Simulation of Microphone Array for Sound Localization using Human Binaural Hearing Model

Viktor Hristov[1], Snejana Pleshkova[2] and Alexander Bekiarski[3]

Abstract – **There are proposed many microphone array structures and algorithms, which are tested and can be characterized with advantages and disadvantages in some concrete applications. Here in this article is proposed to apply and test a combination of a simple two microphone array structure and algorithm for modelling and simulating binaural human perception. The most important thing in this proposal is the simulation of human ears with two microphones and to choose their type to have the beam forming diagram of sensitivity similar each other. Also in simulation algorithm is proposed to apply the elements of human binaural hearing to as much as is possible to known the complex and not fully understand the processes in the human brain on auditory perception and in particular human perception of the direction from which sound waves arrive. The results from simulations are presented to show the characteristics of sound source localization (SSL) achieved with proposition to apply human binaural hearing model for the purposes of sound direction of arrival and localization.**

Keywords – **Human binaural hearing model, Sound source localization (SSL), Microphone arrays.**

I. Introduction

Investigation of the human hearing system and attempts for its simulation are one the most popular themes nowadays. If the sound signals are received from a person it is easy to determinate the proper direction only with the help of the human ears and human brain. But if the human is replaced with mobile robot - combination of a proper microphone array structure and the corresponding sound localization algorithm is needed. Both working together are important to achieve the appropriate accuracy of direction of sound arrival and the corresponding SSL. [1]

Microphone array structure and chosen algorithm working together are important to achieve the appropriate accuracy of direction of sound wave arrival and the corresponding SSL. A number of factors affect the spatial aspects of how a sound is perceived. The “Duplex theory” was the first comprehensive analysis of the physics of binaural perception [2, 3, 4, 5 and 7]. If the imaginary or artificial head as model of the human head in a mobile robot human hearing model of mobile robot microphone array had completely spherical and uniform surface, the interaural time difference (ITD) produced by the sound source that arrives from an azimuth of θ radians can be approximately described with the following equation using diffraction theory [6]:

$$\tau = \left(\frac{a}{c}\right) 2\sin\theta, \qquad (1)$$

where

τ is the interaural time difference (ITD);

a - the radius of the imaginary of human head;

c = 343 m/s - the speed of sound waves in the air.

The goal of this article is to propose, apply and test a combination of a simple two microphone array structure and algorithm for modelling and simulating binaural human perception.

II. Development of SSL Geometrical Model

The geometrical model of microphone array based on the human binaural hearing is proposed in Fig.1. It is based on the human hearing aid and binaural SSL. It consists of two microphones M_1 and M_2, with the identical parameters. They are situated on the short distance d between them, with the value similar to the distance of the ears in the human head. The positions of the two microphones are shown in Fig. 1 as coordinates x_{m1}, y_{m1} and x_{m2}, y_{m2}, respectively (assuming x as horizontal, y as vertical coordinate and the origin of coordinate system in the left upper corner in Fig.1). Sound source, producing corresponding sound source signal S, (usually a speaking person or speaker producing sound waves from an audio system) is positioned on the other end of the room. It can be described with the coordinates (x_{src} and y_{src}).

Usually the number of sound waves arriving to the human ears or to each of the microphones in the microphone array is also infinite in real situations, but it is very difficult or impossible to describe the summary sound wave arriving to each of microphones in the microphone array. Therefore, in this article based on human hearing model, a simplified geometric model (Fig. 1) is proposed for sound waves propagation to each of two microphones in a microphone array, assuming the existence only of one direct and two reflected sound waves arriving to each the left and to right microphones, respectively. In Fig. 1 are shown for simplicity only the trajectories of direct and two reflected sound waves arriving from the sound source to the left standing microphone M_1 in microphone array. The corresponding trajectories lengths of the direct, two reflected sound waves arriving to each the left microphone M_1 and reflection angles are indicated in Fig. 2 in the following way:

