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Hearing impairment simulator based on auditory excitation pattern playback: WHIS

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ABSTRACT A new version of Wakayama-University hearing impairment simulator (WHIS) was developed based on the idea of auditory excitation pattern (EP) playback instead of direct simulation of loudness recruitment in conventional methods. The EPs of normal hearing (NH) and hearing-impaired (HI) listeners were calculated by a frame-based version of the gammachirp filterbank (GCFB) in which the cochlear input-output (IO) function was properly controlled by a parameter referred to as the compression health α. WHIS synthesizes simulated hearing loss (HL) sounds to make the EPs of an NH listener sufficiently close to the EPs of a target HI listener by applying the active and passive level reduction of input signals. The active reduction can be simply formulated by using the composite function of the IO function of HI and the inverse NH IO function of NH. Passive reduction was determined to maintain the hearing level which appears in the audiogram of the HI listener. Two methods were proposed for sound synthesis: a direct time-varying filter method and a filterbank analysis-synthesis method. WHIS was compared with a Cambridge version of the HL simulator (CamHLS) in terms of differences in the IO functions and the spectral distance in the EP spectrograms. WHIS yielded a significantly smaller distance than CamHLS. The results imply that the EP playback is an effective method for the HL simulation.

INDEX TERMS Hearing loss, Hearing impairment, Auditory filterbank, Cochlear model, Peripheral dysfunction

I. INTRODUCTION

Recently, the number of elderly hearing-impaired (HI) persons has increased in many countries. It is crucial to develop next-generation assistive devices that can compensate for the difficulties faced by individual HI listeners. For this purpose, it is essential to effectively specify the dysfunctions without a heavy experimental load. Many psychoacoustic experiments have been conducted to clarify dysfunctions using relatively simple stimulus sounds, such as sinusoids and noise [1]. In addition, many speech sound experiments have been performed, although they have mainly been restricted to intelligibility tests, such as speech-in-noise tests. However, in experiments conducted with elderly HI listeners, it is not easy to specify whether the deterioration factor is located on the periphery, located in the auditory pathway, or is due to cognition. This is because of the huge variability among HI listeners in terms of both audiograms and cognitive factors.

To resolve this problem, at least partially, a hearing loss (HL) simulator was developed to specify the effects of the peripheral dysfunction, such as elevation of the absolute threshold (AT) and loudness recruitment [1], on speech intelligibility [2]. Normal-hearing (NH) listeners could evaluate the speech intelligibility of the HL-simulated sounds, which might correspond to what HI listeners perceive. Moore and Glasberg [3] initially applied roex auditory filters to an HL simulator for loudness recruitment and a more precise simulation of the frequency selectivity. The roex auditory filters were psychoacoustically estimated by conducting notched-noise masking experiments [4] for both NH and HI listeners [5], [6]. Spectrum smearing was introduced to evaluate the effect of bandwidth widening in an HI listener's auditory filter [7], [8]. In addition, a unified version was developed to include loudness recruitment and spectrum smearing [9]. This Cambridge version of the HL simulator is referred to as CamHLS. CamHLS was used in a study of the upper limit of temporal delay in hearing aids [10]. Recently, CamHLS was used in the baseline system of the "Clarity Prediction Challenge" (CPC) [11], which is a competition conducted to develop a new objective measure for hearing-aid signal processing.

There are also other HL simulators. For example, HeLPS v2 [12] is commercially available and includes the simulation of loudness recruitment. We also developed another type of HL simulator [13]–[15], which is referred to as the Wakayama-University Hearing Impairment Simulator (WHIS), which is based on a dynamic compressive gammachirp filterbank (GCFB) [16]. The gammachirp filter in GCFB is a time-domain filter, unlike the roex filter, that accounts for the NN masking thresholds fairly well [17], [18]. The gammachirp requires fewer coefficients than the roex filter [19]. GCFB was also used in an HL simulator functionally similar to CamHLS [20] and a real-time HL simulator [21].

The original version of WHIS was designed to control the degree of the compression in the cochlear input-output (IO) function rather than to simulate loudness recruitment directly, as in other simulators. The purpose of the study is to further extend WHIS to synthesize simulated sounds for an NH listener from auditory representations of the so-called excitation patterns (EPs) [1] of an HI listener. The purpose of WHIS is to make the EPs of the NH listener sufficiently close to the EPs of the HI listener using a method for "excitation pattern playback." It is somewhat similar to a classic study of pattern playback [22] in which sounds were reproduced from visual images of speech spectrograms. Because of the nonlinear characteristics in the cochlear spectral analysis, a sophisticated algorithm is required for EP playback in WHIS.

In this paper, GCFB and its improvement are explained first as the basis of WHIS. Then, the new algorithm of WHIS is explained to derive a mathematical formulae for the EP playback. The performance of WHIS is evaluated in comparison with CamHLS.

II. GCFB AS THE BASIS OF WHIS

WHIS was developed based on a compressive gammachirp filter (cGC) [23] and a dynamic compressive gammachirp filterbank (GCFB) [16]. For the implementation of the new WHIS, GCFB should be improved to meet the following WHIS specifications: 1) fast frame-based processing for an interactive user interface, 2) clear definition of the cochlear output level relative to the AT, and 3) incorporation of the audiograms and cochlear IO functions of HI listeners. In this section, a cGC filter, GCFB, and the modeling of hearing loss are explained in detail to provide an introduction to the basis of the new WHIS.

A. COMPRESSIVE GC FILTER

The background of cGC developed from the original gammachirp was reviewed by [15]. The absolute frequency response of a cGC [23], $|G_{CC}(f)|$, can be formulated as follows:

$$|G_{CC}(f)| = |G_{CP}(f)| \cdot H_{HPAF}(f). \tag{1}$$

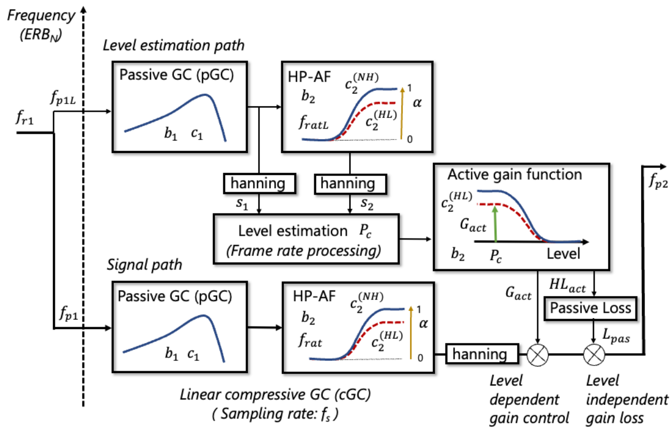

FIGURE 1. Block diagram of the frame-based GCFB (1 ch)

Here

$$|G_{CP}(f)| = a_\Gamma \, |G_T(f)| \, \exp(c_1\theta_1), \tag{2}$$

$$H_{HPAF}(f) = \exp(c_2\theta_2), \tag{3}$$

$$\theta_1 = \arctan\left(\frac{f - f_{r_1}}{b_1 \mathrm{ERB}_N(f_{r_1})}\right), \tag{4}$$

$$\theta_2 = \arctan\left(\frac{f - f_{r_2}}{b_2 \mathrm{ERB}_N(f_{r_2})}\right). \tag{5}$$

$|G_{CC}(f)|$ is a product of a passive gammachirp (pGC), $|G_{CP}(f)|$, and a high-pass asymmetric filter (HP-AF), $H_{HPAF}(f)$, which enables the level-dependent control of bandwidth and gain and is formulated as $\exp(c_1\theta_1)$. $|G_{CP}(f)|$ is a product of a gammatone $|G_T(f)|$ and $\exp(c_1\theta_1)$ which introduces a frequency glide or chirp. The scalar value a_Γ is the amplitude; b_1 and b_2 are bandwidth factors; c_1 and c_2 are chirp factors; and f_{r_1} and f_{r_2} are the asymptotic frequency of pGC and the center frequency of HP-AF, respectively. $\mathrm{ERB_N}(f)$ is an equivalent rectangular bandwidth of NH listeners at frequency f [1].

