The equalization of low frequency response in automotive space

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ABSTRACT Unlike in case of large room such as concert hall, we recognize coloring of reproduced sound in the automotive space due to the acoustic resonance of low frequency up to the relatively high crossover frequency. This crossover frequency is determined by Schroeder's large room limit, which separates large rooms from small ones. To compensate this coloring, firstly we measured the low frequency range response of the specific car up to the crossover frequency using band limited swept sine signal. And then, we discuss the method that equalizes the low frequency response, and we show the result of equalization.

0. INTRODUCTION

Recently there has been a great deal of interest in faithful reproduction of the sound field in automotive space. At low frequency in the small space, unevenness is typically caused by the acoustic response of the automotive space itself as well as the electroacoustic response of the audio system including head unit, power amplifiers, loudspeakers, and etc. This problem is particularly severe in the sound field inside the automotive space, since the acoustic resonances of low frequency occur till the relatively high crossover frequency, which can cause pronounced coloration of the sound because of the high resolution of the human ear at the low frequency.

In this paper we will be concerned with the implementation of an equalization system that uses digital filters, the theoretical basis of which are the principle of deconvolution, to achieve low-frequency equalization in automotive space. Firstly, we measure the impulse responses at the head position of the driver seat in the specific car using band limited swept sine signal, which show significantly higher immunity against distortion and time variance rather than pseudo-noise signals [1]. Also, considering binaural room impulse responses for high-quality auralization purposes requires a signal-to-noise ration greater than 90 dB, which is unattainable with maximum length sequence measurements because of the loudspeaker nonlinearity, but it is fairly easy to reach with swept sine due to the possibility of complete rejection of harmonic distortion [1]. Secondly, we apply the digital filtering methods that equalize the low frequency response. Because the impulse responses are non-minimum phase, the direct inversion of the mixed-phase response is not possible since it leads to unstable filter realizations. For this reason, we introduce the modeling delay and the inversion of mixed-phase response is implemented via least-square based method. This equalization works in the expanded region around a fixed listener position, thanks to the relatively large wavelength of low frequency signal. This region includes the sphere of the listener's head movement from its nominal position. Finally we show the result of the listening test. We can assure that the low frequency response is flattened smoothly and we can hear reduced coloration through the listening test.

1. THE MEASUREMENT OF SOUND FIELD IN AN AUTOMOTIVE SPACE

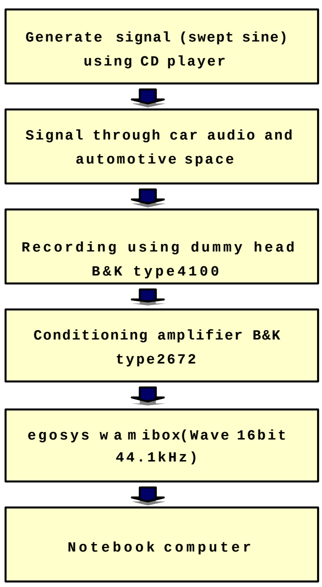

Fig. 1 Sequence of measurement

Fig 1 shows the block diagram of the sequence of measuring the characteristics of sound filed in an automotive space. We used the swept sine signal as a measurement signal. This signal was recorded on CD and played using the CD player installed in the measured car, because this signal do not have to be recorded in perfectly synchronised manner, while outputting input signal like MLS. The input log sweep signal and one of the recorded binaural signals are simultaneously showed in Fig. 2. In this figure anybody can see the excessive low frequency resonance, and we could hear the magnitude variation of radiated sweep signal when we measured this kind of response. We could get an idea of low frequency response equalization from the response.