- l_d direct sound wave trajectory length;
- $l^L = l_1^L + l_2^L$ first reflected sound wave trajectory

[1]Viktor Hristov is with the Faculty of Telecommunications at Technical University of Sofia, 8 Kl. Ohridski Blvd, Sofia 1000, Bulgaria, E-mail: hristov.viktor@gmail.com

[2]Snejana Pleshkova is with the Faculty of Telecommunications at Technical University of Sofia, 8 Kl. Ohridski Blvd, Sofia 1000, Bulgaria, E-mail: snegpl@tu-sofia.bg

[3]Alexander Bekiarski is with the Faculty of Telecommunications at Technical University of Sofia, 8 Kl. Ohridski Blvd, Sofia 1000, Bulgaria, E-mail: aabbv@tu-sofia.bg

- θ_1 reflection angle for first reflected sound wave;
- $l_2^R = l_{21}^R + l_{22}^R + l_{23}^R$ second reflected sound wave trajectory;
- θ_2 angle of the for second reflected sound wave.

The direct and two reflected sound waves with the lengths l_d, l^L and l_2^R produce the corresponding sound signals S_d, S_1 and S_2 at the position of the left microphone *M₁*, arriving from the sound source S. The explained above simplified geometric model (Fig. 1) can be considered also for the right standing microphone M_2, assuming, that it is necessary only to change the index *L* with the index *R* for the corresponding indication of trajectories lengths of the direct, two reflected sound waves arriving to each the right microphone M_2 and reflection angles.

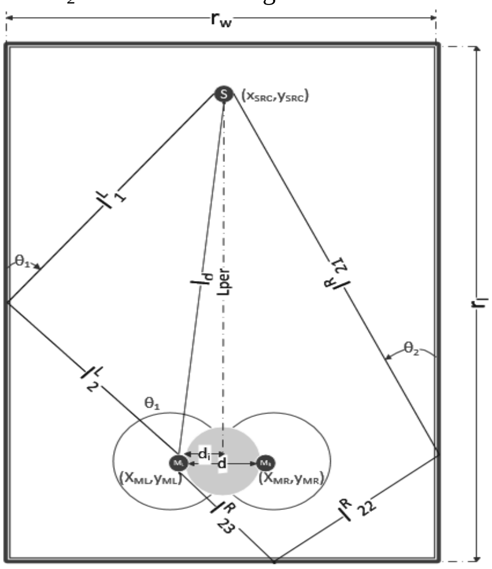

Fig.1. The proposed geometrical model of microphone array

Needed geometric parameters, shown in Fig. 1, are room dimensions (length – r_l and width r_w), distance between microphones – *d*, sound source coordinates (x_{src} and y_{src}) and positions of the two microphones (x_{m1}, y_{m1}, x_{m2} and y_{m2}). Some of these geometric parameters, for example x_{m2} and y_{m2}, can be calculated in the following way from x_{m1} and y_{m1}, if they are defined:

$$x_{m2} = x_{m1} + d\,, \tag{2}$$

$$y_{m2} = y_{m1} = y_m\,, \tag{3}$$

where

x_{m2} and x_{m1} are *x*-coordinates of two microphones M_1 and M_2;
y_{m2} and y_{m1} - *y*-coordinates of two microphones M_1 and M_2;
d - distance between them.

For the calculations of the values of the lengths l_d, l^L and l_2^R of direct and two reflected sound waves, is necessary to known the following additional geometric parameters:

$$L_{per} = |y_{m1} - y_{src}| \tag{4}$$

$$d_i = |x_{m1} - x_{src}| \tag{5}$$

where

L_{per} is the vertical component of the distance between sound source S and microphones M_1 and M_2 if it is assumed the placement of the two microphones M_1 and M_2 in a horizontal line, i.e. the equation (3) is satisfied;

d_i - the horizontal component of the distance between sound source S and microphone M_1.