When the peak frequency of a pGC is f_{p1} and the sound pressure level at the pGC output is estimated as P_{gcp} on a dB scale, the center frequency of HP-AF, the center frequency of HP-AF, f_{r2}, is associated with f_{p1} to introduce the level dependency of a cGC.

$$f_{r2} = f_{rat}(P_{gcp}) \cdot f_{p1}, \tag{6}$$

$$f_{rat}(P_{gcp}) = f_{rat}^{(0)} + f_{rat}^{(1)} \cdot P_{gcp}, \tag{7}$$

where $f_{rat}^{(0)}$ and $f_{rat}^{(1)}$ are coefficients for level dependence.The HF-AF can be level dependent and is represented as

$$H_{HPAF}(c_2, P_{gcp}) = \exp\{c_2 \cdot \theta_2(P_{gcp})\} \tag{8}$$

for a particular frequency. The frequency f is ignored in the following explanation for simplicity.

The parameter values of b_1, c_1, b_2, c_2, $f_{rat}^{(0)}$ and $f_{rat}^{(1)}$ were estimated from the threshold data measured in the notched-noise masking experiments by using the power spectrum model of masking [1], [23], [24]. The parameter values estimated for NH listeners were $b_1 = 1.81$; $c_1 = 2.96$; $b_2 = 2.17$; $c_2 = 2.20$; $f_{rat}^{(0)} = 0.466$; and $f_{rat}^{(1)} = 0.0109$ [18]. These values are also used in this study.

B. GCFB WITH FRAME-BASED PROCESSING

The original version of GCFB was developed using the cGC formula in (1) to simulate cochlear filtering [16]. The level-dependent filtering in this original version required heavy computational costs because the filter coefficients in many channels were updated and convoluted with the input signal at each sample point. The duration required for this sample-by-sample processing was several tens to a hundred times the input signal duration. Therefore, this slow version could not be directly used as a background processor for the interactive human interface necessary in WHIS. Here, GCFB was modified to calculate EPs directly from short-time levels averaged from the filterbank output to improve the processing speed.

1) Block diagram

Figure 1 shows a block diagram for one GCFB channel. As in the original GCFB [16], there are two paths for level estimation (upper block) and signal flow (bottom block), both of which have the same linear passive gammachirp (pGC) and high-pass asymmetric filter (HP-AF). The coefficients of the HP-AF are fixed at $P_{gcp} = 50\,\mathrm{dB}$ in (8). The input signal is filtered by the linear pGC filter and the linear HP-AF to derive the output at the same sampling rate. The processing speed is fast enough for the interactive use. Outputs of the level estimation path were used to estimate the signal level, $P_c(\tau)$, where the RMS level was calculated at the frame time τ with a hanning window length of 1 ms and frame-shift of 0.5 ms. The frame level, $P_c(\tau)$, determined the gain value of an active gain function (right-middle block). The output of the linear filters in the signal path (bottom block) can also be summarized using the same hanning window. The frame-based signal level was, then, controlled by the active gain function to produce the output. As a result, the gain of the auditory filter changes level-dependently while the bandwidth is level independent and determined by the c_2 value of the fixed HP-AF (see Appendix A).

2) GCFB output

Figure 2 shows an example of male speech sound and the outputs of the frame-based GCFB. The output is shown in a spectrographic representation of the sequence of EPs, referred to as EPgram hereafter. Figure 2(b) shows the EPgram using the parameter set of the average hearing level of NH listeners. The spectral features of the speech sound are clearly represented as the relative level from the absolute threshold (AT). It is possible to introduce the peripheral characteristics of HI listeners as described in the next section.

Figures 2(c) and 2(d) show the EPgrams when introducing the average hearing levels of 70-year-old male listeners (hereafter 70-yr) [25] and 80-year-old listeners (hereafter 80-yr) [26]. The average hearing levels are listed in Table 1, and the compression health α, described in Section II-C2, is 0.5. The 70-yr EPgram has less activity, particularly above the 80-th channel. The 80-yr EPgram has much less activity than the NH and 70-yr EPgrams, as expected. The purpose of WHIS is to reproduce the EPgrams of such HI listeners

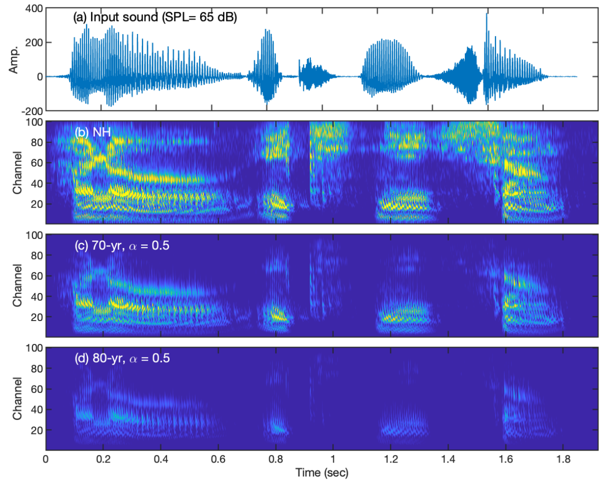

FIGURE 2. EPgrams of male speech sound "Hello ichi ni san" ($L_{eq} = 65\,dB$) (Panel a) analyzed by GCFB with the settings of NH (b), 70-yr condition (c), and 80-yr condition (d).See Table 1. The compression health $\alpha = 0.5$. The horizontal axis is time in ms. The vertical axis is the channel number of GCFB.

TABLE 1. Average hearing levels of 70-year-old male listeners (70-yr) [25] and 80-year-old listeners (80-yr) [26].

Freq.	125	250	500	1000	2000	4000	8000
70-yr	8	8	9	10	19	43	59
80-yr	24	24	27	28	33	48	69

in the peripheral auditory systems of NH listeners by using simulated sound in which the level is reduced adequately and dynamically. The next section explains the method for introducing such HLs into GCFB as the basis of WHIS.

C. MODELING OF HEARING LOSS

1) Active and passive HL

The hearing level measured by a pure tone audiometer is a common measure for diagnosing hearing impairment. This hearing level corresponds to the peripheral HL at the AT. The HL is caused by dysfunctions in active and passive processes. Therefore, the total HL, HL_{total}, is assumed to be the sum of the level-dependent active HL, HL_{act}, and the level-independent passive HL, HL_{pas}, on a dB scale:

$$HL_{total} = HL_{act} + HL_{pas}, \tag{9}$$

where $HL_{total} \geq 0$, $HL_{act} \geq 0$, and $HL_{pas} \geq 0$. Moore and Glasberg [27] proposed a similar equation using the HL caused by the outer hair cell (OHC) HL_{OHC} and the HL caused by the inner hair cell (IHC) HL_{IHC}. Although the main concept for both equations is almost the same, (9) is preferred in this paper because the active and passive processes do not solely function through the OHC and IHC.

