

OCCLUSION REDUCTION SYSTEM FOR HEARING AIDS WITH AN IMPROVED TRANSDUCER AND AN ASSOCIATED ALGORITHM

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ABSTRACT

Many of hearing aid users have complained about discomfort of their own voice and/or the mastication sound. Such discomfort is caused by increased sound pressure at low frequencies when the ear canal is blocked by hearing aid itself. This phenomenon is called “occlusion effect” and is one of the critical issues for hearing aids. This report proposes an occlusion reduction system based on active noise control technique using a new acoustic transducer. The proposed system can reduce the increased sound pressure in the ear canal about 26 dB around 200 Hz. While the proposed system achieved a better performance in the reduction, some distorted sounds are frequently perceived through the system. This secondary issue of the distorted sounds can also be reduced by controlling the feedback loop gain. Finally, a prototype system with the new transducer and a distortion suppressor algorithm is developed and then evaluated.

Index Terms— Hearing aids, Occlusion effect, Active noise control, Improved acoustic transducer, Distortion

1. INTRODUCTION

When the ear canal is completely or partially blocked by hearing aids, one’s own voice and/or mastication sounds are perceived as echoes or hollow sounds [1–4]. This phenomenon is referred to as “occlusion effect” [4] for which hearing aid users complain about discomfort of the annoying or unnatural sounds at low frequencies. Consequently, this discomfort often keeps the users away from wearing hearing aids. Fig. 1 shows a cross-section view of the ear canal with in-the-ear (ITE) type of hearing aid. Vibrations due to the internal body are frequently generated and propagated as bone conduction sounds to a wall of the ear canal. When the ear canal is completely blocked by hearing aid, the sound pressure derived from the bone conduction around 200 Hz increases about 15 to 30 dB or even more, comparing with opened condition [2–4]. There has been an attempt to reduce the occlusion effect by installing a small pipe in hearing aid and venting the internal air to the outside as shown in Fig. 1. By this attempt, however, there is a risk of inducing oscillation sounds due to acoustic feedback in case that the diameter

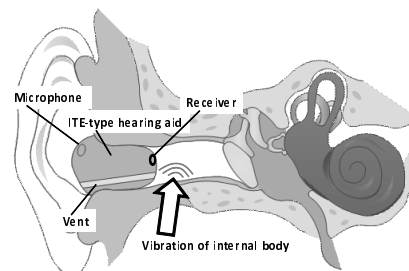


Fig. 1. Cross-section view of the ear canal with ITE type of hearing aid.

of the pipe is too large. Furthermore, this attempt would limit the ability of hearing aids to achieve sufficient gain at low frequencies [4, 5]. The aim of this study is to develop occlusion reduction system for digital hearing aids, which can be regarded as an electrical embodiment of acoustically transparent states in the ear canal blocked by hearing aid completely. This study should be applicable to any types of in-ear headphone or hearing protector.

Several studies have been conducted for occlusion reduction systems in hearing aids based on active noise control technique [2, 6]. However, amount of the reduction of those systems were not sufficient to resolve the severe occlusion such as 30 dB around 200 Hz [3]. One of the reasons why their performance was limited is a relatively large phase lag of transfer function between the receiver and the microphone [7]. To overcome this issue, a system using adaptive filter technique was proposed [3]. However, it was unrealistic to implement the system at this time as the computational cost was extremely high.

This report presents an occlusion reduction system for digital hearing aids utilizing a new acoustic transducer to reduce the increased sound pressure in the blocked ear canal. Although the performance of reduction might be improved by the system, some distorted or click-like sounds are frequently observed due to the insufficient capability of the maximum output power in the receiver. The distorted sounds can be suppressed if a large-sized receiver is employed. However, it would be difficult for the receiver to be incorporated into ITE type of hearing aids. It is also considered that the distorted

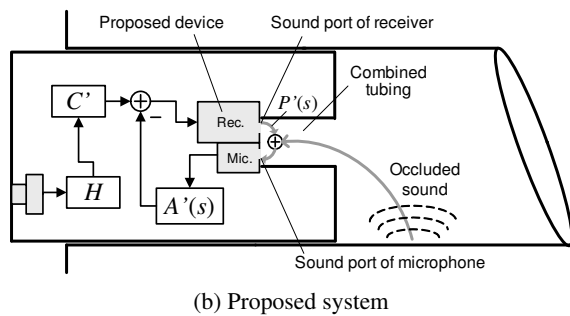
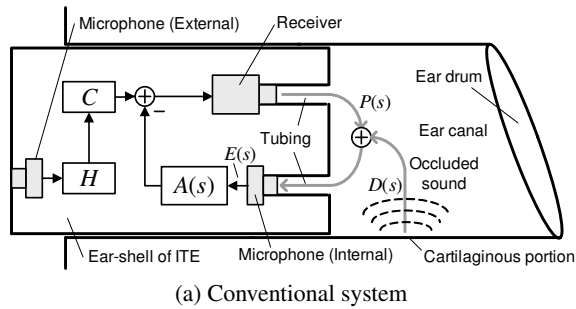


Fig. 2. Configuration of occlusion reduction system.

sounds can be reduced by an algorithm which controls the level of output signal instead of using the large-sized receiver.

