



US009271075B2

(12) **United States Patent**
Matsuo

(10) **Patent No.:** **US 9,271,075 B2**
(45) **Date of Patent:** **Feb. 23, 2016**

(54) **SIGNAL PROCESSING APPARATUS AND SIGNAL PROCESSING METHOD**

381/122, 107, 91; 704/226, 233, E21.004, 704/E21.002, 225

See application file for complete search history.

(71) Applicant: **FUJITSU LIMITED**, Kawasaki-shi (JP)

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(72) Inventor: **Naoshi Matsuo**, Yokohama (JP)

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(73) Assignee: **FUJITSU LIMITED**, Kawasaki (JP)

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(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 54 days.

(21) Appl. No.: **13/685,079**

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(22) Filed: **Nov. 26, 2012**

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(65) **Prior Publication Data**

US 2013/0156221 A1 Jun. 20, 2013

EP	1450353	A1	8/2004
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(30) **Foreign Application Priority Data**

Dec. 15, 2011 (JP) 2011-274742

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(51) **Int. Cl.**
H04R 3/00 (2006.01)
H04R 1/02 (2006.01)
H04R 1/40 (2006.01)

Primary Examiner — Xu Mei

(74) Attorney, Agent, or Firm — Fujitsu Patent Center

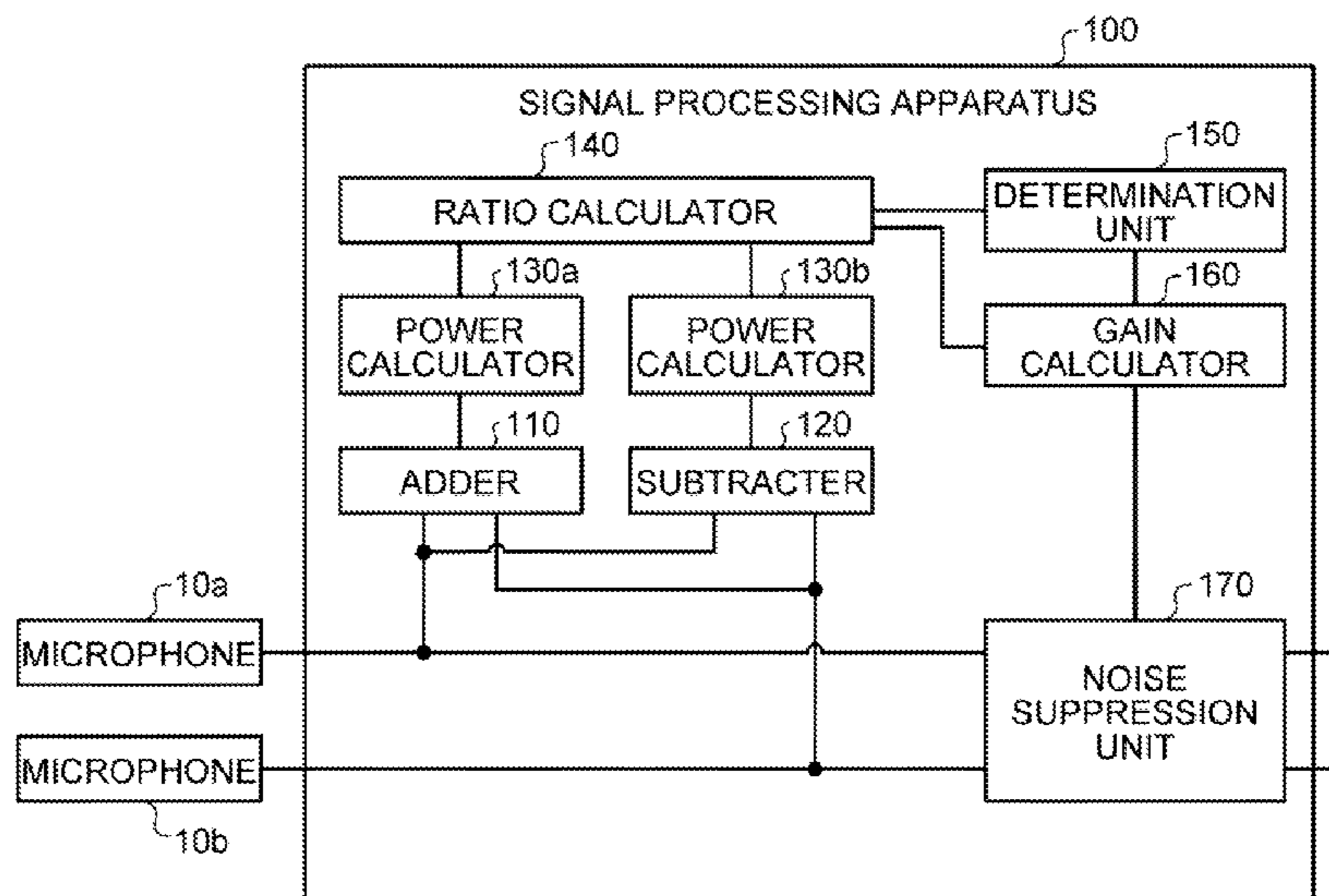
(52) **U.S. Cl.**
CPC *H04R 3/005* (2013.01); *H04R 1/406* (2013.01); *H04R 2201/403* (2013.01); *H04R 2430/20* (2013.01)

(57) **ABSTRACT**

A signal processing apparatus includes an adder that acquires a plurality of input signals from a plurality of microphones and calculates an added value obtained by adding the input signals together, a subtracter that acquires a plurality of input signals from the plurality of microphones and calculates a subtracted value obtained by subtracting one input signal from the other input signal, and a determination unit that determines whether noise is included in the input signals based on the added value and the subtracted value.

(58) **Field of Classification Search**
CPC H04R 3/005; H04R 2410/05; H04R 2410/07; H04R 25/407; H04R 25/505; H04R 2225/43; H04R 2499/11; H04R 2430/20; G10L 21/02; G10L 21/0208; G10L 21/0232; G10L 2021/0216; G10L 2021/0272; G10L 2021/02165–2021/02166; G10L 25/78; G10L 15/20
USPC 381/94.1–94.3, 92, 317, 71.1, 71.11,

