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THE MATHEMATICAL MODEL OF THE AUTOMATED SYSTEM OF CORRECTION POSITION OF THE PSEUDO SOUND SOURCE IN SPACE

Ph. D. V. V. Usyk, I. S. Belikov, National technical university, Kharkiv

This article discusses about math model automatic system of enhancing the interactivity of human involvement in formating apparent sound source in space in real time.

Keywords: *acoustic system, mathematical model, stereophony, apparent sound source, low pass filter.*

Introduction

The purpose of this research is development of the mathematical model of right and left channel of processing acoustic signal in automated system of the pseudo sound source (PSS) positioning.

In the development of automated positioning system of PSS, the structural scheme has been proposed and it's been determined that changing of PSS positioning in space depends on introducing time delays in one of the stereo system channels. Introducing of delay in several milliseconds, for example, in left channel of acoustic system (AS) causes attenuation of perception of sound of this channel and displaces PSS into right loudspeaker [1].

PSS displacing is able in using intensitive stereophony, gain amplitude of one of the channels' signal causes PSS displacing into sounding loudspeaker, that allows person always be in optimal listening zone when moving in the room where person is listening multimedia content.

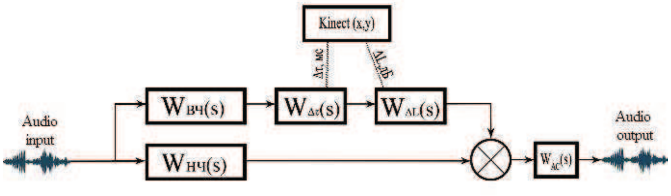
PSS positioning in sounding of area is calculated based on the properties of the human hearing. In placing listener in front of stereobase, playable signals from the right loudspeaker and the left loudspeaker don't differ in time ($\Delta\tau=0$) and in level ($\Delta L=0$). In this condition sounding of both loudspeakers merges into a single sound image, that accords to originally designed sound by sound engineer [3].

This variant of PSS correction is possible for moving listener in space too, or for listener that has not ideal position between loudspeakers of stereo systems.

The narrative of main researching material

Structural scheme of one of the automated system's channels of managing apparent sound source for phase inversion acoustic system is shown in picture 1. The transfer function $W_\Sigma(s)$ of channel generally looks (1):

$$W_\Sigma = \left(W_{НЧ}(s) + W_{ВЧ}(s)\cdot W_{\Delta\tau}(s)\cdot W_{\Delta L}(s)\right)\cdot W_{AC}(s). \quad (1)$$



$W_{ВЧ}(s)$ – transfer function of high pass filter; $W_{НЧ}(s)$ – transfer function of low pass filter; $W_{\Delta\tau}(s)$ – transfer function of time delay' block; $W_{\Delta L}(s)$ – transfer function of managing signal gain' block; Kinect (x,y) – block of receiving the data about displacing listener' head coordinates in space; Δτ,ms – value of the required time delay of audio channel; ΔL,dB – value of gain audio signal according to displacing PSS; $W_{AC}(s)$ – transfer function of acoustic system.

Picture 1 – Structural scheme of the channel of automated managing PSS system

Further in article, the scheme of routing processing of audio signal (picture 1) is considered for left channel of stereo signal. For second channel the values of time delays $\Delta\tau$ and gain amplitude ΔL is being filtered in the same way. The authors had previously found that for managing PSS displacement in space, it's necessary to make time and amplitude corrections in high frequencies area (300 – 20 000 Hz).

Prefiltration of acoustic signal uses low and high pass filters of Butterworth second order frequencies. The transfer function of low pass filter has the form:

$$W_{НЧ}(s) = \frac{1}{B_1^2(s)}, \quad (2)$$

where $B_1(s)=(1+s)$ – Butterworth polynomial.

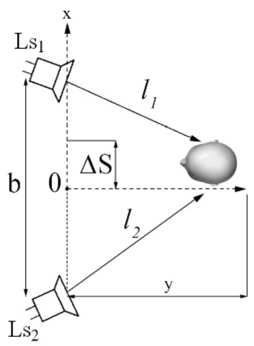
The transfer function of high pass filter has the form:

$$W_{ВЧ}(s) = \frac{s^2}{B_1^2(s)}, \quad (3)$$

The transfer function of time delay' block has the form:

$$W_{\Delta\tau}(s) = e^{-\tau s} \quad (4)$$

where τ – value of AS left channel's delay relative to the right in ms. The schematic location of the listener's head and deviation of his location from the center of AS base is shown in picture 2.



Picture 2 - Deviation of the listener's head location from the center of AS stereobase.

The value of signal delay is determined by the formula:

$$\tau = \frac{\Delta l_1}{c}, \quad (5)$$

where Δl_1 – Deviation of the listener's head from central location of AS stereobase, c – the speed of sound that equals 330 ms/s.

