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(54) ACTIVE NOISE CONTROL DEVICE

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(57) ABSTRACT

An active noise control device controls a speaker so as to output a canceling sound for canceling noise transmitted from a vibration source. The active noise control device includes a control signal generating unit configured to perform signal processing on a basic signal corresponding to a predetermined frequency by a feedback filter and a extraction filter, which is an adaptive notch filter, to generate a control signal that control the speaker, a secondary path filter updating unit configured to update sequentially and adaptively a secondary path filter, and a feedback filter setting unit configured to set the feedback filter based on the secondary path filter.

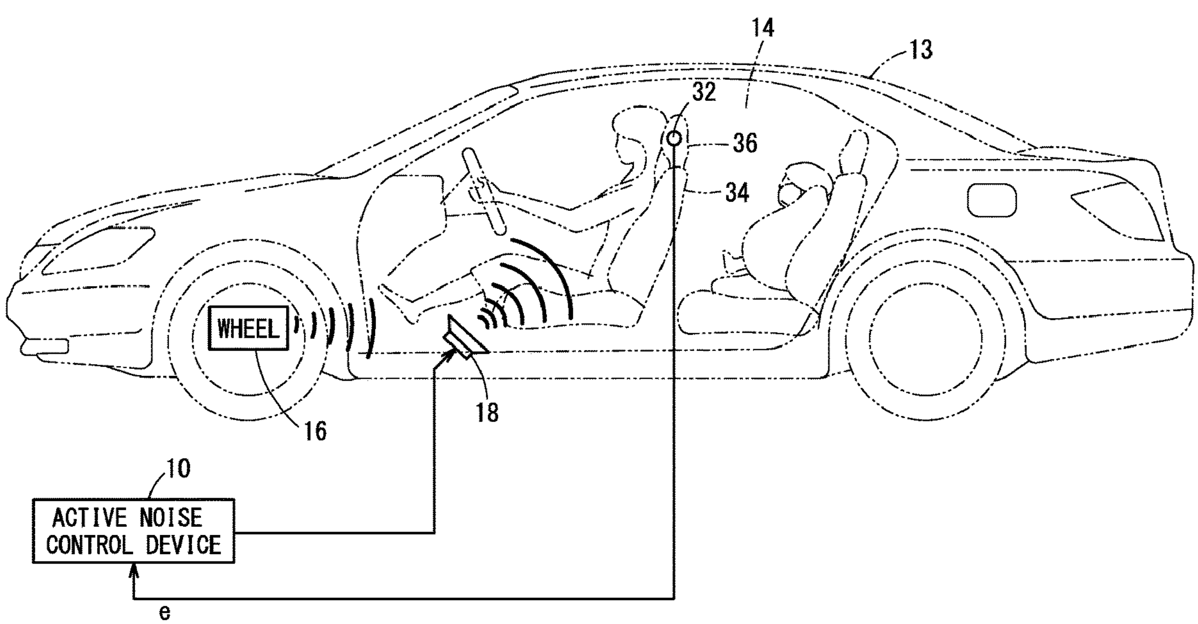


FIG. 1

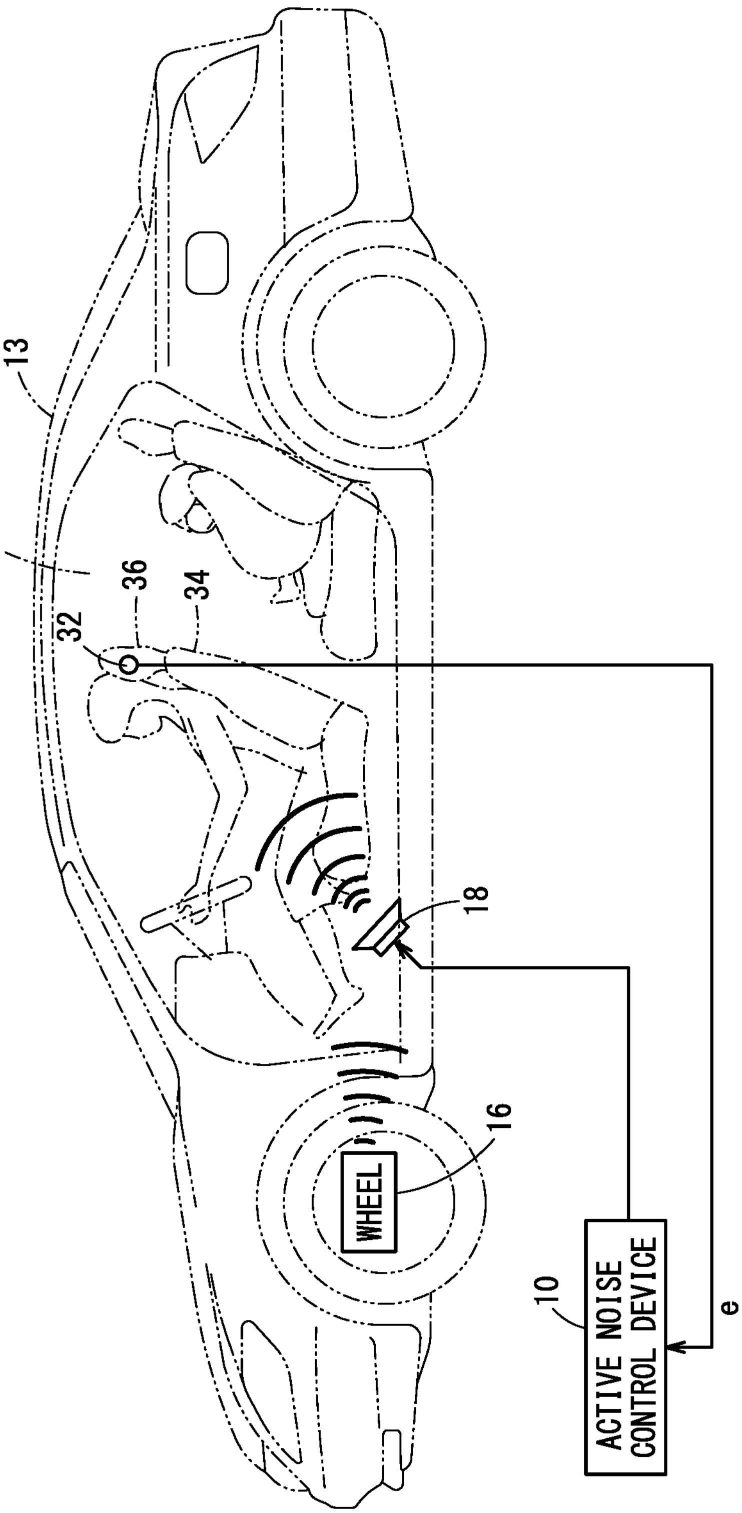


FIG. 2

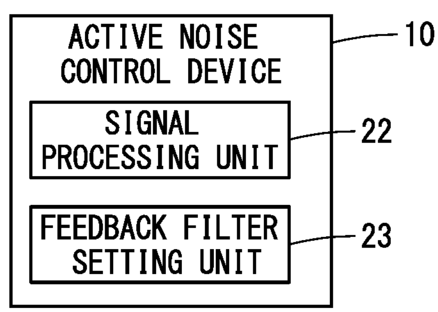