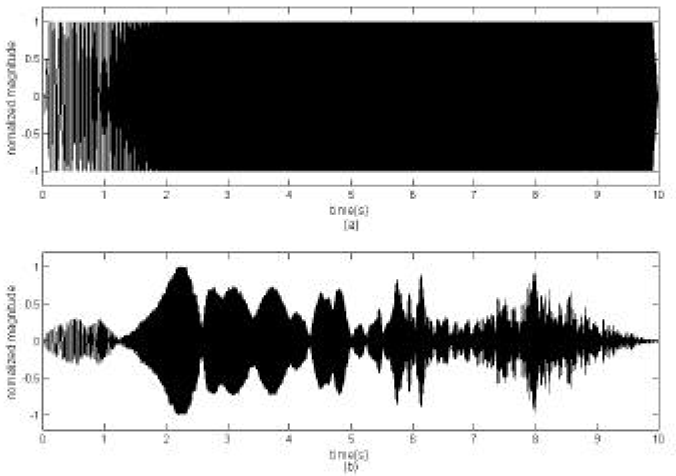

Fig. 2 (a) input log sweep signal, (b) one of the recorded binaural signals

2. THE LOW FREQUENCY RESPONSE EQUALIZATION USING DIGITAL SIGNAL PROCESSING

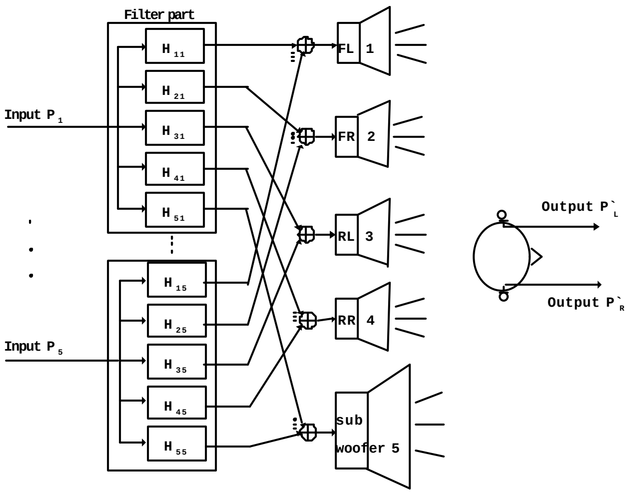

Fig. 3 5-channel sound reproduction system in the automotive space

Fig. 3 shows 5-channel sound reproduction system in the automotive space using one pair of the front speaker, one pair of the rear speaker, and one subwoofer. Eq. (1) is the matrix form of

Fig. 3.

$$\begin{bmatrix} P_L' \\ P_R' \end{bmatrix} = \begin{bmatrix} C_{1L} & C_{2L} & C_{3L} & C_{4L} & C_{5L} \\ C_{1R} & C_{2R} & C_{3R} & C_{4R} & C_{5R} \end{bmatrix} \begin{bmatrix} H_{11} & H_{12} & H_{13} & H_{14} & H_{15} \\ H_{21} & H_{22} & H_{23} & H_{24} & H_{25} \\ H_{31} & H_{32} & H_{33} & H_{34} & H_{35} \\ H_{41} & H_{42} & H_{43} & H_{44} & H_{45} \\ H_{51} & H_{52} & H_{53} & H_{54} & H_{55} \end{bmatrix} \begin{bmatrix} P_1 \\ P_2 \\ P_3 \\ P_4 \\ P_5 \end{bmatrix} = \mathbf{CHP} \tag{1}$$

This paper concentrates on faithful reproduction of conventional stereo recording especially low frequency range below cross over frequency determined by Schroeder's large room limit [2]. The cross over frequency is calculated using Eq. (3).

$$f_s \approx 2000\sqrt{\frac{T}{V}} \tag{2}$$

where T (sec) is the reverberation time and V (m^3) is volume of the space. For example, an automotive space with volume of 5 m^3 and the reverberation time of 0.2 sec will have the crossover frequency of 400 Hz. Also in case of low frequency range the wavelength is very large compared to the distance between left and right ears of human head. Therefore the difference of phase between the signals coming to both ears is negligible,

$$\mathbf{CH} \approx \begin{bmatrix} 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 \end{bmatrix} \tag{3}$$

Eq (3) in the low frequency range can be pursued. This approximation can be understood easily, considering IID (Interaural Intensity Difference), ITD (Interaural Time Difference) is the general situation in case of conventional stereo reproduction. The filter introduced above give birth to the modeling delay in the low frequency range, so in the high frequency range we add the delay component which has the same value with the modeling delay.

Of course we can even adjust delay component like Eq. (4) to compensate asymmetry of listening space in the automotive space and distance between left and right ears of human head.

$$\mathbf{CH} = \begin{bmatrix} e^{jwD1} & e^{jwD2} & e^{jwD3} & e^{jwD4} & e^{jwD5} \\ e^{jw(D1+\Delta)} & e^{jw(D2+\Delta)} & e^{jw(D3+\Delta)} & e^{jw(D4+\Delta)} & e^{jw(D5+\Delta)} \end{bmatrix} \tag{4}$$

where D_i is the delay of each filter i (i=1,2,3,4,5), Δ is the time difference due to head. But in this paper we only consider the case of Eq (3), because the introduction of the delay is can also be implemented in other pre stage. To satisfy Eq. (3) $H_{ii} = \left(\left(C_{iL} + C_{iR}\right)/2\right)^{-1}, H_{ij}(i \neq j) = 0 \quad i = 1,2,3,4,5$ in the Eq. (1) is satfied. When five loudspeakers are driven simultaneously with same impulse-like signal, both of P_L' and P_R' have same value ‘5’. This has almost same effect with the situation described as Eq. (5).