10 Claims, 10 Drawing Sheets



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FIG. 1

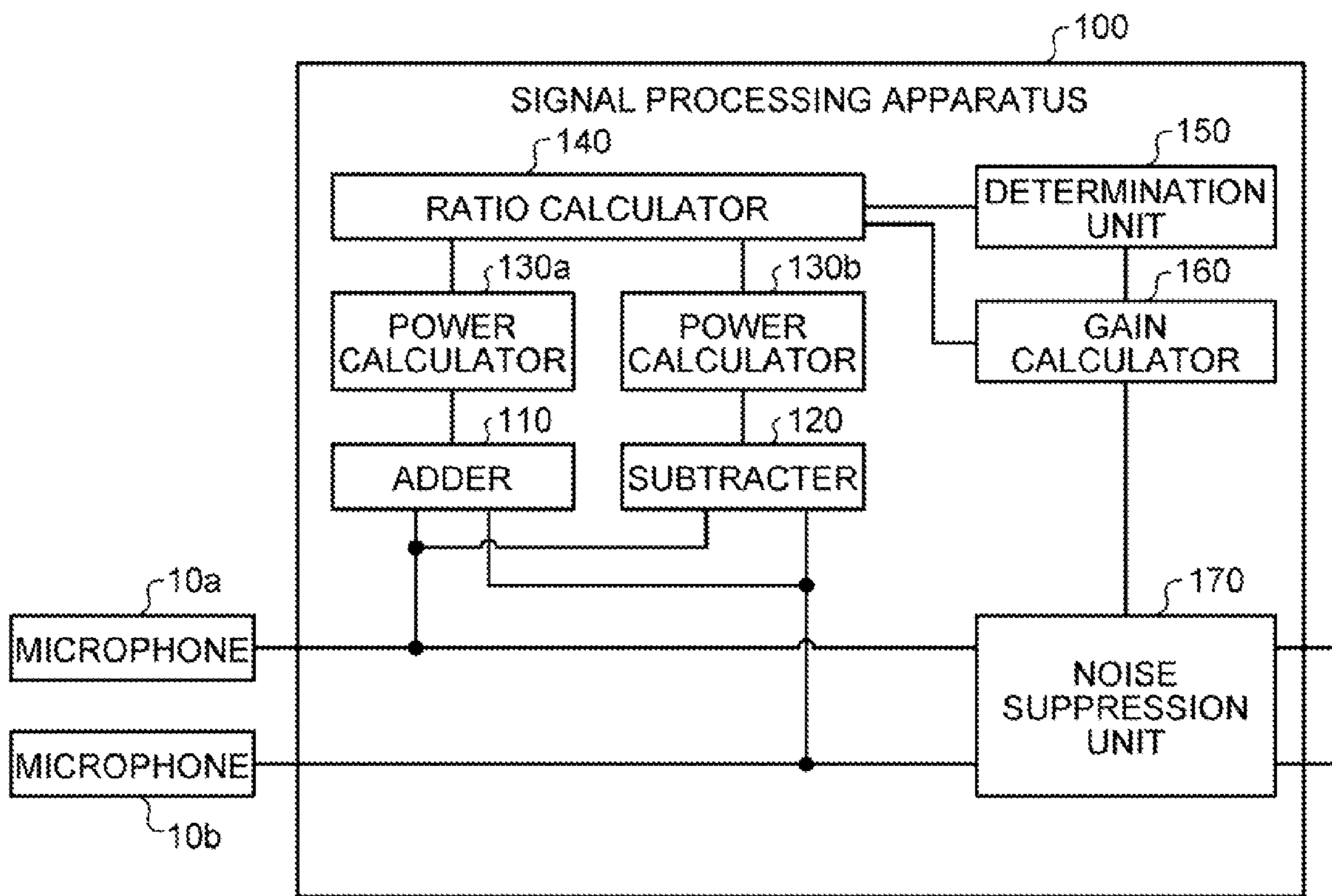


FIG.2

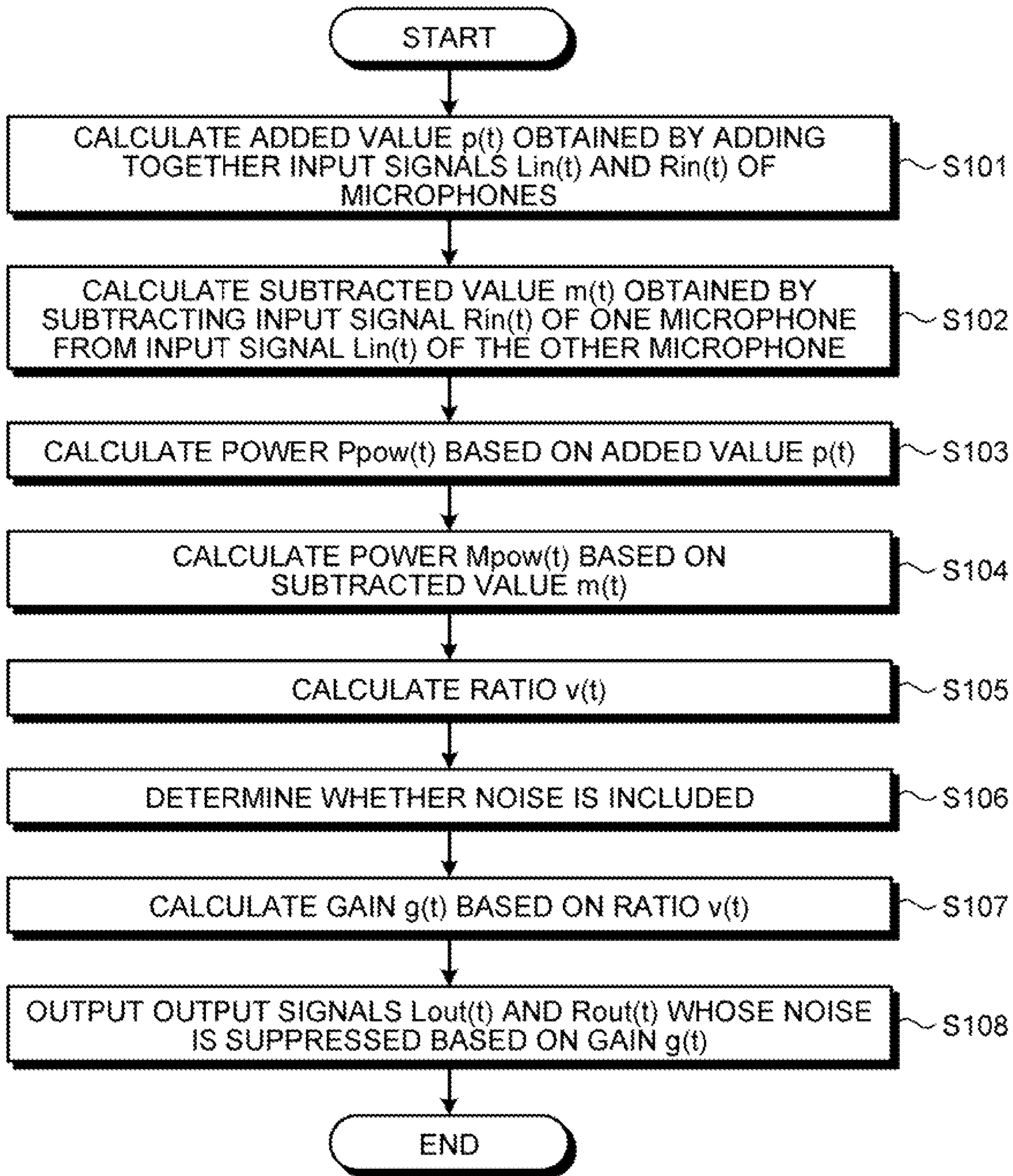


FIG. 3

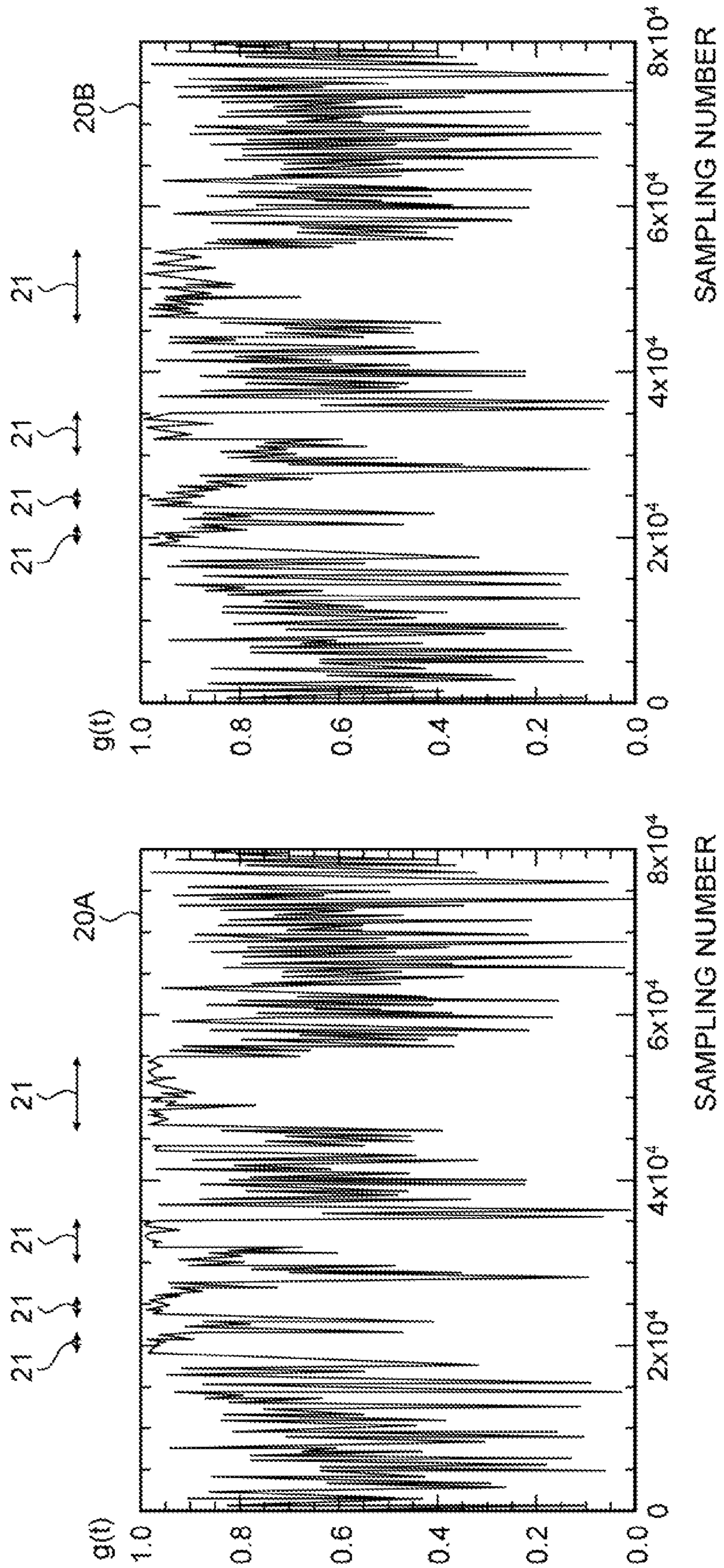


FIG.4

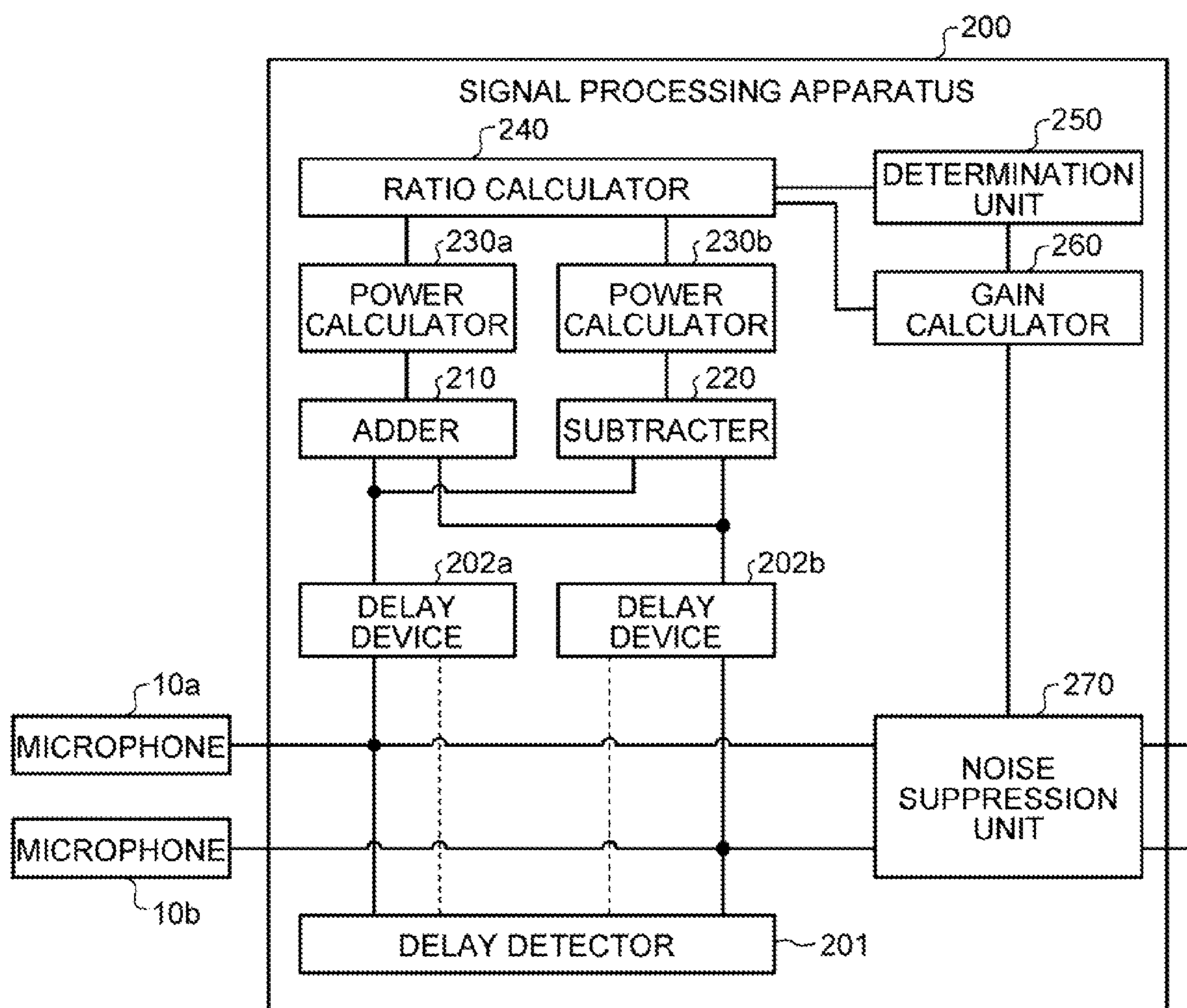


FIG.5

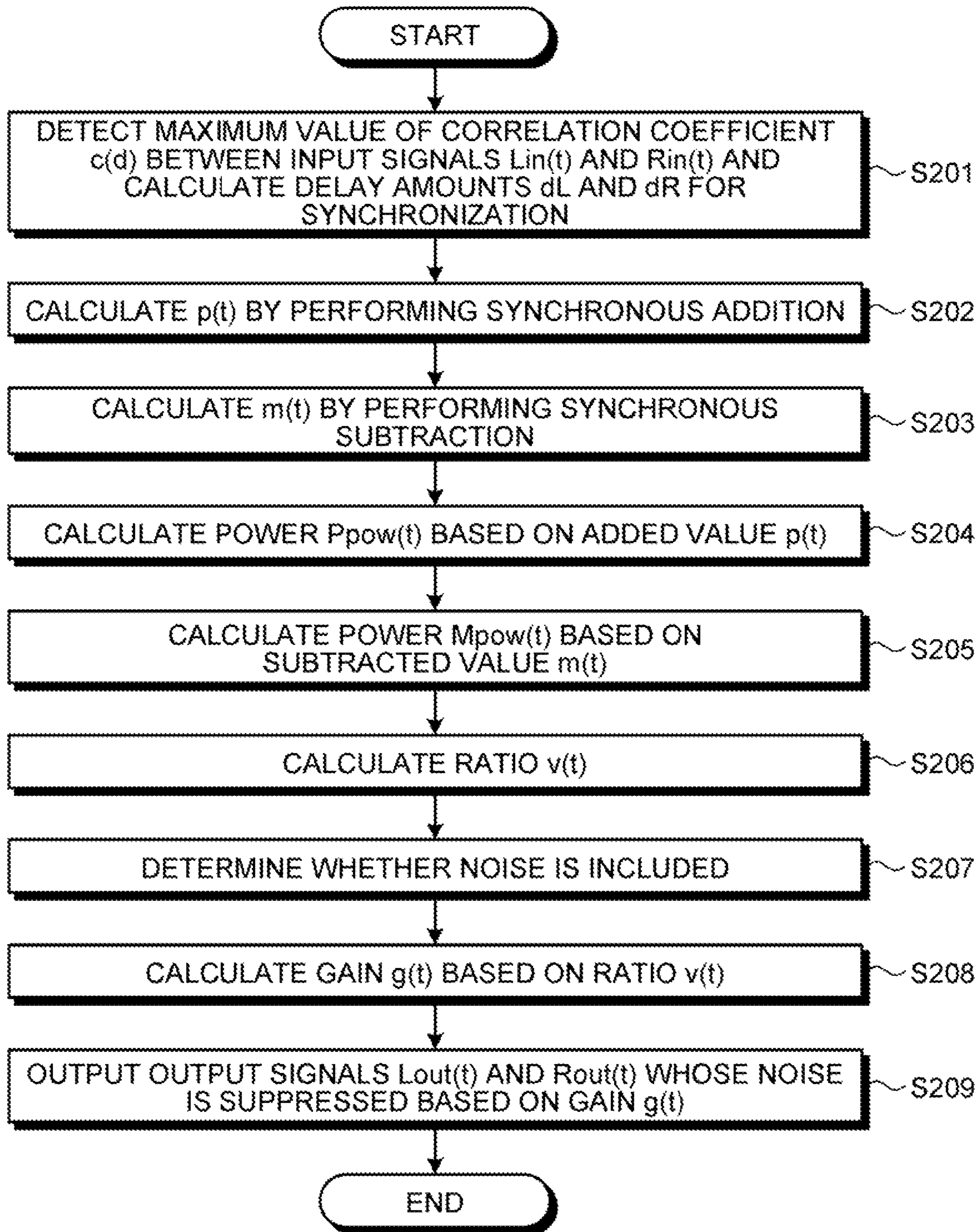


FIG.6

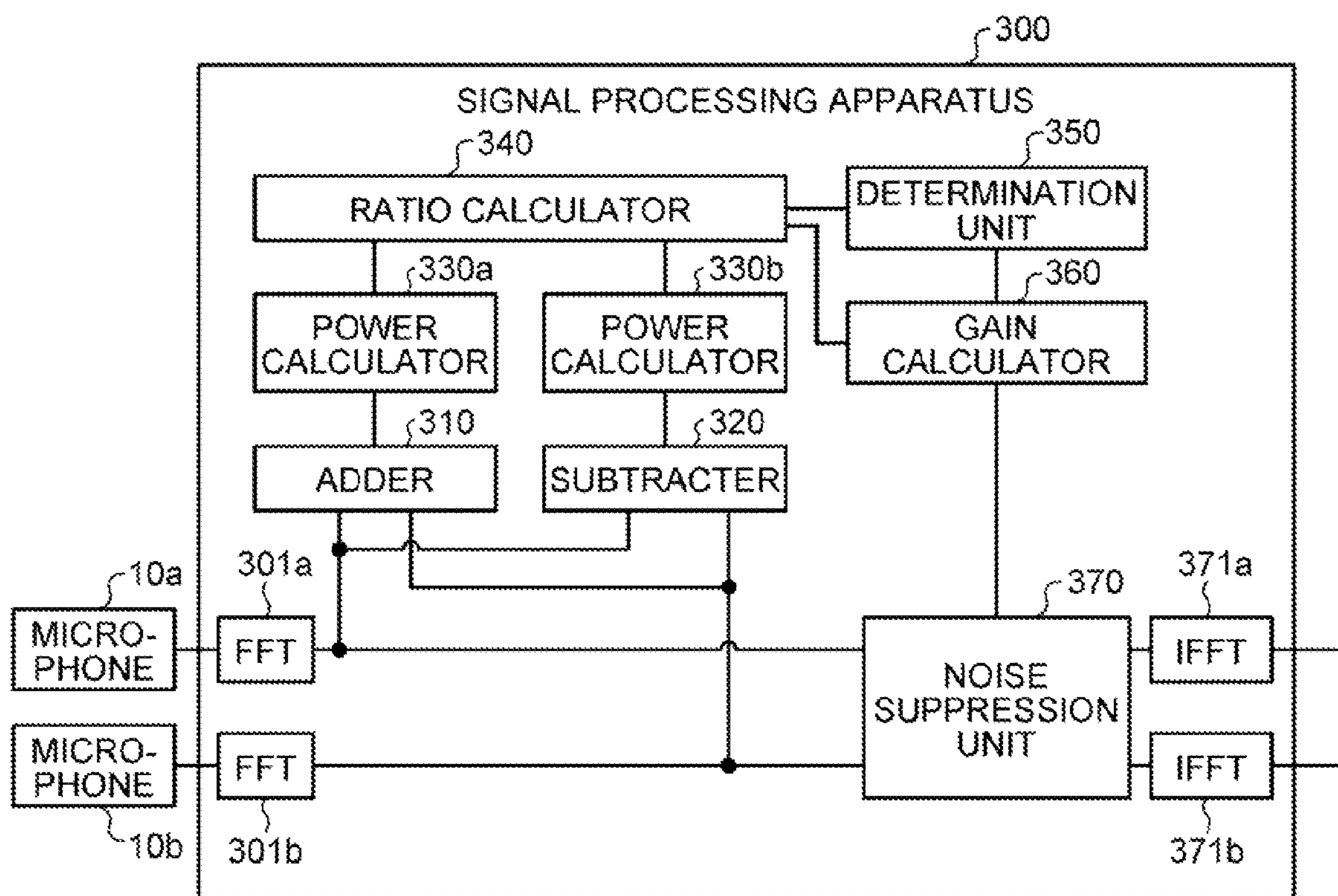


FIG. 7

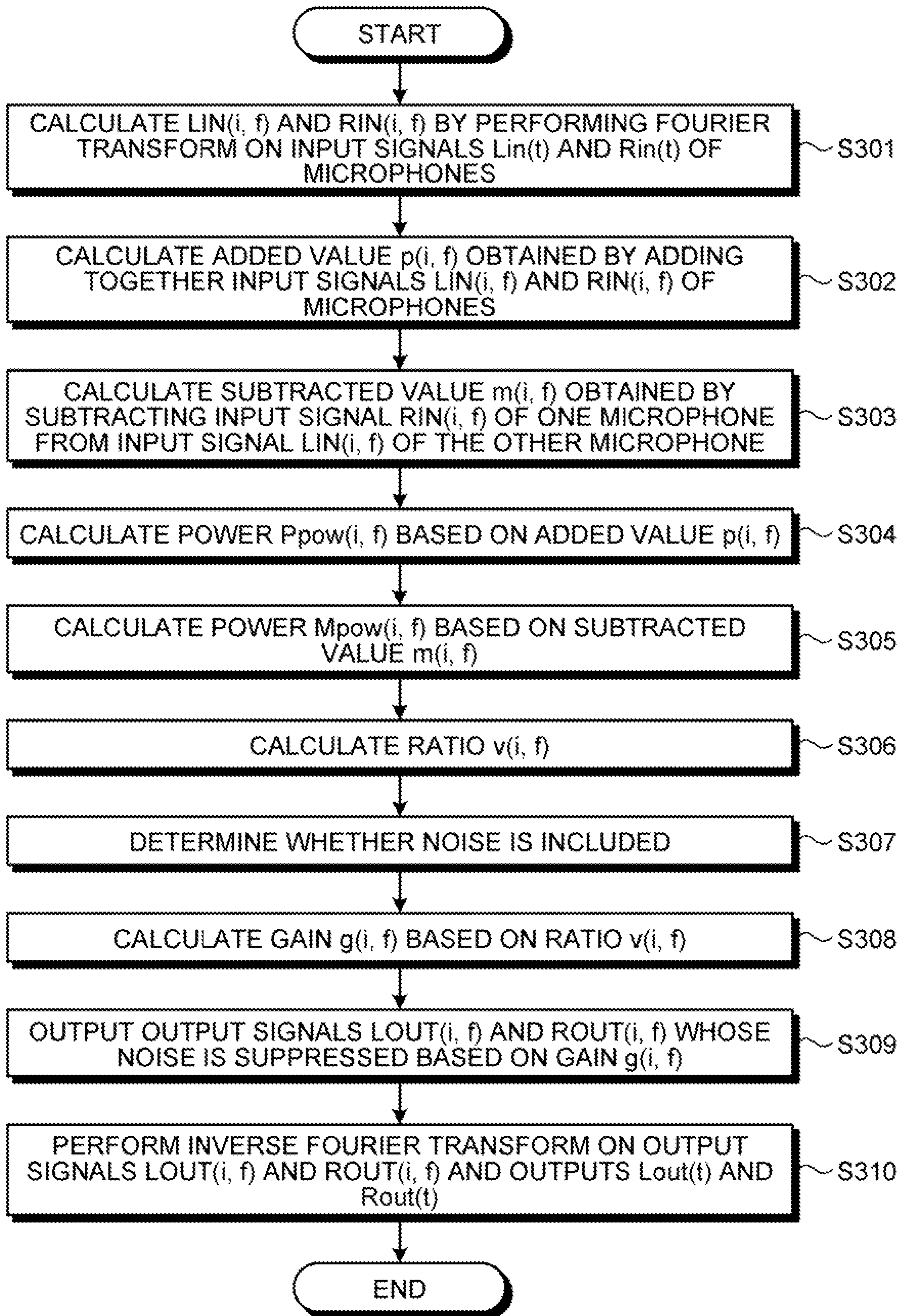


FIG.8

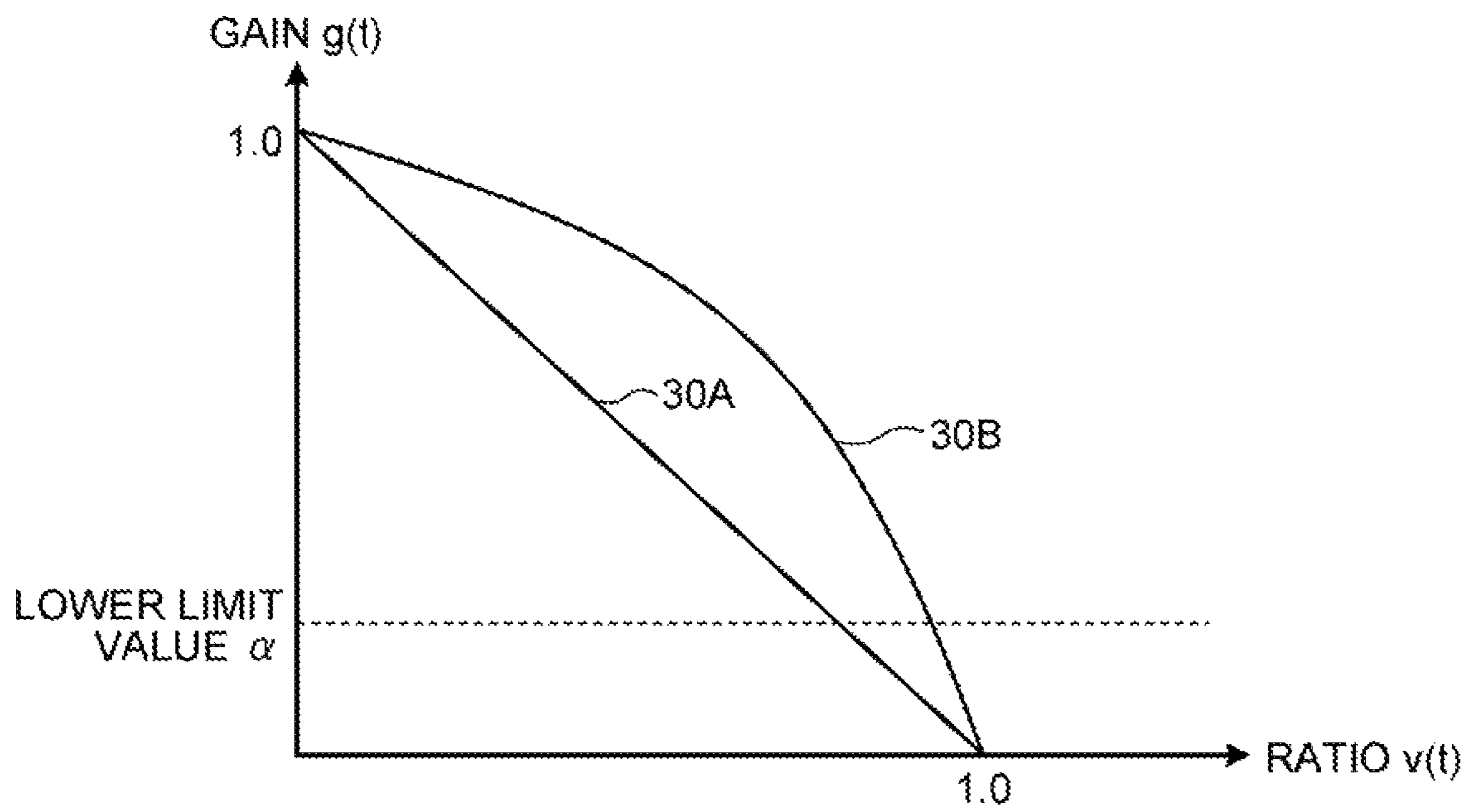


FIG. 9

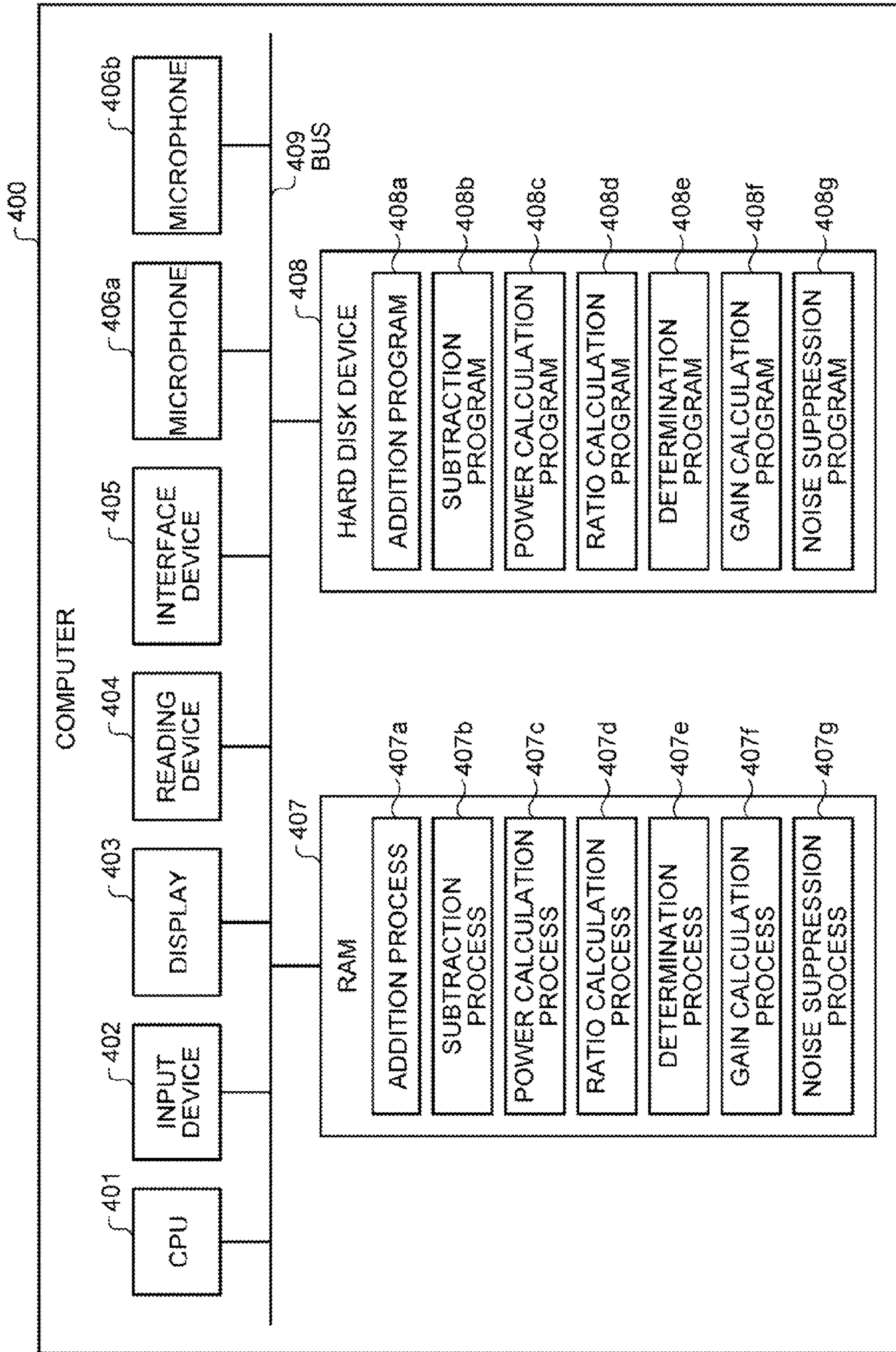
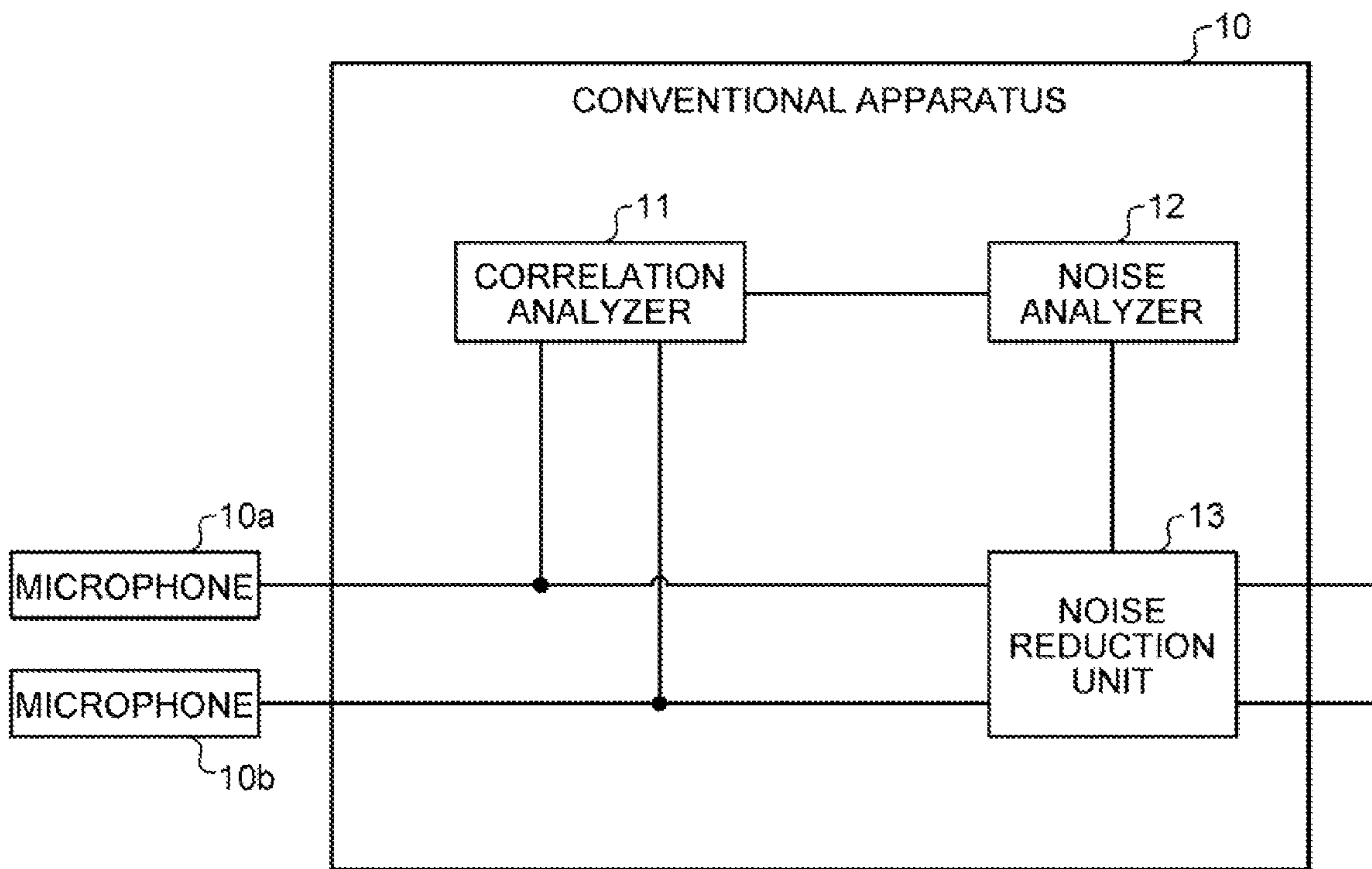


FIG. 10



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SIGNAL PROCESSING APPARATUS AND
SIGNAL PROCESSING METHODCROSS-REFERENCE TO RELATED