2) Introduction of compression health

Compressive gammachirp (cGC) comprises pGC, which represents a passive filter, and HP-AF, which represents an active mechanism [23]. The dynamic range of HP-AF, which is determined by the coefficient c_2, corresponds to the effectiveness of the active process. Therefore, the dysfunction of the active process could be modeled by reducing the c_2

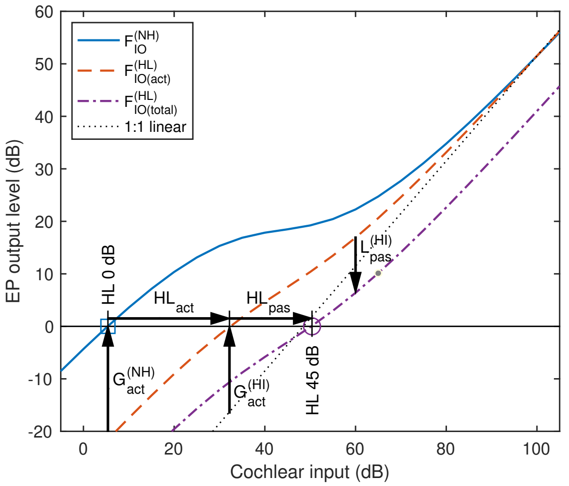

FIGURE 3. Schematic plot of the cochlear IO functions at an audiogram frequency. The abscissa is the sound pressure level (SPL, dB) at the cochlear input. The ordinate is the output level (dB) relative to the absolute threshold (AT) (horizontal solid line at 0 dB). The black dotted line shows a linear relationship or 1:1. The blue solid line represents the IO function, $F_{IO}^{(NH)}$, when using $c_2^{(NH)}$. The label "HL 0 dB" represents the input level corresponding to the AT of the average NH listener as defined in [28]. The orange dashed line shows the IO function of an HI listener, $F_{IO(act)}^{(HI)}$, when using $c_2^{(HI)}$ and $\alpha = 0.5$ without any passive HL. The purple dashed-and-dotted line shows the IO function, $F_{IO(total)}^{(HI)}$, of the HI listener whose hearing level is 45 dB as an example.

value. Then, the c_2 value of an HI listener, $c_2^{(HI)}$, is smaller than the c_2 value of an NH listener, $c_2^{(NH)}$. To simulate this relationship, a coefficient α $\{\alpha | 0 \leq \alpha \leq 1\}$ was introduced as

$$c_2^{(HI)} = \alpha \cdot c_2^{(NH)}. \tag{10}$$

At $\alpha = 1$, there is no dysfunction in the active process and this is the case for NH listeners. At $\alpha = 0$, the active function is completely damaged. The value for an individual HI listener can be somewhat in the middle and frequency dependent. The parameter α is referred to as "compression health" as in the previous WHIS [15] although the definition and value are slightly different.

3) Estimation of α

The value of $c_2^{(NH)}$ in (10) is 2.20, as described at the end of Section II-A. The value of $c_2^{(HI)}$ of an HI listener can be estimated by the same procedure used in NH listeners [18] and can also be estimated indirectly from the cochlear IO function measured by forward masking experiments [1]. The α value is determined as the ratio between $c_2^{(HI)}$ and $c_2^{(NH)}$. However, these psychoacoustic methods require heavy experiments for HI listeners. Developing short tests that are usable in clinical settings is still desirable.

D. COCHLEAR IO FUNCTION AND HL

Figure 3 shows a schematic graph of the cochlear IO functions to explain the effects of the active and passive dysfunctions. The IO function can be calculated from the active process of the HP-AF in (8). The horizontal axis is the input level (SPL dB), P_{in}, to the cochlea, and the vertical axis is the output level (dB), P_{out}, relative to the AT level.

1) IO function for an NH listener

The solid line shows the IO function of an NH listener when using $c_2^{(NH)}$ as

$$\begin{aligned} P_{out} &= F_{IO}^{(NH)}(P_{in}) \\ &= G_{act}^{(NH)}(P_{in}) + P_{in} \\ &= H_{HPAF}(c_2^{(NH)}, P_{in}) + P_{in} \end{aligned} \tag{11}$$

where $G_{act}^{(NH)}$ is the active gain, and $H_{HPAF}(c_2^{(NH)}, P_{in})$ is the transfer function of the HP-AF in (8) on a dB scale. The label "HL 0 dB" represents the input level corresponding to the AT of the average NH listener as defined in a standard of audiometry [28]. The HL 0 dB is achieved with the active gain, $G_{act}^{(NH)}$, which is approximately 45 dB, at this point. The active gain is level dependent on the cochlear input level as represented in $H_{HPAF}(c_2^{(NH)}, P_{in})$. The curve is almost linear when $P_{in} < 30$ dB, is compressive when 30 dB $< P_{in} <$ 70 dB, and is almost linear again when $P_{in} > 70$ dB. This compressive IO function is sufficiently above the AT level and supports the hearing level of NH listeners.

2) IO function for an HI listener

Let us consider a case in which the HL is solely caused by the active dysfunction. When using $c_2^{(HI)}$ with an α value of 0.5 instead of $c_2^{(NH)}$ in (10), the IO function becomes the red dashed line as

$$\begin{aligned} P_{out} &= F_{IO(act)}^{(HI)}(P_{in}) \\ &= G_{act}^{(HI)}(P_{in}) + P_{in} \\ &= H_{HPAF}(c_2^{(HI)}, P_{in}) + P_{in} \end{aligned} \tag{12}$$

This curve is steeper and less compressive. As a result, the output level above the AT is restricted to $P_{in} > 32$ dB. The active gain at the AT level, $G_{act}^{(HL)}$ is much smaller than $G_{act}^{(NH)}$. The difference between $F_{IO}^{(NH)}$ and $F_{IO(act)}^{(HI)}$ results in the elevation of the AT for HL_{act} from HL 0 dB on the horizontal 0-dB line.

When introducing passive dysfunction, the IO function moves downward for $L_{pas}^{(HI)}$ to the purple dashed-and-dotted line. The intercept point of the IO function to the 0-dB line is the total HL of an HI listener, HL_{total}, which is "HL 45 dB" in this case for example. The passive loss corresponds to the AT elevation for HL_{pas}. HL_{total} is the sum of HL_{act} and HL_{pas} as described in (9).

3) Gain calculation in GCFB

In the implementation of GCFB shown in Fig. 1, the peak gain of pGC is normalized to 0 dB independently of the α value. As described in Eqs. 11 and 12, the HP-AF applies the level-dependent active gain to the output of pGC. The signal level, $P_c(\tau)$, is time varying and estimated in the level estimation circuit. The total gain is represented as

$$G_{total}(P_c(\tau)) = G_{act}(P_c(\tau)) - L_{pas}^{(HI)} \quad (13)$$

on a dB scale. Note that $G_{act}(P_c(\tau))$ is level dependent while the passive loss, $L_{pas}^{(HI)}$, is a positive constant that is determined from HL_{pas} and the IO function, $F_{IO(total)}^{(HI)}$. This process is performed in every filterbank channel n_{ch} and every frame time τ with the estimated signal level, $P_c(n_{ch}, \tau)$. The resulting EPgrams can be calculated for both NI and HI listeners, as shown in Fig. 2.

The characteristics of this version of GCFB are explained later. The IO functions are shown in Figs. 7 (a1)–(a4) and described in Section IV together with those of WHIS. The bandwidth of the cGC filter in GCFB is explained in Appendix A.

III. NEW IMPLEMENTATION OF WHIS

As described in Section II-B2, the purpose of WHIS is to reproduce the EPgrams of a target HI listener (e.g. Fig. 2(d) for 80-yr) in the cochlea of NH listeners.

A. OBJECTIVE OF WHIS

Figure 4 shows the schematic plots of the IO functions of an HI listener (a), an NH listener (b), and WHIS (c). The EP output of the HI listener $EP^{(HI)}$ is derived from the IO function shown in Fig.4(a) which is the same IO function in Fig. 3. For hearing loss simulation, WHIS in Fig. 4(c) is applied to the input sound to reduce the sound pressure level (SPL) on a dB scale. The process comprises active reduction, R_{act}, and passive reduction, R_{pas}, which are explained in the next section. The output of WHIS is provided to the IO function of the NH listener in Fig. 4(b). As the input SPL is reduced, the IO function is shifted rightward from the original NH curve (blue dotted line) to the simulated IO function (purple dashed-and-dotted line). The output is represented as $EP^{(NH+WHIS)}$. The objective of WHIS is to minimize the difference between $EP^{(HI)}$ and $EP^{(NH+WHIS)}$.

B. IMPLEMENTATION OF WHIS

Figure 5 shows a block diagram for one channel of the analysis section of the new WHIS. The blocks of the linear cGC filters (pGC + HP-AF) and the level-estimation circuit in the left half are exactly the same as those in GCFB as shown in Fig. 1. In practice, the same software is used in this part. The main difference lies in the use of the gain reduction function, which is the difference between the inverse IO function as described in the next section.

