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DESIGN OF THE SENSORY SYSTEM FOR AUDIO SOURCE LOCALIZATION

Petr Dostálek
Tomas Bata University in Zlin, Faculty of Applied Informatics
Nad Stráněmi 4511, 760 05 Zlín
Czech Republic

ABSTRACT
This paper deals with design of the sensory system for audio source localization using microphone array. It describes main requirements, hardware design of the microphone units with automatic gain control preamplifiers with integrated anti-aliasing filters and finally its connection to personal computer. Properties of designed sensory system were verified by real experiment in which standard personal computer equipped with Advantech PCI-1716 multifunction data acquisition card represented evaluation unit. For audio data acquisition and analysis was created software application with user-friendly graphical user interface which is able to visualize measured sound waves and perform frequency analysis. All acquired data from microphone units can be archived to the standard wav files for further investigation and analysis.
Keywords: microphone array, audio source localization, direction of arrival, signal processing.

1. INTRODUCTION
The first device designed by human for audio localization was created in 1880 by Professor Mayer. This instrument for navigation improvement in fog was called by its author Mayer’s topophone. On the basis of its construction originate number of similar devices but with questionable practical usage. The biggest interest in audio location systems occurs in the period between World War 1 and World War 2. They were primarily used for detection a localization of the aircraft engine sound. Measured data about aircraft position was directly transferred to air-defense artillery which can aim at target before visual contact. 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. Since it was impossible to continuously enlarge horn dimensions for better gain achieving, static dishes and walls based on spherical reflection surface was developed. These systems were able to detect aircrafts at longer distances. After radio locator invention in 1934 audio location devices were not further developed in this area because they were completely replaced by RADAR systems with better detection and ranging properties [1].

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 design of the input part of the each localization system – sensory system based on microphone array. It describes circuit solution of the microphone preamp units, automatic gain control amplifier and output anti-aliasing filter which are necessary for signal conditioning to correct voltage levels before further process using data acquisition card.

2. SENSORY SYSTEM DESIGN

Sensory system was designed with a respect to easy portability, configurability and connectivity with evaluation unit. These requirements best fulfill modular architecture schematic of which is depicted in figure 1. It consists of the following main components: three microphone units with integrated preamplifier, 3 channel automatic gain control amplifier with output anti-aliasing filter and evaluation unit – in this case standard personal computer equipped with multifunction Advantech PCI-1716 data acquisition card. Components are connected together with shielded cables to avoid interference leakage to acquired signal.

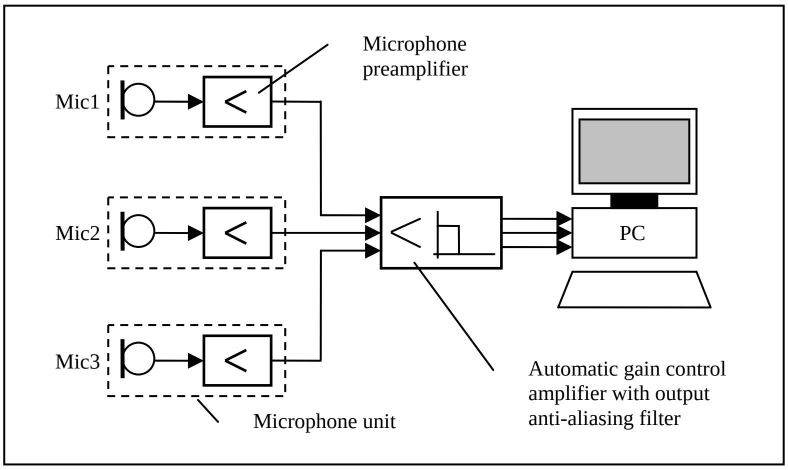

Figure 1. Block schematics of the sensory system

2.1. Microphone units

Sound field is measured with three microphone units each equipped with three omnidirectional electret microphones. They are installed on the small triangular base from cuprextit with the side size of 20mm. This construction improves sensitivity and signal-to-noise ration. Each microphone is connected with summing preamplifier with the gain of 40dB which is mounted in the base of the microphone unit. Output signal level is sufficient for transmission through shielded cable to distance about 10m. Photograph of the completed three microphone units is in the figure 2.