Using equations (4) and (5) is possible to calculate the distance L_{di} (for i=1,2) between sound source *S* and each of microphones M_1 and M_2 as follow:

$$L_{di} = \frac{1}{2}\sqrt{d_i^{\,2} + 4L_{per}^{\;2}} \quad \text{for } i=1,2 \tag{6}$$

The angle of arrival θ_0 of the direct sound wave, for example to the microphone M_1, can be calculated using equations (4) and (6):

$$\theta_0 = \arcsin\left(\frac{L_{per}}{L_d}\right) \tag{7}$$

In the similar way, using the ordinary geometric relations is possible to calculate also each of the angles of arrival θ_n to the microphones M_1 or M_2 of reflected sound waves after n=1,2,3…. number of reflections from the walls of the room (Fig. 2). Therefore, the resultant signals S_{M1} and S_{M2}, received from microphones M_1 and M_2, respectively are the sum of corresponding signals produced by the arrived direct and reflected sound waves to each of them, after n=1,2,3…. number of reflections:

$$\begin{aligned} S_{M1} &= k_0^1 S_d^1 + k_1^1 S_1^1 + k_2^1 S_2^1 + \ldots + k_n^1 S_n^1 \\ S_{M2} &= k_0^2 S_d^2 + k_1^2 S_1^2 + k_2^2 S_2^2 + \ldots + k_n^2 S_n^2\,, \end{aligned} \tag{8}$$

where

$k_0^1, k_1^1, k_2^1, \ldots, k_n^1$ and $k_0^2, k_1^2, k_2^2, \ldots, k_n^2$ are the attenuation indexes of the corresponding parts (direct and reflected) sound signals produced by direct and reflected sound waves arrived to each of the microphones M_1 and M_2.

The attenuation indexes are important for description of the interaural intensity difference (IID) in algorithms of sound localization. Indirectly in the resultant signals S_{M1} and S_{M2} in equation (8) exist as the lengths of the reflected sound wave trajectories and corresponding time difference of arrival, as estimation of interaural time difference (ITD).

The values of the defined in equation (8) attenuation indexes $k_0^1, k_1^1, k_2^1, \ldots, k_n^1$ and $k_0^2, k_1^2, k_2^2, \ldots, k_n^2$ can be determined (in decibels) from microphone beamforming diagram of chosen type of microphones M_1 and M_2, for the corresponding values of angles of arrival θ_n to the microphones M_1 or M_2 of direct and reflected sound waves after n=0,1,2,3…. number of reflections from the walls of the room.

For simplification, in the determination of the values of the attenuation indexes $k_0^1, k_1^1, k_2^1, \ldots, k_n^1$ and $k_0^2, k_1^2, k_2^2, \ldots, k_n^2$, is assumed to ignore the losses in reflection and absorption in the walls of the room.

The resultant signals S_{M1} and S_{M2} of microphones M_1 and M_2, derived by equation (8) can be used as input audio information in development and testing the algorithms of SSL. Simulating the appropriate situations and scenarios of positions for different cases for sound waves reflections from the room walls and in room constructions and dimensions is possible.

The proposed simplified planar geometrical model of sound waves propagation from sound source to the corresponding microphones in microphone array.

III. Simulation of SSL Geometrical Model

The simulations proposed to test the proposed simplified planar geometrical model of sound waves propagation from sound source to the corresponding microphones M_1 and M_2 in microphone array presented in Fig.2.

It is based on the human binaural hearing model, arranged following the algorithm and then realized as the corresponding Matlab program.