$$\begin{bmatrix} P'_{L,e} \\ P'_{R,e} \end{bmatrix} = \begin{bmatrix} H'_{11} & H'_{12} \\ H'_{21} & H'_{22} \end{bmatrix} \begin{bmatrix} P_{L,dummy} \\ P_{R,dummy} \end{bmatrix} \tag{5}$$

where $P_{L,dummy}$ is the impulse response of dummy head left ear, $P_{R,dummy}$ is the impulse response of dummy head right ear. In Eq. (5) both of $P_{L,e}'$ and $P_{R,e}'$ have same value '5' with $H_{11}'=5P_{L,dummy}^{-1}$, $H_{22}'=5P_{R,dummy}^{-1}$, $H_{12}'=H_{21}'=0$. In this paper, we analyze the appropriateness of this equalization method through the listening test after equalizing in the manner of Eq. (5). The complex realization shown in Fig. 6 refers to the author's another paper [3].

To equalize the low frequency range we put Kirkeby's algorithm [4] into practice. In Kirkeby' algorithm the duration of the inverse filters is shortened, thereby provide a way to avoid the undesirable wrap-around effect usually associated with filter design methods based on sampling in the frequency domain. Also, overloading the loudspeaker and amplifier can be avoided.

3. EQUALIZATION RESULTS AND LISTENING TEST

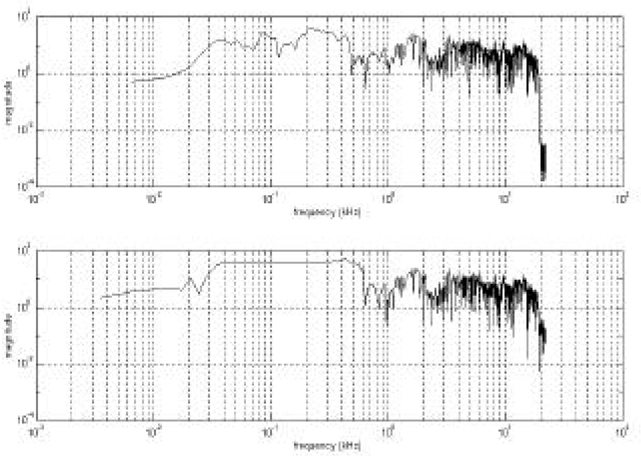

Fig. 5 Frequency response of before-equalization (up) and after-equalization (down)

Fig. 5 shows the frequency response of before-equalization (up) and after-equalization (down), we can confirm that the flattened frequency response from 40 Hz to 400 Hz. These two responses are convolved to ordinary music sample, and then the listening test is carried out. Various music samples like contrabass solo, electric bass solo are selected. The number of test persons was twelve. The questions like below are asked.

a) Which one produces the low frequency range faithfully?

b) Could you describe the difference of two music samples?

Of course, test persons did not know which one is after-equalization or before-equalization music samples before listening test, and they could listen to music samples as many times as they want.

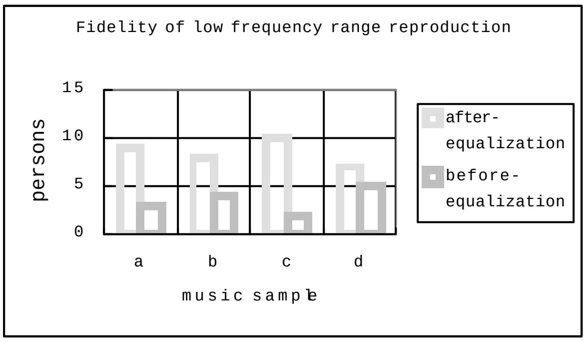

Fig. 6 Fidelity of low frequency range reproduction

Fig. 6 is the result of question a). As we expected, more test persons answer that after-equalized music sample produce produces the low frequency range more faithfully.

4. CONCLUSION

In this paper we introduced the equalization of the low frequency response in the automotive space. And from the results of the listening test we confirmed the appropriateness of this equalization. The filter realization is very easy and simple because we concentrated only on low frequency range. Also, the equalization will have the large applicable area thanks to the relatively large wavelength. If once the impulse responses from each loudspeaker to listener's head position are obtained, the equalization filter realization can be fixed. In the future we will consider the high frequency range more than just delay, and we will make the full system in the real car audio system and then carry this kind of listening test in the real car.

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