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(71) Applicant (for all designated States except US):
SORIZEN CORPORATION [KR/KR]; 3f, 751-5,
Bangbae-dong, Seocho-gu, Seoul 137-060 (KR).

(72) Inventor; and

(75) Inventor/Applicant (for US only): **MIN, Hoon** [KR/KR];
#302, 661-24, Myeonmok7-dong, Jungnang-gu, Seoul
131-207 (KR).

(74) Agent: **JIMYUNG PATENT FIRM**; 6F, Jungwoo
Bldg., 1689-1, Seocho4-dong, Seocho-gu, Seoul 137-882
(KR).

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(54) Title: NOISE CANCELLING APPARATUS

(57) Abstract: A noise cancelling apparatus is disclosed. In accordance with the apparatus, a noise cancellation signal is generated such that a residual noise signal is maintained to be below a minimum audible acoustic pressure level to reduce a noise, and an external noise is cancelled to hear only a desired audio signal when using a portable device using an earphone or a headphone.



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Description

NOISE CANCELLING APPARATUS

Technical Field

- [1] The present invention relates to a noise cancelling apparatus, and more particularly, to a noise cancelling apparatus wherein a noise cancellation signal is generated such that a residual noise signal is below a minimum audible acoustic pressure level to effectively reduce a noise.

Background Art

- [2] A noise generated by a noise source propagates through a vibration of air. The propagating noise vibrates a tympanic membrane in a human ear enabling a person to hear the noise.
- [3] In order to remove an undesirable noise, various technologies for cancelling the noise are used. In accordance with a conventional noise cancelling technology, a phase of a noise signal propagating through the air is inverted by 180 degrees, and the noise signal is superimpose over the inverted noise signal to cancel the noise.
- [4] However, when a person is wearing an earphone, problems described below may occur. The noise generated by the noise source and propagating through the air hits a body of the earphone, a bone and/or a skin of the person, some components of the noise are reflected or diffracted to the air, and some other components of the noise are absorbed by or penetrate the body of the earphone, the bone and/or the skin of the person.
- [5] The noise that vibrates the tympanic membrane in an external auditory canal while a propagating path of the noise propagating through the air is blocked propagates as a vibration using the body of the earphone, the bone and/or the skin of the person as a medium. Therefore, a frequency characteristic of the noise that vibrates the tympanic membrane in the external auditory canal while a propagating path of the noise propagating through the air is blocked differs from the original noise.
- [6] The noise with the differing frequency characteristic may be cancelled by reflecting the frequency characteristic to compensate the noise signal, inverting the phase of the noise signal 180 degrees to generate a noise cancelling signal, and superimposing the noise signal over the noise cancelling signal to cancel the noise.
- [7] However, a residual noise signal remains even when the noise signal and the noise cancelling signal are superimposed. As a result, the residual noise signal acts as a new noise source.

Disclosure of Invention

Technical Problem

- [8] It is an object of the present invention to provide a noise cancelling apparatus wherein a noise cancellation signal is generated such that a residual noise signal is below a minimum audible acoustic pressure level to effectively reduce a noise.

Technical Solution

- [9] To achieve the above-described objects of the present invention, there is provided a noise cancelling apparatus, comprising: an earphone unit including a speaker unit, the earphone unit collecting a noise signal propagating through an air to be outputted as a first signal, and collecting an output signal of the speaker unit and a distorted noise signal within an external auditory canal to be outputted as a second signal; a first signal processor for generating a first noise cancellation signal by processing the first signal based on a compensation characteristic information; a second signal processor for generating the compensation characteristic information based on the second signal and an audio signal to be outputted through the speaker unit and transmitting the compensation characteristic information to the first signal processor, and generating a second noise cancellation signal based on the first noise cancellation signal received from the first signal processor; and a third signal processor for generating the output signal based on the second noise cancellation signal and the audio signal.
- [10] Preferably, the earphone unit comprises: a first microphone for collecting the noise signal to be outputted as the first signal; and a second microphone for collecting the output signal of the speaker unit and the distorted noise signal to be outputted as the second signal.
- [11] Preferably, the earphone unit further comprises an isolation wall for blocking the output signal of the speaker unit from propagating to the first microphone using the air in the earphone unit as a medium.
- [12] Preferably, the earphone unit further comprises a vibration absorbing module for blocking the output signal of the speaker unit from propagating to the first microphone using the earphone unit and the isolation wall as the medium.
- [13] Preferably, the earphone unit further comprises a sound absorbing module for blocking an external non-audio signal from propagating to the first microphone.
- [14] Preferably, the first signal processor comprises: a first filter for extracting a signal of a predetermined band from the first signal; a first ADC for converting an analog signal extracted by the first filter to a digital signal; and a distortion characteristic compensation filter for receiving the compensation characteristic information from the second signal processor, and generating the first noise cancellation signal having an inverted phase of that of the distorted noise signal by processing the digital signal being outputted by the first ADC.
- [15] Preferably, the compensation characteristic information comprises at least one of a

phase coefficient, a delay coefficient, a section amplitude deviation coefficient and a band frequency characteristic coefficient.

- [16] Preferably, the distortion characteristic compensation filter comprises: a phase compensator for compensating a phase of the first signal based on the phase coefficient; a delay compensator for compensating a delay of the first signal based on the delay coefficient; an amplitude compensator for compensating an amplitude of the first signal based on the section amplitude deviation coefficient; a band pass filter unit including one or more band pass filters, the band pass filter unit compensating a band frequency characteristic of the first signal based on the band frequency characteristic coefficient; an amplifier unit including one or more amplifiers amplifying an output of each of the one or more band pass filters; and an adder for adding outputs of the one or more amplifiers.
- [17] Preferably, the first signal processor comprises: a second filter for extracting a signal of a predetermined band from the second signal; a second ADC for converting an analog signal extracted by the second filter to a digital signal; and an adaptive signal processor for generating the compensation characteristic information by processing the digital signal being outputted by the second ADC and the audio signal, transmitting the compensation characteristic information to the first signal processor, receiving the first noise cancellation signal from the first signal processor, and generating the second noise cancellation signal by processing the first noise cancellation signal.
- [18] Preferably, the compensation characteristic information comprises at least one of a phase coefficient, a delay coefficient, a section amplitude deviation coefficient and a band frequency characteristic coefficient.
- [19] Preferably, the adaptive signal processor comprises: a residual signal detector for detecting a residual noise signal in the external auditory canal based on the second signal and the audio signal; a phase difference detector for calculating the phase coefficient of the residual noise signal; a delay coefficient detector for calculating the delay coefficient of the residual noise signal; a section amplitude deviation detector for calculating the section amplitude deviation coefficient of the residual noise signal; a band frequency spectrum analyzer for calculating the band frequency characteristic coefficient of the residual noise signal; and an adaptive filter for converting the first noise cancellation signal to the second noise cancellation signal according to a variation of the residual noise signal to minimize the residual noise signal to be no more than a minimum audible acoustic pressure level.
- [20] Preferably, the third signal processor comprises: an adder for adding the second noise cancellation signal being outputted from the second signal processor and the audio signal; a DAC for converting an output signal of the adder to an analog signal; and an amplifier for amplifying an output signal of the DAC.