C. CALCULATION OF INPUT LEVEL REDUCTION

1) Active level reduction

Let us consider active level reduction first. To modify the IO function of the NH listener, $F_{IO}^{(NH)}$, to the IO function of the HI listener, $F_{IO(act)}^{(HI)}$, in Fig. 4(b), the input level should be reduced for the horizontal difference between the two IO functions. This value corresponds to the active level reduction, R_{act}, shown in Fig. 4(b) with the horizontal arrow

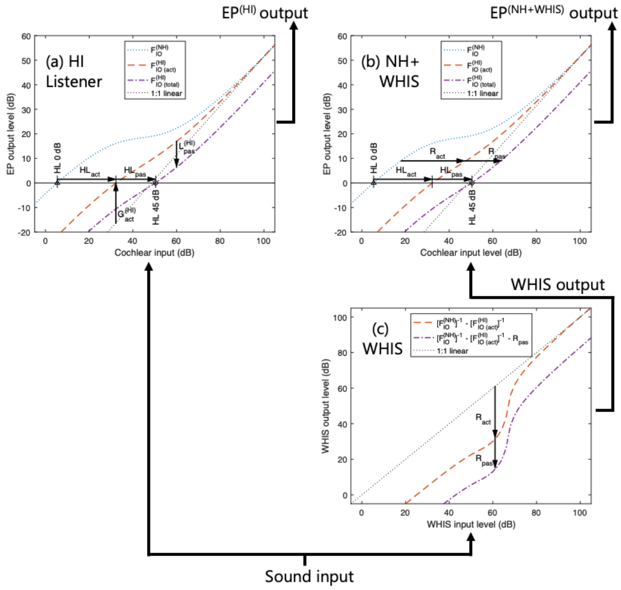

FIGURE 4. Schematic plots of the IO functions of an HI listener (a), an NH listener (b), and WHIS (c). The input sound is processed in accordance with the IO functions to obtain the output EP levels for the HI listener, $EP^{(HI)}$, and the NH listener with WHIS, $EP^{(NH+WHIS)}$. The objective of WHIS is to minimize the difference between $EP^{(HI)}$ and $EP^{(NH+WHIS)}$.

and in Fig. 4(c) with the vertical arrow. R_{act} at a certain output level P_{out} (e.g., 10 dB in Fig. 4(b)) can be calculated from the inverse functions of Eqs. 11 and 12 as

$$R_{act}(P_{out}) = F_{IO(act)}^{(HI)^{-1}}(P_{out}) - F_{IO(act)}^{(NH)^{-1}}(P_{out}), \quad (14)$$

which is represented as the red dashed line in Fig. 4(c).

The target of the HL simulation is to reproduce the EP level of the HI listener, $EP^{(HI)}$. Therefore, P_{out} is $EP^{(HI)}$ which can be calculated from the frame-based level, $P_c(\tau)$, in the level estimation circuit in Fig. 5 with the HL setting using $c_2^{(HI)}$. It is represented as

$$P_{out} = EP^{(HI)} = F_{IO(act)}^{(HI)}(P_c(\tau)). \quad (15)$$

By substituting this equation into (14), we derive

$$\begin{aligned} R_{act}(P_c(\tau)) &= F_{IO(act)}^{(HI)^{-1}}\{F_{IO(act)}^{(HI)}(P_c(\tau))\} \\ &\quad - F_{IO(act)}^{(NH)^{-1}}\{F_{IO(act)}^{(HI)}(P_c(\tau))\} \quad (16) \\ &= P_c(\tau) - F_{IO}^{(NH)^{-1}}\{F_{IO(act)}^{(HI)}(P_c(\tau))\}. \end{aligned}$$

Thus, the active level reduction, $R_{act}(P_c(\tau))$, can be simply formulated by using the composite function of the IO function of HI, $F_{IO(act)}^{(HI)}$, in (12) and the inverse IO function of NH, $F_{IO}^{(NH)^{-1}}$, calculated from (11). This procedure is unexpectedly simple because the IO functions are appropriately defined by the HP-AF in (8).

HL_{act} is the difference between $F_{IO}^{(NH)}$ and $F_{IO(act)}^{(HI)}$ at the AT level, i.e., $P_{out} = 0$, as shown in Fig. 4(b). This is calculated from R_{act} as

$$HL_{act} = R_{act}(P_{c_{AT}}) \quad (17)$$

where $P_{c_{AT}}$ is the input level that yields the R_{act} value at the AT level.

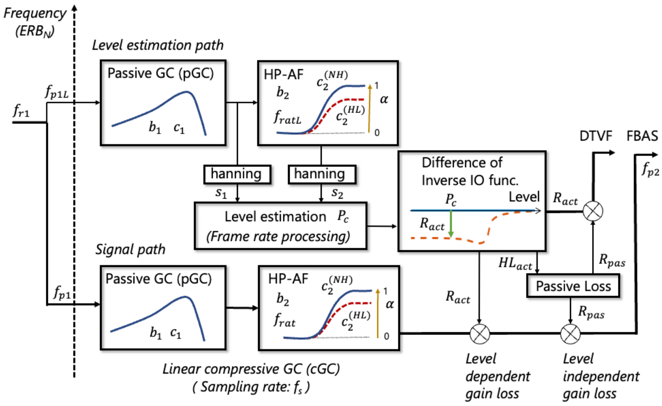

FIGURE 5. Block diagram of the analysis section in WHIS.

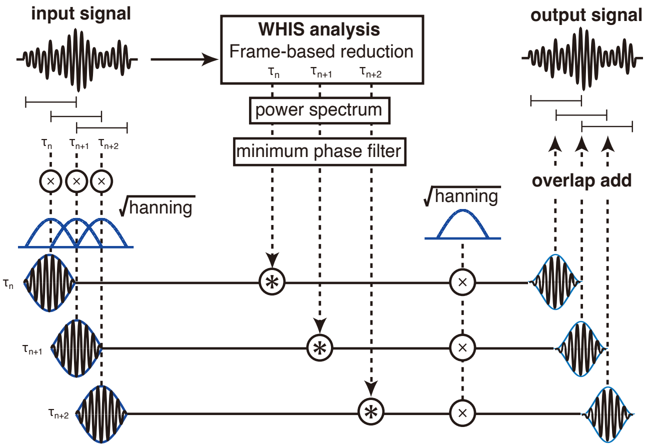

FIGURE 6. Signal synthesis using a direct time-varying filter, DTVF.

2) Passive level reduction

Passive level reduction, R_{pas}, is level independent and is equivalent to the passive HL, HL_{pas}, shown in Figs. 3 and 4. HL_{pas} is calculated from the relationship between the total HL, HL_{total}, and the active HL, HL_{act}, in (9) and (17) as

$$\begin{aligned} R_{pas} = HL_{pas} &= HL_{total} - HL_{act} \\ &= HL_{total} - R_{act}(P_{c_{AT}}). \end{aligned} \tag{18}$$

By this constraint, the total HL, HL_{total}, is maintained at the measured value of the HI listener even if the ratio between R_{act} and R_{pas} is dependent on the α value.

3) Total level reduction

The total level reduction for the HL simulation is the sum of (16) and (18) as

$$\begin{aligned} R_{total}(P_c(\tau)) &= R_{act}(P_c(\tau)) + R_{pas} \\ &= R_{act}(P_c(\tau)) - R_{act}(P_{c_{AT}}) + HL_{total}. \end{aligned} \tag{19}$$

In WHIS, $R_{total}(n_{ch}, P_c(\tau))$ is calculated for each filterbank channel n_{ch} with the estimated signal level $P_c(n_{ch}, \tau)$.