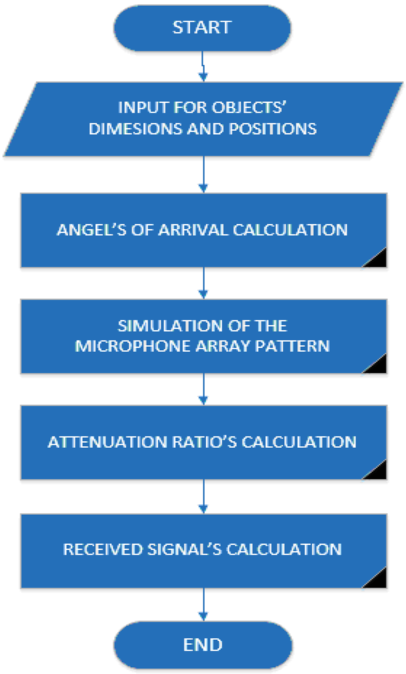

Fig.2. Simulation model is arranged as an appropriate algorithm.

In the first part of the algorithm and the corresponding Matlab program are defined some necessary input parameters for room dimensions - length – r_l and width r_w, also the distance between microphones – d, coordinates of sound source (x_{src} and y_{src}) and positions of the two microphones (x_{m1}, y_{m1}, x_{m2} and y_{m2}).
With these defined parameters the different angles of arrival are calculated depending on the situations if there is no reflection, there is one reflection or there are two reflections.

The generation of microphone beamforming diagram of microphones M_1 and M_2 is done using the existing in Matlab class of microphones as elements of basic class Microphone array. In this case is chosen to use the class "Custom Microphone Element" as the type of microphones M_1 and M_2, which allow to define their beamforming diagram of the type "cardioid", similar to the human ear hearing sensitivity and with appropriate orientation like in the human ears on human head. These conditions are involved in the following module of Matlab program and the definition of beamforming diagram of the left (microphone M_1) and the right (microphone M_2) orientation is prepared with chosen "+" or "-" in the lines 4 and 5 in the module of Matlab program:

```
h = phased.CustomMicrophoneElement;
h.PolarPatternFrequencies = [500 1000];
h.PolarPattern = mag2db([...
0.5±0.5*cosd(h.PolarPatternAngles);...
0.6±0.4*cosd(h.PolarPatternAngles)]);
resp = step(h,[500 1500 2000],[0 0;40 50]');
plotResponse(h,500,'RespCut','Az','Format','Polar');
```

From the generated beamforming diagrams of the left microphone M_1 and the right microphone M_2, shown in Fig. 3 is possible to obtain the information about the values of attenuation indexes $k_0^1, k_1^1, k_2^1, ..., k_n^1$ and $k_0^2, k_1^2, k_2^2, ..., k_n^2$ in the correspondence of calculated from equation (7) angles of arrival θ_n to the microphones M_1 or M_2 of direct and reflected sound waves after n=0,1,2,3…. number of reflections from the walls of the room. The values of attenuation indexes $k_0^1, k_1^1, k_2^1, ..., k_n^1$ and $k_0^2, k_1^2, k_2^2, ..., k_n^2$ are necessary in the equation (8) for definition of the resultant signals S_{M1} and S_{M2} received from microphones M_1 and M_2 as the sum of corresponding signals produced from the arrived to each of two microphones direct and reflected sound waves.

IV. Experimental Results

Experimental results after the simulations carried out with the proposed and developed simplified planar geometrical model of sound waves propagation from sound source to the corresponding microphones M_1 and M_2 in microphone array based on the human binaural hearing model are presented in the following way.

The initial parameters, needed for the calculations with equation (7) of the angles of arrival θ_n to the microphones M_1 or M_2 of direct and reflected sound waves after n=0,1,2,3…. are room dimensions (r_l=4000 mm and width r_w=6000 mm), sound source coordinates (x_{src}=2000mm and y_{src}=500 mm) and positions of the two microphones (x_{m1}=1900 mm, y_{m1}=5000 mm, x_{m2}=2100 mm and y_{m2}=5000 mm).

For example, in concrete simulation the values of the calculated angle of arrival of direct sound wave, i.e. without reflection, is $\theta_0 = 89$ degree (marked with red arrow in Fig.3), which correspond to the index of attenuation $k_0^1 = -6.02dB$ (Fig.3).