Advantageous Effects

- [21] In accordance with the present invention, an external noise may be effectively cancelled to hear only a desired audio signal when using a portable device using an earphone or a headphone.
- [22] In accordance with the present invention, since a high volume against the external noise is not necessary, a hearing of a person may be protected, a hardness in hearing may be prevented, and a power consumption of the portable device may be reduced.
- [23] In accordance with the present invention, since the external noise may be cancelled without the audio signal, a concentration on work or study may be improved without being affected by noisy environment.

Brief Description of Drawings

- [24] Fig. 1 is a block diagram illustrating a noise cancelling apparatus in accordance with the present invention.
- [25] Fig. 2 is a block diagram illustrating in detail an earphone unit, a first signal processor, a second signal processor and a third signal processor of a noise cancelling apparatus in accordance with the present invention.
- [26] Fig. 3 is a diagram illustrating an earphone unit of a noise cancelling apparatus in accordance with the present invention.
- [27] Fig. 4 is a diagram illustrating a distortion characteristic compensation filter of a noise cancelling apparatus in accordance with the present invention.
- [28] Fig. 5 is a diagram illustrating an adaptive signal processor of a noise cancelling apparatus in accordance with the present invention.
- [29] Fig. 6 is a diagram illustrating envelopes of frequency spectrums according to an acoustic source and a wearing state of an earphone in accordance with the present invention.

Best Mode for Carrying out the Invention

- [30] A noise cancelling apparatus in accordance with the present invention will be described in detail with reference to accompanying drawings. Hereinafter, a signal having 't' as its variable such as $A(t)$ denotes a signal in analog domain, a signal having 'n' as its variable such as $B(n)$ denotes a signal in digital domain.
- [31] Fig. 1 is a block diagram illustrating a noise cancelling apparatus in accordance with the present invention.
- [32] Referring to Fig. 1, the noise cancelling apparatus in accordance with the present invention comprises an earphone unit 100, a first signal processor 200, a second signal processor 300 and a third signal processor 400.
- [33] The earphone unit 100 includes a speaker unit, collects a noise signal propagating through an air to be outputted as a first signal, and collects an output signal of the

speaker unit and a distorted noise signal within an external auditory canal to be outputted as a second signal wherein the distorted noise signal is a distorted version of the noise signal.

[34] The first signal processor 200 generates a first noise cancellation signal by processing the first signal based on a compensation characteristic information transmitted from the second signal processor 300.

[35] The second signal processor 300 generates the compensation characteristic information based on the second signal and an audio signal to be outputted through the speaker unit and transmits the compensation characteristic information to the first signal processor 200. The second signal processor 300 also generates a second noise cancellation signal based on the first noise cancellation signal received from the first signal processor 200.

[36] The third signal processor 400 generates the output signal based on the second noise cancellation signal and the audio signal to be outputted through the speaker unit.

[37] Fig. 2 is a block diagram illustrating in detail the earphone unit, the first signal processor, the second signal processor and the third signal processor of the noise cancelling apparatus in accordance with the present invention.

[38] In Fig. 2, $Na(t)$ denotes the noise signal, and $Na(n)$ denotes a digital-to-analog converted $Na(t)$. $O_1(n)$ and $O_2(n)$ denote the first noise cancellation signal and the first noise cancellation signal, respectively. $s(n)$ denotes the audio signal to be outputted through the speaker unit 130, $O(n)$ denotes a sum of $O_2(n)$ and $s(n)$. $O_1(n)$, $O_2(n)$, $s(n)$ and $O(n)$ are digital signals. $O(t)$ denotes an analog-to-digital converted $O(n)$.

[39] In addition, $Na'(t)$ denotes the distorted noise signal, and $Na'(n)$ denotes a digital-to-analog converted $Na'(t)$. $Nm(t)$ denotes a signal generated by collecting $Na'(t)$ and $O(t)$ by the second microphone 120, and $Nm(n)$ denotes a digital-to-analog converted $Nm(t)$.

[40] Referring to Fig. 2, the earphone unit 100 comprises a first microphone 110, a second microphone 120 and a speaker unit 130.

[41] The first microphone 110 collects and outputs the noise signal $Na(t)$ as the first signal.

[42] The second microphone 120 collects in external auditory canal and outputs a signal $Nm(t)$ obtained by adding the output signal $O(t)$ of the speaker unit 130 and the distorted noise signal $Na'(t)$ which is a distorted signal of the noise signal $Na(t)$ as the second signal.

[43] At least one of a phase, an amplitude, a delay and a frequency characteristic of the noise signal $Na(t)$ is distorted during a propagation thereof, wherein the distorted version of the noise signal $Na(t)$ is denoted as the distorted noise signal $Na'(t)$.

[44] That is, the second microphone 120 collects and outputs the signal $Nm(t)$ which is a

sum of the distorted noise signal $Na'(t)$ and the output signal $O(t)$ of the speaker unit 130 during the propagation within the external auditory canal which is a space between an auricle and a tympanic membrane and which undergoes a diffraction, a reflection, a resonance and a superimposition generated by a transfer function $H_{SPK}(t)$ of the speaker unit 130 and an acoustic transfer function $H_{ear}(t)$ of the external auditory canal.

[45] $H_{ear}(t)$ is determined by an acoustic characteristic according to a structure of the external auditory canal, and $H_{SPK}(t)$ is determined by a characteristic of the speaker unit 130.

