A NOVEL APPROACH FOR THE EQUALIZATION OF LOW FREQUENCY RESPONSE IN THE AUTOMOTIVE SPACE

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ABSTRACT

In a different way from large space such as concert hall, we experience the sound coloring of the reproduced sound in the automotive space. It comes from the well-separated acoustic modes in the low frequency range up to the relatively high crossover frequency. The unwanted sound coloring can be reduced by equalization. But this is not a simple matter because the binaural responses are different in each single person and drivers are likely to move their head during driving. In this paper, we introduce a novel approach for the equalization of the low frequency response to compensate the coloring based on minimum phase inversion. We compare the proposed approach with the conventional least square based inversion and show the superiority of this approach with the experimental results. We also confirm that from the results of listening test.

1. INTRODUCTION

Recently there has been a great deal of interest in faithful reproduction of the sound field in automotive space. In the small space like automobile, the frequency unevenness in the low frequency occurs, which 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 unevenness degrades the sound reproduction fidelity. This problem is particularly severe inside the automotive space, since the well-separated acoustic resonances in the low frequency occur up to the relatively high crossover frequency. It can cause the distinct coloration of the sound because of the high resolution of the human ear at the low frequency. The crossover frequency is determined by Schroeder's formula, which is based on the density of acoustic mode along the frequency axis [1]. It separates two frequency ranges in which the transmission function of a given space has different statistical properties. It also allows us to distinguish small rooms from large ones for a given frequency as far as their acoustical behavior is concerned. Sound transmission at high frequencies or in an acoustically large room, is characterized by strongly overlapping resonances, which lead to simple and unspecific statistical properties of the frequency response. In contrast, at low frequency or in an acoustically small room, the resonances are well separated. It reflects the acoustic characteristics of the individual rooms more. Therefore, if we can equalize the low frequency resonances effectively, we can reproduce the low frequency range of music signal more faithfully in the automotive.

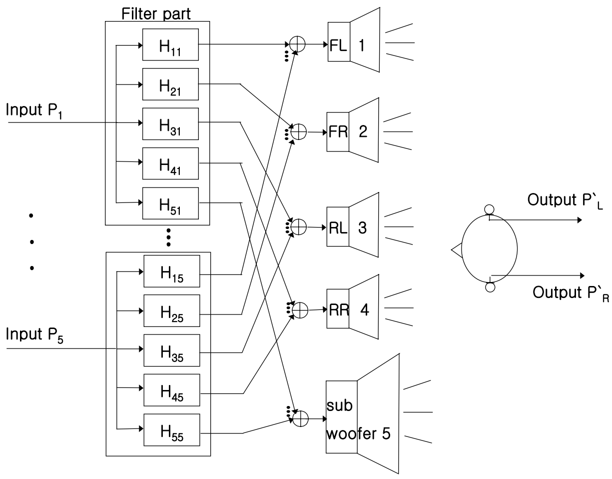

Figure 1:5-channel reproduction system in the automotive space

Typically equalization filters are implemented using band-pass filter bank. Such equalizers can be adjusted so that the modulus of the frequency response is relatively flat. It may, in practice, be difficult to adjust that equalizer to achieve such an objective because of the interaction between the responses of the individual band-pass filters, particularly in low order filters. High-order filters introduce the considerable phase shift, which degrades the transient response of the system, and generates some kinds of distortion.

To tackle the degradation, some researchers have proposed least square based methods [2]. These methods inverse the room transfer function or the loudspeaker response to equalize. These methods, however, show severe distortion in phase with even small deviation between measurement point and installed point while it equalizes magnitude and phase almost perfectly without any deviation [3].

In this paper, we propose a novel equalization system based on minimum phase inverse to achieve the low-frequency equalization in automotive space. The proposed equalizer is robust to the perturbations such as driver's head movement and expands the effective region to enlarge the listening area for driver. We show the results in experiments and confirm the faithful reproduction of low frequency sound through the listening test.

