Hearing Aid System Using Basilar Hardware Membrane Model

Yoko Takase*, Yugo Hayashi, Yoshifumi Sekine
Nihon University, Japan
ysekine@ ecs.cst.nihon-u.ac.jp

Abstract

Our purpose is to apply a basilar hardware membrane model (BMM) to hearing-aid systems to solve some of the problem in existing hearing-aid systems. In this study, we construct a BMM using Simulink. Also, we experiment on the response characteristics of our BMM implementation to C6713DSK, using real sounds and hearing-loss sounds, which imitate a hearing-impaired person's hearing ability. It is shown that our BMM can emphasize characteristic frequencies, and hearing-loss sounds are improved so that it may become easier for a hearing-impaired person to hear. These results suggest that our BMM using Simulink is effective in hearing-aid systems.

1. Introduction

Voice information is very important for conversation. Generally, a hearing-aid is used in order to alleviate a hearing impairment. However, more than half of hearing-aid users feel that existing hearing-aids are uncomfortable, because users find it troublesome to have to constantly adjust the hearing-aid due to varying conversation conditions.

G. von Békésy [1] discovered that the basilar membrane positions itself selectively to specific frequencies of speech sounds. B. M. Johnstone et al. [2] measured the displacement of the basilar membrane to the stimulus of various sound pressure levels and made clear the quality factor of resonance in the basilar membrane varied depending on sound pressure of an input. J. L. Flanagan [3] derived mathematical models from Békésy's data for approximating basilar membrane displacement.

We already have constructed the basilar hardware membrane model (BMM) based on Flanagan's mathematical models in consideration of Johnstone's experimental data. Also, we have examined the feature extract function of this model in order to apply the hearing function to engineering models [4]. At present, we are studying how to apply the basilar membrane model to a hearing-aid system in order to solve problems in existing hearing aid systems, because we think that hearing impairments can be alleviated by a system with characteristics closer to living body's characteristics. When the BMM is applied to hearing-aid systems, the basilar membrane model for DSP is an effective method to cope with individual differences, as well as to improve convenience, and to minimize power consumption and mounting area.

Generally, a system is used with DSP by developing an algorithm using programming languages. This method can be proceeding efficiently. However, this method needs time for programming, testing, and correcting, and consideration must be given to the appropriate hardware. System-level design environments, which can verify functions realized obtained using both design and verification, are proposed in order to reduce time for design and verification. We use The MathWorks's MATLAB/Simulink [5] as a system-level design environment, which can visually design and verify based on the block diagram using the graphical user interface (GUI). This can directly convert the Simulink model into C code using an exclusive tool. Thereby, we can farther simplify the development of the system for DSP [7] [9].

In this paper, we construct the BMM using Simulink in order to apply the BMM to hearing-aid systems. Also, we experiment with response characteristics of our BMM implementation to C6713DSK. Firstly, we show the composition of the BMM using Simulink. Secondly, we show that our BMM compare to Johnstone's data. Lastly, we show the benefits to hearing-aid by using our BMM integrated with the C6713DSK when real sounds and hearing-loss sounds are inputted.

2. The BMM

2.1 Transfer function of our BMM

In this section, we describe the two properties of the basilar membrane model. Firstly it can simulate the characteristic of a basilar membrane in that the gain and Q change with the differences in input sound pressure. Secondly it has the sudden interception property in the high frequency domain. Second-order transfer functions of the band-pass filter (BPF), the low-pass filter (LPF), and the band-elimination filter (BEF), that is $H_L(s)$, $H_B(s)$ *and* $H_N(s)$ respectively can be written as below:

$$H_L(s) = \frac{C\omega_0^2}{S^2 + \frac{\omega_0}{Q_L}s + \omega_0^2} \tag{1}$$

$$H_B(s) = \frac{C\frac{\omega_0}{Q_B}s}{s^2 + \frac{\omega_0}{Q_B}s + \omega_0^2} \tag{2}$$

$$H_N(s) = \frac{s^2 + \omega_0'^2}{s^2 + \frac{\omega_0'}{Q_N}s + \omega_0'^2} \tag{3}$$

Where ω_0 is the resonance angular frequency of the LPF and BPF, Q_L and Q_B are quality factors of the LPF and BPF, ω_0' is the resonance angular frequency of the BEF, Q_N a quality factor of the BEF, and C is the constant.