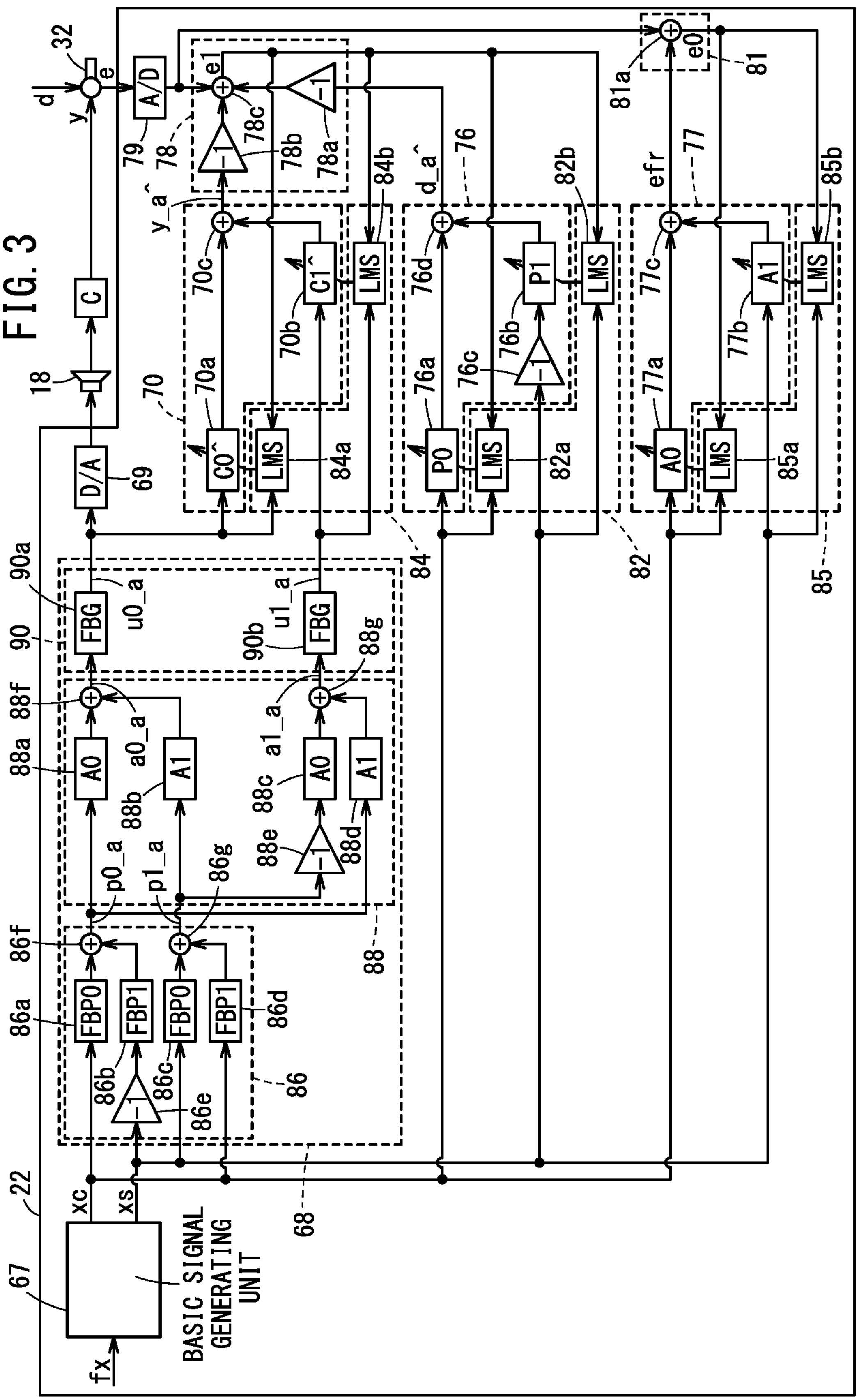

ACTIVE NOISE CONTROL DEVICE
10
SIGNAL PROCESSING UNIT
22
FEEDBACK FILTER SETTING UNIT
23

FIG. 3


BASIC SIGNAL GENERATING UNIT
67
fx
xc
xs
22
68
86
86a
86b
86c
86d
86e
86f
86g
FBP0
FBP1
-1
88
88a
88b
88c
88d
88e
88f
88g
A0
A1
p0_a
p1_a
a0_a
a1_a
90
90a
90b
FBG
u0_a
u1_a
69
D/A
18
C
32
d
y
e
79
A/D
78
78a
78b
78c
e1
y_a^
70
70a
70b
70c
C0^
C1^
LMS
84
84a
84b
76
76a
76b
76c
76d
d_a^
P0
P1
82
82a
82b
77
77a
77b
77c
efr
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85a
85b
81
81a
e0

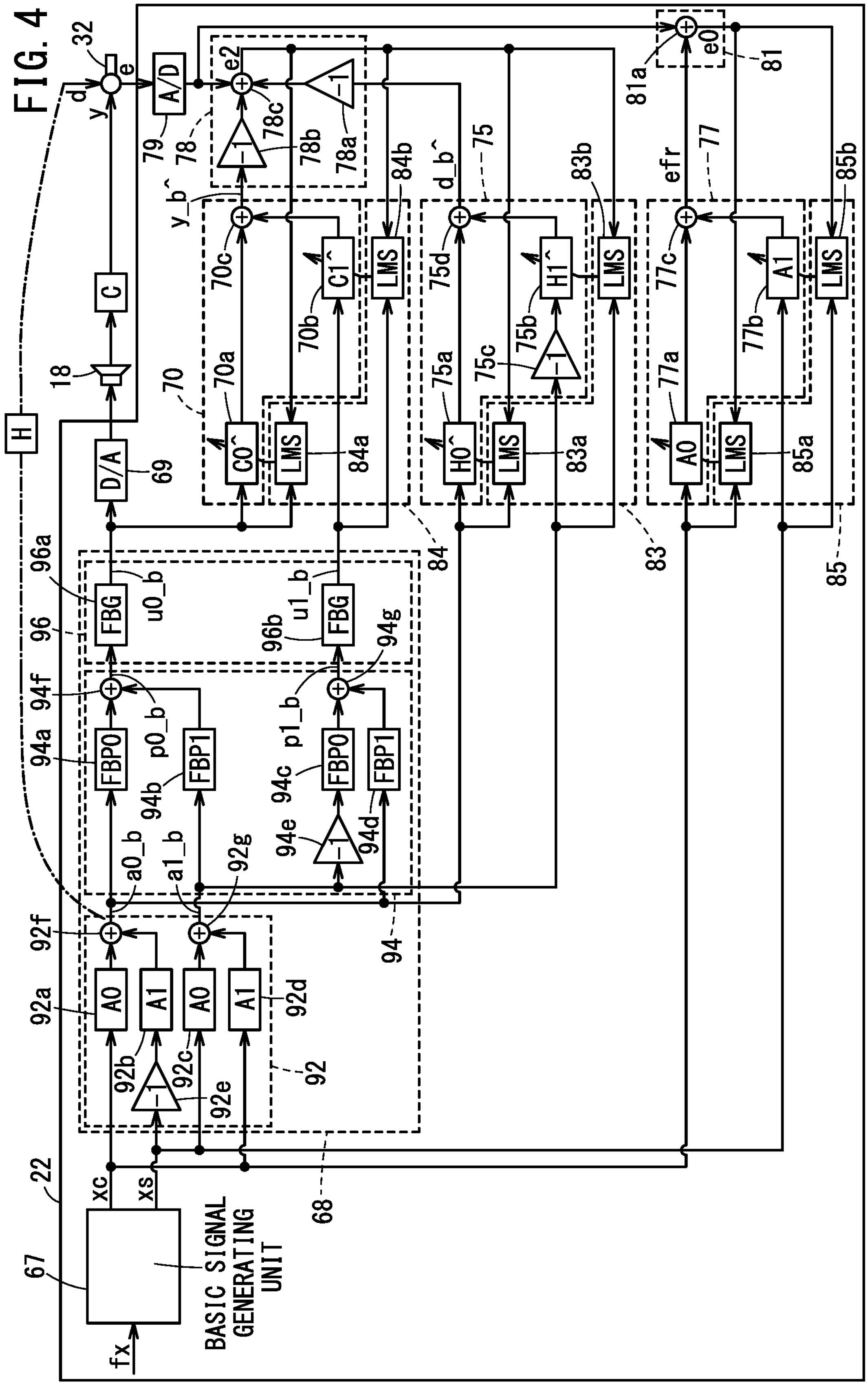
BASIC SIGNAL GENERATING UNIT
fx
xc
xs
A0
A1
FBP0
FBP1
FBG
D/A
A/D
C
H
C0^
C1^
H0^
H1^
LMS
-1
e
d
y
e2
e0
efr
y_b^
d_b^
a0_b
a1_b
p0_b
p1_b
u0_b
u1_b

FIG. 4

ACTIVE NOISE CONTROL DEVICE

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is based upon and claims the benefit of priority from Japanese Patent Application No. 2021-044977 filed on Mar. 18, 2021, the contents of which are incorporated herein by reference.