[46] The speaker unit 130 outputs the signal $O(t)$. The signal $O(t)$ is obtained by adding the audio signal $s(n)$ and the second noise cancellation signal $O_2(n)$, and subjecting the added signal to a digital-to-analog conversion and an amplification.

[47] Still referring to Fig. 2, the first signal processor 200 comprises a first filter 210, a first ADC 220 and a distortion characteristic compensation filter 230.

[48] The first filter 210 extracts a signal of a predetermined band from the first signal.

[49] Specifically, the noise signal $Na(t)$ collected by the first microphone 110 is inputted to the first filter 210. The first filter 210 only passes a certain band of the signal in an audible frequency band ranging from 20Hz to 20KHz wherein an energy of the noise signal $Na(t)$ is concentrated.

[50] The first ADC 220 converts an analog signal that has band-passed through the first filter 210 to a digital signal $Na(n)$ which may be processed in a digital system and inputs the digital signal $Na(n)$ to the distortion characteristic compensation filter 230.

[51] The distortion characteristic compensation filter 230 receives the compensation characteristic information from the second signal processor 300, and generates the first noise cancellation signal having an inverted phase of that of the distorted noise signal by processing the digital signal being outputted by the first ADC 220.

[52] Specifically, the distortion characteristic compensation filter 230 is provided with the compensation characteristic information of the distorted noise signal $Na'(n)$, which includes a phase coefficient representing a phase difference, a delay coefficient representing a delay, a section amplitude deviation coefficient representing a section amplitude deviation and a band frequency characteristic coefficient representing a frequency characteristic, by the second signal processor 300 and compensates the phase, the delay, the amplitude and the frequency characteristic of the noise signal $Na(n)$ to generate the first noise cancellation signal $O_1(n)$ by a feedback control.

[53] A detailed description of the distortion characteristic compensation filter 230 will be given with reference to Fig. 4.

[54] Still referring to Fig. 2, the second signal processor 300 comprises a second filter 310, a second ADC 320 and an adaptive signal processor 330.

[55] As described above, the second microphone 120 inputs the signal $Nm(t)$ to the

second filter 310.

[56] The signal $N_m(t)$ may be represented by equation 1.

[57] [Equation 1]

[58] $N_m(t) = [O(t) \times H_{SPK}(t) + N_a'(t)] \times H_{ear}(t)$

[59] , where $H_{ear}(t)$ is the acoustic transfer function of the external auditory canal, $H_{SPK}(t)$ is the transfer function of the speaker unit 130, $N_a'(t)$ is the distorted noise signal in an analog domain, and $O(t)$ is the output signal of the speaker unit 130.

[60] The second filter 310 extracts a signal of a predetermined band from the second signal.

[61] Specifically, the second filter 310 only passes the certain band of the signal in the audible frequency band ranging from 20Hz to 20KHz.

[62] The second ADC 320 converts the analog signal $N_m(t)$ that has band-passed through the second filter 310 to a digital signal $N_m(n)$ and inputs the digital signal $N_m(n)$ to the adaptive signal processor 330.

[63] The adaptive signal processor 330 generates the compensation characteristic information by processing the digital signal $N_m(n)$ being outputted by the second ADC 320 and the audio signal to be outputted by the speaker unit 130, transmits the compensation characteristic information to the first signal processor 200, receives the first noise cancellation signal from the first signal processor 200, and generates the second noise cancellation signal by processing the first noise cancellation signal.

[64] Specifically, the adaptive signal processor 330 generates the compensation characteristic information of the distorted noise signal $N_a'(n)$ which includes the phase coefficient, the delay coefficient, the section amplitude deviation coefficient and the band frequency characteristic coefficient based on the digital signal $N_a(n)$. Thereafter, the adaptive signal processor 330 transmits the compensation characteristic information to the first signal processor 200. Next, the adaptive signal processor 330 receives and processes the first noise cancellation signal $O_1(n)$ transmitted from the first signal processor 200 to generate the second noise cancellation signal $O_2(n)$. The second noise cancellation signal $O_2(n)$ is then transmitted to the third signal processor 400.

[65] A detailed description of the adaptive signal processor 330 will be given with reference to Fig. 5.

[66] Still referring to Fig. 2, the third signal processor 400 comprises an adder 410, a DAC 420 and an amplifier 430.

[67] The adder 410 adds the second noise cancellation signal $O_2(n)$ being outputted from the second signal processor 300 and the audio signal $s(n)$ to be outputted through the speaker unit 130 to generate the output signal $O(n)$.

[68] The DAC 420 converts the output signal $O(n)$ of the adder 410 to an analog signal.

[69] The amplifier 430 amplifies the analog signal being outputted by the DAC 420 to be

suitable for the speaker unit 130.

[70] Fig. 3 is a block diagram illustrating the earphone unit of the noise cancelling apparatus in accordance with the present invention.

[71] Referring to Fig. 3, the first microphone 110 is disposed close to the speaker unit 130 suitable for collecting the noise signal $N_a(t)$.

[72] The second microphone 120 is disposed in front of the speaker unit 130. The second microphone 120 is disposed in a position suitable for collecting the output signal $O(t)$ and the distorted noise signal $N_a'(t)$ in the external auditory canal.

[73] In order to minimize an effect to the first microphone 110 by the output signal $O(t)$ that is outputted from the speaker unit 130, the earphone unit 100 may include an isolation wall 150. The isolation wall 150 is disposed between the speaker unit 130 and the first microphone 110 to block a flow of the air. Specifically, the isolation wall 150 blocks the output signal $O(t)$ of the speaker unit 130 from propagating to the first microphone 110 using the air as a medium.

[74] In addition, the earphone unit 100 may comprise a vibration absorbing module 160 in order to block the output signal $O(t)$ of the speaker unit 130 from propagating to the first microphone 110 using a earphone body 140 and the isolation wall 150 as the medium. The vibration absorbing module 160 reduces a vibration of the earphone body 140 and the isolation wall 150 to isolate the first microphone 110 from the vibration.

[75] Moreover, the earphone unit 100 may comprise a sound absorbing module 170 for blocking an external non-audio signal such as a wind and a contact from propagating to the first microphone 110. The vibration absorbing module 170 may have structure wherein the vibration absorbing module 170 surround an outer surface of the first microphone 110.

[76] Fig. 4 is a diagram illustrating the distortion characteristic compensation filter of the noise cancelling apparatus in accordance with the present invention.