4) Lower limit of α

When using WHIS, users can arbitrarily set the α value in (10). Although it is desirable to determine α from the experiments, the estimation is not very easy, as described in Section II-C3. When the α value and HL_{total} are small, HL_{act} calculated from (16) and (17) may become greater than HL_{total}. This case no longer satisfies the constraint of $HL_{pas} \geq 0$ in (9). Therefore, the α value is automatically increased to the lower limit which satisfies the condition $HL_{total} = HL_{act}$ and $HL_{pas} = 0$. An example of the relationship between HL_{act} and HL_{total} is shown in an audiogram of Fig. 10 in Appendix B. The curve of HL_{act} cannot lie below that of HL_{total}.

D. SYNTHESIS

The analysis part in Fig. 5 produces two types of outputs, which correspond to the two synthesis methods.

1) Direct time-varying filter

One of the synthesis methods is to apply a nonlinear, time-varying filter to input signals, as used in the previous WHIS [15]. This is referred to as direct time-varying filtering (DTVF) hereafter. The filter coefficients are calculated from the frame-based level reduction $R_{total}(n_{ch}, P_c(\tau))$ in (19). Figure 6 presents a schematic of signal processing. The input signal is divided into frames with a square-root hanning window, $w(t) = \sqrt{0.5 + 0.5\cos(2\pi t/T)}$ $\{t | -T/2 \leq t \leq T/2\}$, where the frame length, T, is 20 ms and the frame shift is 10 ms. The framed signal is convoluted with a minimum phase filter, which is described in the next paragraph. The filtered signal is windowed again with the same square-root hanning window, $w(t)$. Then, the frames are overlapped and added to produce the output signal. When the minimum phase filter is an impulse, the input and output signals are identical as this procedure is equivalent to processing with a hanning window with half overlapping.

The minimum phase filter is derived from the output of the path labeled DTVF in Fig. 5. $R_{total}(n_{ch}, P_c(\tau))$ in (19) is interpreted as the spectral distribution of the reduction function along the filter channel, n_{ch} (i.e., on the $\mathrm{ERB_N number}$ axis). This distribution is converted into the power spectrum on the linear frequency axis by using a warping function from Cam on $\mathrm{ERB_N number}$ to Hz. Then, the minimum phase filter is derived from this power spectrum using the cepstrum method.

WHIS with this DTVF synthesis method is referred to as $\mathrm{WHIS_d}$. The results of the preliminary listening tests indicated that $\mathrm{WHIS_d}$ did not produce noticeable distortion in the output sounds, which is attributable to a single time-varying filter between the input and output for each frame. The filter has a minimum phase response that does not produce preecho, which might be perceived as distortion.

2) Filterbank synthesis

Filterbank synthesis is an alternative method. The output sound is synthesized by an overlap-and-add method, which is commonly used and is similar to that in [16], [29]. In the current implementation for fast processing, the phase delay of the output waveform from the individual filterbank channel is compensated for a constant reciprocal to the center frequency of the corresponding gammachirp filter. Then, the compen-

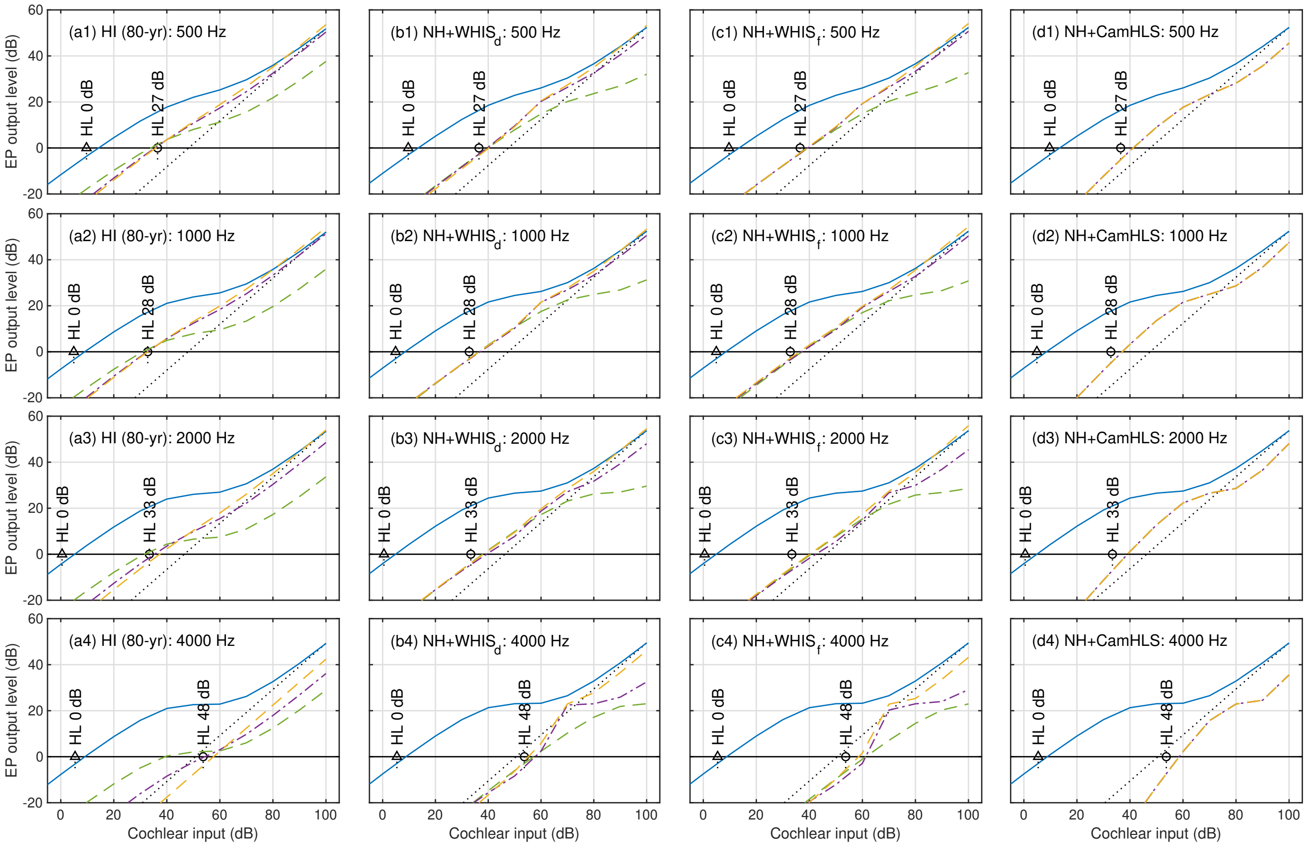

FIGURE 7. Simulation results on the IO functions at 500, 1000, 2000, and 4000 Hz. The abscissa, the ordinate, and HL 0 dB are the same as in Fig. 3. The HL value in each panel corresponds to the average HL of 80-yr listeners [26] at that frequency. The panels show the IO functions of the HI listener (a1)–(a4), $\mathrm{NH} + \mathrm{WHIS_d}$ (b1)–(b4), $\mathrm{NH} + \mathrm{WHIS_d}$ (c1)–(c4), and $\mathrm{NH} + \mathrm{CamHLS}$ (d1)–(d4). Blue solid line: the IO function of the NH condition. Green dashed line: HL with a compression health α of 1. Purple dashed-and-dotted line: HL with α of 0.5. Orange dashed line: HL with α of 0. Black dotted line: The linear relationship (1:1).

sated waveforms are added together to synthesize the output. The WHIS process with this filterbank analysis synthesis (FBAS) method is referred to as $\mathrm{WHIS_f}$. The process in the single channel is shown in the path labeled FBAS in Fig. 5. The amplitude of the output waveform of the linear cGC filter is reduced by $R_{act}(n_{ch}, P_c(\tau))$ and $R_{pas}(n_{ch})$, as defined in (19). The process is performed with adequate resampling from the frame rate to the signal sampling rate.