BACKGROUND OF THE INVENTION

Field of the Invention

[0002] The present invention relates to an active noise control device.

Description of the Related Art

[0003] JP 2007-025527 A discloses an active noise reduction device. The active noise reduction device generates a signal for controlling a speaker. Thus, an interference sound is output from the speaker. The sound pressure of noise such as road noise is reduced by the interference sound.

SUMMARY OF THE INVENTION

[0004] An active noise control device as disclosed in JP 2007-025527 A generates a control signal that controls a speaker based on a transfer characteristic between the speaker and a microphone. In the active noise control device, the transfer characteristic between the speaker and the microphone is fixed. Therefore, when the transfer characteristic changes, there are problems such that the active noise control device cannot reduce the sound pressure of the noise.

[0005] An object of the present invention is to solve the above-described problems.

[0006] According to an aspect of the present invention, an active noise control device controls a speaker based on a component of a frequency band around a predetermined frequency of an error signal output from a detector that detects, at a control point, a synthetic sound of a noise transmitted from a vibration source and a canceling sound output from the speaker to cancel the noise, and includes a basic signal generating unit configured to generate a basic signal corresponding to the predetermined frequency, a control signal generating unit configured to perform signal processing on the basic signal by a feedback filter and an extraction filter, which is an adaptive notch filter, to generate a control signal that controls the speaker, an estimated canceling sound signal generating unit configured to perform signal processing on the control signal by a secondary path filter, which is an adaptive notch filter, to generate an estimated canceling sound signal, an extraction signal generating unit configured to perform signal processing on the basic signal by the extraction filter to generate an extraction signal, a virtual error signal generating unit configured to generate a virtual error signal from the error signal and the estimated canceling sound signal, a differential signal generating unit configured to generate a differential signal from the error signal and an extraction signal, a secondary path filter updating unit configured to update sequentially and adaptively the secondary path filter based on the control signal and the virtual error signal in a manner that a magnitude of the virtual error signal is minimized, an extraction filter updating unit configured to update sequentially and adaptively the extraction filter based on the basic signal and the differential signal in a manner that a magnitude of the differential signal is minimized, and a feedback filter setting unit configured to set the feedback filter based on the secondary path filter.

[0007] The active noise control device according to the present invention can reduce noise even if the transfer characteristic changes.

[0008] The above and other objects, features, and advantages of the present invention will become more apparent from the following description when taken in conjunction with the accompanying drawings in which a preferred embodiment of the present invention is shown by way of illustrative example.

BRIEF DESCRIPTION OF THE DRAWINGS

[0009] FIG. **1** is a diagram illustrating an outline of active noise control executed by an active noise control device;

[0010] FIG. **2** is a schematic diagram illustrating a configuration of an active noise control device;

[0011] FIG. **3** is a control block diagram of a signal processing unit; and

[0012] FIG. **4** is a control block diagram of a signal processing unit.

DESCRIPTION OF THE INVENTION

First Embodiment

[0013] FIG. **1** is a diagram illustrating an outline of active noise control executed by an active noise control device **10**.

[0014] A wheel **16** is vibrated by a force received from a road surface when the vehicle travels. This vibration is transmitted to the vehicle body via the suspension, and road noise is generated in a vehicle compartment **14** of a vehicle **13**. The road noise has a peak from 40 to 50 Hz. The range of frequencies from 40 to 50 Hz is a range of frequencies excited by acoustic resonance characteristics of a closed space such as the vehicle compartment **14**. Narrow band components with a constant bandwidth around the peak frequency produce a roaring sound, also called drumming noise. Drumming noise is likely to cause discomfort to vehicle occupants.

[0015] The active noise control device **10** according to the present embodiment causes a speaker **18** provided in the vehicle compartment **14** to output a canceling sound. Thus, the sound pressure of the drumming noise at a control point in the vehicle compartment **14** is reduced.

[0016] FIG. **2** is a schematic diagram illustrating a configuration of the active noise control device **10**. The active noise control device **10** includes a signal processing unit **22** and a feedback filter setting unit **23**.

[0017] The active noise control device **10** includes a computation unit and a storage unit (not shown). The signal processing unit **22** and the feedback filter setting unit **23** described above are realized by the computation unit.

[0018] The computation unit may be configured by a processor such as a CPU (Central Processing Unit) or a GPU (Graphics Processing Unit).

[0019] The computation unit includes a determination unit and a control unit which are not illustrated. The determination unit and the control unit are realized by the computation unit executing a program stored in the storage unit.

[0020] At least a part of the determination unit and the control unit may be realized by an integrated circuit such as an ASIC (Application Specific Integrated Circuit) or an FPGA (Field-Programmable Gate Array). In addition, at least a part of the determination unit and the control unit may be configured by an electronic circuit including a discrete device.

[0021] The storage unit can be configured by a volatile memory (not illustrated) and a nonvolatile memory (not illustrated). Examples of the volatile memory include, for example, a RAM (Random Access Memory) or the like. Examples of the nonvolatile memory include, for example, a ROM (Read Only Memory), a flash memory, or the like. Data or the like may be stored, for example, in the volatile memory. Programs, tables, maps, and the like are stored, for example, in the nonvolatile memory. At least a part of the storage unit may be provided in the processor, the integrated circuit, or the like as described above.

[Configuration of Signal Processing Unit]

[0022] FIG. **3** is a control block diagram of the signal processing unit **22**. The signal processing unit **22** performs feedback signal processing. In the feedback signal processing, a control signal u**0**_*a* is generated. The control signal u**0**_*a* is a signal for causing the speaker **18** to output a canceling sound that cancels the drumming noise. The control signal u**0**_*a* is generated on the basis of an error signal e output from a microphone **32** provided at the control point. Hereinafter, a sound transfer path from the speaker **18** to the microphone **32** is referred to as a secondary path, and a transfer characteristic of the secondary path is denoted by C.

[0023] In the present embodiment, the control point is near the ears of the vehicle occupant. For this purpose, as shown in FIG. **1**, the microphone **32** is provided at a headrest **36** of a seat **34** in the vehicle compartment **14**. The error signal e is a signal output from the microphone **32** that has detected a synthetic sound of the noise d at the control point and the canceling sound y at the control point.

[0024] The signal processing unit **22** includes a basic signal generating unit **67**, a control signal generating unit **68**, an estimated canceling sound signal generating unit **70**, an estimated noise signal generating unit **76**, an extraction signal generating unit **77**, a virtual error signal generating unit **78**, a differential signal generating unit **81**, an adjustment filter updating unit **82**, a secondary path filter updating unit **84**, and an extraction filter updating unit **85**.

[0025] The basic signal generating unit **67** generates a basic signal xc ($=\cos(2\pi\times fx\times t)$) and xs ($=\sin(2\pi\times fx\times t)$). The basic signal xc is a cosine signal of a control target frequency fx. The basic signal xs is a sine signal of the control target frequency fx. Here, t denotes time. The control target frequency fx is each set in advance to the peak frequency of the drumming noise and a frequency near the peak frequency.

[0026] The control signal generating unit **68** generates control signals u**0**_*a* and u**1**_*a*. The control signals u**0**_*a* and u**1**_*a* are generated by performing signal processing on the basic signals xc and xs by a feedback filter FB and an extraction filter A. The control signal generating unit **68** includes a phase adjusting unit **86**, a signal extraction unit **88**, and a gain adjusting unit **90**.

[0027] The feedback filter FB is indicated by FB=FBG (FBP**0**+iFBP**1**) using a gain FBG, a filter coefficient FBPO, and a filter coefficient FBP**1**. Here, i denotes an imaginary number. Further, $FBP0^2+FBP1^2=1$. The feedback filter FB is set by the feedback filter setting unit **23**. The setting of the feedback filter FB will be described later in detail. The extraction filter A will be described in detail together with an extraction signal generating unit **77** described later.

