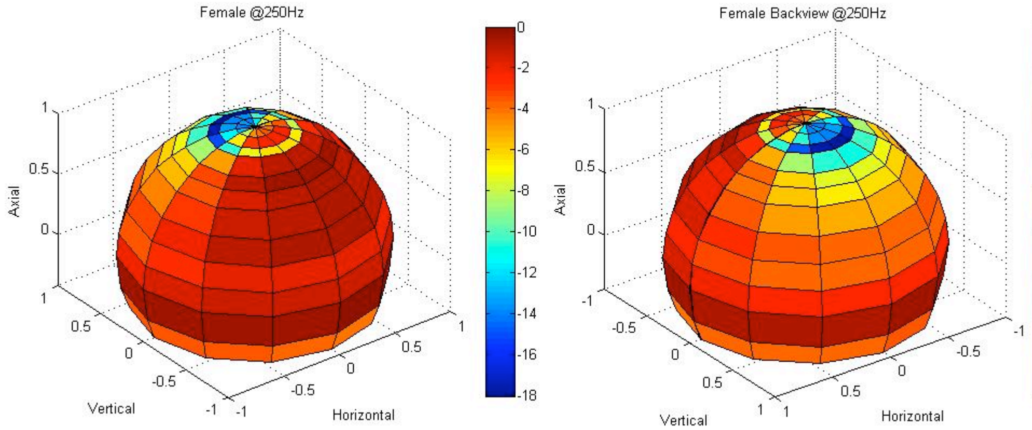


Male, at 10 kHz.

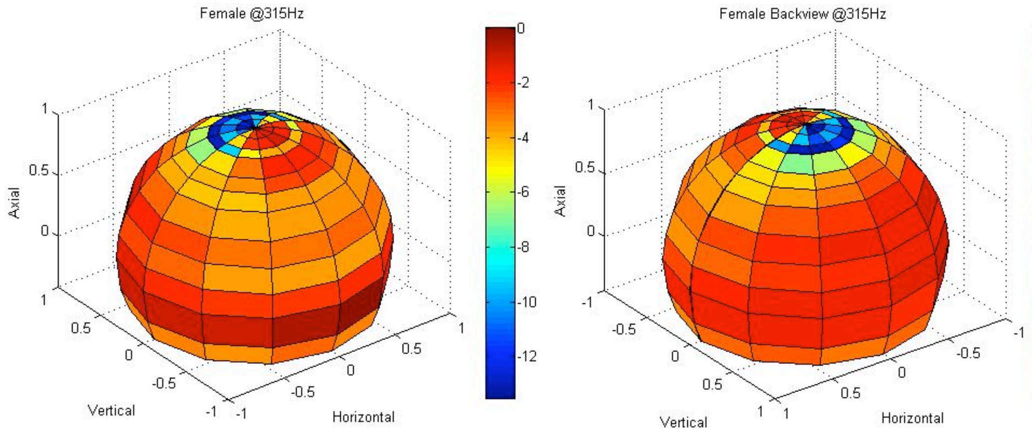
Figure 18: *Female directivity model. The range is from 200 Hz to 10 kHz in 1/3 octave band. Left one is the front view and right one is back view. The colorbar in the middle shows the relative level.*



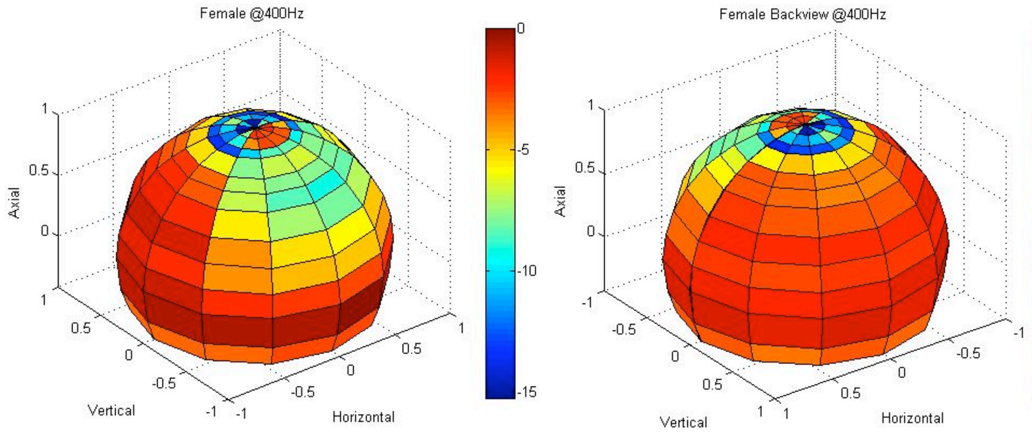
Female, at 200 Hz.



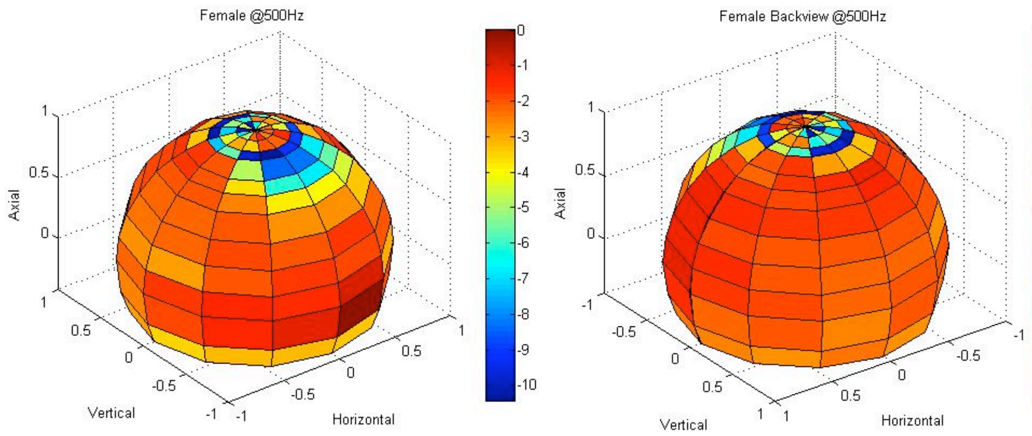
Female, at 250 Hz.