2. OCCLUSION REDUCTION SYSTEM

2.1. Conventional systems

Figure 2 (a) depicts a configuration of conventional occlusion reduction system incorporated in ITE type of hearing aid. The system consists of a microphone (external), a receiver, a digital signal processor and a microphone (internal) for the specific purpose of picking up occluded sound. $P(s)$ is a transfer function between the receiver and the internal microphone. H represents general process of the hearing aid, and $A(s)$ indicates the negative feedback control filter. The predicted sound pressure reduction in the ear canal using the feedback loop is shown as (1),

$$\frac{E(s)}{D(s)} = \frac{1}{1 + A(s)P(s)}, \quad (1)$$

where $E(s)$ is the error signal picked up by the internal microphone and $D(s)$ is a signal of the occluded sound generated in the ear canal. The feedback control filter $A(s)$ should be designed to have as much negative feedback gain as possible without any oscillations. At the same time, the open loop transfer function $A(s)P(s)$ should be satisfied with the Nyquist stability criterion [8]. In practical implementation, the feedback control filter $A(s)$ can be expressed as $A(z)$ in terms of z-transform since it is configured with a digital filter. The latency of the DA/AD convertors causes the negative

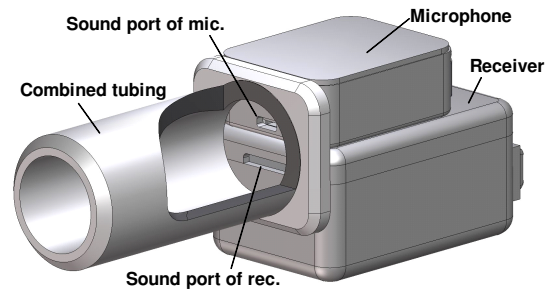


Fig. 3. Appearance of the proposed acoustic transducer.

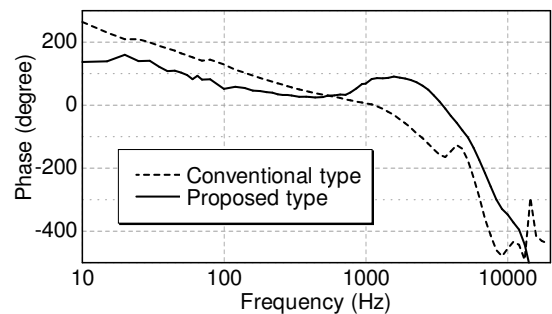


Fig. 4. Phase responses of $P(s)$ and $P'(s)$.

effect to the phase lag. The feedback loop would also reduce the output signal of hearing aid. However, this effect can be corrected by a pre-compensation filter C as shown in Fig. 2, which can be obtained as the inverse of (1).

The conventional system shown in Fig. 2 (a) has two separate tubings at the sound ports of the receiver and the internal microphone. Each sound port is connected to space in the ear canal independently. However, it is difficult for the conventional system to obtain feedback gain sufficiently since the path between the receiver and the internal microphone is complicated and the phase lag of the transfer function $\angle P(s)$ becomes larger (indicated as dashed line in Fig. 4).

2.2. Proposed system

Figure 2 (b) depicts a configuration of proposed occlusion reduction system in the same manner of Fig. 2 (a). A major difference between Fig. 2 (a) and (b) is that the proposed system introduces a new structure of the receiver and the internal microphone. Figure 3 shows an appearance of the proposed acoustic transducer. The transducer has only one tubing that is commonly used as the sound port of the receiver and the internal microphone. This structure improves phase response $\angle P'(s)$ as the path between the receiver and the internal microphone is shorter and simpler. Figure 4 shows a comparison between phase response $\angle P(s)$ in conventional system and $\angle P'(s)$ in the proposed system. From the figure, it is observed that $\angle P'(s)$ lower than 4 kHz is flatter than $\angle P(s)$. It can be said that the proposed system have a larger nega-