$\mathrm{WHIS_f}$ can accommodate the temporal smearing method within a single framework although it is difficult for $\mathrm{WHIS_d}$. The method is similar to that used by [30] in which temporal characteristics of HI listeners could be simulated. The envelope is extracted from the filterbank output by Hilbert transformation or rectification and is filtered using a lowpass filter designed to reduce the temporal resolution. The original carrier component and the reduced envelope are used to synthesize the output sound.

In the preliminary listening tests, the output sounds in $\mathrm{WHIS_f}$ are slightly distorted, even without any temporal smearing. The distortion level is slightly higher than that in $\mathrm{WHIS_d}$. The phase compensation across the filter channels is not perfectly performed, probably because the temporal response of the cGC filter in Fig.5 is determined not only by the center frequency but also by the compression health α. Thus, to achieve better quality, more sophisticated processing is needed.

IV. EVALUATION OF WHIS

The simulated HL sounds of speech are evaluated to clarify the potential and limit of the three HL simulators: $\mathrm{WHIS_d}$, $\mathrm{WHIS_f}$, and CamHLS [9] described in Section I. The evaluation was performed for the similarities in the IO functions and EPgrams between $EP^{(HI)}$ and $EP^{(NH+WHIS)}$, as shown in Fig. 4.

A. IO FUNCTION

The IO function of sinusoidal signals is one of the important characteristics for simulating cochlear signal processing. Figures 7 (a1)–(a4) show the IO functions derived from GCFB under the average hearing level of 80-yr listeners shown in Table 1 [26] at 500, 1000, 2000, and 4000 Hz. The intercept points of the IO functions and the 0-dB line were sufficiently close to the hearing levels of the NH and 80-yr listeners

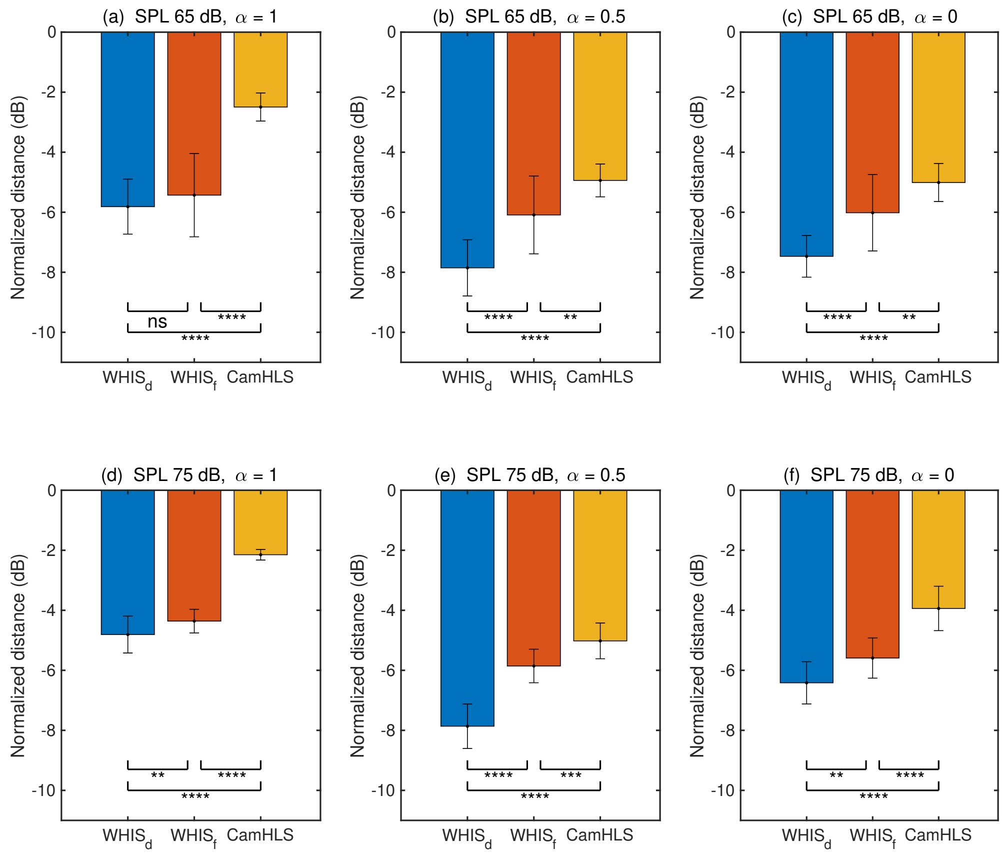

FIGURE 8. Normalized RMS spectral distance d_s (dB). Bars and error bars represent the mean and standard deviation (SD). The conditions for the SPL and α are labeled at the top of each panel. The HL simulator conditions are $WHIS_d$ (blue), $WHIS_f$ (red), and CamHLS (orange). Tukey's HSD test, ****: $p < 0.0001$, ***: $p < 0.001$, **: $p < 0.01$, ns: not significant.

as intended. The differences between them are less than 5 dB, which is the resolution of a common audiometer [28], independent of the α value except for the condition of 4 kHz and $\alpha = 1$. The question here is whether these IO functions are properly simulated by the combination of WHIS and GCFB under the NH setting.

Figures 7 (b1)–(b4) and (c1)–(c4) show the IO functions derived by $\mathrm{WHIS_d}$ and $\mathrm{WHIS_f}$, respectively. There are very small differences between the corresponding IO functions, and therefore, the synthesis method does not affect the IO functions of sinusoids. When compared with Fig. 7 (a1)–(a4), the IO functions for α in (10) of 0.0 and 0.5 are similar, while those for α of 1.0 are very different. The locations of the compressive regions are higher in Figs. 7 (b1)–(b4) and (c1)–(c4) than in Figs. 7 (a1)–(a4). This is caused by the difference in the gain function in (13) and the reduction function in (19). Therefore, it is difficult to simulate the HL when the active process of an HI listener is fairly healthy as in an NH listener. This is the case for some hidden hearing-loss patients [31]. This is not solely a problem of WHIS. The same problem occurs in other HL simulators that simply simulate loudness recruitment.

Figures 7 (d1)–(d4) show the IO functions derived by CamHLS in which the IO function is automatically determined from the given audiogram in the default set. There is one IO function for the HI condition in each frequency because it was difficult to control it due to insufficient information. However, the IO functions are similar to those in $\mathrm{WHIS_d}$ and $\mathrm{WHIS_f}$ at $\alpha = 0.5$. This result implies that WHIS can also simulate loudness recruitment, as CamHLS was developed for this purpose [9].

B. DISTANCE BETWEEN EPGRAMS

1) Method

The spectral distances between EPgrams for the 80-yr HI listener and the NH listener with WHIS or CamHLS were evaluated using speech sounds. The reference EPgrams, $S^{(HI)}(n_{ch}, \tau)$, were calculated from $EP^{(HI)}$ in Fig. 4(a) and Fig. 7 (a1)–(a4). GCFB was set to the 80-yr hearing level (Table 1) with α values of 1.0, 0.5, and 0.0. The simulated EPgrams, $S^{(Sim)}(n_{ch}, \tau)$, were calculated from $EP^{(NH+WHIS)}$ in Fig. 4(b) when using $\mathrm{WHIS_d}$ (Fig. 7 (b1)–(b4)), $\mathrm{WHIS_d}$ (Fig. 7 (c1)–(c4)), and CamHLS (Fig. 7 (d1)–(d4)). WHIS and CamHLS were set to the 80-yr hearing level. WHIS was also set with α values of 1.0, 0.5, and 0.0. GCFB was set to the NH hearing level (HL 0 dB) with an α

value of 1.

Speech sounds of 20 words pronounced by 2 males and 2 females were used for the evaluation. The speech sounds were normalized at the SPLs of 65 dB and 75 dB in L_{eq} (i.e., RMS level). The normalized distance, d_s (dB), was calculated after the time alignment of the input and output sounds as

$$d_s = 20\log_{10}\frac{\mathrm{rms}\{S^{(Sim)}(n_{ch},\tau) - S^{(HI)}(n_{ch},\tau)\}}{\mathrm{rms}\{S^{(HI)}(n_{ch},\tau)\}} \tag{20}$$

As a result, the d_s values of 20 words were derived for each simulation condition.