[0028] The phase adjusting unit **86** generates phase adjustment signals p**0**_*a* and p**1**_*a*. The phase adjustment signals p**0**_*a* and p**1**_*a* are generated by performing signal processing on the basic signals xc and xs by the phase adjustment filter FBP.

[0029] The phase adjusting unit **86** includes a first phase adjustment filter **86***a*, a second phase adjustment filter **86***b*, a third phase adjustment filter **86***c*, a fourth phase adjustment filter **86***d*, an inverting amplifier **86***e*, an adder **86***f*, and an adder **86***g*.

[0030] The first phase adjustment filter **86***a* has the filter coefficient FBP**0**. The second phase adjustment filter **86***b* has the filter coefficient FBP**1**. The third phase adjustment filter **86***c* has the filter coefficient FBP**0**. The fourth phase adjustment filter **86***d* has the filter coefficient FBP**1**.

[0031] The second phase adjustment filter **86***b* receives the basic signal −xs whose polarity has been inverted by the inverting amplifier **86***e*. The basic signal xc whose amplitude is adjusted by the first phase adjustment filter **86***a* and the basic signal −xs whose amplitude is adjusted by the second phase adjustment filter **86***b* are added by the adder **86***f*. Thus, the phase adjustment signal p**0**_*a* is generated.

[0032] The basic signal xs whose amplitude is adjusted by the third phase adjustment filter **86***c* and the basic signal xc whose amplitude is adjusted by the fourth phase adjustment filter **86***d* are added by the adder **86***g*. Thus, the phase adjustment signal p**1**_*a* is generated.

[0033] The signal extraction unit **88** performs signal processing on the phase adjustment signal p**0**_*a* and the phase adjustment signal p**1**_*a* using the extraction filter A. Thus, the extraction signals a**0**_*a* and a**1**_*a* are generated.

[0034] The signal extraction unit **88** includes a first extraction filter **88***a*, a second extraction filter **88***b*, a third extraction filter **88***c*, a fourth extraction filter **88***d*, an inverting amplifier **88***e*, an adder **88***f*, and an adder **88***g*.

[0035] The first extraction filter **88***a* has a filter coefficient A**0**. The second extraction filter **88***b* has a filter coefficient A**1**. The third extraction filter **88***c* has filter the coefficient A**0**. The fourth extraction filter **88***d* has the filter coefficient A**1**.

[0036] The phase adjustment signal p**0**_*a* whose amplitude has been adjusted by the first extraction filter **88***a* and the phase adjustment signal p**1**_*a* whose amplitude has been adjusted by the second extraction filter **88***b* are added by the adder **88***f*. Thus, the extraction signal a**0**_*a* is regenerated.

[0037] The phase adjustment signal −p**1**_*a* whose polarity is inverted by the inverting amplifier **88***e* is input to the third extraction filter **88***c*. The phase adjustment signal −p**1**_*a* whose amplitude is adjusted by the third extraction filter **88***c* and the phase adjustment signal p**0**_*a* whose amplitude is adjusted by the fourth extraction filter **88***d* are added by the adder **88***g*. Thus, an extraction signal a**1**_*a* is generated.

[0038] The gain adjusting unit **90** performs signal processing on the extraction signals a**0**_*a* and a**1**_*a* using the gain filters FBG. Thus, control signals u**0**_*a* and u**1**_*a* are generated.

[0039] The gain adjusting unit **90** includes a first gain adjustment filter **90***a* and a second gain adjustment filter **90***b*.

The first gain adjustment filter **90***a* has the gain FBG. The second gain adjustment filter **90***b* has the gain FBG.

[0040] The amplitude of the extraction signal a**0**_*a* is adjusted by the first gain adjustment filter **90***a*. Thus, the control signal u**0**_*a* is generated. The amplitude of the extraction signal a**1**_*a* is adjusted by the second gain adjustment filter **90***b*. Thus, the control signal u**1**_*a* is generated. The control signal u**0**_*a* is converted into an analog signal by a digital-to-analog converter **69** and output to the speaker **18**.

[0041] In the estimated canceling sound signal generating unit **70** described below, the control signal u**0**_*a* is used as a real component, and the control signal u**1**_*a* is used as an imaginary component.

[0042] The estimated canceling sound signal generating unit **70** performs signal processing on the control signals u**0**_*a* and u**1**_*a* by a secondary path filter C^. Thus, an estimated canceling sound signal y_a^ is generated.

[0043] In the estimated canceling sound signal generating unit **70**, an adaptive notch filter (for example, a SAN (Single-frequency Adaptive Notch) filter) is used as the secondary path filter C^. The secondary path filter C^ is updated by the secondary path filter updating unit **84** described later. As a result, the secondary path filter C^ converges on the sound transfer characteristic C in the secondary path. The secondary path filter C^ is indicated by C^=C**0**^+iC**1**^ using filter coefficients C**0**^ and C**1**^. Here, i denotes an imaginary number.

[0044] The estimated canceling sound signal generating unit **70** includes a first secondary path filter **70***a*, a second secondary path filter **70***b*, and an adder **70***c*.

[0045] The first secondary path filter **70***a* has the filter coefficient C**0**^. The second secondary path filter **70***b* has the filter coefficient C**1**^. The control signal u**0**_*a* whose amplitude is adjusted by the first secondary path filter **70***a* and the control signal u**1**_*a* whose amplitude is adjusted by the second secondary path filter **70***b* are added by the adder **70***c*. Thus, the estimated canceling sound signal y_a^ is generated.

[0046] The estimated noise signal generating unit **76** performs signal processing on the basic signal xc and the basic signal xs by an adjustment filter P. Thus, an estimated noise signal d_a^ is generated. In the estimated noise signal generating unit **76**, an adaptive notch filter (for example, a SAN filter) is used as the adjustment filter P for adjusting characteristics of the basic signal xc and the basic signal xs. The adjustment filter P is updated by the adjustment filter updating unit **82** described later. The adjustment filter P is indicated by P=P**0**+iP**1** using a filter coefficient P**0** and a filter coefficient P**1**. Here, i denotes an imaginary number.

[0047] The estimated noise signal generating unit **76** includes a first adjustment filter **76***a*, a second adjustment filter **76***b*, an inverting amplifier **76***c*, and an adder **76***d*. The first adjustment filter **76***a* has the filter coefficient P**0**. The second adjustment filter **76***b* has the filter coefficient P**1**.

[0048] The second adjustment filter **76***b* receives the basic signal −xs whose polarity has been inverted by the inverting amplifier **76***c*. The basic signal xc whose amplitude is adjusted by the first adjustment filter **76***a* and the basic signal −xs whose amplitude is adjusted by the second adjustment filter **76***b* are added by the adder **76***d*. Thus, the estimated noise signal d_a^ is generated.

[0049] The extraction signal generating unit **77** performs signal processing on the basic signal xc and the basic signal xs using an extraction filter A. Thus, an extraction signal efr is generated. In the extraction signal generating unit **77**, an adaptive notch filter (for example, a SAN filter) is used as the extraction filter A. The extraction filter A is updated and optimized by the extraction filter updating unit **85** described later. The extraction filter A has filter coefficients A**0** and A**1** which match the basic signals xc and xs to the amplitude and phase of the drumming noise.

[0050] The extraction signal generating unit **77** includes a first extraction filter **77***a*, a second extraction filter **77***b*, and an adder **77***c*. The first extraction filter **77***a* has the filter coefficient A**0**. The second extraction filter **77***b* has the filter coefficient A**1**.

[0051] The basic signal xc whose amplitude is adjusted by the first extraction filter **77***a* and the basic signal xs whose amplitude is adjusted by the second extraction filter **77***b* are added by the adder **77***c*. Thus, the extraction signal efr is generated.

[0052] The virtual error signal generating unit **78** generates a virtual error signal e**1** based on the error signal e, the estimated noise signal d_a^, and the estimated canceling sound signal y_a^. The virtual error signal generating unit **78** includes an inverting amplifier **78***a*, an inverting amplifier **78***b*, and an adder **78***c*.