APPLICATION

This application is based upon and claims the benefit of priority of the prior Japanese Patent Application No. 2011-274742, filed on Dec. 15, 2011, the entire contents of which are incorporated herein by reference.

FIELD

The embodiments discussed herein are directed to a signal processing apparatus and the like.

BACKGROUND

When wind hits a microphone, a diaphragm included in the microphone vibrates largely, so that noise is included in an input signal. Here, if noise is included in the input signal, when speech recognition is performed or a hands-free phone call is made, the accuracy of the speech recognition and the quality of the phone call degrade. Therefore, in conventional speech recognition and a hands-free phone call, a conventional apparatus which suppresses irregular noise generated when wind hits the microphone is used.

FIG. 10 is a diagram illustrating an example of the conventional apparatus. As illustrated in FIG. 10, a conventional apparatus 10 is connected to microphones 10a and 10b. The microphones 10a and 10b are spaced apart from each other by a predetermined distance. The conventional apparatus 10 includes a correlation analyzer 11, a noise analyzer 12, and a noise reduction unit 13.

The microphone 10a is a microphone which converts collected sound into an input signal $Lin(t)$ and outputs the input signal $Lin(t)$ to the conventional apparatus 10. The microphone 10b is a microphone which converts collected sound into an input signal $Rin(t)$ and outputs the input signal $Rin(t)$ to the conventional apparatus 10. Here, t corresponds to a sampling number of the input signal.

The correlation analyzer 11 calculates a correlation value $r(t)$ between the input signal $Lin(t)$ and the input signal $Rin(t)$ by using the formula (1). In the formula (1), the correlation analyzer 11 sequentially changes the value of i from 1 to n and calculates a correlation value $r(t)$. For example, the value of n is 128. The correlation analyzer 11 outputs the correlation value $r(t)$ to the noise analyzer 12.

$$r(t) = \frac{\sum Lin(t-i)Rin(t-i)}{((\sum Lin(t-i)^2)(\sum Rin(t-i)^2))^{1/2}} \quad (1)$$

When the value of the correlation value $r(t)$ is small, the noise analyzer 12 determines that there is noise of wind and outputs control information to turn down volume to the noise reduction unit 13.

The noise reduction unit 13 adjusts the volumes of the input signals $Lin(t)$ and $Rin(t)$ and outputs the input signals $Lin(t)$ and $Rin(t)$ to an external apparatus. For example, when the noise reduction unit 13 receives the control information to turn down volume, the noise reduction unit 13 turns down the volumes of the input signals $Lin(t)$ and $Rin(t)$ and outputs the input signals $Lin(t)$ and $Rin(t)$ to the external apparatus.

