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UTILIZATION OF BEAMFORMING IN SOUND SOURCE LOCALIZATION APPLICATIONS

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Abstract: *This paper deals with utilization of beamforming in acoustic source localization using static microphone array with fixed geometry. It describes main principle of delay and sum beamformer and its practical implementation in localization system. Experimental part propose hardware design of the sensory system consisting of microphone array with preamplifiers and low pass anti-aliasing filters and its connection with evaluation system based on standard personal computer equipped with Advantech PCI-1716 multichannel data acquisition device.*
Key words: *sound source localization, beamforming, signal processing, microphone array*

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.

This paper deals with both design of the input part of the each localization system which is sensory system based on microphone array and software implementation of detection algorithms. Theoretical part describes main principle of delay and sum type beamformer operation and its application in sound source localization system. Next part proposes design of the sensory system consisting of microphone array with fixed geometry and preamplifiers including low pass anti-aliasing filters necessary for signal conditioning before it enters data acquisition device. Last part of the paper deals with software implementation for evaluation unit.

2. DELAY AND SUM BEAMFORMER

Principle of delay and sum beamformer operation is obvious from Fig. 1. Input signals from microphone array x[k] are delayed by time which depends on sensory system geometrical configuration and sound source angle. Because of we can consider that signals from microphone units are same except time-shift, we obtain by setting of appropriate delays H to each audio channel after summing maximum level of useful signal while signal to noise ratio will be improved.

Beamformer output y[k] can be computed by equation (1) where x[k] is input signal from microphone array, H is delay introduced to signal path by beamformer and M is number of microphone units.

$$y[k] = \sum_{m=1}^{M} x_m[k - H_m] \tag{1}$$

For linear uniform microphone array and on assumption that sound source is in much larger distance than is each sensor spacing d_S, time delay in each microphone unit signal for direction of sound wave arrival α can be computed by equation (2), where k is microphone unit index and v is sound speed in air. Reference unit is microphone with index 1 which has zero time shifts for all source angles.

$$t_k = \frac{\sin\alpha \cdot d_s}{v} \cdot (k-1) \tag{2}$$

Sound source localization using delay and sum beamformer is based on computation of the beamformer output signal level for each sound source azimuth angle. RMS value of the n samples length output signal and j azimuth angle is:

$$V_{RMS}[j] = \sqrt{\frac{1}{n}\sum_{k=1}^{n} y[k,j]^2}\ . \tag{3}$$

Maximum RMS value of beamformer output and correspondig angle indicates sound source azimuth:

$$\alpha = \arg\max_j (V_{RMS}[j])\ . \tag{4}$$

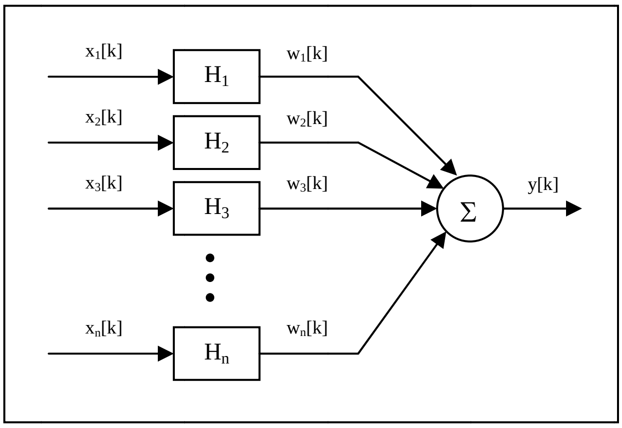

Fig. 1. Delay and sum beamformer block diagram

3. EVALUATION SYSTEM DESCRIPTION

Hardware and software structure of the sound source localization system utilizing beamforming is depicted in the Fig. 2. It consists of microphone array, preamplifiers, low-pass anti-aliasing filters and standard personal computer equipped with data acquisition device.

Microphone array contains 15 omnidirectional condenser microphone units MCE100 soldered on the printed circuit board with dimensions of 200 x 200 mm. Figure 3 shows actual array geometry designed for audio source frequencies in the range of 1500 Hz to 2000 Hz. Operation with wider frequency range is possible but with negative effect on the directional characteristic. Normalised directional characteristic of the electronically steered microphone array for different frequencies is shown in the Fig. 4. Weights of the beamformer are set for maximum gain at azimuth of 180 degrees.

Each microphone cartridge is connected with preamplifier units followed by 4-th order active low-pass anti-aliasing filter with Sallen-Key topology based on low power quad operational amplifiers LM324 (***, 2000). Filter parts was designed using Bessel approximation with cut-off frequency of 2000 Hz and gain of 20 dB in passband. This type of the filter provides linear curve of the phase in the wide frequency range and advantageous step response without overshot (Mancini, 2002).

Last part of the system is standard personal computer with AMD Athlon64 processor. It is equipped with multifunction data acquisition card Advantech PCI-1716 dedicated for PCI bus. This card provides sixteen analog inputs in single-ended or eight analog inputs in differential mode with input impedance of 100MΩ. Each input is through analog multiplexor connected to analog-to-digital converter with 16-bit resolution and maximum sampling rate equal to 250 kHz (***, 2001).