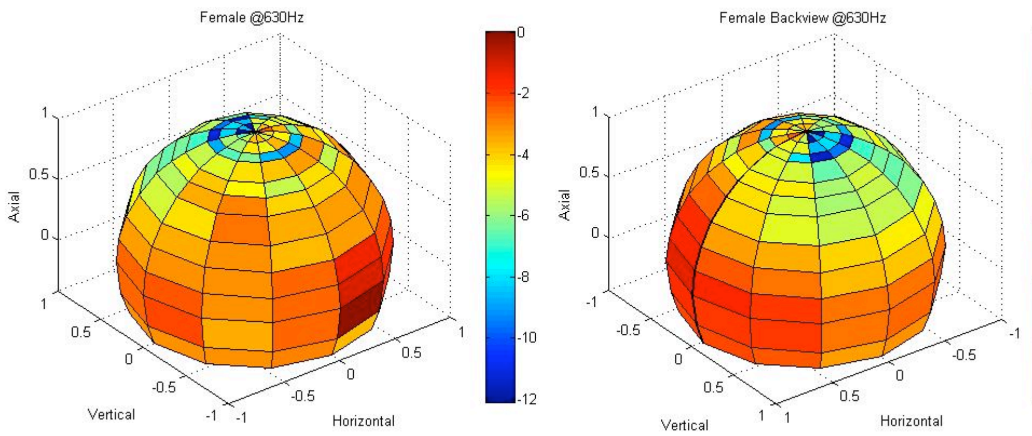
Female, at 315 Hz.



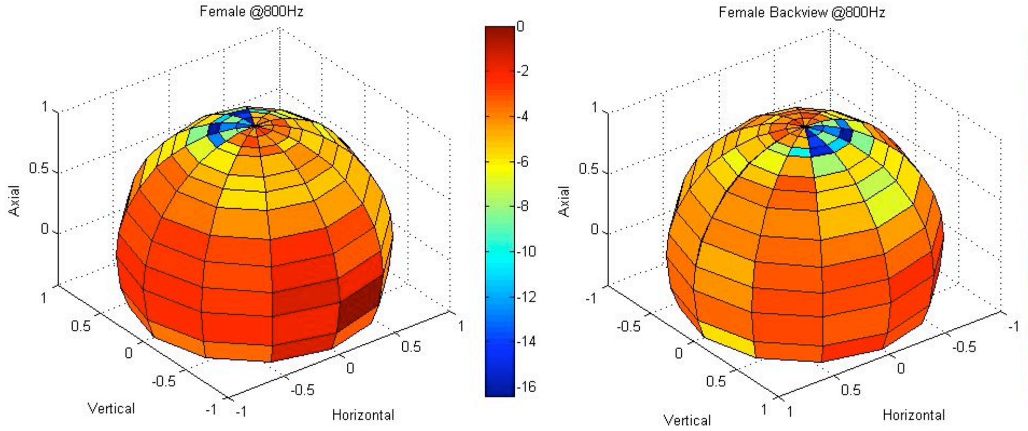
Female, at 400 Hz.



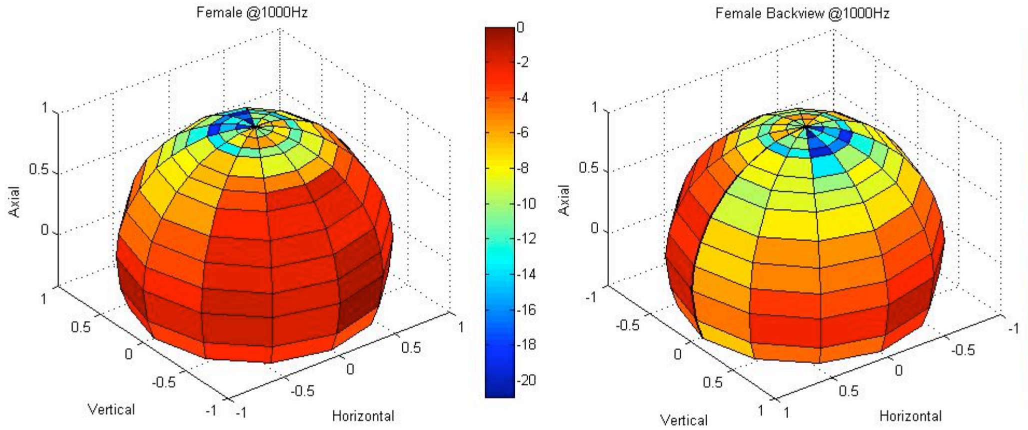
Female, at 500 Hz.



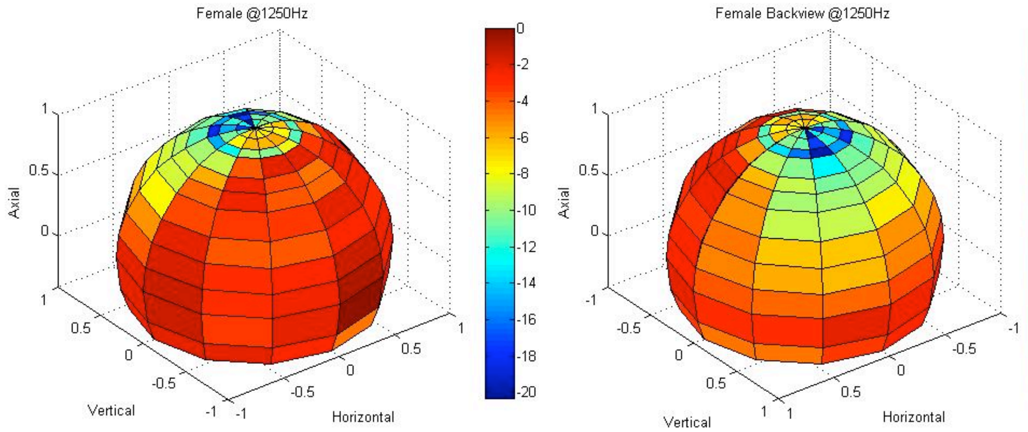
Female, at 630 Hz.