2. ACOUSTIC AND ELECTROACOUSTIC CIRCUMSTANCE IN AN AUTOMOTIVE SPACE

Fig. 1 shows conventional 5-channel sound reproduction system in an automobile, which consists of one pair of the front speaker, one pair of the rear speaker and one subwoofer. Eq. (1) is the matrix representation of Fig. 1.

$$\begin{bmatrix} P_L^{'} \\ P_R^{'} \end{bmatrix} = \begin{bmatrix} C_{1L} & C_{2L} & C_{3L} & C_{4L} & C_{5L} \\ C_{1R} & C_{2R} & C_{3R} & C_{4R} & C_{5R} \end{bmatrix} \cdot$$

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

where
$$\begin{bmatrix} P_1 \\ P_2 \\ P_3 \\ P_4 \\ P_5 \end{bmatrix} = \begin{bmatrix} M_{1L} & M_{1R} \\ M_{2L} & M_{2R} \\ M_{3L} & M_{3R} \\ M_{4L} & M_{4R} \\ M_{5L} & M_{5R} \end{bmatrix} \begin{bmatrix} P_L \\ P_R \end{bmatrix} = \mathbf{MP_{in}},$$

$C_{i,j}$ (i=1,2,3,4,5 j=L, R) is an impulse response from i^{th} loudspeaker to j ear and $M_{i,j}$ (i=1,2,3,4,5 j=L, R) is the transfer function from j input channel of stereo signal to i^{th} input signal. In this paper, we used

$$\mathrm{M} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \\ 1 & 0 \\ 0 & 1 \\ 1/2 & 1/2 \end{bmatrix} \tag{2}$$

,which is used conventionally.

This paper concentrates on the high fidelity reproduction of the conventional stereo recording, which is not the binaurally recorded source, especially in the low frequency range below the crossover frequency determined by Schroeder's formula. The crossover frequency is calculated using Eq. (3).

$$f_s = c\sqrt{\frac{6}{A}} \approx 2000\sqrt{\frac{T}{V}}, \tag{3}$$

Especially in the frequency range under the crossover frequency 400 Hz in case of a typical automotive space with volume of 5 m^3 and the reverberation time of 0.2s, the wavelength is much larger than the distance between left and right ears of human head so that the difference of phase between the signal at both ears is nigligible,i.e. $C_{iL} \approx C_{iR} \approx C_i$. Therefore,

$$\mathbf{CH} \approx \begin{bmatrix} 1 & 1 & 1 & 1 & 1 \\ 1 & 1 & 1 & 1 & 1 \end{bmatrix} \tag{4}$$

can be satisfied. This condition is ideal for the reproduction of stereo recording. To satisfy Eq. (4) approximately,

$$H_{ii} = C_i^{-1}, H_{ij}(i \neq j) = 0 \qquad i = 1,2,3,4,5 \tag{5}$$

in the Eq. (1) should be satisfied. Fig. 2 shows sound reproduction situation from input P_i to driver's ears through loudspeaker i when Eq. (5) is applied. C_i represents both $C_{i,L}$ and $C_{i,R}$ with their average value [4]. This is explained in detail afterwards.

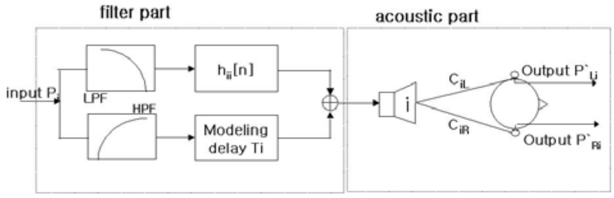

Figure 2:Low-frequency inverse filter adopted sound reproduction

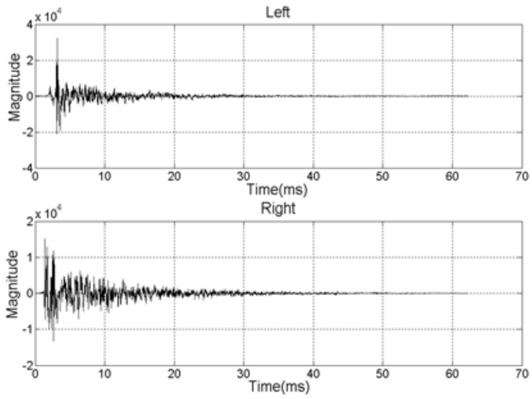

Figure 3:Time response in the measured automotive space using dummy head from front right (FR)