[77] Referring to Fig. 4, the distortion characteristic compensation filter comprises a phase compensator 231, a delay compensator 232, an amplitude compensator 233, a band pass filter unit 234, an amplifier unit 235 and an adder 236.

[78] As described above, the compensation characteristic information generated by the second signal processor 300 may comprise at least one of a phase coefficient, a delay coefficient, a section amplitude deviation coefficient and a band frequency characteristic coefficient.

[79] When $P_m(n)$ denotes the phase coefficient, $D_m(n)$ denotes the delay coefficient, $A_m(n)$ denotes the section amplitude deviation coefficient and $F_m(n)$ denotes the band frequency characteristic coefficient, the phase compensator 231 compensates a phase of the first signal based on the phase coefficient $P_m(n)$. That is, the phase compensator 231 compensates the noise signal $N_a(n)$ by reflecting the phase coefficient $P_m(n)$.

- [80] The delay compensator 232 compensates a delay of the first signal based on the delay coefficient $Dm(n)$. That is, the delay compensator 232 compensates the noise signal $Na(n)$ by reflecting the delay coefficient $Dm(n)$.
- [81] The amplitude compensator 233 compensates an amplitude of the first signal based on the section amplitude deviation coefficient $Am(n)$. That is, the amplitude compensator 233 compensates the noise signal $Na(n)$ by reflecting the section amplitude deviation coefficient $Am(n)$.
- [82] The band pass filter unit 234 includes first through k^{th} band pass filters as shown in Fig. 4. The band pass filter unit 234 compensates a band frequency characteristic of the first signal based on the band frequency characteristic coefficient $Fm(n)$. That is, the band pass filter unit 234 compensates the noise signal $Na(n)$ by reflecting the band frequency characteristic coefficient $Fm(n)$. One or more suitable band pass filter may be used according to the band.
- [83] The amplifier unit 235 includes first through k^{th} amplifiers $AMP_1 - AMP_k$ as shown in Fig. 4. The amplifier unit 235 amplifies an output of each of the one or more band pass filters.
- [84] The adder 236 for adds outputs of the one or more amplifiers.
- [85] As shown in Fig. 4, the noise signal $Na(n)$ is sequentially passed through the phase compensator 231, the delay compensator 232, the amplitude compensator 233, the band pass filter unit 234, the amplifier unit 235 and the adder 236 to generate the first noise cancellation signal $O_1(n)$.
- [86] That is, the distortion characteristic compensation filter 230 applies the phase coefficient $Pm(n)$ to the phase compensator 231 to compensate the phase of the noise signal $Na(n)$, applies the delay coefficient $Dm(n)$ to the delay compensator 232 to compensate the delay of the phase compensated signal, applies the section amplitude deviation coefficient $Am(n)$ to the amplitude compensator 233 to compensate the amplitude of the phase and delay compensated signal, and applies the band frequency characteristic coefficient $Fm(n)$ to the band pass filter unit 234 and the amplifier unit 235 such that the phase, delay and amplitude compensated signal has different amplitudes according to the band. The outputs signals of the band pass filter unit 234 are added by the adder 236 and the output signal of the adder 236 is then inverted to generate the first noise cancellation signal $O_1(n)$.
- [87] The order may be changed according to a design of the distortion characteristic compensation filter 230.
- [88] Fig. 5 is a diagram illustrating an adaptive signal processor of the noise cancelling apparatus in accordance with the present invention.
- [89] Referring to Fig. 5, the adaptive signal processor comprises a phase difference detector 331, a delay coefficient detector 332, a section amplitude deviation detector

333, a band frequency spectrum analyzer 334, an adaptive filter 335, and residual signal detectors 336, 337, 338, 339 and 340.

- [90] The residual signal detectors 336, 337, 338, 339 and 340 detect a residual noise signal in the external auditory canal based on the second signal and the audio signal.
- [91] The phase difference detector 331 calculates the phase coefficient $P_m(n)$ of a residual noise signal detected by the residual signal detectors 336, 337, 338, 339 and 340.
- [92] The delay coefficient detector 332 calculates the delay coefficient $D_m(n)$ of the residual noise signal detected by the residual signal detectors 336, 337, 338, 339 and 340.
- [93] The section amplitude deviation detector 333 calculates the section amplitude deviation coefficient $A_m(n)$ of the residual noise signal detected by the residual signal detectors 336, 337, 338, 339 and 340.
- [94] The band frequency spectrum analyzer 334 calculating the band frequency characteristic coefficient $F_m(n)$ of the residual noise signal detected by the residual signal detectors 336, 337, 338, 339 and 340.
- [95] The adaptive filter 335 converts the first noise cancellation signal $O_1(n)$ to the second noise cancellation signal $O_2(n)$ according to a variation of the residual noise signal to minimize the residual noise signal to be no more than a minimum audible acoustic pressure level.
- [96] A detailed description of the noise cancelling apparatus in accordance with the present invention, the residual signal detectors 336, 337, 338, 339 and 340 in particular, is given below with reference to Figs. 2, 4 and 5
- [97] The noise signal $N_a(n)$ propagating through the air and the first noise cancellation signal $O_1(n)$ generated by the distortion characteristic compensation filter 230 may be represented by Equation 2.
- [98] [Equation 2]
- [99] $O_1(n) = N_a(n) \times H_{PDAF}(n)$
- [100] where $H_{PDAF}(n)$ is a transfer function of the distortion characteristic compensation filter.
- [101] On the other hand, the digital signal $N_a'(n)$, which is obtained by subjecting the distorted noise signal $N_a'(t)$ to the analog-to-digital conversion, and the digital signal $O(n)$, which is obtained by subjecting the output signal $O(t)$ of the speaker unit 130 to the analog-to-digital conversion, are added and then transformed by the transfer function $H_{SPK}(t)$ of the speaker unit 130 and the acoustic transfer function $H_{ear}(t)$ of the external auditory canal to obtain the signal $N_m(n)$, which is obtained by subjecting the signal $N_m(t)$ collected by the second microphone 120 to the analog-to-digital conversion.
- [102] The signal $N_m(n)$ may be expressed by equation 3

[103] [Equation 3]

$$[104] \quad Nm(n) = [O(n) \times H_{SPK}(n) + N_a'(n)] \times H_{ear}(n)$$

[105] ,where $H_{SPK}(t)$ is the transfer function of the speaker unit 130 and $H_{ear}(t)$ is the acoustic transfer function of the external auditory canal, $N_a'(n)$ is the distorted noise signal in digital domain, and $O(n)$ is the output signal of the speaker unit in digital domain.