[0053] The error signal e converted into a digital signal by an analog-to-digital converter **79**, the estimated noise signal −d_a^ whose polarity is inverted by the inverting amplifier **78***a*, and the estimated canceling sound signal −y_a^ whose polarity is inverted by the inverting amplifier **78***b* is added by the adder **78***c*. Thus, the virtual error signal e**1** is generated.

[0054] The differential signal generating unit **81** generates a differential signal e**0** based on the error signal e and the extraction signal efr. The differential signal generating unit **81** includes an adder **81***a*. The error signal e and the extraction signal efr are added by the adder **81***a*. As a result, the differential signal e**0** is generated.

[0055] The adjustment filter updating unit **82** sequentially and adaptively updates the adjustment filter P by an adaptive algorithm (for example, an LMS (Least Mean Square) algorithm) so that the virtual error signal e**1** is minimized.

[0056] The adjustment filter updating unit **82** includes a first adjustment filter coefficient updating unit **82***a* and a second adjustment filter coefficient updating unit **82***b*. The first adjustment filter coefficient updating unit **82***a* and the second adjustment filter coefficient updating unit **82***b* update the filter coefficient P**0** and the filter coefficient P**1** based on the following expressions. In the expressions, n denotes the number of time steps (time step number, n=0, 1, 2, . . .). The signal processing unit **22** performs signal processing at predetermined periods. The time step indicates the length of each period. The time step number indicates how many periods (times) the signal processing is performed. In the expressions, $\mu 0_P$ and $\mu 1_P$ indicate step size parameters.

$$P0_{n+1}=P0_n-\mu 0_P\times e1_n\times xc_n$$

$$P1_{n+1}=P1_n-\mu 1_P\times e1_n\times xs_n$$

[0057] The secondary path filter updating unit **84** sequentially and adaptively updates the secondary path filter C^ by an adaptive algorithm (for example, LMS algorithm) so that the virtual error signal e**1** is minimized.

[0058] The secondary path filter updating unit **84** includes a first secondary path filter coefficient updating unit **84***a* and a second secondary path filter coefficient updating unit **84***b*. The first secondary path filter coefficient updating unit **84***a* and the second secondary path filter coefficient updating unit **84***b* update the filter coefficient C**0**^ and the filter coefficient C**1**^ based on the following expressions. In the expressions, n denotes the time step number (n=0, 1, 2, . . .), and $\mu 0_C$ and $\mu 1_C$ denotes step size parameters.

$$C0\hat{}_{n+1}=C0\hat{}_n-\mu 0_C\times e1_n\times u0\_a_n$$

$$C1\hat{}_{n+1}=C1\hat{}_n-\mu 1_C\times e1_n\times u1\_a_n$$

[0059] The extraction filter updating unit **85** sequentially and adaptively updates the extraction filter A using an adaptive algorithm (for example, an LMS algorithm) so that the differential signal e**0** is minimized.

[0060] The extraction filter updating unit **85** includes a first extraction filter coefficient updating unit **85***a* and a second extraction filter coefficient updating unit **85***b*. The first extraction filter coefficient updating unit **85***a* and the second extraction filter coefficient updating unit **85***b* update the filter coefficient A**0** and the filter coefficient A**1** based on the following expressions. In the expressions, n denotes the time step number (n=0, 1, 2, . . .), and $\mu 0_A$ and $\mu 1_A$ denote step size parameters.

$$A0_{n+1}=A0_n-\mu 0_A\times e0_n\times xc_n$$

$$A1_{n+1}=A1_n-\mu 1_A\times e0_n\times xs_n$$

[Setting of Feedback Filter FB]

[0061] The feedback filter setting unit **23** sets the feedback filter FB based on the secondary path filter C^. Hereinafter, setting of the feedback filter FB will be described.

[0062] A sensitivity function S, which is a transfer function of the error signal e and the noise d, is expressed by the following expression. The sensitivity function S indicates a reduction amount of noise d.

$$S=\frac{E}{D}=\frac{1}{1+C\cdot FB}$$

[0063] In the expression, E is a frequency characteristic of the error signal e, and D is a frequency characteristic of the noise d. When the secondary path filter C^ is substituted for the transfer characteristic C of the secondary path, the feedback filter FB is expressed by the following expression.

$$FB=\frac{1-S}{S}\cdot\frac{1}{C\hat{}}$$

[0064] The value of the sensitivity function S is predetermined. For example, when the acoustic pressure of drumming noise is reduced by approximately 6 dB, the sensitivity function S is approximately 0.5. When the sensitivity function S=0.5, the feedback filter setting unit **23** sets the filter coefficient FBP**0** to a value obtained by normalizing the real part of 1/C^ with |1/C^|, and sets the filter coefficient FBP**1** to a value obtained by normalizing the imaginary part of 1/C^ with |1/C^|.

[0065] The feedback filter setting unit **23** sets the gain FBG so as to gradually increase from the initial value to 1/|C^|. In a state in which the number of updating of the secondary path filter C^ is small and learning has not progressed, the value of 1/|C^| may rapidly increase. Therefore, by gradually increasing the gain FBG, it is possible to prevent a sound that the vehicle occupant feels uncomfortable from being output from the speaker **18** at a high sound pressure. Here, the initial value of the gain FBG is not 0, but is set to a value small enough to prevent the speaker **18** from emitting a sound that the vehicle occupant feels uncomfortable. This is because if the initial value of the gain FBG is set to 0, the learning of the secondary path filter C^ does not proceed much.

[0066] Further, the feedback filter setting unit **23** may set the gain FBG to have an initial value when the gain |C^| of the secondary path filter C^ is equal to or smaller than a predetermined value. Since the gain FBG is set to have the initial value until the learning of the secondary path filter C^ proceeds, it is possible to prevent the speaker **18** from emitting a sound that the vehicle occupant feels uncomfortable.

[0067] Further, when the amount of change in gain or the amount of change in phase due to the updating of the secondary path filter C^ is equal to or larger than a predetermined amount, the feedback filter setting unit **23** may return the gain FBG to the initial value. When the position of the microphone **32** changes, the transfer characteristic C of the secondary path may change greatly. In this case, the secondary path filter C^ is relearned. Therefore, by once setting the gain FBG to have an initial value and gradually increasing the gain FBG from the initial value to 1/|C^|, it is possible to prevent a sound that the vehicle occupant feels uncomfortable from being output from the speaker **18** at a high sound pressure.

[Operational Effects]

[0068] In the active noise control device **10** according to the present embodiment, the control signal generating unit **68** performs signal processing on the basic signal xc and the basic signal xs by the feedback filter FB and the extraction filter A. Thus, the control signal u**0**_*a* for controlling the speaker **18** is generated. Further, the feedback filter setting unit **23** sets the feedback filter FB based on the secondary path filter C^. Furthermore, the secondary path filter updating unit **84** sequentially and adaptively updates the secondary path filter C^. As a result, even when the transfer characteristic C of the secondary path changes, the secondary path filter C^ can follow the transfer characteristic C. As a result, the control signal u**0**_*a* is generated in accordance with the change in the transfer characteristic C, so that the sound pressure of drumming noise can be reduced.

[0069] Further, in the active noise control device **10** according to the present embodiment, the feedback filter setting unit **23** sets the feedback filter FB based on the secondary path filter C^ and a predetermined noise reduction amount (sensitivity function S). As a result, it is possible to reduce the amount of calculation when setting the feedback filter FB, and to suppress the load on the computation unit.

[0070] In the active noise control device **10** according to the present embodiment, the feedback filter setting unit **23** gradually increases the gain FBG of the feedback filter FB from a predetermined initial value to a gain 1/|C^|. As a result, it is possible to prevent a sound that the vehicle occupant feels uncomfortable from being output from the speaker **18** at a high sound pressure.

[0071] Further, in the active noise control device **10** according to the present embodiment, the feedback filter setting unit **23** sets the gain FBG of the feedback filter FB to a predetermined initial value when the gain |C^| of the secondary path filter C^ is equal to or smaller than a predetermined value. As a result, the gain FBG is set to have the initial value until the learning of the secondary path filter C^ proceeds, and thus it is possible to prevent the speaker **18** from emitting a sound that the vehicle occupant feels uncomfortable.