Using Eqs. (1), (2) and (3), the transfer function *F(s)* of the basilar membrane model is expressed as follows:

$$F(s) = H_L(s) \cdot H_B(s) \cdot H_N(s) \tag{4}$$

2.2 BMM using Simulink for DSP

Figure 1 shows the block diagram of the BMM. The whole BMM consists of the parallel connection of a single-channel BMM shown in Fig. 1, because the basilar membrane has the property that selectively responds to the frequency of at a specific position. The single channel of the BMM consist of the filter part, which can simulate the vibration characteristics of the basilar membrane, and the look-up-table that controls the Q, depending on the different input sound pressures. This filter part corresponds to the function of the outer hair cells [6] in the human auditory system. The input of the look-up-table is taken from the output of the BPF. The Q of LPF is controlled by the Q characteristics approximated to Johnstone's data.

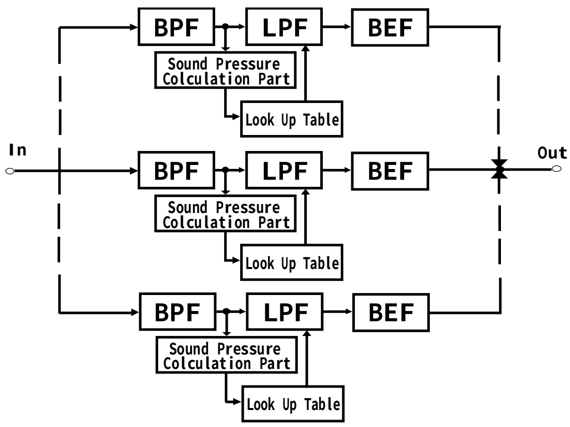

Fig.1. Block diagram of the BMM

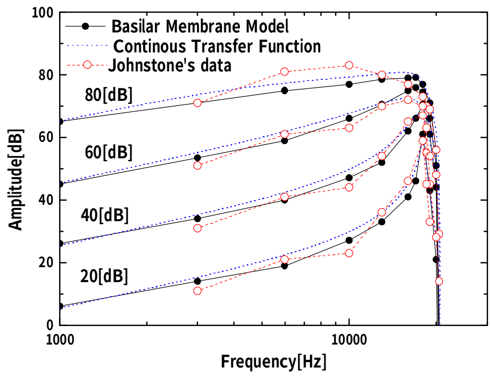

Fig. 2. Frequency response characteristics of our BMM for a single channel at various input levels compared with Johnstone's data and transfer function of BMM.

2.3 Response characteristics of our BMM

Figure 2 shows the frequency response characteristics of our BMM for a single channel at various input levels compared to Johnstone's data and transfer function of BMM. The horizontal axis shows the frequency in Hz, and the vertical axis shows the amplitude in dB. The broken line shows Johnstone's experimental data, and the dotted line shows *F(s)*. Where, Q_L was set to 36.3, 7.7, 1.2, and 0.2, when the input level was 20 dB, 40 dB, 60 dB, and 80 dB respectively (dB re. 20 μPa). ω_0 for the BPF, LPF was 18 kHz, ω'_0 for the BEF was 20.6 kHz. C for the LPF was 110, C for the BPF was 1.0, and moreover, Q_B and Q_N were equal to 3.6 and 0.9 respectively. The results show that the transfer function *F(s)* and BMM using Simulink can approximate Johnstone's data well, i.e. our BMM can imitate Q and the variation of the gain according to a difference in the input sound pressure, and in abrupt cutoff property in the high region.

Figure 3 shows Q characteristics of the QC compared with Johnstone's data. The horizontal axis shows sound pressure in Pa, and the vertical axis shows of Q for LPF. The result shows that the characteristics of the QC can approximate Johnstone's data well.