[106] The speaker unit 130 outputs the signal $O(t)$ obtained by subjecting the signal $O(n)$ which is obtained by adding the second noise cancellation signal $O_2(n)$ generated by applying the first noise cancellation signal $O_1(n)$ generated by the distortion characteristic compensation filter 230 to the adaptive filter 335 and the audio signal $s(n)$. Therefore, an audio transformed by the transfer function of the speaker unit 130 $H_{SPK}(t)$ is heard in the external auditory canal.

[107] The digital signal $Nm(n)$ may be expressed as equation 4

[108] [Equation 4]

$$[109] \quad Nm(n) = [O(n) \times H_{SPK}(n) + N_a'(n)] \times H_{ear}(n) \\ = [(O_2(n) + s(n)) \times H_{SPK}(n) + N_a'(n)] \times H_{ear}(n)$$

[110] Referring to Fig. 5, after amplifying $s(n)$ to a proper amplitude by the amplifier 338, $s(n)$ is subtracted from $Nm(n)$ through the subtractor 339. After subjecting the second noise cancellation signal $O_2(n)$ that has passed through the adaptive filter 335 to a time lead or a time lag compensation by the time lead/lag unit 336, the output signal of the time lead/lag unit 336 is amplified to a proper amplitude by the amplifier 337. The residual noise signal $e(n)$ is then obtained by subtracting an output signal of the amplifier 337 from an output signal of the subtractor 339.

[111] The residual noise signal $e(n)$ may be expressed as equation 5.

[112] [Equation 5]

$$[113] \quad e(n) = Nm(n) - A_1 \times s(n) - A_2 \times O_2(n) \times H_T(n)$$

[114] ,where A_1 and A_2 are gains of the amplifiers 338 and 337, respectively, and $H_T(n)$ is a time lead/lag function of $O_2(n)$.

[115] The adaptive filter 335 generates an optimized signal, i.e., the second noise cancellation signal $O_2(n)$ in order to minimize the residual noise signal $e(n)$, which is a remaining portion of the first noise cancellation signal $O_1(n)$ after removing the noise therefrom, to be no more than a minimum audible acoustic pressure level (2×10^{-5} N/m² or 10^{-6} W/cm² in case of a pure sound of 1KHz which is a smallest sound a healthy person can hear).

[116] In order to generate the second noise cancellation signal $O_2(n)$, the adaptive filter 335 carries out a linear feedback control or a linear feedforward control which minimizes the residual noise signal $e(n)$ by applying an LMS (Least Mean Square) algorithm or

an improved version or variation thereof.

[117] The second noise cancellation signal $O_2(n)$ may be expressed as equation 6.

[118] [Equation 6]

[119] $O_2(n) = O_1(n) \times H_{AP}(n) + e(n)$

[120] , where $H_{AP}(n)$ is a transfer function of the adaptive filter 335.

[121] When $s(n)$ is zero, $O(n)$ is equal to $O_2(n)$.

[122] When equations 4 and 6 are applied under the assumption that $s(n)$ is zero, $Nm(n)$ may be expressed as equation 7.

[123] [Equation 7]

[124]
$$\begin{aligned} Nm(n) &= [O_2(n) \times H_{SPK}(n) + N_a'(n)] \times H_{ear}(n) \\ &= [[O_1(n) \times H_{AP}(n) + e(n)] \times H_{SPK}(n) + N_a'(n)] \times H_{ear}(n) \\ &= e(n) \end{aligned}$$

[125] In order to simplify the equation, $H_{AP}(n)$, $H_{SPK}(n)$ and $H_{ear}(n)$ are assumed to be '1' and equation 7 may be expressed as equation 8.

[126] [Equation 8]

[127] $e(n) = [O_1(n) + e(n)] + N_a'(n)$

$e(n) - e(n) = 0 = O_1(n) + N_a'(n)$

[128] That is, as expressed by equation 8, a sum of the first noise cancellation signal $O_1(n)$ and the distorted noise signal $N_a'(n)$ is zero.

[129] However, $e(n)$ is not zero due to the transfer function $H_{SPK}(t)$ of the speaker unit 130 and the acoustic transfer function $H_{ear}(t)$ of the external auditory canal in reality.

[130] Therefore, the transfer function $H_{AP}(n)$ of the adaptive filter 335 should be set in a manner that the residual noise signal $e(n)$ remaining due to the transfer function $H_{SPK}(t)$ and the acoustic transfer function $H_{ear}(t)$ becomes zero.

[131] In addition, since the distorted noise signal $N_a'(n)$ changes with time, the residual noise signal $e(n)$ also changes with time.

[132] Therefore, the adaptive filter 335 should generate the second noise cancellation signal $O_2(n)$ to vary with time such that the residual noise signal $e(n)$ is zero or below the minimum audible acoustic pressure level. The transfer function $H_{AP}(n)$ should be set in a manner that the adaptive filter 335 generates the optimized second noise cancellation signal $O_2(n)$ adaptive to the change of the residual noise signal $e(n)$.

[133] The adaptive signal processor 330 generates and provides the distortion characteristic compensation filter 230 with the compensation characteristic information such as the phase coefficient $Pm(n)$, the delay coefficient $Dm(n)$, the section amplitude deviation coefficient $Am(n)$ and the band frequency characteristic coefficient $Fm(n)$ so that the distortion characteristic compensation filter 230 may carry out the feedback control.