2) Result

Figure 8 shows the results of the normalized spectral distance d_s (dB). First of all, the mean distances derived from $\mathrm{WHIS_d}$ and $\mathrm{WHIS_f}$ were significantly smaller than those from CamHLS independently of the α value and the SPL. This result implies that WHIS can simulate the EPgrams of the HI listener better than CamHLS. The distances of $\mathrm{WHIS_d}$ were significantly smaller than those of $\mathrm{WHIS_f}$ except for the condition of SPL 65 dB and α of 1 as shown in Fig. 8(a). This result implies that $\mathrm{WHIS_d}$ is better than $\mathrm{WHIS_f}$ in speech processing. The distances were generally smaller at SPL 65 dB than at SPL 75 dB. This is probably because the difference in the IO functions between $EP^{(HI)}$ and $EP^{(NH+WHIS)}$ or $EP^{(NH+CamHLS)}$ in Fig. 7 was greater at higher input levels.

3) Preliminary results on sound quality

In the preliminary listening, the sound quality of CamHLS was worse than that of $\mathrm{WHIS_d}$ and $\mathrm{WHIS_f}$. This is mainly because spectral smearing is introduced into CamHLS to simulate the bandwidth widening in the HI listeners [7], [8]. The process of the spectral smearing produces distortion, which is described in the software (control_impaired_ear.m) as follows: *"The 'spectral smearing' software has the perceptual effect of adding a certain 'grittiness' to the audio quality."* This distortion can be avoided and the spectral distance can be reduced by removing the spectral smearing process. However, the method for controlling the process was not clearly defined in the software. As described in Section III-D2, the sound quality of $\mathrm{WHIS_f}$ was slightly worse than that of $\mathrm{WHIS_d}$. The perceptual distortion may be related to the spectral distance, at least partially. Formal listening experiments and objective evaluations to examine the relationship are planned in the next study.

V. CONCLUSIONS

A new version of WHIS was developed based on the idea of the EP playback instead of the direct simulation of loudness recruitment in conventional methods. WHIS synthesizes simulated HL sounds to make the EPs of an NH listener sufficiently close to the EPs of an HI listener by applying the active and passive level reduction of input signals. The active reduction can be simply formulated by using the composite function of the HI IO function and the inverse NH IO function. The passive reduction was determined to maintain the hearing level as shown in the audiogram of the HI listener. Two methods, DTVF and FBAS, were used for sound synthesis. WHIS was compared with CamHLS in terms of differences in the IO functions and the spectral distance between EPgrams of the HI listener and the NH listener with using the simulator. WHIS yielded a significantly smaller distance than CamHLS. The results imply that the EP playback is an effective method for the HL simulation. The software of WHIS and GCFB is available in our GitHub repository [32].

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REFERENCES

[1] B.C.J. Moore, An introduction to the psychology of hearing, Brill, Leiden, The Netherlands, 6th edition, 2013.
[2] E. Villchur, "Simulation of the effect of recruitment on loudness relationships in speech," J. Acoust. Soc. Am., vol. 56, no. 5, pp. 1601–1611, 1974.
[3] B.C.J. Moore and B.R. Glasberg, "Simulation of the effects of loudness recruitment and threshold elevation on the intelligibility of speech in quiet and in a background of speech," J. Acoust. Soc. Am., vol. 94, no. 4, pp. 2050–2062, 1993.
[4] R.D. Patterson, "Auditory filter shapes derived with noise stimuli," J. Acoust. Soc. Am., vol. 59, no. 3, pp. 640–654, 1976.
[5] R.D. Patterson, I. Nimmo-Smith, D.L. Weber, and R. Milroy, "The deterioration of hearing with age: Frequency selectivity, the critical ratio, the audiogram, and speech threshold," J. Acoust. Soc. Am., vol. 72, no. 6, pp. 1788–1803, 1982.
[6] B.R. Glasberg and B.C.J. Moore, "Auditory filter shapes in subjects with unilateral and bilateral cochlear impairments," J. Acoust. Soc. Am., vol. 79, no. 4, pp. 1020–1033, 1986.
[7] T. Baer and B.C.J. Moore, "Effects of spectral smearing on the intelligibility of sentences in noise," J. Acoust. Soc. Am., vol. 94, no. 3, pp. 1229–1241, 1993.
[8] T. Baer and B.C.J. Moore, "Effects of spectral smearing on the intelligibility of sentences in the presence of interfering speech," J. Acoust. Soc. Am., vol. 95, no. 4, pp. 2277–2280, 1994.
[9] Y. Nejime and B.C.J. Moore, "Simulation of the effect of threshold elevation and loudness recruitment combined with reduced frequency selectivity on the intelligibility of speech in noise," J. Acoust. Soc. Am., vol. 102, no. 1, pp. 603–615, 1997.
[10] M.A. Stone and B.C.J. Moore, "Tolerable hearing aid delays. i. estimation of limits imposed by the auditory path alone using simulated hearing losses," Ear and Hearing, vol. 20, no. 3, pp. 182–192, 1999.
[11] Clarity Challenge 2022, "https://claritychallenge.org," (Last: 24 Oct 2022).
[12] P.M. Zurek and J.G. Desloge, "Hearing loss and prosthesis simulation in audiology," The Hearing Journal, vol. 60, no. 7, pp. 32–38, 2007.
[13] T. Irino, T. Fukawatase, M. Sakaguchi, R. Nisimura, H. Kawahara, and R.D. Patterson, "Accurate estimation of compression in simultaneous masking enables the simulation of hearing impairment for normal-hearing listeners," in Basic Aspects of Hearing, pp. 73–80. Springer, 2013.
[14] M. Nagae, T. Irino, R. Nisimura, H. Kawahara, and R.D. Patterson, "Hearing impairment simulator based on compressive gammachirp filter," in APSIPA ASC 2014, 2014.
[15] T. Irino and R.D. Patterson, "The gammachirp auditory filter and its application to speech perception," Acoust. Sci. and Technol., vol. 41, no. 1, pp. 99–107, 2020.
[16] T. Irino and R.D. Patterson, "A dynamic compressive gammachirp auditory filterbank.," IEEE Trans. ASLP, vol. 14, no. 6, pp. 2222–2232, 2006.
[17] T. Irino and R.D.. Patterson, "A time fomain, level-dependent auditory filter: the gammachirp," J. Acoust. Soc. Am., vol. 101, no. 1, pp. 412–419, 1997.