3. COVENTIONAL LEAST SQUARE BASED INVERSE IN LOW FREQUENCY RANGE

One of the exemplary least square based inverse methods for the low frequency equalization is Kirkeby's algorithm in [2]. The main idea of this algorithm is to minimize a cost function as

$$J = E + \beta V, \tag{6}$$

where E is a measure of the performance error, which is the cost function of the conventional least square inverse, and V is a measure of the 'effort', which is usually defined as the total energy of the loudspeaker input signal [2]. The second term in Eq. (6) prevents overcompensation in the frequency region near Nyquist frequency or in deep null points by control of input signal. Therefore, it provides more improved sound quality than normal least square based algorithms do [2]. The positive real number β is a regularization parameter that determines how much weight to assign to the effort term. By varying β from zero to infinity, equalizer response, H in Eq.(5), changes gradually. In this paper we used the value of β , '0.001'.

As we mentioned previous chapter, C_i should be the representative of both $C_{i,L}$ and $C_{i,R}$ such as their average [4]. In that case, the difference between the exact binaural responses and the averaged one can be catastrophic to the deconvolution procedure using the inverse filtering like Kirkeby's algorithm. Fig. 4 shows binarual time responses applying Kirkeby's algorithm, which is the result of inverse FFT of low pass filtered $C_{iL}H_{ii}$ and $C_{iR}H_{ii}$ (i=2). As you can see in Fig. 4 the serious problem was occurred, that is heard by 'chime like sound'. The reason is revealed in Fig. 5 as an unexpected resonance around 300Hz. This outstanding problem may originate in the serious dissimilarity between the binaural responses as shown in Fig. 3.

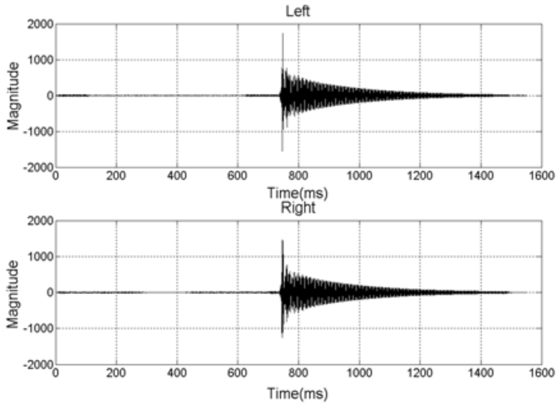

Figure 4:Binarual time responses applying Kirkeby's algorithm, which is the result of inverse FFT of low passed $C_{iL}H_{ii}$ and $C_{iR}H_{ii}$, i=2