[0072] Further, in the active noise control device **10** according to the present embodiment, the feedback filter setting unit **23** sets the gain FBG of the feedback filter FB to have a predetermined initial value when the amount of change in the gain of the secondary path filter C^ or the amount of change in the phase of the secondary path filter C^ is equal to or greater than a predetermined amount. As a result, it is possible to prevent a sound that the vehicle occupant feels uncomfortable from being output from the speaker **18** at a high sound pressure.

Second Embodiment

[0073] In the active noise control device **10** according to the present embodiment, the configuration of the signal processing unit **22** is partially different from that of the signal processing unit **22** according to the first embodiment. In addition, a method of setting the feedback filter FB by the feedback filter setting unit **23** is different from the feedback filter setting unit **23** of the first embodiment.

[Configuration of Signal Processing Unit]

[0074] The signal processing unit **22** performs feedback signal processing. In the feedback signal processing, a control signal u**0**_*b* is generated. The control signal u**0**_*b* is a signal for causing the speaker **18** to output a canceling sound that cancels the drumming noise. The control signal u**0**_*b* is generated on the basis of an error signal e output from the microphone **32** provided at the control point. Hereinafter, a sound transfer path from the wheel **16** to the microphone **32** is referred to as a primary path, and a transfer characteristic of the primary path is denoted by H. Further, a sound transfer path from the speaker **18** to the microphone **32** is referred to as a secondary path, and a transfer characteristic of the secondary path is denoted by C.

[0075] FIG. **4** is a control block diagram of the signal processing unit **22**. The signal processing unit **22** includes a basic signal generating unit **67**, a control signal generating unit **68**, an estimated canceling sound signal generating unit **70**, an estimated noise signal generating unit **75**, an extraction signal generating unit **77**, a virtual error signal generating unit **78**, a differential signal generating unit **81**, a primary path filter updating unit **83**, a secondary path filter updating unit **84**, and an extraction filter updating unit **85**.

[0076] The basic signal generating unit **67** generates basic signals xc ($=\cos(2\pi \times fx \times t)$) and xs ($=\sin(2\pi \times fx \times t)$). The basic signal xc is a cosine signal of the control target frequency fx. The basic signal xs is a sine signal of the control target frequency fx. Here, t denotes time. The control target frequency fx is set near the peak frequency of the drumming noise in advance.

[0077] The control signal generating unit **68** performs signal processing on the basic signal xc and the basic signal xs by the feedback filter FB and the extraction filter A. Thus, the control signals u**0**_*b* and u**1**_*b* are generated. The control signal generating unit **68** includes a signal extraction unit **92**, a phase adjusting unit **94**, and a gain adjusting unit **96**.

[0078] The feedback filter FB is indicated by FB=FBG (FBP**0**+iFBP**1**) using the gain FBG, a filter coefficient FBP**0**, and a filter coefficient FBP**1**. Here, i denotes an imaginary number. Further, $FBP0^2+FBP1^2=1$. The feedback filter FB is set by the feedback filter setting unit **23**. The setting of the feedback filter FB will be described later in detail.

[0079] The signal extraction unit **92** performs signal processing on the basic signal xc and the basic signal xc by an extraction filter A. Thus, the extraction signals a**0**_*b* and a**1**_*b* are generated.

[0080] The signal extraction unit **92** includes a first extraction filter **92***a*, a second extraction filter **92***b*, a third extraction filter **92***c*, a fourth extraction filter **92***d*, an inverting amplifier **92***e*, an adder **92***f*, and an adder **92***g*.

[0081] The first extraction filter **92***a* has a filter coefficient A**0**. The second extraction filter **92***b* has a filter coefficient A**1**. The third extraction filter **92***c* has the filter coefficient A**0**. The fourth extraction filter **92***d* has the filter coefficient A**1**.

[0082] The second extraction filter **92***b* receives the basic signal −xs whose polarity is inverted by the inverting amplifier **92***e*. The basic signal xc whose amplitude is adjusted by the first extraction filter **92***a* and the basic signal −xs whose amplitude is adjusted by the second extraction filter **92***b* are added by the adder **92***f*. Thus, the extraction signal a**0**_*b* is generated.

[0083] The basic signal xs whose amplitude is adjusted by the third extraction filter **92***c* and the basic signal xc whose amplitude is adjusted by the fourth extraction filter **92***d* are added by the adder **92***g*. Thus, the extraction signal a**1**_*b* is generated.

[0084] The phase adjusting unit **94** performs signal processing on the extraction signals a**0**_*b* and a**1**_*b* by the phase adjustment filter FBP. As a result, the phase adjustment signals p**0**_*b* and p**1**_*b* are generated.

[0085] The phase adjusting unit **94** includes a first phase adjustment filter **94***a*, a second phase adjustment filter **94***b*, a third phase adjustment filter **94***c*, a fourth phase adjustment filter **94***d*, an inverting amplifier **94***e*, an adder **94***f*, and an adder **94***g*.

[0086] The first phase adjustment filter **94***a* has a filter coefficient FBP**0**. The second phase adjustment filter **94***b* has a filter coefficient FBP**1**. The third phase adjustment filter **94***c* has the filter coefficient FBP**0**. The fourth phase adjustment filter **94***d* has the filter coefficient FBP**1**.

[0087] The extraction signal a**0**_*b* whose amplitude is adjusted by the first phase adjustment filter **94***a* and the extraction signal a**1**_*b* whose amplitude is adjusted by the second phase adjustment filter **94***b* are added in the adder **94***f*. Thus, the phase adjustment signal p**0**_*b* is generated.

[0088] The third phase adjustment filter **94***c* receives the extraction signal −a**1**_*b* whose polarity is inverted by the inverting amplifier **94***e*. The extracted signal −a**1**_*b* whose amplitude is adjusted by the third phase adjustment filter **94***c* and the extraction signal a**0**_*b* whose amplitude is adjusted by the fourth phase adjustment filter **94***d* are added by the adder **94***g*. Thus, the phase adjustment signal p**1**_*b* is generated.

[0089] The gain adjusting unit **96** performs signal processing on the phase adjustment signals p**0**_*b* and p**1**_*b* by the gain filter FBG. Thus, the control signals u**0**_*b* and u**1**_*b* are generated.

[0090] The gain adjusting unit **96** includes a first gain adjustment filter **96***a* and a second gain adjustment filter **96***b*.

The first gain adjustment filter **96**_a_ has a gain FBG. The second gain adjustment filter **96**_b_ has the gain FBG.

[0091] The amplitude of the phase adjustment signal p**0**_b is adjusted by the first gain adjustment filter **96**_a_. Thus, the control signal u**0**_b is generated. The amplitude of the phase adjustment signal p**1**_b is adjusted by the second gain adjustment filter **96**_b_. Thus, the control signal u**1**_b is generated. The control signal u**0**_b is converted into an analog signal by the digital-to-analog converter **69** and output to the speaker **18**.

[0092] In the estimated canceling sound signal generating unit **70** described below, the control signal u**0**_b is used as a real component, and the control signal u**1**_b is used as an imaginary component.

[0093] The estimated canceling sound signal generating unit **70** performs signal processing on the control signals u**0**_b and u**1**_b by a secondary path filter C^. Thus, the estimated canceling sound signal y_b^ is generated.

[0094] In the estimated canceling sound signal generating unit **70**, an adaptive notch filter (for example, a SAN filter) is used as the secondary path filter C^. The secondary path filter C^ is updated by the secondary path filter updating unit **84**, which will be described later, so as to converge on a sound transfer characteristic C of the secondary path. The secondary path filter C^ is indicated by C^=C**0**^+iC**1**^ using the filter coefficients C**0**^ and C**1**^. Here, i denotes an imaginary number.

[0095] The estimated canceling sound signal generating unit **70** includes a first secondary path filter **70**_a_, a second secondary path filter **70**_b_, and an adder **70**_c_.