- [134] When the gains of the amplifier 338 for amplifying the audio signal $s(n)$ and the amplifier 337 for amplifying the second noise cancellation signal $O_2(n)$ are set as zero in order to generate the compensation characteristic information, the residual noise signal $e(n)$ outputted through the subtractor 339 for the audio signal $s(n)$ and the subtractor 340 for the second noise cancellation signal $O_2(n)$ becomes equal to $Nm(n)$.
- [135] This process may be expressed as equation 9.
- [136] [Equation 9]
- [137]
$$e(n) = Nm(n) - 0 \times s(n) - 0 \times O_2(n) \times H_T(n) = Nm(n)$$
- [138] The phase coefficient $Pm(n)$ may be calculated through the phase difference detector 331 and the delay coefficient $Dm(n)$ may be calculated through the delay coefficient detector 332 from The residual noise signal $e(n)$ obtained by equation 9.
- [139] In addition, the section amplitude deviation coefficient $Am(n)$ may be calculated through the section amplitude deviation detector 333 and the band frequency characteristic coefficient $Fm(n)$ may be calculated through the band frequency spectrum analyzer 334.
- [140] The second noise cancellation signal $O_2(n)$ and the audio signal $s(n)$ are added by an adder 410 to obtain the signal $O(n)$.
- [141] The signal $O(n)$ is converted to an analog signal by a DAC 420 and the analog signal is then amplified by a terminal amplifier 430. The output signal $O(t)$ of the terminal amplifier 430 is outputted through the speaker unit 130.
- [142] Fig. 6 is a diagram illustrating envelopes of frequency spectrums according to an acoustic source and a wearing state of the earphone unit in accordance with the present invention.
- [143] As shown in Fig. 6, the noise signal $Na(t)$ propagating through the air and the noise signal in the external auditory canal according to the wearing state, i.e. the distorted noise signal $Na'(t)$ have different frequency characteristics.
- [144] That is, as shown in Fig. 6, the distorted noise signal $Na'(t)$ has a larger amplitude in a low frequency band and a smaller amplitude in a high frequency band compared to the noise signal $Na(t)$.
- [145] Also shown in Fig. 6, the amplitude in the high and low frequency bands of the distorted noise signal $Na'(t)$ varies when the earphone unit is worn loosely and tightly.
- [146] Since the conventional noise cancelling technology inverts the phase of the noise signal propagating through the air by 180 degrees and superimposes the noise signal and the inverted noise signal to cancel the noise, the amplitude of the noise signal in the external auditory canal cannot be adjusted in the high and low frequency bands.
- [147] However, in accordance with the present invention, since the noise signal is cancelled based on the distorted noise signal $Na'(t)$, the noise may be cancelled

according to the wearing state.

- [148] While the present invention has been described and illustrated herein with reference to the preferred embodiments thereof, it will be apparent to those skilled in the art that various modifications and variations can be made therein without departing from the spirit and scope of the invention. Thus, it is intended that the present invention covers the modifications and variations of this invention that come within the scope of the appended claims and their equivalents.

Industrial Applicability

- [149] As described above, in accordance with the present invention, an external noise may be effectively cancelled to hear only a desired audio signal when using a portable device using an earphone or a headphone.
- [150] In accordance with the present invention, since a high volume against the external noise is not necessary, a hearing of a person may be protected, a hardness in hearing may be prevented, and a power consumption of the portable device may be reduced.
- [151] In accordance with the present invention, since the external noise may be cancelled without the audio signal, a concentration on work or study may be improved without being affected by noisy environment.

Claims

- [1] A noise cancelling apparatus, comprising:
an earphone unit including a speaker unit, the earphone unit collecting a noise signal propagating through an air to be outputted as a first signal, and collecting an output signal of the speaker unit and a distorted noise signal within an external auditory canal to be outputted as a second signal;
a first signal processor for generating a first noise cancellation signal by processing the first signal based on a compensation characteristic information;
a second signal processor for generating the compensation characteristic information based on the second signal and an audio signal to be outputted through the speaker unit and transmitting the compensation characteristic information to the first signal processor, and generating a second noise cancellation signal based on the first noise cancellation signal received from the first signal processor; and
a third signal processor for generating the output signal based on the second noise cancellation signal and the audio signal.
- [2] The apparatus in accordance with claim 1, wherein the earphone unit comprises:
a first microphone for collecting the noise signal to be outputted as the first signal; and
a second microphone for collecting the output signal of the speaker unit and the distorted noise signal to be outputted as the second signal.
- [3] The apparatus in accordance with claim 2, wherein the earphone unit further comprises an isolation wall for blocking the output signal of the speaker unit from propagating to the first microphone using the air in the earphone unit as a medium.
- [4] The apparatus in accordance with claim 3, wherein the earphone unit further comprises a vibration absorbing module for blocking the output signal of the speaker unit from propagating to the first microphone using the earphone unit and the isolation wall as the medium.
- [5] The apparatus in accordance with claim 2, wherein the earphone unit further comprises a sound absorbing module for blocking an external non-audio signal from propagating to the first microphone.
- [6] The apparatus in accordance with claim 1, wherein the first signal processor comprises:
a first filter for extracting a signal of a predetermined band from the first signal;
a first ADC for converting an analog signal extracted by the first filter to a digital signal; and
a distortion characteristic compensation filter for receiving the compensation

- characteristic information from the second signal processor, and generating the first noise cancellation signal having an inverted phase of that of the distorted noise signal by processing the digital signal being outputted by the first ADC.
- [7] The apparatus in accordance with claim 6, wherein the compensation characteristic information comprises at least one of a phase coefficient, a delay coefficient, a section amplitude deviation coefficient and a band frequency characteristic coefficient.
- [8] The apparatus in accordance with claim 7, wherein the distortion characteristic compensation filter comprises:
- a phase compensator for compensating a phase of the first signal based on the phase coefficient;
 - a delay compensator for compensating a delay of the first signal based on the delay coefficient;
 - an amplitude compensator for compensating an amplitude of the first signal based on the section amplitude deviation coefficient;
 - a band pass filter unit including one or more band pass filters, the band pass filter unit compensating a band frequency characteristic of the first signal based on the band frequency characteristic coefficient;
 - an amplifier unit including one or more amplifiers amplifying an output of each of the one or more band pass filters; and
 - an adder for adding outputs of the one or more amplifiers.
- [9] The apparatus in accordance with claim 1, wherein the first signal processor comprises:
- a second filter for extracting a signal of a predetermined band from the second signal;
 - a second ADC for converting an analog signal extracted by the second filter to a digital signal; and
 - an adaptive signal processor for generating the compensation characteristic information by processing the digital signal being outputted by the second ADC and the audio signal, transmitting the compensation characteristic information to the first signal processor, receiving the first noise cancellation signal from the first signal processor, and generating the second noise cancellation signal by processing the first noise cancellation signal.
- [10] The apparatus in accordance with claim 9, wherein the compensation characteristic information comprises at least one of a phase coefficient, a delay coefficient, a section amplitude deviation coefficient and a band frequency characteristic coefficient.
- [11] The apparatus in accordance with claim 10, wherein the adaptive signal

processor comprises:

a residual signal detector for detecting a residual noise signal in the external auditory canal based on the second signal and the audio signal;

a phase difference detector for calculating the phase coefficient of the residual noise signal;

a delay coefficient detector for calculating the delay coefficient of the residual noise signal;

a section amplitude deviation detector for calculating the section amplitude deviation coefficient of the residual noise signal;

a band frequency spectrum analyzer for calculating the band frequency characteristic coefficient of the residual noise signal; and

an adaptive filter for converting the first noise cancellation signal to the second noise cancellation signal according to a variation of the residual noise signal to minimize the residual noise signal to be no more than a minimum audible acoustic pressure level.