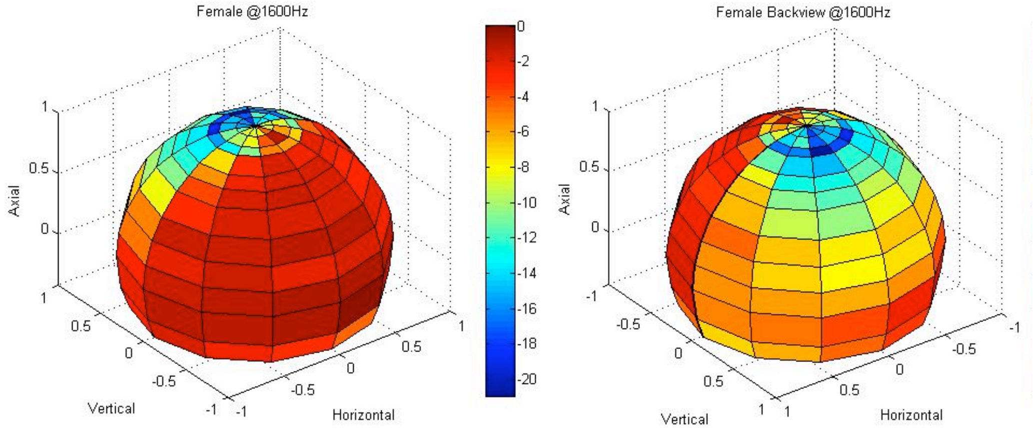
Female, at 800 Hz.



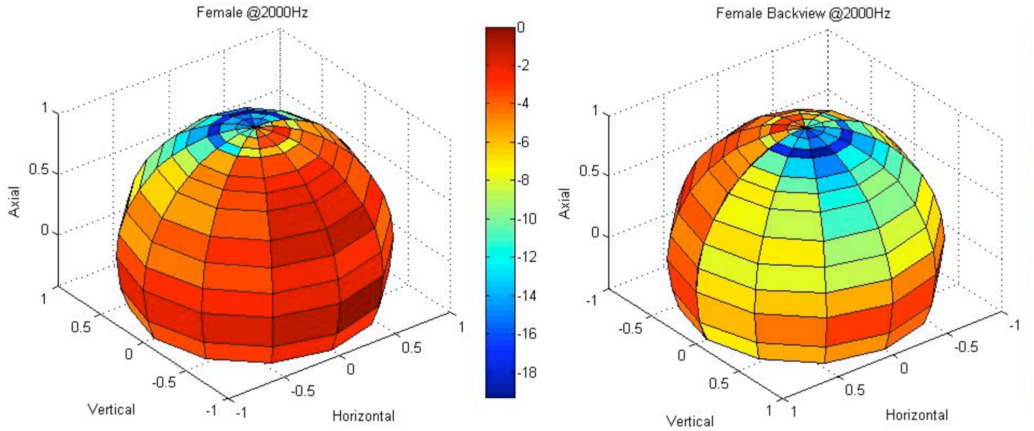
Female, at 1 kHz.



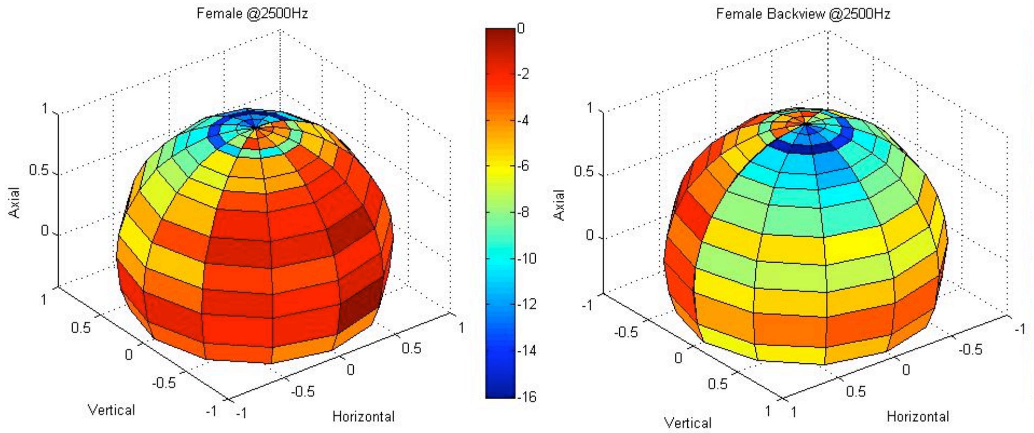
Female, at 1.25 kHz.



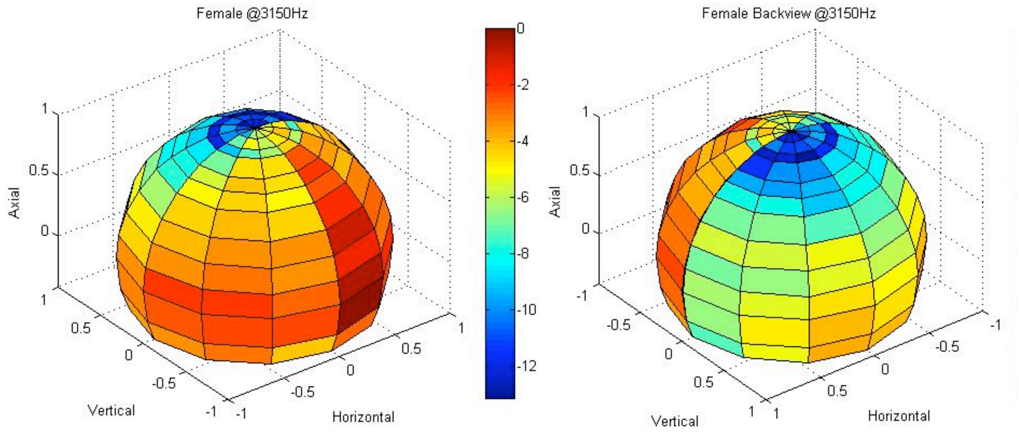
Female, at 1.6 kHz.