In the similar way, the angle of arrival of sound wave after one reflection is $\theta_1 = 49$ degree (corresponding to 131 degree in Fig. 3 and marked with green arrow) with attenuation index $k_1^1 = -1.70dB$. Also the angle of arrival of sound wave after two reflections is $\theta_2 = 22$ degree, which correspond to the attenuation index $k_2^1 = -28.01dB$.

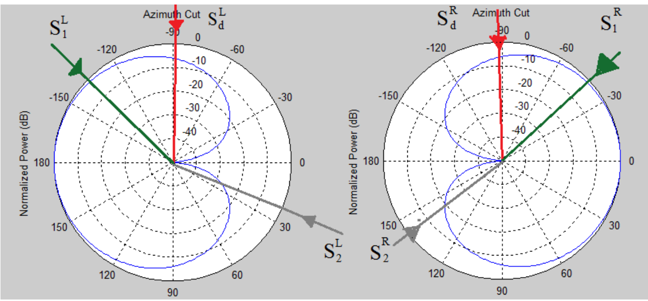

Fig.3. Experimental results

The used above values of the angles of arrival $\theta_0, \theta_1, \theta_2$ and the attenuation indexes k_0^1, k_1^1, k_2^1 are calculated executing the developed for these simulations Matlab program, based on the proposed algorithm, shown in Fig.2. The results in Table I are for these calculated from Matlab program angles of arrival $\theta_0, \theta_1, \theta_2$ and the attenuation indexes k_0^1, k_1^1, k_2^1.

The values of the attenuation indexes calculated by the Matlab program k_0^1, k_1^1, k_2^1 are substituted in the equation (8) for determine the resultant signal S_{M1} received from microphone M_1 as the sum of corresponding signals produced by the arrived to the microphone M_1 direct sound wave and reflected sound waves after n=1,2,3…. number of reflections:

$$S_{M1} = -6.02 S_d^1 - 1.57 S_1^1 - 28.01 S_2^1 \qquad (9)$$

TABLE I
THE CALCULATED IN MATLAB PROGRAM ANGLES OF ARRIVAL AND ATTENUATION INDEXES

Name	Value	Min	Max
k0	-6.0206	-6.0206	-6.0206
k1	-1.5708	-1.5708	-1.5708
k2	-28.0138	-28.01...	-28.01...
theta0	89	89	89
theta1	49	49	49
theta2	22	22	22
xm1	1.9000	1.9000	1.9000
xm2	2.1000	2.1000	2.1000
xsrc	2	2	2
ym1	5	5	5
ym2	5	5	5
ysrc	0.5000	0.5000	0.5000

The similar equation to the equation (9) can be achieved for the right microphone M_2 using the similar results from Matlab program of the attenuation indexes k_0^2, k_1^2, k_2^2 substituted in the equation (8) for determine the resultant signal S_{M2} received from microphone M_2 as the sum of corresponding signals produced by the arrived to the microphone M_2 direct sound wave and reflected sound waves after n=1,2,3…. number of reflections.

V. CONCLUSION

The aim of the proposed in this article geometrical model of sound waves propagation from sound source with combination of a simple two microphone array based on the human binaural hearing model is fully achieved.

The equations derived from the proposed geometrical model lead to achieve the important descriptions of direct and reflected sound waves propagation after the defined simple cases of two numbers of reflections from the walls of the room, especially the resultant signals received from two microphones in the in the microphone array. These equations hold the information of interaural intensity difference (IID) as defined attenuation indexes and information of interaural time difference (ITD) as the lengths of the reflected sound wave trajectories. Used microphones are with similar receiving characteristics like the human ears and their type is chosen to have the beam forming diagram of sensitivity similar to human ears beam forming diagram of sensitivity. The results, derived as equations and simulations, will be used for further deep analyses and will be extended and complicated for the different positions of sound source, increasing the number of sound waves reflections.

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