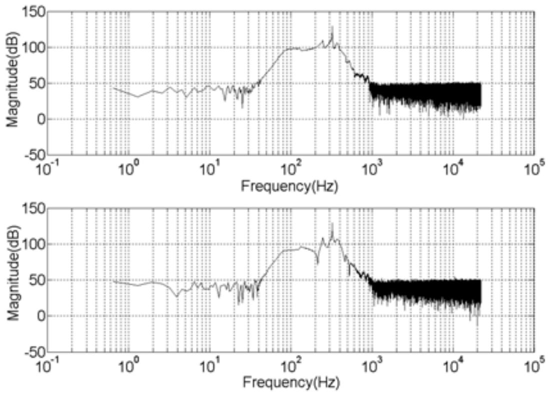

Figure 5:Frequency response of Fig. 4

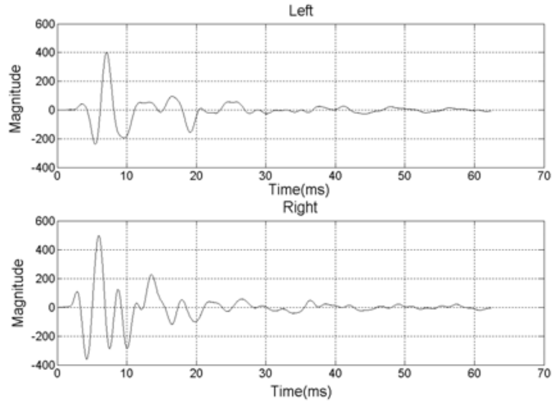

Figure 6:Binaural time responses applying proposed algorithm, which is the result of inverse FFT of low passed $C_{deconv,i,L}(n)=C_{iL}(n)G_{mp,i}(n)$ and $C_{deconv,i,R}(n)=C_{iR}(n)G_{mp,i}(n)$, i=2

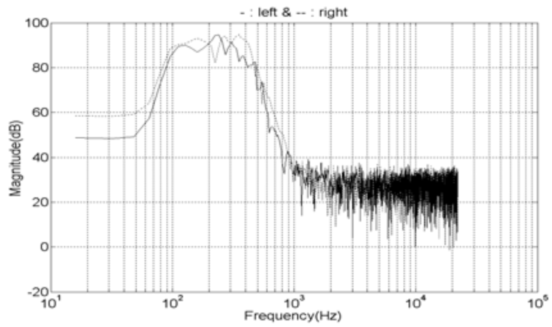

Figure 7:Binaural frequency responses applying proposed algorithm, which is the result of low passed

If we consider that drivers usually move their head while driving, this time domain binaural response dissimilarity may become more serious.

4. PROPOSED MINIMUM PHASE INVERSE IN LOW FREQUENCY RANGE

The change of the driver's head position causes perturbations in the relative arrival times of reflections, which produces phase variations. However, the spectra keep the similarity [3]. The similarity of spectra means the similarity of the minimum phase responses so that we concentrated on the minimum phase responses for a more robust inverse filtering method. Therefore, this approach aims to deconvolve the largest similar part of each binaural response. The procedure obtaining the minimum phase response is as follows [5]:

1. Compute the discrete Fourier transform (DFT) C_i of $C_i=(C_{iL}+C_{iR})/2$.
2. Compute the logarithm of the frequency response, as

$$\hat{C}_i(n) = \log|C_i(n)| + j\arg(C_i(n)) \tag{8}$$

3. Compute the even part of the complex cepstrum $\hat{c}(k)$, as

$$\hat{c}_e(k) = \frac{1}{N}\sum_{n=0}^{N-1}\log|C_i(n)|e^{j(2\pi/N)kn} , \tag{9}$$

where k=0,1,2,⋯,N-1.

4. Calculate the corresponding complex cepstrum of the minimum phase sequence

$$\hat{c}_{mp}(k) = \begin{cases} \hat{c}_e(k), & k=0,\ N/2, \\ 2\hat{c}_e(k), & 1 \le k < N/2, \\ 0 & N/2 < k \le N-1. \end{cases} \tag{10}$$

5. Compute the DFT of $\hat{c}_{mp}(k)$, as

$$\hat{C}_{mp,i}(n) = \sum_{n=0}^{N-1}\hat{c}_{mp}(k)e^{-j(2\pi/N)kn} \tag{11}$$

6. Infer the minimum phase part of $C_i(n)$, as

$C_{deconv,i,L}(n)=C_{iL}(n)G_{mp,i}(n)$ and $C_{deconv,i,R}(n)=C_{iR}(n)G_{mp,i}(n)$, i=2

$$C_{mp,i}(n) = \exp\left[\hat{C}_{mp,i}(n)\right] \tag{12}$$

Once $C_{mp,i}(n)$ is known, the $C_{deconv,i}(n)$, deconvolved component of both C_{iL} and C_{iR} , can be computed as $C_{deconv,i,L}(n)=C_{iL}(n)G_{mp,i}(n)$ and $C_{deconv,i,R}(n)=C_{iR}(n)G_{mp,i}(n)$, where $G_{mp,i}(n)$ represents the minimum phase inverse, given by $G_{mp,i}(n)=1/C_{mp,i}(n)$. In the time domain, this is equivalent to deconvolution $c_{dconv,i,L}(k)=c_{i,L}(k)*g_{mp,i}(k)$ and $c_{dconv,i,R}(k)=c_{i,R}(k)*g_{mp,i}(k)$, with $g_{mp,i}(k)$ being the inverse DFT of $G_{mp,i}(k)$.