[18] R.D. Patterson, M. Unoki, and T. Irino, "Extending the domain of center frequencies for the compressive gammachirp auditory filter," J. Acoust. Soc. Am., vol. 114, no. 3, pp. 1529–1542, 2003.
[19] M. Unoki, T. Irino, B. Glasberg, B.C.J. Moore, and R.D. Patterson, "Comparison of the roex and gammachirp filters as representations of the auditory filter," J. Acoust. Soc. Am., vol. 120, no. 3, pp. 1474–1492, 2006.
[20] H. Hu, J. Sang, M.E. Lutman, and S. Bleeck, "Simulation of hearing loss using compressive gammachirp auditory filters," in ICASSP 2011. IEEE, 2011, pp. 5428–5431.
[21] N. Grimault, T. Irino, S. Dimachki, A. Corneyllie, R.D. Patterson, and S. Garcia, "A real time hearing loss simulator," Acta Acustica united with Acustica, vol. 104, no. 5, pp. 904–908, 2018.
[22] F.S. Cooper, A.M. Liberman, and J.M. Borst, "The interconversion of audible and visible patterns as a basis for research in the perception of speech," Proc. Nation. Acad. Sci., vol. 37, no. 5, pp. 318–325, 1951.
[23] T. Irino and R.D. Patterson, "A compressive gammachirp auditory filter for both physiological and psychophysical data," J. Acoust. Soc. Am., vol. 109, no. 5, pp. 2008–2022, 2001.
[24] H. Fletcher, "Auditory patterns," Reviews of modern physics, vol. 12, no. 1, pp. 47, 1940.
[25] ISO 7029:2017, "Acoustics — statistical distribution of hearing thresholds related to age and gender," ISO, 2017.
[26] T. Tsuiki, S. Sasamori, Y. Minami, T. Ichinohe, K. Murai, S. Murai, and H. Kawashima, "Age effect on hearing: a study on japanese," Audiology Japan (in Japanese), vol. 45, no. 3, pp. 241–250, 2002.
[27] B.C.J. Moore, B.R. Glasberg, and T. Baer, "A model for the prediction of thresholds, loudness, and partial loudness," Journal of the Audio Engineering Society, vol. 45, no. 4, pp. 224–240, 1997.
[28] ANSI/ASA S3. 6 - 2018, "Specification for audiometers," 2018.
[29] T. Irino and M. Unoki, "An analysis/synthesis auditory filterbank based on an IIR implementation of the gammachirp," J. Acoust. Soc. Jpn. (E), vol. 20, no. 6, pp. 397–406, 1999.
[30] R. Drullman, J.M. Festen, and R. Plomp, "Effect of temporal envelope smearing on speech reception," J. Acoust. Soc. Am., vol. 95, no. 2, pp. 1053–1064, 1994.
[31] M.C. Liberman, "Hidden hearing loss," Scientific Am., vol. 313, no. 2, pp. 48–53, 2015.
[32] "https://github.com/amlab-wakayama/," (15 Apr 2023).
[33] T. Matsui, T. Irino, M. Nagae, H. Kawahara, and R.D. Patterson, "The effect of peripheral compression on syllable perception measured with a hearing impairment simulator," in Physiology, Psychoacoustics and Cognition in Normal and Impaired Hearing, pp. 307–314. Springer, Cham, 2016.
[34] T. Irino, Soichi Higashiyama, and Hanako Yoshigi, "Speech clarity improvement by vocal self-training using a hearing impairment simulator and its correlation with an auditory modulation index.," in Proc. Interspeech 2020, 2020, pp. 2507–2511.
[35] T. Irino, H. Tamaru, and A. Yamamoto, "Speech intelligibility of simulated hearing loss sounds and its prediction using the Gammachirp Envelope Similarity Index (GESI)," in Proc. Interspeech 2022, 2022, pp. 3929–3933.
[36] J. Hasegawa, M. Hashi, T. Matsui, and T. Irino, "Application of a hearing loss simulator to education, clinic, and research and its evaluation by speech-language-hearing therapists (in japanese)," in Proc. Japanese Association of Speech-Language-Hearing Therapists, 2019, pp. 1–P03–4.

APPENDIX A BANDWIDTH OF THE CGC FILTER

As described in Section II-B, the cGC filter which comprises pGC and HP-AF is implemented as a linear filter for speed-up. The filter shapes of HI and NH listeners are determined from the α value in (10). Although the filter gain is controlled level-dependently by the active gain function shown in Fig. 1, the bandwidth is level independent in this implementation.

Figure 9 shows the bandwidth of the cGC filter as a function of α at signal frequencies of 1 kHz and 4 kHz. The vertical axis is the relative bandwidth normalized by the bandwidth at $\alpha = 1$ (i.e., a completely healthy NH condition). The lower curves show that as α decreases from 1.0 to 0, the bandwidth increases gradually from 1.0 to 1.4 times. At $\alpha = 0$ (i.e., a completely damaged HI condition), the bandwidth is equal to that of the pGC filter because $c_2^{(HI)} = 0$ in (10) and the frequency response of the HP-AF filter is unity (0 dB), as indicated in (3).

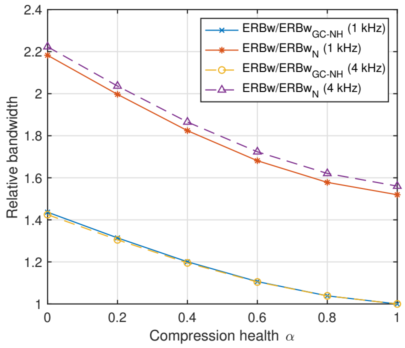

FIGURE 9. Bandwidth of the cGC filter as a function of the compression health, α. Solid lines: 1 kHz. Dashed lines: 4 kHz. The two lower lines show the bandwidth relative to the case when $\alpha = 1$. The two upper lines show the bandwidth relative to the standard $\mathrm{E}RB_N$ bandwidth of the average NH listener [1].

When the bandwidth is normalized by the standard $\mathrm{ERB_N}$ bandwidth [1], the relative bandwidth is 1.6 times wider (the two upper lines) than the value calculated above (the two lower lines). This is because the bandwidth of the cGC filter estimated with the NN masking paradigm by [18] is wider than the standard $\mathrm{ERB_N}$ bandwidth.

As a result, the bandwidth difference between NH and HI listeners is linearly introduced into this version of GCFB by controlling the α value. This approximation might not be entirely unreasonable, at least when simulating the HI listener's filter, because the level-dependent active gain is smaller in HI listeners than in NH listeners.

APPENDIX B APPLICATIONS OF WHIS

HL simulators have many potential applications. For example, Zurek and Desloge [12] indicated that HL and prosthesis simulations are useful in counseling, hearing aid fitting, training, hearing conservation, testing warning signals, and setting job requirements. This appendix introduces several experiments and practices performed with WHIS as examples for further studies.

A. SEVERAL EXAMPLES USING WHIS

Perceptual experiments on speech sounds are one of the most important applications. For this purpose, WHIS has been developed to minimize the distortion and noise, which might affect the experimental results. For example, an early version of WHIS was used to measure the effect of compression loss on syllable recognition [33]. It was also used in vocal self-training experiments to examine whether the speech clarity toward HI listeners was improved [34]. Speech intelligibility experiments using the current version of WHIS were performed in the laboratory and crowdsourced remote environments to develop a new objective measure, GESI

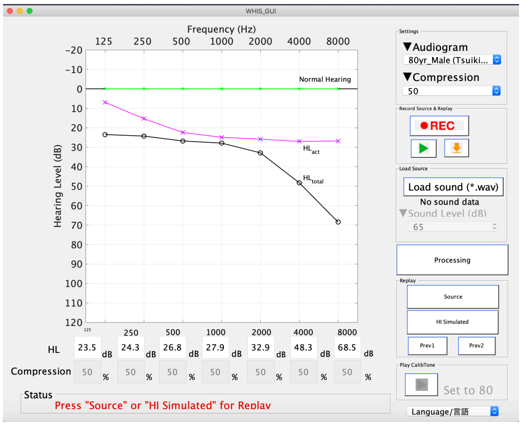

FIGURE 10. GUI of WHIS. The main panel represents the audiogram of an HI listener (HL_{total}; black line), the active hearing loss (HL_{act}; magenta line), and the audiogram of an NH listener ($HL = 0$; green line). There are several sets of control buttons on the right.

[35]. The stimuli in these experiments were produced by using a batch program in WHIS.

B. INTERACTIVE USE WITH GUI

For interactive usage, a graphical user interface (GUI) was developed for WHIS as shown in Fig. 10. This GUI version has been used in a training program for speech-language-hearing therapists for several years [36]. The GUI has a main audiogram panel and several sets of control buttons. After calibration of the SPL, the user chooses an audiogram, which represents the hearing level, and a value of the compression health α. The audiogram is plotted with the black line labeled with HL_{total} in the main panel. Then, R_{act} and HL_{act} (the magenta line) are calculated from (16) and (17) in accordance with the α value. Note that the α value at each audiogram frequency is automatically compensated to satisfy $HL_{act} \leq HL_{total}$, as described in Section III-C4. At $\alpha = 1$, HL_{act} (the magenta line) is 0 dB (the green line). The difference between HL_{total} and HL_{act} corresponds to R_{pas} as shown in (18). After the parameter setting is finished, the user can record speech sound or load prerecorded speech, and then listen to the simulated HL sound.

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