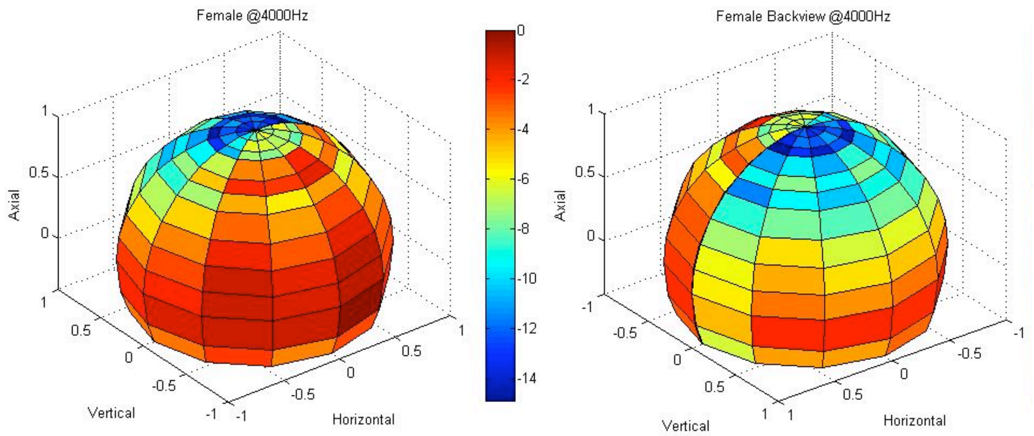
Female, at 2 kHz.



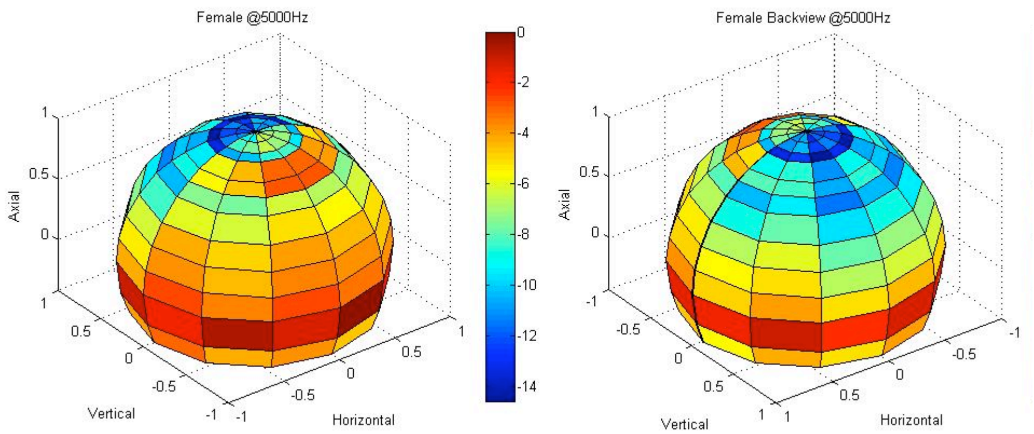
Female, at 2.5 kHz.



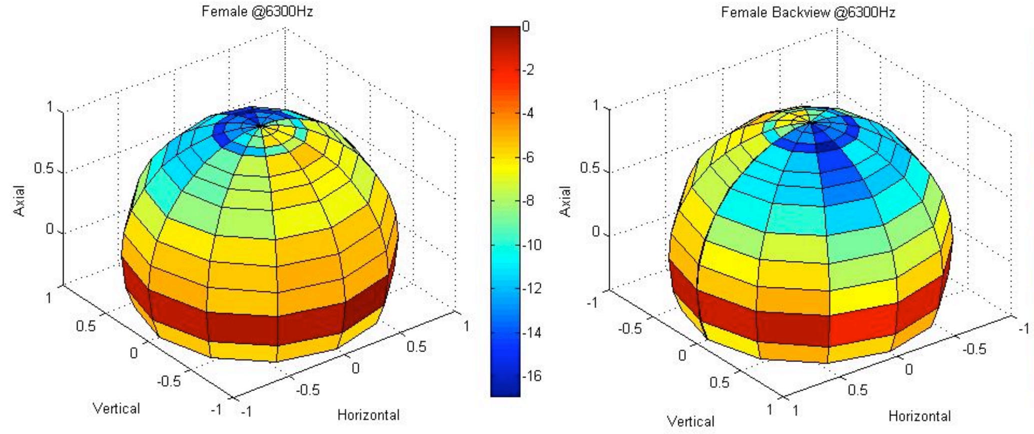
Female, at 3.15 kHz.