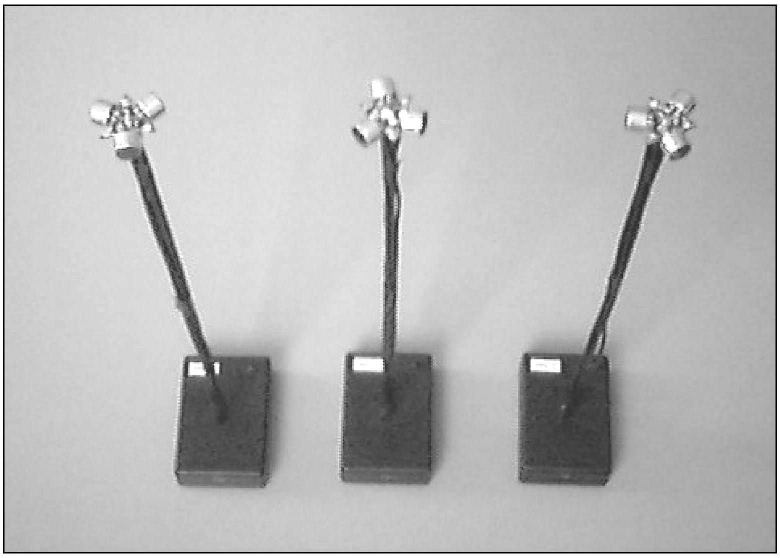

Figure 2. Completed microphone units

2.2. Automatic gain control amplifier with anti-aliasing filter

This part of the sensory system is composed of two main components: three channel automatic gain control amplifier combined with 4th order anti-aliasing filter. The purpose of this last amplification stage is to adapt signal voltage levels to the appropriate level suitable for analog inputs of the data acquisition card. In the case that input signal has too high voltage level the gain of the amplifier is automatically lowered to prevent overdriving of the card inputs. After amplifier stage is connected 4th order low-pass filter designed using Bessel approximation which function as anti-aliasing filter with cutoff frequency of 20000Hz. Filter with these properties is described by transfer function

$$H(s) = \frac{105}{4.0101\cdot 10^{-21} s^4 + 5.0393\cdot 10^{-15} s^3 + 2.8497\cdot 10^{-9} s^2 + 8.3556\cdot 10^{-4} s + 105}. \qquad (1)$$

This type of the filter has linear curve of the phase characteristic in wide frequency range and advantageous step response with small overshot. On the other hand its drawback is smaller slope of the stop-band part of the frequency characteristic in comparison with Chebyshev or Butterworth approximations. Frequency and phase characteristics of the 4th order Bessel low pass filter simulated in Matlab6.5 environment is depicted in figure 3. Step response of the filter is in figure 4.