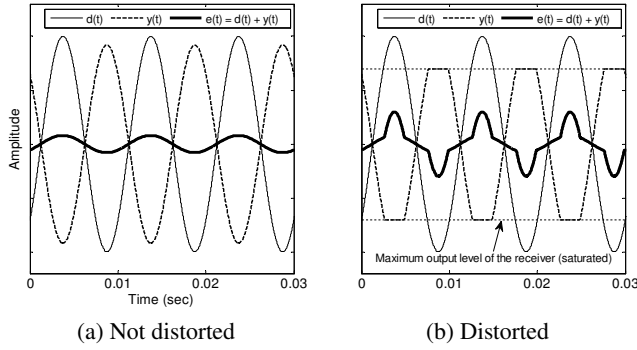


Fig. 5. Graphical explanations for generation of the distorted sounds.

tive feedback gain in $A'(s)$ without any oscillations compared with conventional system.

3. SUPPRESSING DISTORTED SOUNDS

3.1. Distorted sounds in the proposed system

When a larger feedback gain is applied to the proposed system, some distorted or click-like sounds can be frequently perceived as annoying sound under particular conditions such as uttering, jaw moving, yawning, etc. Figure 5 shows graphical explanations of how the distorted sounds are generated. The solid, dashed, and bold solid lines in the figure indicate the occluded sound $d(t)$, the receiver output sound $y(t)$, and sound in the ear canal $e(t) = d(t) + y(t)$, respectively. When the occlusion reduction system works properly, the sound similar to phase-inverted signal of the occluded sound is generated from the receiver, and then the sound pressure in the ear canal is reduced as shown in Fig. 5 (a). However, if the occluded sound pressure exceeds the maximum output level of the receiver, the receiver output sound $y(t)$ is saturated. Accordingly, the sound in the ear canal $e(t)$ gets distorted severely as shown in Fig. 5 (b).

3.2. Algorithm for suppressing distorted sounds

Figure 6 shows a block diagram of proposed system including proposed distortion suppressor algorithm. The proposed algorithm controls a broadband gain of the feedback loop $k(n)$ in order to prevent $y'(n)$ from saturation by the maximum output level of the receiver. Algorithm 1 shows a specific process of the distortion suppressor at “block A” shown in Fig. 6. In Algorithm 1, L_{th} is amplitude detection threshold for $y'(n)$, α and β are constants controlling the variation speed of gain $k(n)$, and k_{min} and k_{max} indicate lower and upper limits of gain $k(n)$, respectively. When amplitude of the signal $y'(n)$ exceeds L_{th} , gain $k(n)$ decreases as following equation;

$$k(n) = k(n-1) - \alpha \frac{|y'(n)|}{\sum y'(n)/M}. \quad (2)$$

Algorithm 1 Distortion suppressor algorithm

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 $y'(n) \leftarrow \mathbf{0}_{1 \times M}$ 
for all  $n$  do
   $y'(n) \leftarrow [|y'(n)|, |y'(n-1)|, \dots, |y'(n-M)|]$ 
  if  $|y'(n)| > L_{th}$  then {Decreasing the gain  $k(n)$ }
     $k(n) \leftarrow k(n-1) - \alpha \frac{|y'(n)|}{\sum y'(n)/M}$ 
    if  $k(n) < k_{min}$  then {Lower limiting}
       $k(n) \leftarrow k_{min}$ 
    end if
  else {Increasing the gain  $k(n)$ }
     $k(n) \leftarrow k(n-1) + \beta$ 
    if  $k(n) > k_{max}$  then {Upper limiting}
       $k(n) \leftarrow k_{max}$ 
    end if
  end if
end for

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When $y'(n)$ increases rapidly, the gain $k(n)$ decreases faster. Gain $k(n)$ is recovered to the maximum feedback loop gain when $y'(n)$ is maintained below L_{th} .

3.3. Computer simulation

A computer simulation by MATLAB was carried out to evaluate the proposed distortion suppressor algorithm. Transfer function $P'(s)$ for a subject was measured and represented by an FIR filter (1024 taps). A male voice sound / i / was prepared as the increased sound pressure by one's own voice $d(t)$. The sampling rate for feedback loop was 128 kHz, however, the rate for distortion suppressor algorithm was 16 kHz with consideration for limited instruction set of a DSP (On Semiconductor: Ezairo 7100) for 128 kHz. Table 1 lists parameters used in the simulation.

Figure 7 (a) shows a waveform of the distorted signal $y'(n)$ in case that the proposed algorithm was not applied. On the other hand, the signal $y'(n)$ was not saturated in case that the proposed algorithm was applied as shown in Fig. 7 (b). It can be said that gain $k(n)$ was controlled appropriately and the distorted sound was hardly generated in the ear canal. From Fig. 7 (c), gain $k(n)$ decreased about 6 dB at maximum and the performance of the occlusion reduction was degraded temporarily. Subjective evaluation of the proposed algorithm is described in Sec. 4.3.