Fig. 6 shows the time responses applying the proposed algorithm, which is the result of inverse FFT of low passed $C_{deconv,i,L}(n)=C_{iL}(n)G_{mp,i}(n)$ and $C_{deconv,i,R}(n)=C_{iR}(n)G_{mp,i}(n)$ (i=2). Fig. 7 shows the frequency responses applying proposed algorithm, which is the result of low passed $C_{deconv,i,L}(n)=C_{iL}(n)G_{mp,i}(n)$ and $C_{deconv,i,R}(n)=C_{iR}(n)G_{mp,i}(n)$ (i=2).

From the comparison between Fig. 4 and Fig. 6, we can confirm that proposed algorithm has much shorter unwanted ringing due to the difference of C_{iL}, C_{iR}, with C_i. Fig. 7 shows much more flattened low frequency response. Those say the superiority of the proposed method to the conventional least square method.

5. RESULTS OF SUBJECTIVE LISTENING TESTS

From the measured binaural impulse response of a loudspeaker, the minimum phase based inverse filter is derived, and it is convolved with the low pass filtered music signal. The other high pass filtered music signal is convolved with just the impulse response of the loudspeaker. Fig. 2 illustrates those operations. To synthesis the binaural listening materials, both the low passed one and the high passed one, is summed and then is convolved with the binaural impulse responses, C_{iL} and C_{iR} , respectively. The left part of the synthesized music signals is applied to FL and RL in Fig. 1. The right one is for FR and RR in Fig. 1. The average of left and right is used for subwoofer as Eq. (2). In the same way, the other binaural listening materials are synthesized for the corresponding loudspeakers and then these five materials are mixed down into a stereo two-track wave file using CoolEdit software. The listening test was carried out using a high quality headphone 'Sennheiser HD 540 reference Ⅱ' while playing the wave file on the computer. As a sound card we used 'Echo Layla24' which has much higher quality digital to analog converter relative to ordinary sound cards. As a digital filter the 6^{th} order Butterworth filter is used. Considering the crossover frequency in the crossover network, we used the band pass filter from 80Hz to 400Hz for the front and rear loudspeakers, and used one from 20Hz to 80Hz for the subwoofer instead of a low pass filter itself. The inquiries for listening test are prepared to compare the equalized music signal by the proposed method with the non-equalized signal. Two music samples <1. contrabass solo 'Best of Chesky jazz and more audiophile tests, volum 2'> and <2. Full instrument using sample ' I love paris by Johnny Frigo'> are used. Especially sample 1 is used for evaluation of bass resonance from 'The Ultimate Demonstration Disc' [6]. The number of test persons was seven and they are ones who do not participate in this research. The questions were as follows:

a) Which one produces the low frequency range faithfully?

b) Which one do you like?

Of course, test persons does not know which one was after-equalization or before-equalization music samples before listening test, and they could listen to music samples as many times as they want. As we expected, most of test persons (1: 7 persons, 2: 6 persons) answered that after-equalized music samples produced the low frequency range more faithfully, and that majority of the persons (1: 6 persons, 2: 6 persons) preferred after-equalized music sample.

6. CONCLUSION

In this paper, we proposed the minimum phase based equalization method for the low frequency response in the automotive space. From the results, we confirmed the excellence of this equalization quantitatively and qualitatively. The filter realization is very easy and simple because we concentrated only on minimum phase response of low frequency range. The equalization will also have the wide applicable area around driver due to the relatively large wavelength and similarity of spectral magnitude around that area. Once the impulse responses from each loudspeaker to listener's head position are obtained, the equalization filter realization can be fixed.

In this paper we considered only 5-channel loudspeaker system, but this approach can be applied to any system if we can measure the binaural response in a car. 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.

7. REFERENCES

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