

디지털 보청기 적합 검증을 위한 전기음향 시험장치 개발

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Digital Hearing Aid Fitting Program Testing System Development

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Abstract

DSP chip parameters of a digital hearing aid (HA) should be optimally selected or fitted for hearing impaired persons. The more precise parameter fitting guarantees the better compensation of the hearing loss (HL). Digital HAs adopt DSP chips for more precise fitting of various HL threshold curve patterns. A specific DSP chip such as Gennum GB3211 was designed and manufactured in order to match up to about 4.7 billion different possible HL cases with combination of 7 limited parameters. This paper deals with a digital HA fitting program which is developed for optimal fitting of GB3211 DSP chip parameters. The fitting program has completed features from audiogram input to DSP chip interface. The compensation effects of the microphone and the receiver are also included. The paper shows some application examples.

I. Introduction

The main features of a digital hearing aid (HA) IC chip, GB3211, are 4 channel nonlinear compressive active filtering and 4 extra linear biquad filtering. Those 8 active digital filters are used for fitting of various patterns of HL(Hearing Loss)

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threshold curves. In this paper a GB3211 chip interfacing and fitting program was developed for digital HAs in which the most optimal DSP chip parameters were selected from 4.7 billion combinations of possible parameters.

II. DSP parameter fitting procedures

2.1 Audiogram Hearing Threshold

The first step of the parameter fitting is to read the HL threshold of the hearing impaired person (Fig. 1).

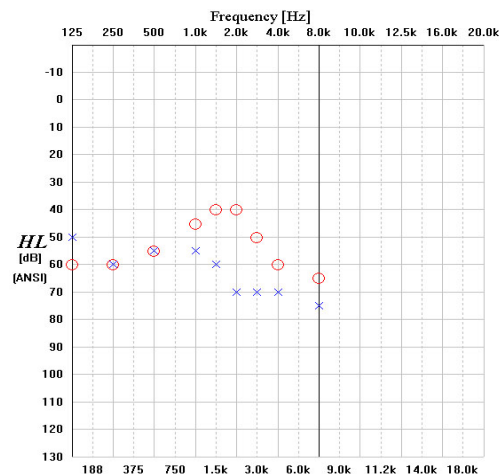


Fig. 1 Air conduction HL thresholds. x(Left Ear), o(Right

Ear).

2.2 Fitting Formula

The second step is to calculate the HA amplification as a function of frequency by a conventional fitting formula such as 'FIG6' [1].

2.3 DSP chip parameters' fitting

The third step is to calculate DSP chip parameters for fitting to the HA amplification function. The parameter fitting in the third step means the proper adjustment of the DSP chip parameters in order for the DSP chip to resemble to the HA amplification function derived by the second step.

2.4 Compensation for the microphone and the receiver

The fourth step is to remove the resonance effects of the microphone and the receiver such as EM4346 microphone and ED3146SAT receiver (Sonicmicroelectronics Co.)

2.5 Compensation for the ear canal resonance

The fifth step is to compensate the insertion loss of the ear canal resonance effect. When an ITE(In-The_Ear) HA is inserted inside an ear canal, the ear canal is blocked with the ITE HA. Then the insertion loss of the ear canal resonance effect happens. Therefore a digital HA should consider the compensation of the insertion loss. The main frequency band of the canal resonance is 1~4 kHz. For this reason the audiogram may be modified as shown in Fig. 2.

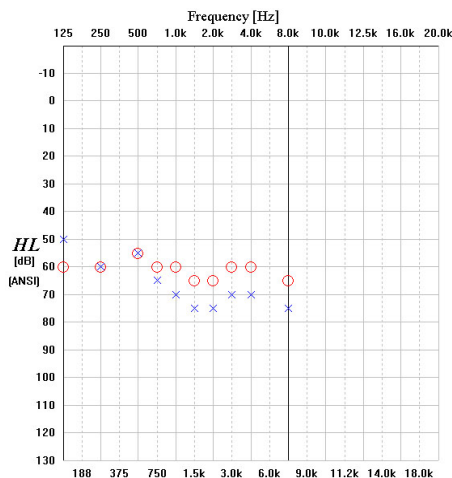
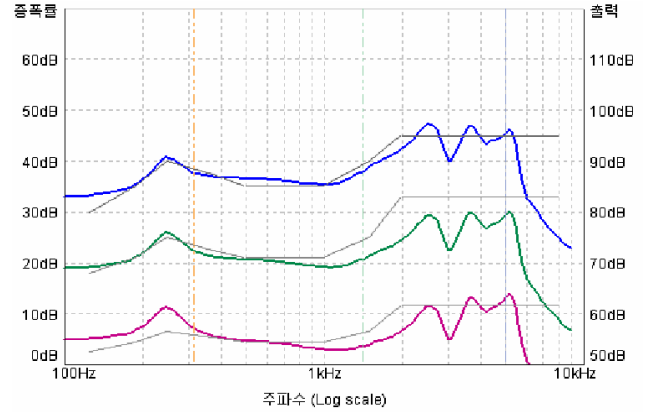