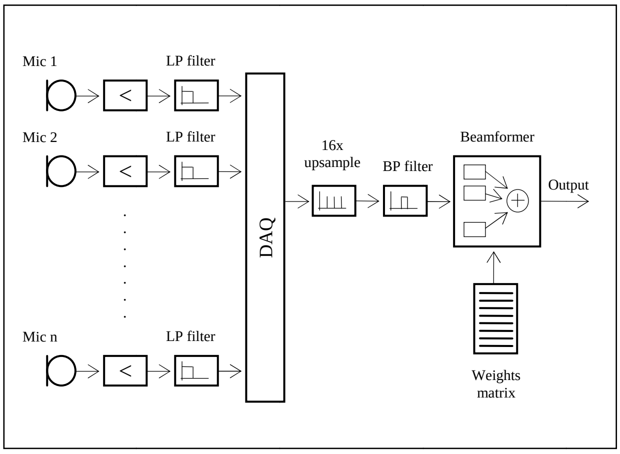

Fig. 2. Sound source localization system structure

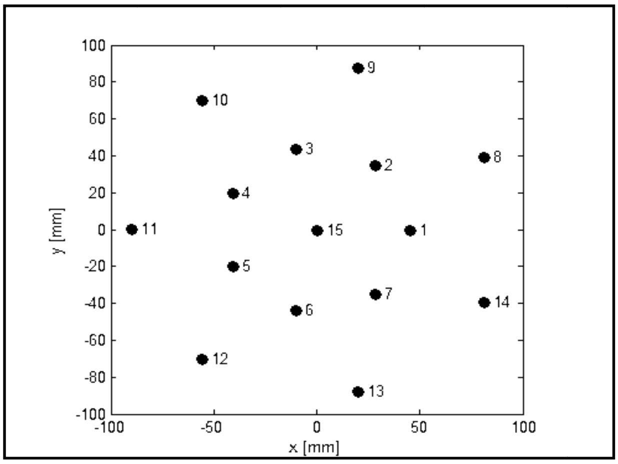

Fig. 3. Microphone array geometry

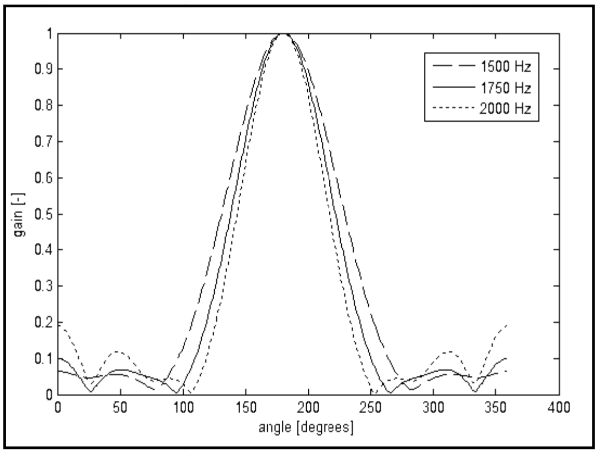

Fig. 4. Normalised directional characteristic of the array

4. SOFTWARE IMPLEMENTATION

Software application for sound source localization was created in Microsoft Visual C++ as Win32 application with utilization of MFC and FFTW library (Frigo & Johnson, 2006). Input audio signal is acquired with DAQ card and via its device driver and ADSAPI interface transfers digitized raw audio data to upsampler routine. Upsampled signal is then filtered by FIR bandpass filter passing only frequencies for which microphone array is designed. Filtered signal enters delay and sum beamformer algorithm which weights are subsequently set for all examined angles of possible sound source azimuth. Maximum RMS value of the beamformer output then corresponds with sound source angle.

5. CONCLUSION

Paper deals with possibility of acoustic source localization using delay and sum beamformer. Developed sensory system consists of microphone array with 15 omnidirectional condenser microphones with optimised geometry for frequency range of 1500 Hz to 2000 Hz and 16 channel preamplifier combined with anti-aliasing filter. Evaluation system is represented by standard personal computer equipped with multifunction DAQ card Advantech PCI-1716 with 16 analog input channels. Practical and simulation experiments show that sound source angle can be determined within 1% accuracy.

6. ACKNOWLEDGMENTS

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7. REFERENCES

Frigo, M. & Johnson, S. G. (2006). FFTW3 for version 3.2.2. *Available from:* http://www.fftw.org/#documentation *Accessed:* 2009-08-02

Mancini, R. (2002). Op Amps for Every One - Design Reference. *Available from:* http://www.ti.com *Accessed*: 2009-08-10

Self, D. (2004). Acoustic Location and Sound Mirrors. *Available from:* http://www.dself.dsl.pipex.com/museum/comms/ear/ear.htm#steer *Accessed*: 2009-08-05

***(2000) http:// www.national.com - National Semi-conductor, LM124 / LM224 / LM324/ LM2902 Low Power Quad Operational Amplifiers, *Accessed on*: 2009-08-10

***(2001) http://www.advantech.com - PCI-1716 250kS/s, 16-bit High-Resolution Multifunction Card, *Accessed:* 2009-08-02