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SOUND SOURCE POSITION DETERMINATION USING HYPERBOLIC LOCALIZATION

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Abstract: *This paper deals with acoustic source localization using sensory system based on microphone array using hyperbolic localization algorithm. It describes to time-difference of sound wave arrival estimation method which is necessary for localization algorithm function since its results represents input data for processing. Experimental part proposes its software implementation in MS Visual C++ development environment.*
Key words: *hyperbolic localization, microphone array, time-difference, cross-correlation, signal processing*

1. INTRODUCTION

Beginning of audio localization is dated to the year 1880 when the first device for this purpose was designed. Its inventor Professor Mayer used it for navigation improvement in fog. This instrument was called by its author Mayer's topophone. The biggest interest in audio location systems occurs in the period between World War 1 and World War 2 where they were primarily used for detection a localization of the aircraft engine sound. Constructions and dimensions of these systems were very various but the basic concept is based on Mayer's topophone improved with next two horns oriented in vertical plane. Due to state of electronics then minimally two people were required for sound analysis originated from horn system. Main problem with these systems was small sensitivity due to limitations in horn dimensions. For better gain static dishes and walls based on spherical reflection surface was developed. After radio locator invention in 1934 audio location devices were not further developed because they were completely replaced (Self, 2004). Nowadays very dynamical development in electronics and computer science enables applying of the sound localization systems in areas where it was impossible due to technical and economical aspects several years ago. These areas include applications in security, teleconferencing, robotic systems and other else where information is coded in audio signal source position.

Paper deals with acoustic source localization using sensory system based on microphone array. It describes time difference of arrival of sound wave computation algorithm based on microphone signal cross-correlation and hyperbolic localization method resulting in coordinates of sound source position.

2. HYPERBOLIC LOCALIZATION

Hyperbolic localization is time-difference of arrival of sound wave to microphone pair evaluation method enabling to find sound source position coordinates. Microphone signal analysis takes place in two basic steps – time-difference determination of each data samples acquired from different microphones followed by computation of the sound source position coordinates.

Suppose that we have microphone array consisting of only two microphone units m_1 and m_2 with coordinates [m_{1x}; m_{1y}] and [m_{2x}; m_{2y}] and sound source S with coordinates [S_x; S_y] as depicted in figure 1.

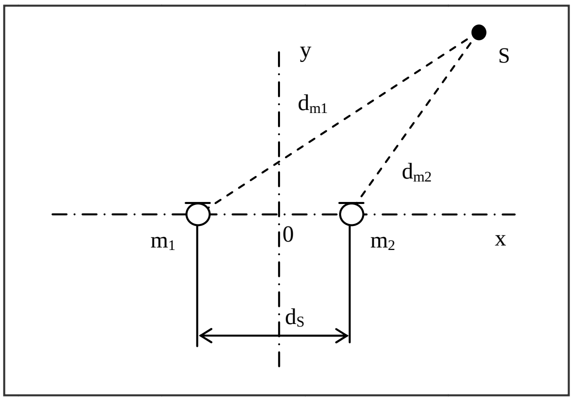

Fig. 1. Simple microphone array geometry

From the figure 1 is resulting that distance of sound source S from each microphone unit m_1 and m_2 can be computed by equations (1) and (2).

$$d_{m1} = \sqrt{(s_x - m_{1x})^2 + (s_y - m_{1y})^2} \quad (1)$$

$$d_{m2} = \sqrt{(s_x - m_{2x})^2 + (s_y - m_{2y})^2} \quad (2)$$

Time-difference of the sound wave arrival to acoustic sensors m_1 and m_2 is:

$$t_{12} = \frac{d_{m1} - d_{m2}}{c}, \quad (3)$$

where d_{m1} and d_{m2} are distances between each microphone and sound source S and c is sound wave speed. Inserting of equations (1) and (2) to equation (3) we obtain final formula describing dependency of time difference t_{12} on sound source position with coordinates [S_x; S_y]:

$$t_{12} = \frac{\sqrt{(s_x - m_{1x})^2 + (s_y - m_{1y})^2} - \sqrt{(s_x - m_{2x})^2 + (s_y - m_{2y})^2}}{c}. \quad (4)$$

By solving of equation (4) we obtain infinitely number of solutions each representing possible sound source position. Because these points have constant distance difference of microphone units they forms hyperbola. Equation (4) has solution if time difference t_{12} fulfills condition:

$$|t_{12}| \le t_{12\max}, \quad (5)$$

where t_{12max} is determined by microphone pair distance d_S:

$$t_{12\max} = \frac{d_S}{c} = \frac{\sqrt{(m_{1x} - m_{2x})^2 + (m_{1y} - m_{2y})^2}}{c}. \quad (6)$$