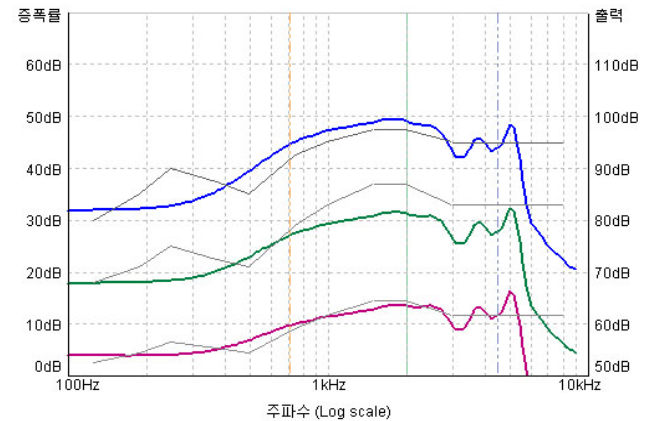
Fig. 2 Audiograms with the compensation of the insertion loss.

x(Left Ear), o(Right Ear).

III. Results



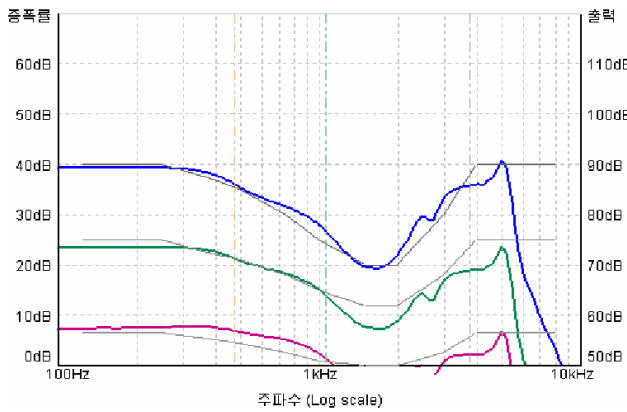
(a) without ear canal compensation(Left Ear)



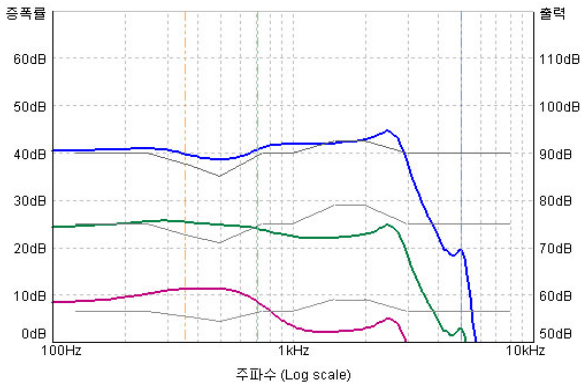
(b) with ear canal compensation(Left Ear)

Fig. 3 The results of the fitting formula (thin lines) and the optimal parameter fitting (thick lines) with the resonance effects' cancellation of the microphone and the receiver by four biquad filters as well as with the compensation of the left ear canal resonance effect. Blue: 40dB Input Level, Green: 60dB Input Level, Violet: 80dB Input Level.

Fig. 3 (b) shows the results of the optimal parameter fitting with the resonance effects' cancellation of the microphone and the receiver by four extra biquad filters as well as with the compensation of the left ear canal resonance effect. And Fig. 4 (b) shows the results of the optimal parameter fitting with the resonance effects' cancellation of the microphone and the receiver by four extra biquad filters as well as with the compensation of the right ear canal resonance effect.