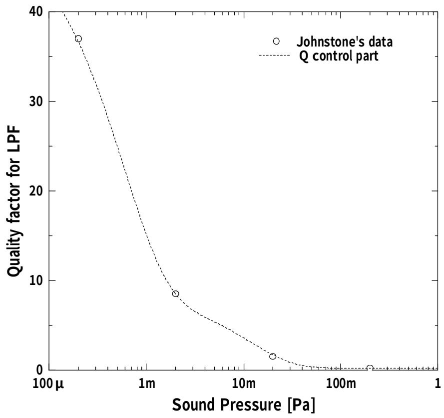

Fig. 3. Q characteristics of QC compared with the Johnstone's data.

3. Response characteristics

3.1 Response characteristics using Japanese vowels.

In this section, we describe some properties of the BMM using Simulink with Japanese vowels in order to clarify the control function of the Q and gain of our BMM. We examined the response of our BMM with the input of Japanese vowels. In this study, 4 channels of the BMM used Simulink.

Figure 4 shows an example of the spectrum of the Japanese vowel /a/ used as the input of the experiments. The horizontal axis shows frequency in Hz, and the vertical axis shows amplitude in dB. Figure 4 (a) shows an example of a loud real voice, and Fig. 4 (b) shows an example of a weak real voice that is 20 dB lower than the frequency at 600Hz in Fig. 4 (a) with the highest level. Figure 5 shows the output response of the BMM for use on C6713DSK, when inputting the waveform shown in Fig. 4. Figure 5 (a) shows the output response of the input of Fig. 4 (a) and, Fig. 5 (b) shows the output response of Fig. 4 (b). These results show that the sound pressure changes significantly. We can say, therefore, that the BMM changes Q according to the difference in the input sound pressure. When we compare 400Hz and 800Hz, respectively in Fig. 4 (a) and Fig. 4 (b) even if the input sound pressure is different, the output level is similar to the nonlinear amplification action of our BMM.

Therefore, our BMM can emphasize frequency constituents with a lower level. It can probably emphasize characteristic frequency of the voice.

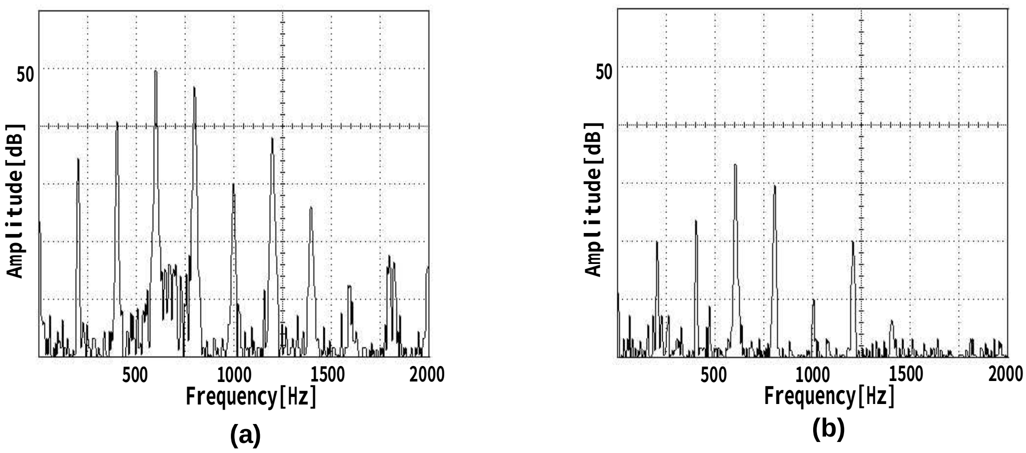

Fig. 4. An example of the spectrum of the Japanese vowel /a/.

