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Advanced Method for Measurement of Parameters of Multi-channel Sound Systems

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Summary

Multi-channel sound systems are very often used in many areas. One of the significant areas is playback of music in the entertainment industry. A specific area is high-quality car sound system. At present, the sound systems in the automotive are realized using multi-channel loudspeaker systems positioned at different distances of the listening positions. These different placements of the loudspeakers cause strong non-uniform frequency distribution of the sound field. The possibility of the placement of the loudspeakers into the optimal position in the car is limited and very often it is not possible. Thus, it is necessary to solve the consequences of the inappropriate position of the loudspeakers on the sound field using some sophisticated digital signal processing.
In this work, we focus on design of advanced method for measurement of parameters of multi-channel sound systems such as frequency response, impulse response, time delay, etc. Based on these parameters the compensation of the non-uniform sound field can be performed.
It will be shown a real measurement in the car cabin with subsequent determination of the parameters of multi-channel sound systems.

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1. Introduction

At present, the development of the sound systems is based on multi-channel solution. Multi-channel sound systems are very often used in many areas. One of the significant areas is entertainment industry especially movies, where with increasing number of channels the perception of surround sound improves and this leads to really comfortable experience.
Another significant area is reproduction of standard stereophonic sound in complicated spatial arrangement (large or small spaces, geometrically complex spaces or other non-stand spaces). The main problem of sound reproduction in these arrangements is the presence of non-uniform sound field. Using multi-channel sound system it is possible to compensate the non-uniform sound field, i.e. to improve the perception of stereophonic sound.
In this work, we focus on design of advanced method for measurement of parameters of multi-channel sound systems. Based on these parameters the compensation of the non-uniform sound field in complicated spatial arrangements can be performed. As complicated spatial arrangement we use a car cabin, where the possibility of the placement of the loudspeakers into the optimal position is not possible. Most of the loudspeakers in the car are positioned in the near-field and the cabin materials are either strongly sound reflective or strongly sound absorptive - glass or seats, etc.
Based on a real measurement in the car cabin we demonstrate the determination of selected parameters of multi-channel sound system used in this car.

2. Method for the measurement of the sound field in the car cabin

2.1. Theoretical background

Basic principle of the measurement of the sound field in multi-channel system is shown in Figure 1.

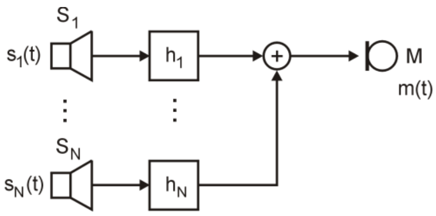


Figure 1. Scheme of the basic principle of the measurement.

The method is based on generating of N appropriate impulse signals ($s_1(t)$, …, $s_N(t)$) followed by obtaining of N impulse responses of the system (h_1, …, h_N).
In the case of one loudspeaker with one microphone, the linear system can be described by equation 1 [1]:

$$m(t) = h(t) * s_n(t) = \int_0^\infty h(t-\tau)\cdot s(\tau)\mathrm{d}t \qquad (1)$$

In the case of multi-channel system with one microphone, the linear system can be described by equation 2:

$$m(t) = \sum_{n=1}^{N} h_n(t) * s_n(t) = \sum_{n=1}^{N} \int_0^\infty h(t-\tau)\cdot s(\tau)\mathrm{d}t \qquad (2)$$

The impulse responses of the system (h_x) represent following properties of the system:

- impulse response of the loudspeakers (includes their directional radiation),
- position of the loudspeakers (includes delay of the signal and influence of the placement of the loudspeaker),
- space geometry (includes reflection and absorption of the signal from barriers),
- environment (includes ambient temperature, pressure and moisture).

Further, it is necessary to include (1) reflections of acoustic waves from the barriers and (2) a microphone array. The microphone array ensures obtaining of the impulse responses in various, but precise positions at the same time.
The method for measurement of the sound field in multi-channel system is shown in Figure 2. Mathematical description of such linear system can be described by equation (3).

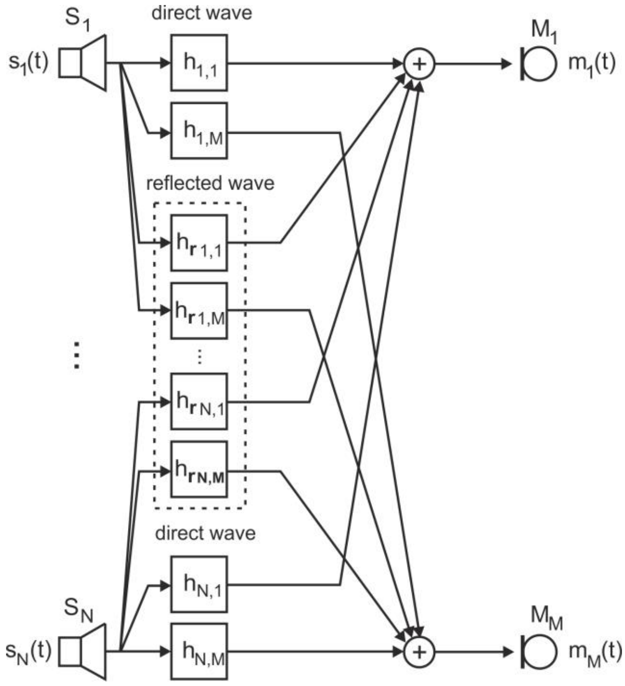


Figure 2. Scheme of the method for measurement of the sound field in multi-channel system.