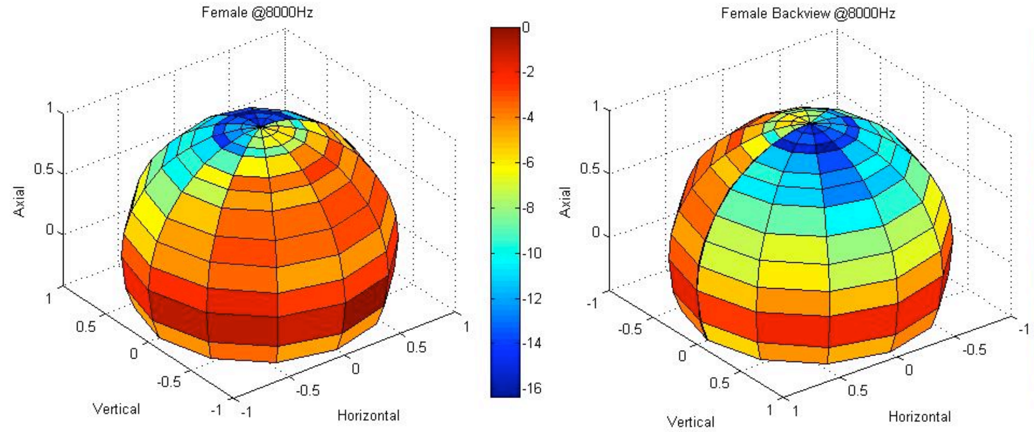
Female, at 4 kHz.



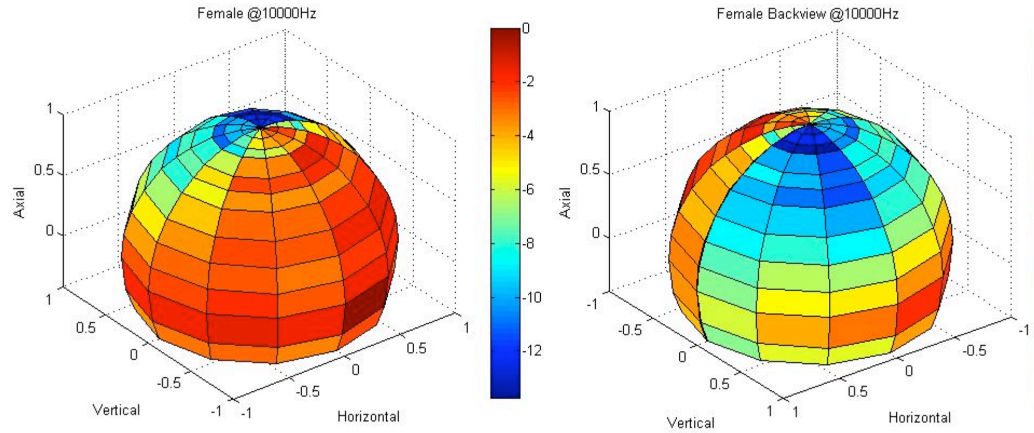
Female, at 5 kHz.



Female, at 6.3 kHz.



Female, at 8 kHz.



Female, at 10 kHz.

3.3. Contrast Test

The contrast test was retested with same setup and equipment but only changed the testing seat. Two vehicle seats as vehicle-related condition and one wooden box as free-field conduction were used. A girl (22 years old, Chinese) was joined in as a volunteer. The measurement was in one day, reading the same text with previous measurements. Figure 19 is some examples of those directivities in 1/3 octave band. Three different frequencies, (200 Hz, 500 Hz and 1 kHz) are chosen as examples to do a simple contrast.

3.3.1. Vehicle-Related Conditions

Two different vehicle seats were used as the vehicle-related condition (see Figure 1 and 2, Chapter 2). The first seat which is called chair one is the same one with the previous measurements. Compared with the 3D directivity model of female in Chapter 3.2.2 (Figure 18), the model of chair one is more random and has more variation. Because Figure 18 is the average directivity of seven female volunteers, some obvious disparities will be reduced and averaged.

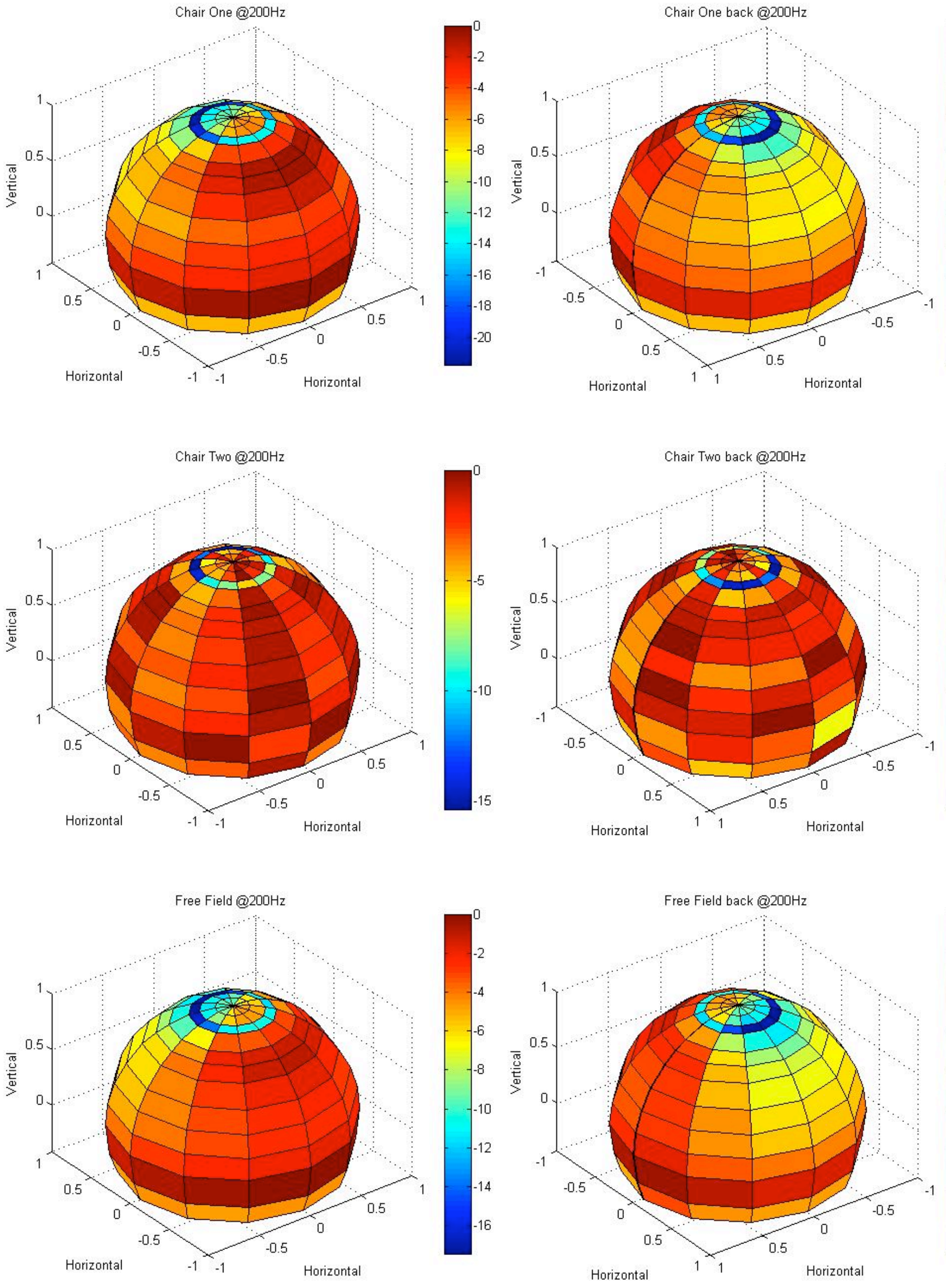
Compared with chair one, chair two has more influence on the sound directivity. The 3D directivity model is more random and has less regularity. One interesting thing is there is not as obvious reduce as chair one in the rear. From those three examples it looks like chair two has better sound spread to the backside than chair one. That may be because of the chair itself, or the shake or wobble of the volunteer during the measurement.