4. EVALUATION

4.1. Prototype system

Prototype of occlusion reduction system including the proposed acoustic transducer in Figs. 2 and 3 was developed for evaluation purpose. General-purpose ear tips made of urethane were chosen as the ear-shell of the hearing aid. An evaluation board of the DSP (On semiconductor: Ezairo 7100)

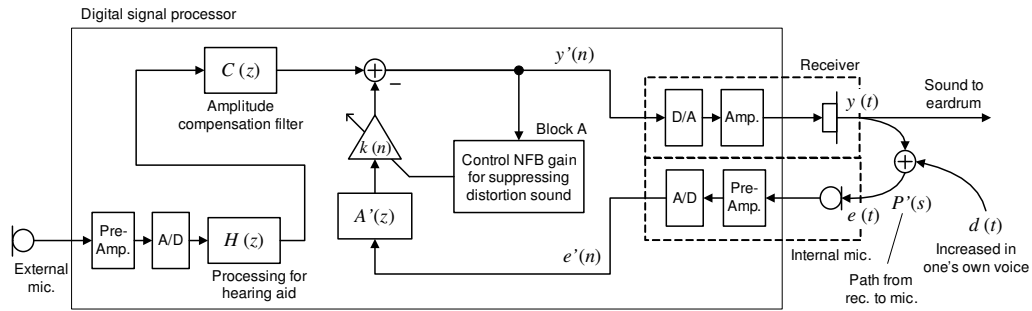


Fig. 6. Signal block diagram of the proposed system including proposed distortion suppressor algorithm.

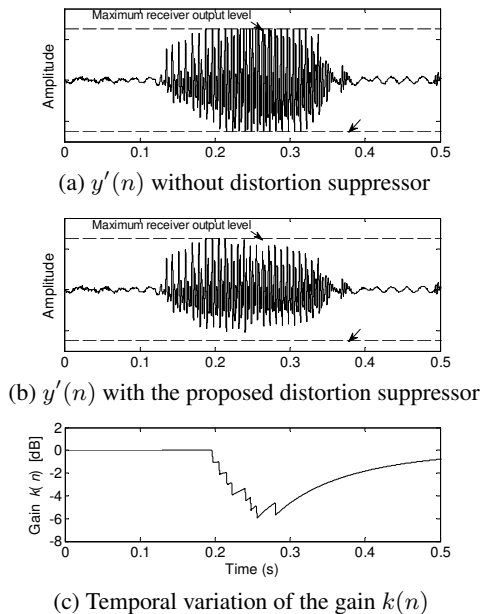


Fig. 7. Results of the computer simulation.

was used. The feedback control filters $A(z)$ and $A'(z)$ could not be installed in the ear shell as the evaluation board was too large. The filters $A(z)$ and $A'(z)$ were designed so as to have as large negative feedback gain as possible within stable conditions respectively. The latency of the AD/DA was $47\mu\text{s}$.

4.2. Comparison in the performance between conventional system and proposed system

Figure 8 shows a measured Nyquist plot of the open loop transfer functions $A(s)P(s)$ and $A'(s)P'(s)$ for a subject who has normal hearing. The Nyquist plot of proposed system resulted in a larger circle around 100 to 400 Hz compared with that of conventional system. This result indicates that the larger negative feedback gain can be applied for the proposed system in the frequency range where the occlusion problems are frequently occurred.

To verify the difference between conventional system and proposed system, the reduction of occluded sounds were mea-

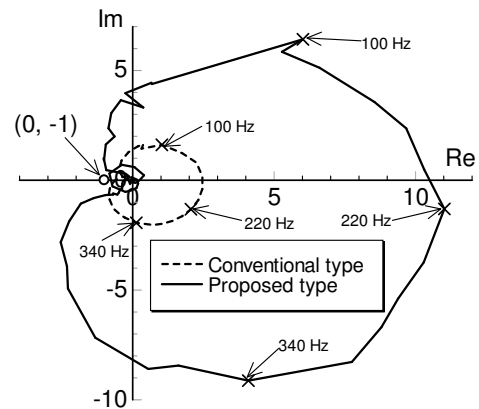


Fig. 8. Nyquist plot of open loop.

sured for both systems without distortion suppressor algorithm. A subject was asked to utter a vowel / i : / continuously. A small electret condenser microphone was prepared to measure the sound pressures in the ear canal. Then, the gain of reduction that is the difference in power spectra at the fundamental frequency of the vowel was calculated in two cases such as ON and OFF for conventional and proposed systems. As shown in Fig. 9, the proposed system had the gain of reduction more than 20 dB around 100 to 300 Hz. Especially, the gain of reduction was 26 dB at 200 Hz, which is about 15 dB larger than that of conventional system.