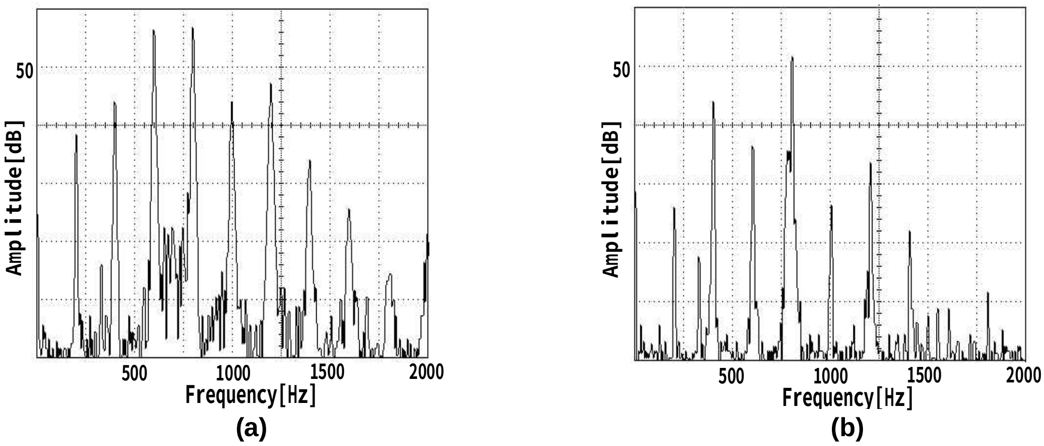

Fig. 5. Output response of the BMM for use on C6713DSK, when inputting the Fig. 4 waveform shown in Fig. 4.

3.2 Response characteristics of hearing-loss sounds.

In this section, we describe the possibility of incorporating our BMM with hearing-aid systems. The hearing-loss sounds, which imitate the characteristics of a hearing-impaired person's hearing ability, were inputted. We examined the validity of a hearing-aid in the model. Figure 6 shows an example of the audiograms of the modeled hearing-loss. Figure 6 (a) shows characteristics of high frequency deafness and Fig. 6 (b) shows characteristics of low tone deafness. The horizontal axis shows frequency in Hz, and the vertical axis shows hearing level in dBHL. These figures show the characteristics that imitate the hearing characteristics assumed most mixed hearing loss as the hearing characteristics of hearing-impaired people.

Figure7 shows an example of the inputted waveform of hearing-loss sounds. Figure 7 (a) shows the waveform of a hearing-loss sound of high frequency deafness and, Fig. 7 (b) shows the waveform of hearing-loss sound of low tone deafness. These hearing-loss sounds were built designed based on Fig. 4 (a).

Figure 8 shows an example of output response of our BMM, when inputting the input waveform of Fig. 7. If Fig. 7 (a), which is the input waveform of artificial hardness-of-hearing sound of high frequency deafness, is compared to output response Figure 8 (a), our BMM can emphasize the characteristic frequency of a high frequency domain. Furthermore, if Fig. 7 (b),

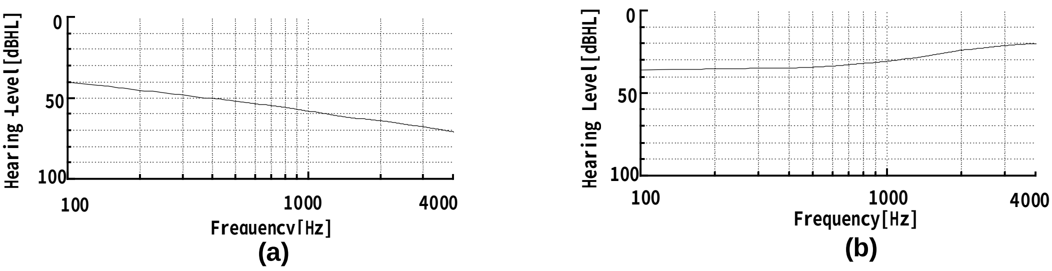

Fig. 6. An example of the audiograms of the modeled hearing loss.