3.3.2. Free-Field Condition

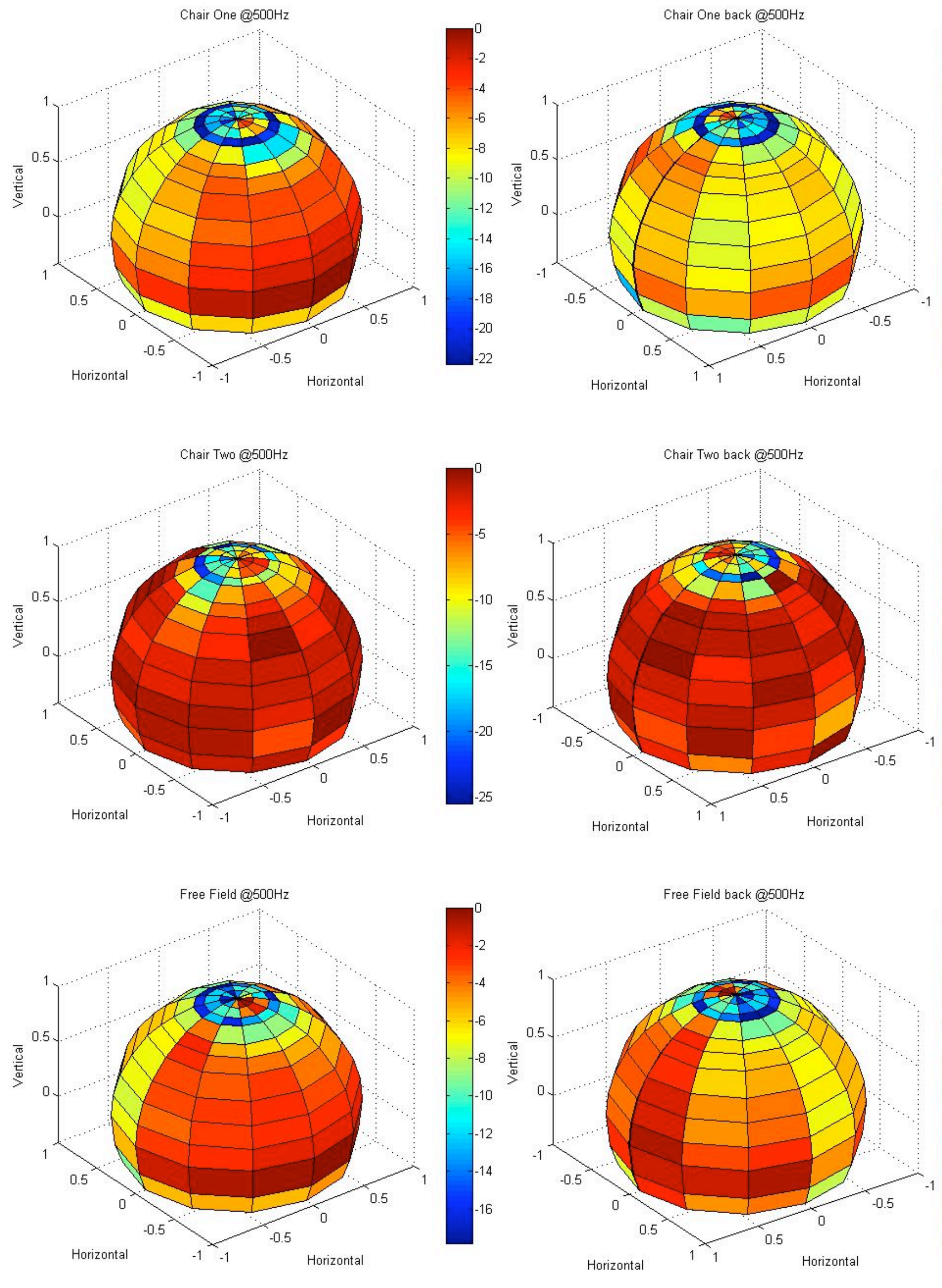
A wooden box was used to measure the free-field condition. The height of wooden box was as same as the vehicle seats but there was no backrest (see Figure 5, Chapter 2), which decreased the sound reflection to the front side and increased the sound radiation to the back side. Although the backrest was gone, the volunteer was told not to move her head to keep the head restraints.