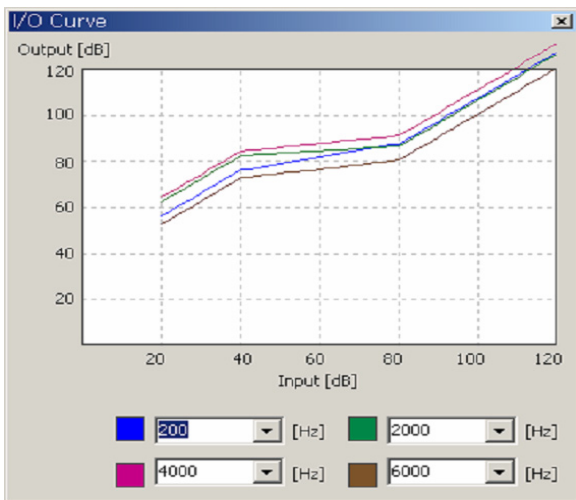


(a) without ear canal compensation (Right Ear)

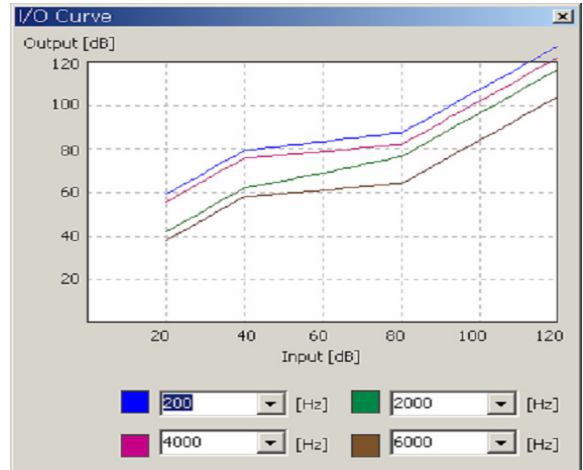


(b) with ear canal compensation (Right Ear)

Fig. 4 The results of the fitting formula (thin lines) and the optimal parameter fitting (thick lines) with the resonance effects' cancellation of the microphone and the receiver by four biquad filters as well as with the compensation of the right ear canal resonance effect. Blue: 40dB Input Level, Green: 60dB Input Level, Violet: 80dB Input Level.



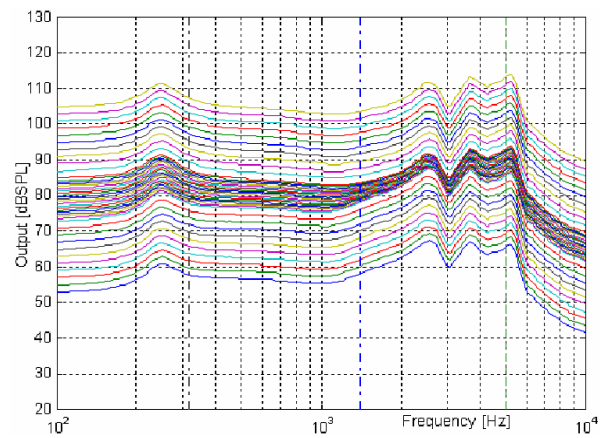
(a) Left Ear



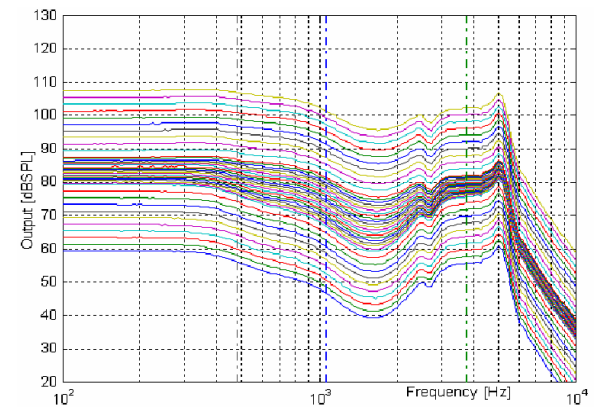
(b) Right Ear

Fig. 5 Nonlinear input/output compression curves for four different frequencies.

Fig. 5 shows the nonlinear input-to-output compression curves for four different frequencies. And Fig. 6 shows nonlinear output sound pressure levels against input sound pressure levels as functions of frequency.



(a) Left Ear



(b) Right Ear

Fig. 6 Nonlinear output sound pressure levels (dB SPL) against input sound pressure levels as functions of frequency.

V. Conclusion

This paper applies a specific DSP chip such as Gennum GB3211 to digital ITE type HA fabrication and shows the result of the optimal parameter fitting program development for the chip. Details of chip parameters are explained. The fitting program has completed features from audiogram input to DSP chip interface. The compensation effects of the microphone and the receiver are also included. The DSP chip parameters such as 4.7 billion possible combinations of parameters should be optimally chosen to resemble to the amount of hearing threshold compensation derived by the fitting formula. Extra digital filters need to be added for the complete fitting to the HA amplification function. Even though the present DSP chip parameter fitting program provides the best fitting for the fitting formula, the final precise fitting should be manually adjusted by the user, that is, the hearing impaired person [2,3].

References

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