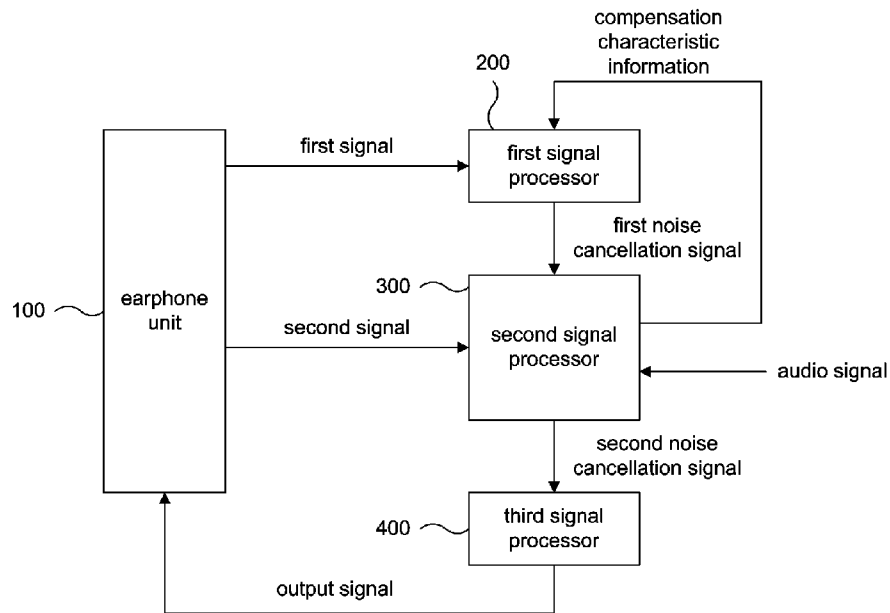
[12] The apparatus in accordance with claim 1, wherein the third signal processor comprises:

an adder for adding the second noise cancellation signal being outputted from the second signal processor and the audio signal;

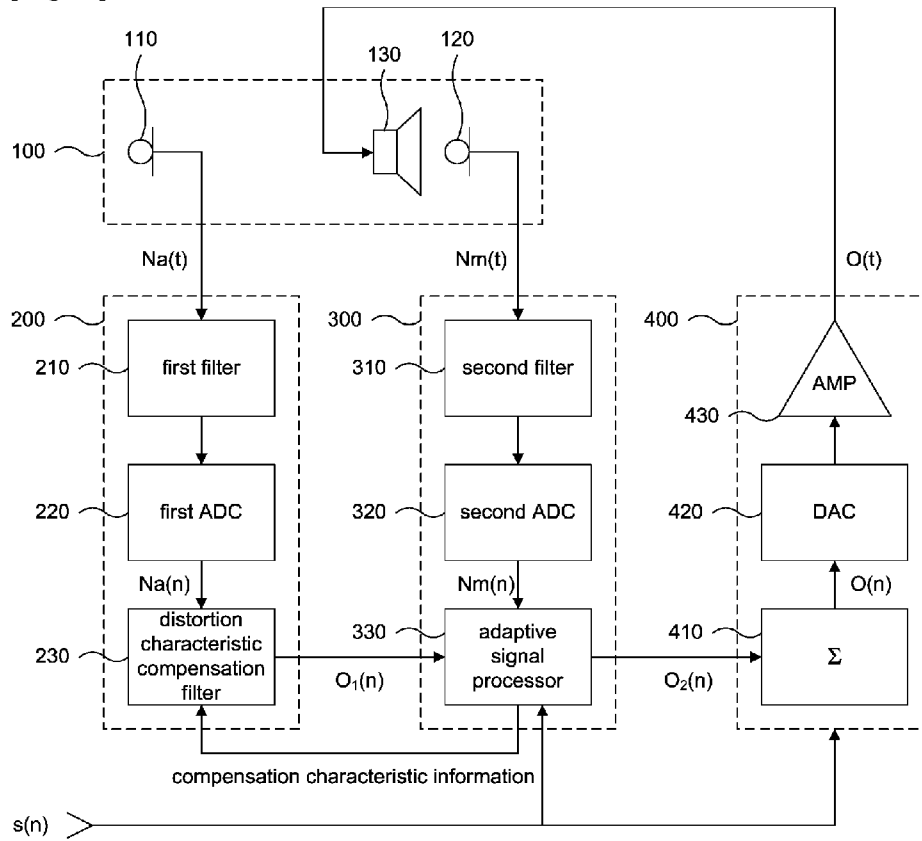
a DAC for converting an output signal of the adder to an analog signal; and

an amplifier for amplifying an output signal of the DAC.