Patent Document 1: Japanese Laid-open Patent Publication No. 2004-289762

Patent Document 2: Japanese Laid-open Patent Publication No. 2009-005133

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However, in the conventional technique described above, there is a problem that noise included in a target input signal such as voice is not sufficiently suppressed.

For example, in the conventional technique, when wind blowing against the microphone is strong, the noise of the input signal can be suppressed. However, when wind blowing against the microphone is weak, the difference of correlation coefficient between a section including voice and a section including wind hitting noise is small, so that the noise is not accurately suppressed.

SUMMARY

According to an aspect of an embodiment, a signal processing apparatus includes: an adder that acquires a plurality of input signals from a plurality of microphones and calculates an added value obtained by adding the input signals together; a subtracter that acquires a plurality of input signals from the plurality of microphones and calculates a subtracted value obtained by subtracting one input signal from the other input signal; and a determination unit that determines whether noise is included in the input signals based on the added value and the subtracted value.

The object and advantages of the invention will be realized and attained by means of the elements and combinations particularly pointed out in the claims.

It is to be understood that both the foregoing general description and the following detailed description are exemplary and explanatory and are not restrictive of the invention, as claimed.

BRIEF DESCRIPTION OF DRAWINGS

FIG. 1 is a diagram illustrating a configuration of a signal processing apparatus according to a first embodiment;

FIG. 2 is a flowchart illustrating a processing procedure of the signal processing apparatus according to the first embodiment;

FIG. 3 is a diagram for explaining an effect of the present invention compared with a conventional apparatus;

FIG. 4 is a diagram illustrating a configuration of a signal processing apparatus according to a second embodiment;

FIG. 5 is a flowchart illustrating a processing procedure of the signal processing apparatus according to the second embodiment;

FIG. 6 is a diagram illustrating a configuration of a signal processing apparatus according to a third embodiment;

FIG. 7 is a flowchart illustrating a processing procedure of the signal processing apparatus according to the third embodiment;

FIG. 8 is a diagram for explaining another process of a gain calculator;

FIG. 9 is a diagram illustrating an example of a computer that executes a signal processing program; and

FIG. 10 is a diagram illustrating an example of a conventional apparatus.

DESCRIPTION OF EMBODIMENTS

Preferred embodiments of the present invention will be explained with reference to accompanying drawings. The present invention is not limited by the embodiments.

[a] First Embodiment

FIG. 1 is a diagram illustrating a configuration of a signal processing apparatus according to a first embodiment. As illustrated in FIG. 1, a signal processing apparatus 100 is

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connected to microphones **10a** and **10b**. The microphones **10a** and **10b** are spaced apart from each other by a predetermined distance.

The signal processing apparatus **100** includes an adder **110**, a subtracter **120**, and power calculators **130a** and **130b**. The signal processing apparatus **100** also includes a ratio calculator **140**, a determination unit **150**, a gain calculator **160**, and a noise suppression unit **170**.

The microphone **10a** is a microphone which converts collected sound into an input signal $Lin(t)$ and outputs the input signal $Lin(t)$ to the signal processing apparatus **100**. The microphone **10b** is a microphone which converts collected sound into an input signal $Rin(t)$ and outputs the input signal $Rin(t)$ to the signal processing apparatus **100**. Here, t corresponds to a sampling number of the input signal.

The adder **110** calculates an added value $p(t)$ by adding the input signal $Lin(t)$ and the input signal $Rin(t)$ together. The added value $p(t)$ is represented by the formula (2). The adder **110** outputs the added value $p(t)$ to the power calculator **130a**.

$$p(t)=Lin(t)+Rin(t) \quad (2)$$

The subtracter **120** calculates a subtracted value $m(t)$ by subtracting the input signal $Rin(t)$ from the input signal $Lin(t)$. The subtracted value $m(t)$ is represented by the formula (3). The subtracter **120** outputs the subtracted value $m(t)$ to the power calculator **130b**.

$$m(t)=Lin(t)-Rin(t) \quad (3)$$

The power calculator **130a** calculates a power $Ppow(t)$ by summing up squared values of the added value $p(t)$ of different sampling numbers. The power $Ppow(t)$ is represented by the formula (4). The power calculator **130a** sequentially changes the value of j from 1 to n and calculates the power $Ppow(t)$. The value of n is a natural number. For example, the value of n is 128. The power calculator **130a** outputs the power $Ppow(t)$ to the ratio calculator **140**.

$$Ppow(t)=\sum p(t-j)^2 \quad (4)$$

The power calculator **130b** calculates a power $Mpow(t)$ by summing up squared values of the subtracted value $m(t)$ of different sampling numbers. The power $Mpow(t)$ is represented by the formula (5). The power calculator **130b** sequentially changes the value of j from 1 to n and calculates the power $Mpow(t)$. The value of n is a natural number. For example, the value of n is 128. The power calculator **130b** outputs the power $Mpow(t)$ to the ratio calculator **140**.

$$Mpow(t)=\sum m(t-j)^2 \quad (5)$$

The ratio calculator **140** calculates a ratio $v(t)$ of the power $Ppow(t)$ and the power $Mpow(t)$. For example, the ratio calculator **140** calculates the ratio $v(t)$ by the formula (6). The ratio calculator **140** outputs the ratio $v(t)$ to the determination unit **150** and the gain calculator **160**. Here, $v(t)$ may be a square root of the ratio of the power $Ppow(t)$ and the power $Mpow(t)$.

$$v(t)=Mpow(t)/Ppow(t) \quad (6)$$

The determination unit **150** determines whether noise is included in the input signals $Lin(t)$ and $Rin(t)$ based on the ratio $v(t)$. When the value of the ratio $v(t)$ is greater than or equal to a predetermined threshold value, the determination unit **150** determines that noise is included in the input signals $Lin(t)$ and $Rin(t)$. The determination unit **150** outputs a determination result to the gain calculator **160**.

The gain calculator **160** calculates a gain based on the ratio $v(t)$. The gain calculator **160** first calculates $g'(t)$ by the for-

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mula (7) and thereafter calculates a gain $g(t)$ by the formula (8).

$$g'(t)=1-v(t) \quad (7)$$

$$g(t)=\max(g'(t),\alpha) \quad (8)$$

In the formula (8), α is a lower limit value greater than or equal to 0 and smaller than or equal to 1. The gain calculator **160** calculates $g'(t)$ and thereafter calculates the larger value of $g'(t)$ and α as the gain $g(t)$. The gain calculator **160** outputs the gain $g(t)$ to the noise suppression unit **170**.

The gain calculator **160** may calculate the gain $g(t)$ when noise is included in the input signals $Lin(t)$ and $Rin(t)$ based on the determination result of the determination unit **150**. On the other hand, when noise is not included in the input signals $Lin(t)$ and $Rin(t)$, the gain calculator **160** may set 1 to the gain $g(t)$ and output the gain $g(t)$ to the noise suppression unit **170**.

The noise suppression unit **170** suppresses noise by multiplying the input signals $Lin(t)$ and $Rin(t)$ by the gain $g(t)$. For example, the noise suppression unit **170** generates output signals $Lout(t)$ and $Rout(t)$ from the input signals $Lin(t)$ and $Rin(t)$ and outputs the generated signals to an external apparatus. For example, the output signals $Lout(t)$ and $Rout(t)$ are represented by the formula (9) and the formula (10).

$$Lout(t)=g(t)Lin(t) \quad (9)$$

$$Rout(t)=g(t)Rin(t) \quad (10)$$

Next, a processing procedure of the signal processing apparatus **100** according to the first embodiment will be described. FIG. 2 is a flowchart illustrating a processing procedure of the signal processing apparatus according to the first embodiment. For example, the process illustrated in FIG. 2 is performed when input signals are received from the microphones **10a** and **10b**.