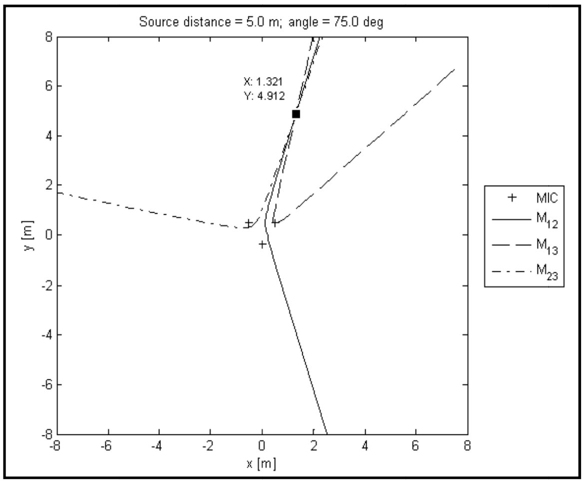

Fig. 2. Hyperbolic localization principle

For time-difference t_{12} computation can be used cross-correlation which is defined by equation (7). Where *x1[k]* is acquired signal from microphone 1, *x2[k]* from microphone 2 and *N* number of acquired data samples.

$$R_{12}[k] = \sum_{n=0}^{N-1} x_1^*[n] x_2[n+k] \tag{7}$$

Because of cross-correlation computation using (7) for larger N takes a lot of processing time it is suitable to use alternate computation in the frequency domain (Smith, 1999):

$$R_{12} = F^{-1}\left\{\left(F\{x_1\}\right)^* \cdot F\{x_2\}\right\} \tag{8}$$

Cross-correlation of the two same shifted signals resulting in coefficients representing conformity of these two signals. Maximum value of correlation coefficient and corresponding shift of *k* samples indicates relative shift of these signals. Time-difference t_{12} for sampling frequency f_S is:

$$t_{12} = \frac{1}{f_S} \arg\max_k \left(R_{12}[k]\right). \tag{9}$$

To find exact sound source position it is necessary to use microphone array with minimally three microphone units in appropriate geometrical configuration. It leads to computation of three time-differences and consequently three hyperbolas which intersect in one common point – sound source position. Principle of sound source coordinates determination for microphone array with 3 microphone units illustrates figure 2.

3. SOFTWARE IMPLEMENTATION

Software application for audio signal analysis and sound source localization was created in Microsoft Visual C++ as Win32 application with utilization of MFC and FFTW library. Input audio signal is acquired with data acquisition device Advantech USB-4716 and via its device driver and ADSAPI interface transfers digitized raw audio data to data buffer (***, 2001). Acquired sound signals are then cross-correlated with utilization of FFTW library. Results of cross-correlation analysis enter the hyperbolic locator routine for sound source position determination and then to the visualization module.

Computational core of the program use FFTW library developed by authors Matteo Frigo and Steven G. Johnson which is released under the GNU General Public License (Frigo & Johnson, 2006). It is used for computations of FFT and IFFT in cross-correlation algorithm in "Audio locator" mode and for frequency spectra analysis in "Audio analysis" mode.

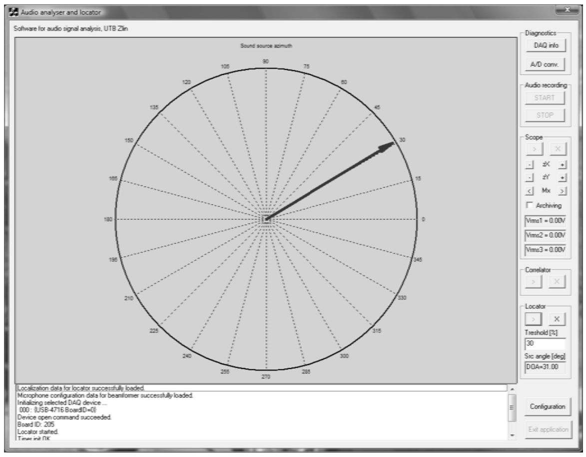

Fig. 3. Program window appearance in "Audio locator" mode

Main window appearance of the developed application switched in "Audio locator" mode is depicted in the figure 3 Main dialog window integrates all necessary program components for program control and results visualization.

Software is running on passive cooled ultra compact computer system Advantech ARK-1382. It is based on Intel Celeron M423 ULV processor working at frequency of 1.06 GHz. Operating memory has capacity of 1 GiB (DDR2 type), main storage is implemented as solid state CF card with capacity of 8 GiB. Communication functions are provided by Ethernet, USB 2.0 and RS232/485 interfaces (***, 2008).

4. CONCLUSION

Paper deals with possibility of acoustic source position determination using hyperbolic localization algorithm for time-differences evaluation in microphone arrays. Developed software application computes in real-time time- differences of sound wave arrival between each microphone pairs and computed sound source position using described hyperbolic localization algorithm presents via graphical user interface to system operator. Evaluation system is based on ultra compact computer system Advantech ARK-1382 connected via USB2.0 with data acquisition device USB-4716. Next research will be focused on optimization of program code to optimal utilization of multi-core processors in computational algorithms for improving system response on localization events.

5. ACKNOWLEDGEMENTS

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