The transfer function of gain (attenuation) signal can be represented as

$$W_{\Delta l}(s) = k \quad (6)$$

where *k* – value of gain (attenuation) acoustic signal of left channel relative to the right in dB.

This variant of intensitive stereophony are encouraged to apply at the same time with time delays if value Δτ exceeds 12 ms for return PSS position in required space. The principle of operation is to gain opposite channel AS, that doesn't have any time delays Δτ [2].

For stereobase, that have value b=0,8…1,8 meters, the value of gain signal *k*, in deviation of listener's head from the AS base center on S=0,5b, is equal 5 dB. For stereobase, that have value b=1,8…2.8 meters, in deviation of listener's head from the base center on 0,5b, value *k*=5…8 dB, accordingly[3]. Coordinates of moving listener's head and detecting image of the person in space is performed by the camera Microsoft Kinect [6-7]. The authors had previously had experimental verification of changing PSS positioning in space, and the method of calibration PSS central location for loudspeakers has been proposed [8].

Normalized transfer function of phase inversion type acoustic system has the form:

$$W_{AC}(s) = \frac{T_0^4 \cdot s^4}{T_0^4 \cdot s^4 + a_1 \cdot T_0^3 \cdot s^3 + a_2 \cdot T_0^2 \cdot s^2 + a_3 \cdot T_0 \cdot s + 1}, \quad (7)$$

where $T_0 = \sqrt{T_B \cdot T_S}$,

where $T_B = \frac{1}{f_b}$, f_b – phase inverter tuning frequency;

$T_s = \frac{1}{f_s}$, f_s – resonant frequency of loudspeaker's head;

$$a_1 = \frac{Q_L + h \cdot Q_{ts}}{\sqrt{h \cdot Q_L \cdot Q_{ts}}};$$

$$a_2 = \frac{h + \left(\alpha + 1 + h^2\right) \cdot Q_L \cdot Q_{ts}}{\sqrt{h \cdot Q_L \cdot Q_{ts}}};$$

$$a_3 = \frac{Q_L \cdot h + Q_{ts}}{\sqrt{h \cdot Q_L \cdot Q_{ts}}},$$

where a – value of normalized parameters of filter elements;

$h = \frac{f_B}{f_S} = \frac{\varpi_B}{\varpi_S} = \frac{\tau_S}{\tau_B}$ – normalized by phase inverter tuning frequency;

Q_{ts} – the total quality factor of the loudspeaker;

$\alpha = \frac{C_{AS}}{C_{AH}} = \frac{L_{CES}}{L_{CEB}}$ – the ratio of flexibilities suspension in the air and inside the body;

$Q_L = \varpi_B \cdot C_{AB} \cdot R_{AL} = \frac{1}{\varpi_B \cdot C_{MEP} \cdot R_{EL}}$ – quality factor, that characterizes slit losses, where R_{AL} – acoustic resistance of emitter; C_{AB} – acoustic flexibility of air in the AS body;

$\varpi_B = 2 \cdot \pi \cdot f_B = \frac{1}{\sqrt{C_{AB} \cdot M_{AP}}} = \frac{1}{\sqrt{C_{MEP} \cdot L_{CEB}}}$ – the circular frequency of phase inverter tuning, M_{AP} – acoustic mass of the passive emitter or air in the phase inverter pipe.

Suppose

$$A_0 = T_0^4;$$
$$A_1 = a_1 \cdot T_0^3;$$
$$A_2 = a_2 \cdot T_0^2;$$
$$A_3 = a_3 \cdot T_0 .$$

Taking into account introduced notations, transfer function of acoustic system will have the form

$$W_{AC}(s) = \frac{A_0 \cdot s^4}{A_0 \cdot s^4 + A_1 \cdot s^3 + A_2 \cdot s^2 + A_3 \cdot s + 1}. \quad (8)$$

In result, the final expression (1) will have the form

$$W_{\Sigma}(s) = \frac{A_0 \cdot s^4 - A_0 \cdot k \cdot s^6 \cdot e^{-\tau s}}{(s^2 + 2s + 1) \cdot (A_0 \cdot s^4 + A_1 \cdot s^3 + A_2 \cdot s^2 + A_3 \cdot s + 1)}. \quad (9)$$

Denotion $k_1 = k \cdot A_0$, we'll get the expression

$$W_{\Sigma}(s) = \frac{-s^4 \cdot (k_1 \cdot e^{-\tau s} + A_0)}{(s^2 + 2s + 1) \cdot (A_0 \cdot s^4 + A_1 \cdot s^3 + A_2 \cdot s^2 + A_3 \cdot s + 1)}. \quad (10)$$

The resulting expression of transfer function of channel will be used for receiving summarized transfer function of whole system for further analysis.

Conclusions

The author is analyzing the resulting mathematical model of automated system, the researching system stability in changing time delay that is entered in each channel of by hardware component and device Kinect, as well as gain coefficient's influence on system stability, being obtained by intensitive stereophony.

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