$$\begin{bmatrix} m_1(t) \\ \vdots \\ m_M(t) \end{bmatrix} = \begin{bmatrix} h_{1,1}(t) & \cdots & h_{1,N}(t) \\ \vdots & \ddots & \vdots \\ h_{M,1}(t) & \cdots & h_{M,N}(t) \end{bmatrix} * \begin{bmatrix} s_1(t) \\ \vdots \\ s_N(t) \end{bmatrix} +$$

$$+ \begin{bmatrix} h_{r1,1}(t) & \cdots & h_{r1,N}(t) \\ \vdots & \ddots & \vdots \\ h_{rM,1}(t) & \cdots & h_{rM,N}(t) \end{bmatrix} * \begin{bmatrix} s_1(t) \\ \vdots \\ s_N(t) \end{bmatrix} \qquad (3)$$

2.2. Real measurement of the time delay

The advanced method for measurement of parameters of multi-channel sound systems described in 2.1 can be used for determination of important parameters of the system, such as frequency response, impulse response, time delay, etc.
One of these key parameters is time delay. Measurement of this parameter is very difficult in the system with multiple reflections. It corresponds to the acoustic distance between the loudspeaker and listening position and it can be different from geometrical distance. The time delay is estimated based on the auto-correlation function (see equation

4) in combination with of appropriate measurement input signal.

$$R_{xy}(\tau) = E[x_1 y_1] = \iint_{-\infty}^{\infty} x_1 y_1 p(x_1 y_1) \mathrm{d}x_1 \mathrm{d}y_1 \quad (4)$$

In Figure 3 is shown the real measurement situation in the car cabin. Geometrical distance between the listening position and loudspeakers is represented by the dash line.

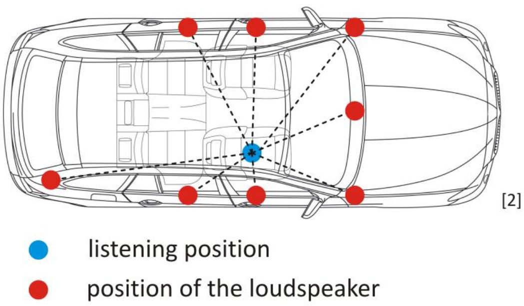
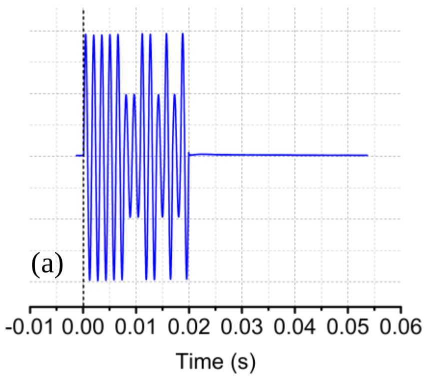


Figure 3. Measurement situation in the car cabin.

For the measurement specific impulse signal was used – the combination of a Barker code and a harmonic signal. Barker codes [3] are a group of discrete signals with very high of auto-correlation coefficient - this attribute allows to easy identification of the path with the larger amount of energy.

For each type of loudspeaker appropriate combination of suitable types of impulse signals and modulation frequencies was used (e.g. for mid-range loudspeaker the B13 + sin. 1 kHz). Signal delay is obtained from the impulse response of each loudspeaker as the difference of the maxima of auto-correlation functions. Figure 4 shows (a) signal from power amplifier, (b) signal from microphone and (c) auto-correlation function for one loudspeaker measured. From the auto-correlation function it can be seen that the value of the time delay for this loudspeaker is 4.13 ms.



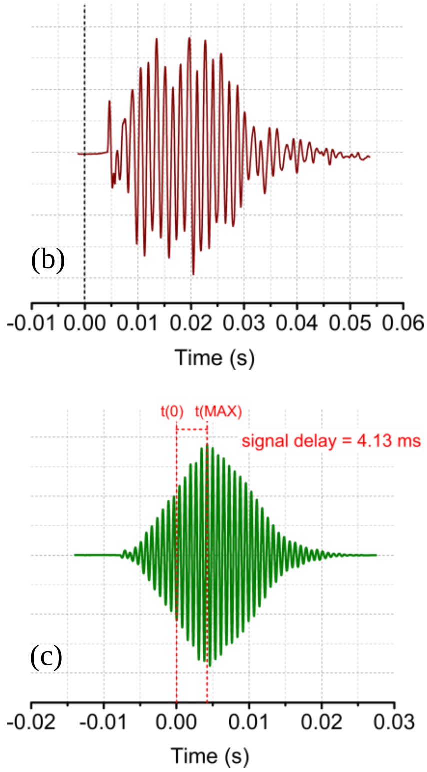


Figure 4. (a) signal from power amplifier, (b) signal from microphone and (c) auto-correlation function for one loudspeaker measured.

3. Conclusions

Novel method for measurement of the acoustic delay of the signals in car cabins was developed. The method is based on using suitable impulse signals which is a combination of a Barker code and a harmonic signal. The value of the time delay in model example was verified in an anechoic chamber and then the method was applied in the real car cabin.

Acknowledgement

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