Variable	Symbol	Value
Sampling rate for feedback loop	F_s	128 kHz
Sampling rate for distortion suppressor	F'_s	16 kHz
Amplitude threshold level	L_{th}	0.38
Decreasing speed of the gain	α	-0.085
Increasing speed of the gain	β	0.030
Number of past sample for weighting	M	4
Lower limit of the gain	k_{min}	0.056
Upper limit of the gain	k_{max}	1.0 (0 dB)

Table 1. Parameter values used in the simulation.

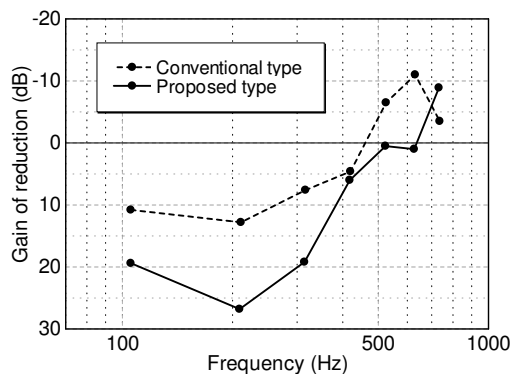


Fig. 9. Gain of occlusion reduction.

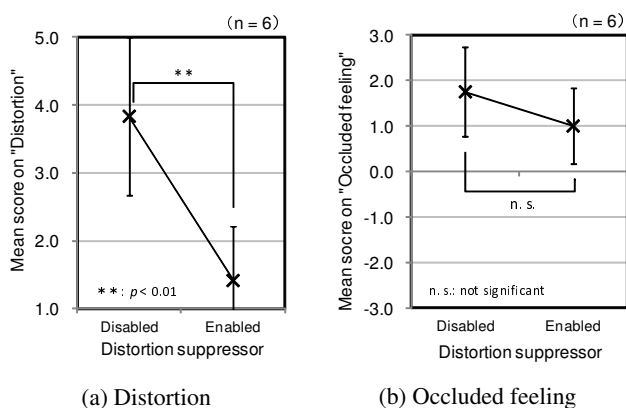


Fig. 10. Results of subjective evaluations.

4.3. Subjective evaluation for distortion suppressor algorithm

Subjective evaluations for the distortion suppressor algorithm in Sec. 3 were conducted. Subjects were six males with normal hearing and asked to pronounce a conversational sentence of five words in Japanese repeatedly with sound pressure level like a natural conversation. Then the subjects evaluated their own voice using two lists of rating words when the distortion suppressor algorithm was enabled or disabled. One list of rating word is about annoyance for the distorted sound and consists of five points; 1: Not annoying at all; 2: A little annoying; 3: Annoying; 4: Very annoying; 5: Extremely annoying. Another list of rating word is about occluded feeling when the system is switched from OFF to ON and consists of seven points; -3: Extremely increased; -2: Very increased; -1: Increased; 0: Moderate; +1: Decreased; +2: Very decreased; +3: Extremely decreased.

Figure 10 shows the results of subjective evaluations. In Fig. 10 (a), the mean scores on the distortion showed 1.4 point when the distortion suppressor algorithm was enabled, and the difference between enabled and disabled options showed 2.4 point. It can be said that the annoyance of the distorted sound was significantly suppressed using the distortion sup-

pressor algorithm. In Fig. 10 (b), however, the mean scores on the occlusion feeling slightly decreased 0.75 point when the distortion suppressor algorithm was enabled. This means that there was no significant difference between enabled and disabled options. A presumed reason could be that the gain of the feedback loop decreased temporarily by the distortion suppressor algorithm.

5. CONCLUSION

Occlusion reduction system based on active noise control technique with the new acoustic transducer and distortion suppressor algorithm was proposed. By the proposed system, the gain of reduction of a subject was 26 dB at 200 Hz, which is about 15 dB larger than that of conventional system. From subjective evaluations for the distortion suppressor algorithm, the annoyance for the distorted sound was significantly reduced, however, the occlusion feeling of the subjects slightly increased due to the temporal decrease of the feedback loop gain. These results suggested that the proposed system can reduce the increased sound pressure sufficiently even for those who experienced severe occlusion effect without any distorted sounds.

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