Compared with the vehicle-related conditions, the free-field condition shows better directivity. Without the backrest, the reflection decreases and the related SPL level of front is not as high as chair one. Assuming the right ahead direction of sound source is 0 degree in horizontal direction, it is distinct that a scarlet mesh as the top color in the colorbar is in that direction. With the changing of the angle to the back side, the related level decreases more fleetly than the vehicle-related condition.

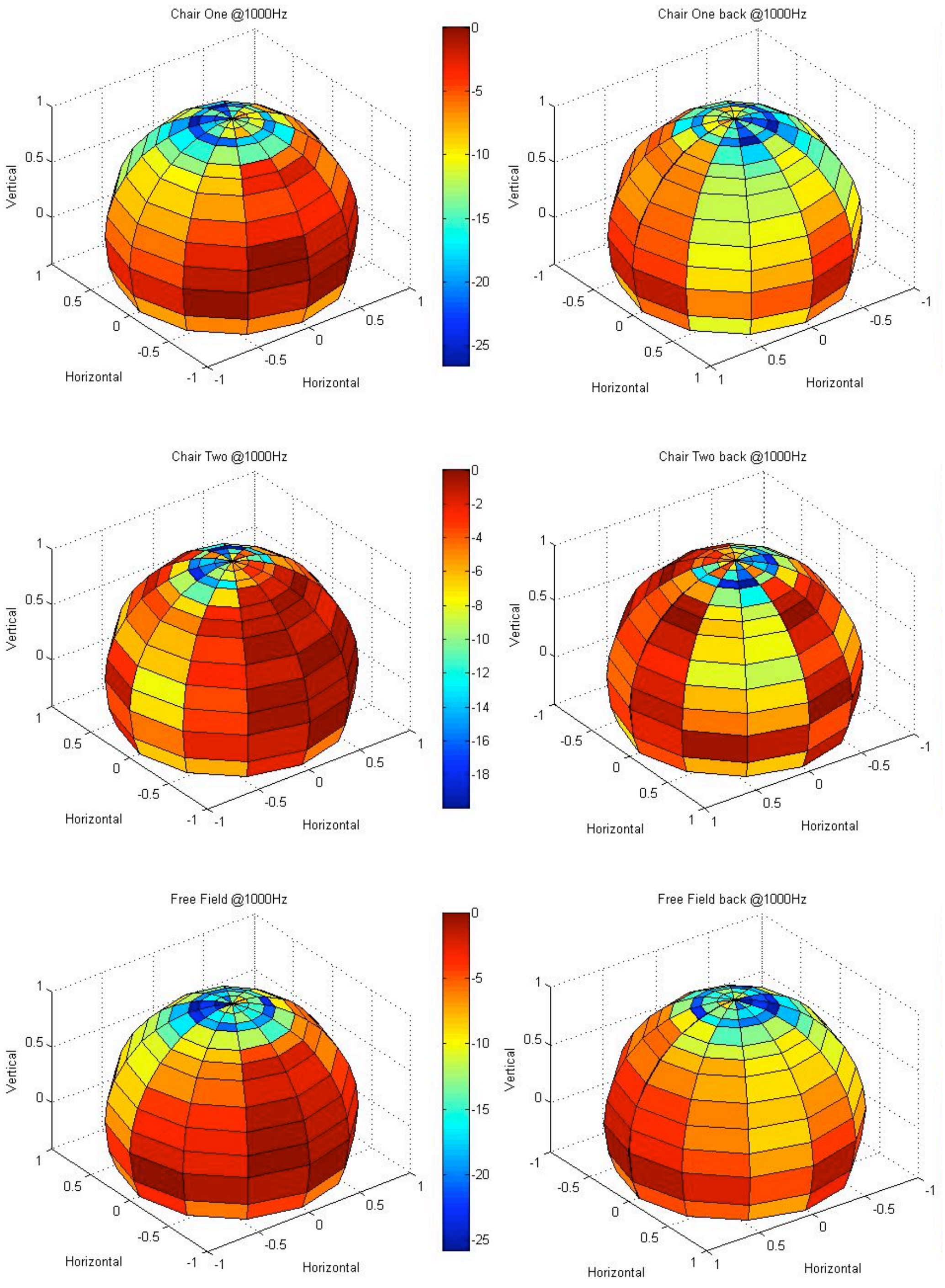
Figure 19: *3D directivity model of the contrast test, including two vehicle chairs and one free field. Left one is the front view and right one is back view. The colorbar in the middle shows the relative level.*



Vehicle-Related Condition and Free-Field Condition at 200 Hz



Vehicle-Related Condition and Free-Field Condition at 500 Hz



Vehicle-Related Condition and Free-Field Condition at 1 kHz

4. Conclusions and summary

The objective of this research project is to establish a statement about the directivity of human speakers sitting in car seats with head restraints. It includes two main parts:

1. To establish the spatial properties of the human voice in vehicle seats with head restraints.

2. To ascertain the difference between the vehicle-related condition and the free-field condition (with or without the vehicle seat).

A simple setup to simulate the vehicle interior is built inside an anechoic chamber. The environment is regarded as anechoic at frequencies above 75 Hz, and the free distance from the volunteer to the wall of room is approximately 4 m. The most influential thing is the car seat with head restraints. A kind of hemisphere model with 3D spatial dimensions is built to show the directivity around human speakers in unit distance.

Figure 17 and 14 (Chapter 3) show the directivity around human speakers with head restraints in unit distance. The variation tendency of the figures is consistent with the theory, and considering the other researches which used the artificial torso as the sound source, undoubtedly those figures are closer to the real condition.