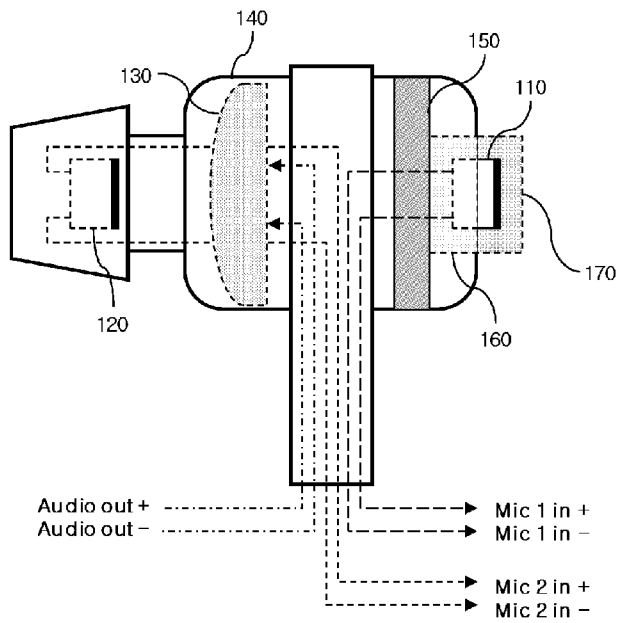
[Fig. 1]



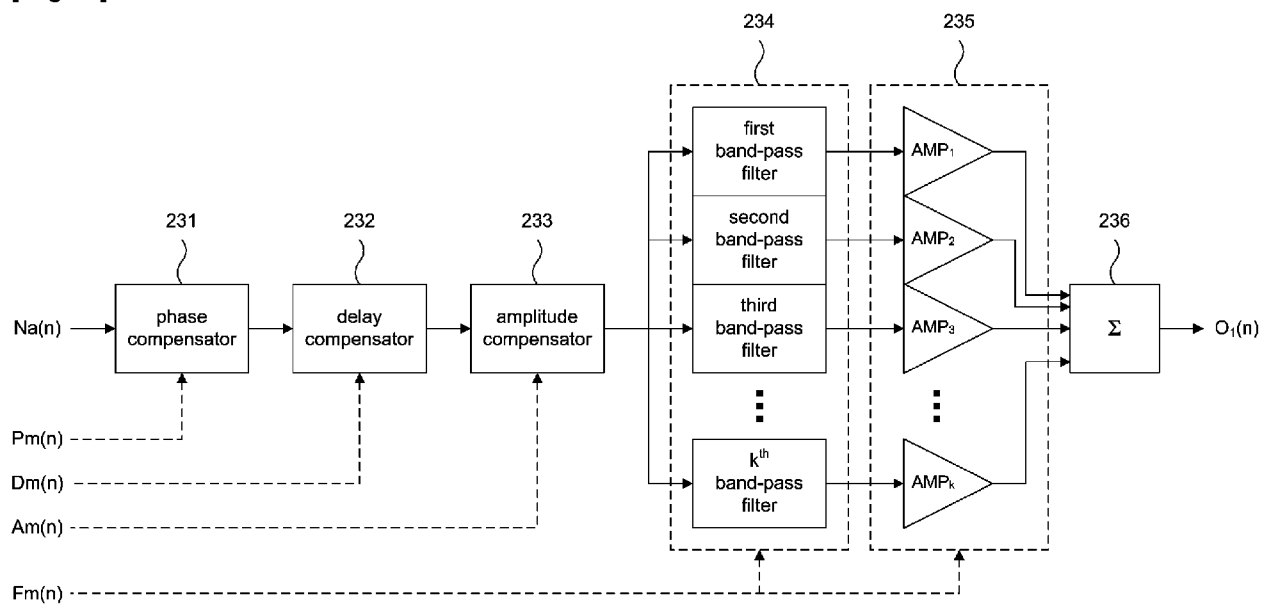
[Fig. 2]



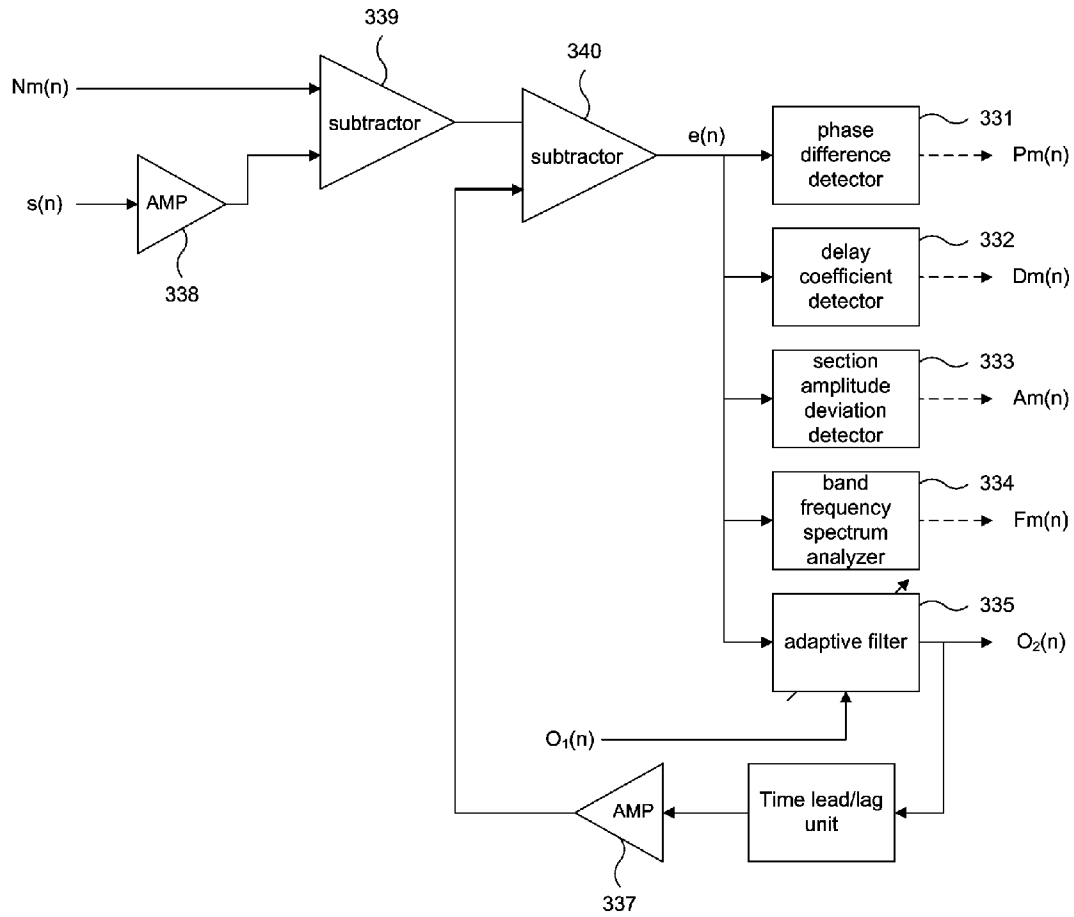
[Fig. 3]



[Fig. 4]



[Fig. 5]



[Fig. 6]