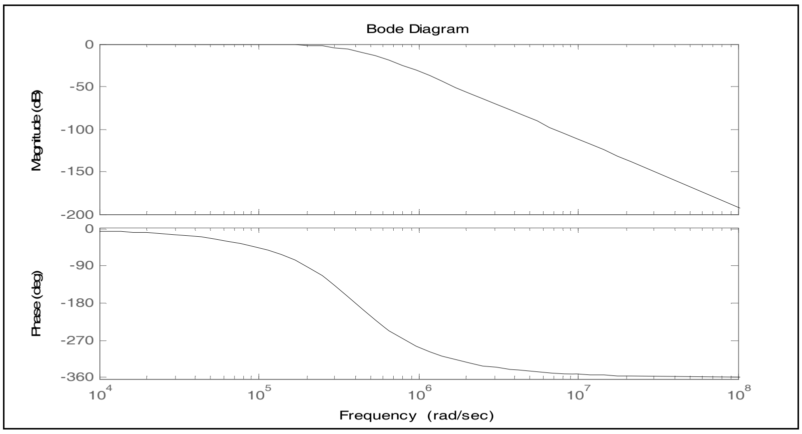

Figure 3. Frequency and phase characteristics of the Bessel filter

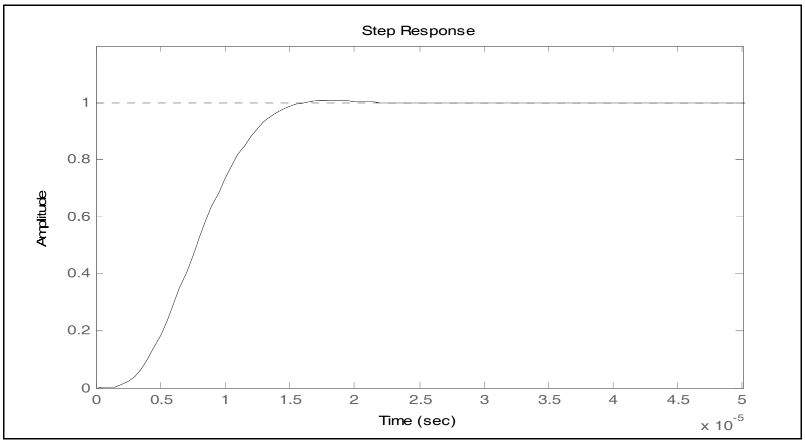

Figure 4. Step response of the Bessel filter

3. SENSORY SYSTEM VERIFICATION

Evaluation system is based on standard personal computer with processor AMD Athlon64 equipped with multifunction data acquisition card Advantech PCI-1716 dedicated for PCI bus interface with full plug and play capability. This card provides sixteen analog inputs in single-ended or eight analog inputs in differential mode. Each input is through analog multiplexor connected to analog-to-digital converter with 16-bit resolution and maximum sampling rate equal to 250 kHz. Integrated FIFO memory with capacity of 1K samples enables efficient data transfer from the card to the system memory without excessive CPU utilization. It is also equipped with two analog outputs, sixteen digital inputs and outputs with TTL compatible logic and finally with 16-bit timer with reference frequency of 10 MHz [2].

For audio data acquisition and analysis software application which is able to visualize measured sound waves and perform frequency analysis was created. It was developed in MS Visual C++ 6.0 software development studio as a win32 application with utilization of MFC library. Frequency analysis is performed by FFTW library which is distributed under GPL license [4]. Example of the 500 samples of audio signal acquired at 80 kHz sampling rate per channel and 16-bit resolution is in figure 5. As a test sound signal was chosen linear chirp which can be generated using “chirp” function of the Matlab’s DSP toolbox.

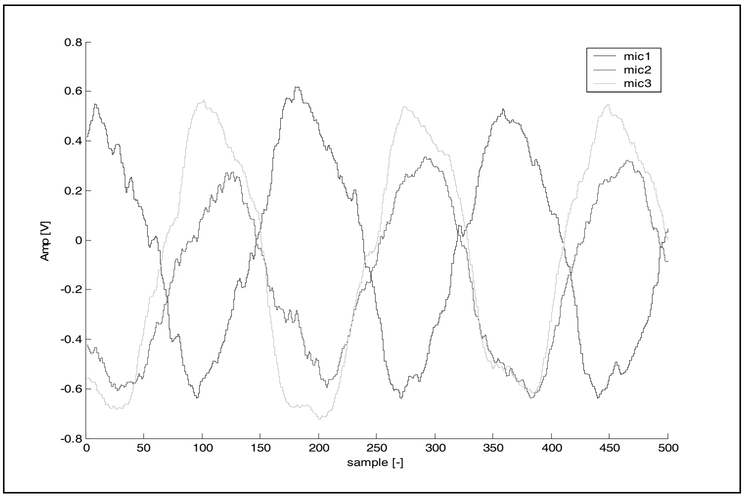

Figure 5. Detail of the 500 samples of acquired sound signal

4. CONCLUSION

This paper deals with design of the sensory system for audio source localization using microphone array. Proposed sensory system has modular architecture which enables easy portability, configurability and connectivity with evaluation unit. It is composed from microphone units with integrated preamplifiers and 3-channel automatic gain control amplifiers with anti-aliasing filter. Properties of designed sensory system were verified by laboratory experiment in which standard personal computer equipped with Advantech PCI-1716 multifunction data acquisition card represented the evaluation unit. For audio data acquisition and analysis software application with user-friendly graphical user interface was created which is able to visualize measured sound waves and perform frequency analysis. All acquired data from microphone units can be archived to standard wav files for further investigation and analysis.

5. ACKNOWLEDGMENT

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