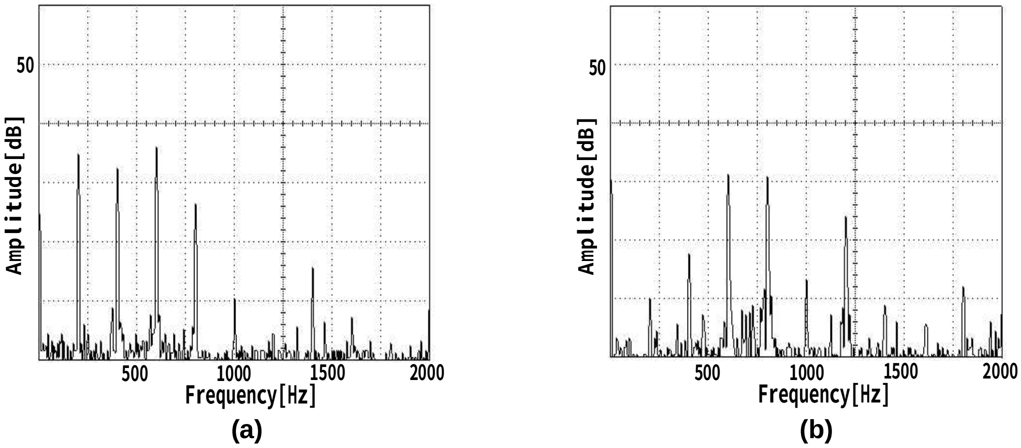

Fig. 7. An example of the inputted waveform of hearing-loss sounds.

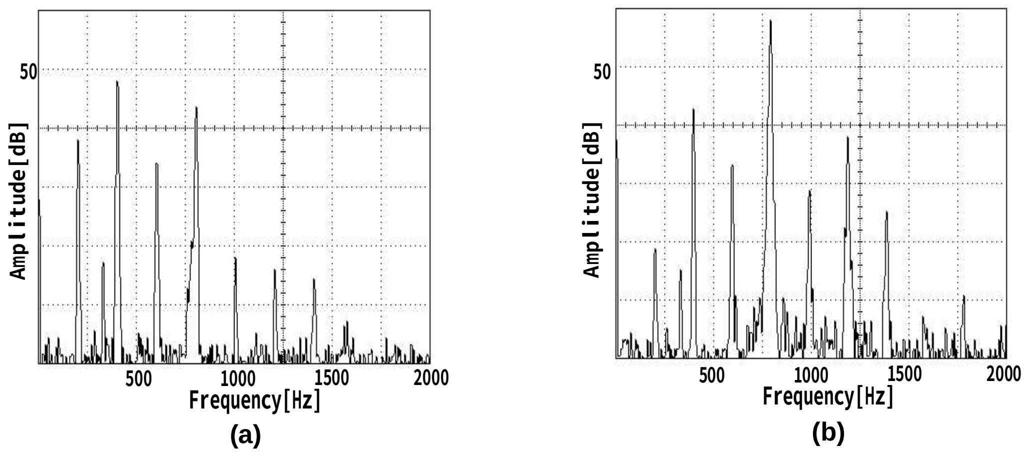

Fig. 8. An example of output response of our BMM, when inputting the waveform of Fig. 7.

which is the input waveform of artificial hardness-of-hearing sound of low tone deafness, is compared to output response Fig. 8 (b), our BMM can emphasize the characteristic frequency of a low frequency domain. These results show our BMM can make up for hearing-loss and output hearing-loss sounds easier to hear. These results suggest that our BMM cooperated with C6713DSK can ameliorate the hearing-loss sounds, even if it is attenuated with the characteristics that imitated the hearing-impaired person's hearing ability characteristics. As a result our BMM is effective in hearing-aid systems.

4. Conclusion

In this paper, we constructed the BMM using Simulink in order to apply the BMM to hearing-aid systems, and experimented with response characteristics of our BMM in cooperated into C6713DSK.

As a result, we showed that our BMM can approximate Johnstone’s data well. Next, we showed that our BMM can emphasize characteristics frequency. Furthermore, when the hearing-loss sounds, which imitate a hearing-impaired person's hearing ability characteristic, were inputted, the output sound became closer to real sounds and it became easer to hear clearly. These results suggest that our BMM using Simulink is effective in hearing-aid systems.

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