[0096] The first secondary path filter **70**_a_ has a filter coefficient C**0**^. The second secondary path filter **70**_b_ has a filter coefficient C**1**^. The control signal u**0**_b whose amplitude is adjusted by the first secondary path filter **70**_a_ and the control signal u**1**_b whose amplitude is adjusted by the second secondary path filter **70**_b_ are added by the adder **70**_c_. Thus, the estimated canceling sound signal y_b^ is generated.

[0097] The estimated noise signal generating unit **75** performs signal processing on the extraction signals a**0**_b and a**1**_b by a primary path filter H^. As a result, an estimated noise signal d_b^ is generated.

[0098] In the estimated noise signal generating unit **75**, an adaptive notch filter (for example, a SAN filter) is used as the primary path filter H^. The primary path filter H^ is updated by the primary path filter updating unit **83**, which will be described later, so as to converge on a sound transfer characteristic H of the primary path. The primary path filter H^ is indicated by H^=H**0**^+iH**1**^ using the filter coefficients H**0**^ and H**1**^. Here, i denotes an imaginary number.

[0099] The estimated noise signal generating unit **75** includes a first primary path filter **75**_a_, a second primary path filter **75**_b_, an inverting amplifier **75**_c_, and an adder **75**_d_. The first primary path filter **75**_a_ has a filter coefficient H**0**^. The second primary path filter **75**_b_ has a filter coefficients H**1**^.

[0100] The second primary path filter **75**_b_ receives the extraction signal −a**1**_b whose polarity is inverted by the inverting amplifier **75**_c_. The extracted signal a**0**_b whose amplitude is adjusted by the first primary path filter **75**_a_ and the extraction signal −a**1**_b whose amplitude is adjusted by the second primary path filter **75**_b_ are added by the adder **75**_d_. As a result, an estimated noise signal d_b^ is generated.

[0101] The extraction signal generating unit **77** performs signal processing on the basic signal xc and the basic signal xs by an extraction filter A. Thus, an extraction signal efr is generated. In the extraction signal generating unit **77**, an adaptive notch filter (for example, a SAN filter) is used as the extraction filter A. The extraction filter A is updated and optimized by the extraction filter updating unit **85** described later. The extraction filter A has filter coefficients A**0** and A**1** which match the basic signals xc and xs to the amplitude and phase of the drumming noise.

[0102] The extraction signal generating unit **77** includes a first extraction filter **77**_a_, a second extraction filter **77**_b_, and an adder **77**_c_. The first extraction filter **77**_a_ has the filter coefficient A**0**. The second extraction filter **77**_b_ has the filter coefficient A**1**.

[0103] The basic signal xc whose amplitude is adjusted by the first extraction filter **77**_a_ and the basic signal xs whose amplitude is adjusted by the second extraction filter **77**_b_ are added by the adder **77**_c_. Thus, an extraction signal efr is generated.

[0104] The virtual error signal generating unit **78** generates a virtual error signal e**2** based on the error signal e, the estimated noise signal d_b^, and the estimated canceling sound signal y_b^. The virtual error signal generating unit **78** includes an inverting amplifier **78**_a_, an inverting amplifier **78**_b_, and an adder **78**_c_.

[0105] The error signal e converted into a digital signal by an analog-to-digital converter **79**, the estimated noise signal −d_b^ whose polarity is inverted by the inverting amplifier **78**_a_, and the estimated canceling sound signal −y_b^ whose polarity is inverted by the inverting amplifier **78**_b_ is added by the adder **78**_c_. Thus, the virtual error signal e**2** is generated.

[0106] The differential signal generating unit **81** generates a differential signal e**0** based on the error signal e and the extraction signal efr. The differential signal generating unit **81** includes an adder **81**_a_. The error signal e and the extraction signal efr are added by the adder **81**_a_ to generate the differential signal e**0**.

[0107] The primary path filter updating unit **83** sequentially and adaptively updates the primary path filter H^ by an adaptive algorithm (for example, an LMS algorithm) so that the virtual error signal e**2** is minimized.

[0108] The primary path filter updating unit **83** includes a first primary path filter coefficient updating unit **83**_a_ and a second primary path filter coefficient updating unit **83**_b_. The first primary path filter coefficient updating unit **83**_a_ and the second primary path filter coefficient updating unit **83**_b_ update the filter coefficient H**0**^ and the filter coefficient H**1**^ based on the following expressions. In the expressions, n denotes the time step number (n=0, 1, 2, . . .), and $\mu 0_H$ and $\mu 1_H$ denote step size parameters.

$$H0\hat{}_{n+1}=H0\hat{}_n-\mu 0_H \times e2_n \times a0\_b_n$$

$$H1\hat{}_{n+1}=H1\hat{}_n-\mu 1_H \times e2_n \times a1\_b_n$$

[0109] The secondary path filter updating unit **84** sequentially and adaptively updates the secondary path filter C^ by an adaptive algorithm (for example, LMS algorithm) so that the virtual error signal e**2** is minimized.

[0110] The secondary path filter updating unit **84** includes a first secondary path filter coefficient updating unit **84**_a_ and a second secondary path filter coefficient updating unit **84**_b_. The first secondary path filter coefficient updating unit **84**_a_ and the second secondary path filter coefficient updating unit **84**_b_ update the filter coefficient C**0**^ and the filter coefficient C**1**^ based on the following expressions. In the expressions, n denotes the time step number (n=0, 1, 2, . . .), and $\mu 0_C$ and $\mu 1_C$ denotes step size parameters.

$$C0\hat{}_{n+1}=C0\hat{}_n-\mu 0_C\times e2_n\times u0\_b_n$$

$$C1\hat{}_{n+1}=C1\hat{}_n-\mu 1_C\times e2_n\times u1\_b_n$$

[0111] The extraction filter updating unit **85** sequentially and adaptively updates the extraction filter A using an adaptive algorithm (for example, an LMS algorithm) so that the differential signal e**0** is minimized.

[0112] The extraction filter updating unit **85** includes a first extraction filter coefficient updating unit **85***a* and a second extraction filter coefficient updating unit **85***b*. The first extraction filter coefficient updating unit **85***a* and the second extraction filter coefficient updating unit **85***b* update the filter coefficient A**0** and the filter coefficient A**1** based on the following expressions. In the expressions, n denotes the time step number (n=0, 1, 2, . . .), and $\mu 0_A$ and $\mu 1_A$ denote step size parameters.

$$A0_{n+1}=A0_n-\mu 0_A\times e0_n\times xc_n$$

$$A1_{n+1}=A1_n-\mu 1_A\times e0_n\times xs_n$$

[Setting of Feedback Filter FB]

[0113] The feedback filter setting unit **23** sets the feedback filter FB based on the primary path filter H^ and the secondary path filter C^. Hereinafter, setting of the feedback filter FB will be described.

[0114] When the primary path filter H^ converges on the transfer characteristic H of the primary path and the secondary path filter C^ converges on the transfer characteristic C of the secondary path, the primary path filter H^ is expressed by the following expression.

$$H\hat{}=C\hat{}\cdot FB$$

[0115] When this expression is solved for the feedback filter FB, the feedback filter FB is expressed by the following expression.

$$FB=H\hat{}/C\hat{}$$

[0116] The feedback filter setting unit **23** sets the filter coefficient FBP**0** to a value obtained by normalizing the real part of H^/C^ with |H^/C^|. The feedback filter setting unit **23** sets the filter coefficient FBP**1** to a value obtained by normalizing the imaginary part of H^/C^ with |H^/C^|.

[0117] The feedback filter setting unit **23** gradually increases the gain FBG from the initial value to |1/C^|. In a state where the number of updating of the primary path filter H^ and the secondary path filter C^ is small and learning has not progressed, the value of |1/C^| may rapidly increase. Therefore, by gradually increasing the gain FBG, it is possible to prevent a sound that the vehicle occupant feels uncomfortable from being output from the speaker **18** at a high sound pressure. Here, the initial value of the gain FBG is not 0, but is set to a value small enough to prevent the speaker **18** from emitting a sound that the vehicle occupant feels uncomfortable. This is because if the initial value of the gain FBG is set to 0, the learning of the secondary path filter C^ does not proceed much.