Through the figures, it could conclude that when the head is restrained, the vehicle seats do not have an obvious influence on the left and right sides. However, there is very clear barrier effect to the rear. That means the communication between passengers in the same row will be clearer than between different rows. The backrest of chair is the main reason. The backrest has obvious effect on the backward radiation voice but no clear resonance radiation. Also the backrest has the reflection effect. However, compared with the free-field condition from the contrast test, the reflection is not obvious. Because when a man or woman sits in the vehicle seat, his or her body will cover the vast majority of the car seat. The bodily form and the material of clothes will be important factors which will influence the reflected sound to the front.

The vehicle seats have more influence on the directivity at high frequencies. To a certain extent, the female voice will get more influence than the male voice because the female voice is more acute has higher frequency content than the male voice.

The Figure 17 (Chapter 3) shows the contrast test results. By the comparison between different car seats and free field, the backrest is determined as the decisive factor to the back side. Also the material and parameters of the vehicle seats will have some influence.

Since this research is to emphasize the spatial dimensions of human speakers in vehicle seats with head restraints, the volunteers were told to keep their heads touching the backrest. Because of the angle of vehicle seat itself, the lowest related level is not at vertical right above (90°) but appear with some angle (around 75°). The head fixed device was not used during this measurement, so small movement like head shake or wobble of the volunteers is unavoidable. For the average value the small movement may have not much influence, however, for once measurement like contrast test that movement will be a big problem.

About the error analysis, the volunteers are the main source. The head shake, the material of clothes, even the bodily form will become the reason to influence the directivity. Expressly, because the directivities in different directions around human speaker were measured one by one during this research, it cost approximately 45 minutes to 1 hour, the volunteers are not similar as the loudspeaker, they would be tired, the throat will be scratchy and unavoidably the voice would change. Although the repeated measurements and average values are used to decrease the influence, it always exists.

One solution is to measure the directivity in all directions at the same time. In fact the predicted plan of this research was that. The predicted setup could be found in Figure 2, which is a model as the fullerene, or called footballene or C60. The fullerene model has 60 nodes and could record the 60 different directions. However, due to the VXI system and some other reasons, finally the bracket arc was used.

For further research, if the acquisition and analytical system could carry more channels, the fullerene model will be a better choice. Recording the directivity in all directions at the same time will decrease the possible error and become easier to analyze. Also more microphone positions will show the directivity more clearly. A head holder should be used to prevent the involuntary head movement, and then the error at some frequencies will be avoided, especially at high frequency. If possible, a test suit should be prepared, which keeps the same reflected index as far as possible.

Appendix Matlab Code

The Matlab code is using to get the SPL from the recordings.

Code one

```
for num=1:408
    number=num2str(num);
    path='D:\...\';
    file1='P_';
    file2=number;
    file3='.wav';
    filename=strcat(path,file1,file2,file3);
    a=GETSPL(filename);
    save(number,'a');
end

function[Lp]=GETSPL(x)
```

Code two

```
% ***********************************************************************
a=wavread(x);
Span=20000;
%    Span     - The upper frequency limit as specified. It is
%                  the maximum frequency insignificantly affected by the
%                  anti-aliasing lowpass filter. The sampling frequency is
%                  2.56 times this frequency (i.e. fs=2,56*Span) (scalar)
P0=20e-6;
%    P0   - the reference sound pressure in air, unit: Pa

aa=a/P0;

% ***********************************************************************
% Initial setup
% ***********************************************************************

Na=size(aa,1)/2.56+1;             %compute protected frequency components
df=Span/(Na-1);        %Frequency resolution
[Nf,M]=size(aa);       %Nf - frequency components
                        %M - the number of active channels
t=(1:length(aa))/Nf/df;             %Building time axis
f=(0:Nf-1)*df;            %Building frequency axis
```

```
fa=f(1:Na);

% ************************************************************************
% Windowing and calibration,
% ************************************************************************

wind=hanning(Nf);                        % Hanning window
w_fact=sqrt(sum(wind.^2)/Nf);      % Compenstion factor for the energy lost when applying
the window
a1=aa.*wind(:,ones(1,M))/w_fact; % Windowing and compensation on the time signal

clear aa;
% ************************************************************************
% Fourier transform
% ************************************************************************
Sx = zeros(length(a1),M);
for i=1:M
    Sx(:,i) = fft(a1(:,i))/Nf;   % fft has to be divided by block size to get the right
                                  % scale for a double sided top-amplitude spectrum
end
clear a1 i;

% ************************************************************************
% The 1/3Oct Bands limits, center frequencies
% ************************************************************************
third_lims = [22 28 35 44 57 71 88 113 141 176 225 283 353 440 565 707 880 ...
    1130 1414 1760 2250 2825 3530 4400 5650 7070 8800 11300 14140 17600 22500 28250 35300];
third_mids = [25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 800 ...
    1000 1250 1600 2000 2500 3150 4000 5000 6300 8000 10000 12500 16000 20000    25000
31500];

firstband = find(fa(1) < third_lims , 1);    % Index of first full 1/3Oct Bands we have data for
lastband = find(fa(end) < third_lims , 1) - 2; % Index of last full 1/3Oct Bands we have data
for

% Truncate the 1/3 octave bands to include only those we have data for
third_mids = third_mids(firstband:lastband);
third_lims = third_lims(firstband:lastband+1);

Sx_oct = zeros(length(third_mids),M);          %for 1/3oct bands
f_oct=third_mids;

for ii=1:M
for nn = 1:length(third_mids)
```

```
    this_llim = third_lims(nn);
    this_ulim = third_lims(nn+1);

    this_llim_indx = find( fa > this_llim , 1);
    this_ulim_indx = find( fa > this_ulim , 1) - 1;
    mm=length(Sx(this_llim_indx:this_ulim_indx,ii));
    Sx_oct(nn,ii)=sum( abs(Sx(this_llim_indx:this_ulim_indx,ii)).^2 ) / mm;
%       Sx_oct(nn,ii)=sum( real(Sx(this_llim_indx:this_ulim_indx,ii)).^2 ) / mm;

end
clear nn this_llim this_ulim this_llim_indx this_ulim_indx mm
end
```

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