[0118] The feedback filter setting unit **23** may set the gain FBG to an initial value when the gain |H^| of the primary path filter H^ or the gain |C^| of the secondary path filter C^ is equal to or less than a predetermined value. Since the gain FBG is set to the initial value until the learning of the primary path filter H^ and the secondary path filter C^ proceeds, it is possible to prevent the speaker **18** from emitting a sound that the vehicle occupant feels uncomfortable.

[0119] When at least one of the following four conditions is satisfied, the feedback filter setting unit **23** may return the gain FBG to the initial value. The four conditions are the following (1) to (4).

[0120] (1) The amount of change in gain due to the updating of the primary path filter H^ is equal to or greater than a predetermined amount.

[0121] (2) The amount of change in phase due to the updating of the primary path filter H^ is equal to or greater than a predetermined amount.

[0122] (3) The amount of change in gain due to the updating of the secondary path filter C^ is equal to or greater than a predetermined amount.

[0123] (4) The amount of change in phase due to the updating of the secondary path filter C^ is equal to or greater than a predetermined amount.

[0124] When the position of the microphone **32** changes, the transfer characteristic C of the secondary path may change greatly. In this case, the secondary path filter C^ is relearned. Then, by once setting the gain FBG to have an initial value and gradually increasing the gain FBG from the initial value to |H^/C^|, it is possible to prevent a sound that the vehicle occupant feels uncomfortable from being output from the speaker **18** at a high sound pressure.

[Advantageous Effects]

[0125] In the active noise control device **10** according to the present embodiment, the control signal generating unit **68** performs signal processing on the basic signal xc and the basic signal xs by the feedback filter FB and the extraction filter A. Thus, the control signal u**0**_*b* for controlling the speaker **18** is generated. Further, the feedback filter setting unit **23** sets the feedback filter FB based on the secondary path filter C^. Furthermore, the secondary path filter updating unit **84** sequentially and adaptively updates the secondary path filter C^. As a result, even when the transfer characteristic C of the secondary path changes, the secondary path filter C^ can follow the transfer characteristic C. As a result, the control signal u**0**_*b* can be generated in accordance with the change in the transfer characteristic C, so that the sound pressure of drumming noise can be reduced.

[0126] In the active noise control device **10** according to the present embodiment, the feedback filter setting unit **23** sets the feedback filter FB based on the primary path filter H^ and the secondary path filter C^. As a result, it is possible to reduce the amount of calculation when setting the feedback filter FB, and to suppress the load on the computation unit.

[Technical Concepts Obtained from Embodiments]

[0127] A description will be given below concerning technical concepts that are capable of being grasped from the above-described embodiments.

[0128] The active noise control device (**10**) controls the speaker (**18**) based on a component of a frequency band around a predetermined frequency of an error signal output from the detector (**32**) that detects, at a control point, a synthetic sound of a noise transmitted from a vibration source and a canceling sound output from the speaker to cancel the noise, and includes the basic signal generating unit (**67**) configured to generate a basic signal corresponding to the predetermined frequency, the control signal generating unit (**68**) configured to perform signal processing on the basic signal by a feedback filter and an extraction filter, which is an adaptive notch filter, to generate a control signal that controls the speaker, the estimated canceling sound signal generating unit (**70**) configured to perform signal processing on the control signal by a secondary path filter, which is an adaptive notch filter, to generate an estimated canceling sound signal, the extraction signal generating unit (**77**) configured to perform signal processing on the basic signal by the extraction filter to generate an extraction signal, the virtual error signal generating unit (**78**) configured to generate a virtual error signal from the error signal and the estimated canceling sound signal, the differential signal generating unit (**81**) configured to generate a differential signal from the error signal and the extraction signal, the secondary path filter updating unit (**84**) configured to update sequentially and adaptively the secondary path filter based on the control signal and the virtual error signal in a manner that a magnitude of the virtual error signal is minimized, the extraction filter updating unit (**85**) configured to update sequentially and adaptively the extraction filter based on the basic signal and the differential signal in a manner that a magnitude of the differential signal is minimized, and the feedback filter setting unit (**23**) configured to set the feedback filter based on the secondary path filter.

[0129] In the above active noise control device, the feedback filter setting unit may increase a gain of the feedback filter gradually from a predetermined initial value.

[0130] In the above active noise control device, the feedback filter setting unit may set a gain of the feedback filter to a predetermined initial value when a gain of the secondary path filter is equal to or less than a predetermined value.

[0131] In the above active noise control device, the feedback filter setting unit may set a gain of the feedback filter to a predetermined initial value when an amount of change in gain or an amount of change in phase of the secondary path filter is equal to or greater than a predetermined amount.

[0132] In the above active noise control device, the feedback filter setting unit may set the feedback filter based on the secondary path filter and a predetermined noise reduction amount.

[0133] The active noise control device may further include the estimated noise signal generating unit (**75**) configured to perform signal processing on the extraction signal by a primary path filter, which is an adaptive notch filter, to generate an estimated noise signal, and the primary path filter updating unit (**83**) configured to update the primary path filter based on the basic signal and the virtual error signal in a manner that the magnitude of the virtual error signal is minimized, wherein the virtual error signal generating unit may generate the virtual error signal from the error signal, the estimated noise signal, and the estimated canceling sound signal, and the feedback filter setting unit may calculate the feedback filter from the primary path filter and the secondary path filter.

[0134] The present invention is not particularly limited to the embodiments described above, and various modifications are possible without departing from the essence and gist of the present invention.

What is claimed is:

1. An active noise control device that controls a speaker based on a component of a frequency band around a predetermined frequency of an error signal output from a detector that detects, at a control point, a synthetic sound of a noise transmitted from a vibration source and a canceling sound output from the speaker to cancel the noise, the active noise control device comprising one or more processors that execute computer-executable instructions stored in a memory, wherein the one or more processors execute the computer-executable instructions to cause the active noise control device to:

generate a basic signal corresponding to the predetermined frequency;

perform signal processing on the basic signal by a feedback filter and an extraction filter, which is an adaptive notch filter, to generate a control signal that controls the speaker;

perform signal processing on the control signal by a secondary path filter, which is an adaptive notch filter, to generate an estimated canceling sound signal;

perform signal processing on the basic signal by the extraction filter to generate an extraction signal;

generate a virtual error signal from the error signal and the estimated canceling sound signal;

generate a differential signal from the error signal and the extraction signal;

update sequentially and adaptively the secondary path filter based on the control signal and the virtual error signal in a manner that a magnitude of the virtual error signal is minimized;

update sequentially and adaptively the extraction filter based on the basic signal and the differential signal in a manner that a magnitude of the differential signal is minimized; and

set the feedback filter based on the secondary path filter.

2. The active noise control device according to claim **1**, wherein the one or more processors cause the active noise control device to increase a gain of the feedback filter gradually from a predetermined initial value.

3. The active noise control device according to claim **1**, wherein the one or more processors cause the active noise control device to set a gain of the feedback filter to a predetermined initial value when a gain of the secondary path filter is equal to or less than a predetermined value.

4. The active noise control device according to claim **1**, wherein the one or more processors cause the active noise control device to set a gain of the feedback filter to a predetermined initial value when an amount of change in gain or an amount of change in phase of the secondary path filter is equal to or greater than a predetermined amount.

5. The active noise control device according to claim **1**, wherein the one or more processors cause the active noise control device to set the feedback filter based on the secondary path filter and a predetermined noise reduction amount.

6. The active noise control device according to claim **1**, wherein the one or more processors cause the active noise control device to:

perform signal processing on the extraction signal by a primary path filter, which is an adaptive notch filter, to generate an estimated noise signal;

update the primary path filter based on the basic signal and the virtual error signal in a manner that the magnitude of the virtual error signal is minimized;

generate the virtual error signal from the error signal, the estimated noise signal, and the estimated canceling sound signal; and

calculate the feedback filter from the primary path filter and the secondary path filter.

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