As illustrated in FIG. 2, the signal processing apparatus **100** calculates the added value $p(t)$ obtained by adding together the input signals $Lin(t)$ and $Rin(t)$ of the microphones **10a** and **10b** (step S101). The signal processing apparatus **100** calculates the subtracted value $m(t)$ obtained by subtracting the input signal $Rin(t)$ of one microphone **10b** from the input signal $Lin(t)$ of the other microphone **10a** (step S102).

The signal processing apparatus **100** calculates the power $Ppow(t)$ based on the added value $p(t)$ (step S103). The signal processing apparatus **100** calculates the power $Mpow(t)$ based on the subtracted value $m(t)$ (step S104).

The signal processing apparatus **100** calculates the ratio $v(t)$ (step S105) and determines whether noise is included (step S106). The signal processing apparatus **100** calculates the gain $g(t)$ based on the ratio $v(t)$ (step S107). The signal processing apparatus **100** outputs the output signals $Lout(t)$ and $Rout(t)$ whose noise is suppressed based on the gain $g(t)$ (step S108).

Next, effects of the signal processing apparatus **100** according to the first embodiment will be described. In the signal processing apparatus **100**, the determination unit **150** determines whether noise is included in the input signals $Lin(t)$ and $Rin(t)$ based on the added value $p(t)$ calculated by the adder **110** and the subtracted value $m(t)$ calculated by the subtracter **120**. Therefore, it is possible to accurately determine whether noise is included, so that noise included in the input signals can be sufficiently suppressed by using the determination result.

Generally, the lower the frequency, the stronger the correlation between the input signals. This is because the lower the frequency, the smaller the variation of waveforms in the time

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axis direction, so that the waveforms have similar shapes. An extreme example is a direct current. Therefore, in order to strengthen the correlation, the adder **110** sums up the input signals. On the other hand, in order to suppress the correlation, the subtracter **120** subtracts one input signal from the other input signal.

Therefore, for example, in a case where the input signals of the microphone have a correlation, such as cases of voice and noise in a driving car, a relationship of the formula (11) is established.

$$\text{“power of } (Lin(t)+Rin(t)) \text{”} \gg \text{“power of } (Lin(t)-Rin(t)) \text{”} \quad (11)$$

On the other hand, in a case where the correlation between the input signals of the microphone is weak, such as case of wind hitting noise, a relationship of the formula (12) is established.

$$\text{“power of } (Lin(t)+Rin(t)) \text{”} \approx \text{“power of } (Lin(t)-Rin(t)) \text{”} \quad (12)$$

Therefore, in the signal processing apparatus **100** according to the first embodiment, noise can be accurately detected by using the ratio of the “power of (Lin(t)-Rin(t))” based on the “power of (Lin(t)+Rin(t))”. Also, an appropriate gain can be calculated by using the ratio of the “power of (Lin(t)-Rin(t))” based on the “power of (Lin(t)+Rin(t))”, and a noise component included in the input signals can be suppressed by using the gain.

FIG. 3 is a diagram for explaining an effect of the present invention compared with a conventional apparatus. Signals **20A** in FIG. 3 indicates a relationship between the sampling number and the gain $g(t)$ of the conventional apparatus. Signals **20B** in FIG. 3 indicates a relationship between the sampling number and the gain $g(t)$ of the signal processing apparatus **100**. Sections **21** in the signals **20A** and **20B** are sections including voice. When comparing the signals **20A** and **20B**, in the signals **20B**, the gain $g(t)$ in sections including only wind hitting noise is reduced while the gain $g(t)$ in the sections **21** is substantially maintained at 1. Therefore, in the present invention, it is possible to turn down volume of a part of input signals including noise without turning down volume of whole input signals in voice sections.

By the way, in the signal processing apparatus **100** according to the first embodiment, the determination unit **150** determines whether noise is included based on the value of the ratio $v(t)$. However, it is not limited to this. The determination unit **150** may obtain a difference between the $P_{pow}(t)$ and the $M_{pow}(t)$, and when the difference value is smaller than a predetermined threshold value, the determination unit **150** may determine that noise is included in the input signals.

The signal processing apparatus **100** may be configured by integrating the determination unit **150** and the gain calculator **160** together.

[b] Second Embodiment

Next, a signal processing apparatus according to a second embodiment will be described. FIG. 4 is a diagram illustrating a configuration of the signal processing apparatus according to the second embodiment. As illustrated in FIG. 4, a signal processing apparatus **200** is connected to the microphones **10a** and **10b**. The microphones **10a** and **10b** are spaced apart from each other by a predetermined distance. The description about the microphones **10a** and **10b** is the same as that in the first embodiment.

The signal processing apparatus **200** includes a delay detector **201** and delay devices **202a** and **202b**. The signal processing apparatus **200** also includes an adder **210**, a subtracter **220**, and power calculators **230a** and **230b**. The signal

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processing apparatus **200** also includes a ratio calculator **240**, a determination unit **250**, a gain calculator **260**, and a noise suppression unit **270**.

The delay detector **201** calculates a correlation coefficient $c(d)$ between the input signal $Lin(t)$ and the input signal $Rin(t)$ by the formula (13). The delay detector **201** changes a delay amount d and detects a delay amount d_{max} by which the value of correlation coefficient $c(d)$ is the maximum.

$$c(d) = \frac{\sum Lin(t)Rin(t-d)}{(\sum Lin(t)^2)^{1/2}(\sum Rin(t-d)^2)^{1/2}} \quad (13)$$

After detecting the d_{max} , the delay detector **201** calculates a delay amount dL applied to the input signal from the microphone **10a** and a delay amount dR applied to the input signal from the microphone **10b** in order to synchronize the input signals inputted from the microphones **10a** and **10b**. A calculation example of the delay amounts dL and dR will be described below.

When the value of the d_{max} is greater than or equal to 0, the delay detector **201** sets the value of dL to 0 and sets the value of dR to the value of the d_{max} . Then, the delay detector **201** outputs the dL to the delay device **202a** and outputs the dR to the delay device **202b**.

On the other hand, when the value of the d_{max} is smaller than 0, the delay detector **201** sets the value of dL to the absolute value of the d_{max} and sets the value of dR to 0. Then, the delay detector **201** outputs the dL to the delay device **202a** and outputs the dR to the delay device **202b**.

The delay device **202a** performs a delay process of the input signal $Lin(t)$ from the microphone **10a**. The delay device **202a** outputs an input signal $Lin(t-dL)$ obtained by performing a delay process on the input signal $Lin(t)$ to the adder **210** and the subtracter **220**.

The delay device **202b** performs a delay process of the input signal $Rin(t)$ from the microphone **10b**. The delay device **202b** outputs an input signal $Rin(t-dR)$ obtained by performing a delay process on the input signal $Rin(t)$ to the adder **210** and the subtracter **220**.

The adder **210** calculates an added value $p(t)$ by adding together the input signals $Lin(t-dL)$ and $Rin(t-dR)$ whose delay amounts are adjusted. The adder **210** outputs the added value $p(t)$ to the power calculator **230a**.

The subtracter **220** calculates a subtracted value $m(t)$ by subtracting the input signal $Rin(t-dR)$ from the input signal $Lin(t-dL)$. The subtracter **220** outputs the subtracted value $m(t)$ to the power calculator **230b**.

The descriptions about the power calculators **230a** and **230b**, the ratio calculator **240**, the determination unit **250**, the gain calculator **260**, and the noise suppression unit **270** are the same as those of the power calculators **130a** and **130b**, the ratio calculator **140**, the determination unit **150**, the gain calculator **160**, and the noise suppression unit **170** illustrated in FIG. 1.

Next, a processing procedure of the signal processing apparatus **200** according to the second embodiment will be described. FIG. 5 is a flowchart illustrating the processing procedure of the signal processing apparatus according to the second embodiment. For example, the process illustrated in FIG. 5 is performed when input signals are received from the microphones **10a** and **10b**.

The signal processing apparatus **200** detects the maximum value of the correlation coefficient $c(d)$ between the input signal $Lin(t)$ and the input signal $Rin(t)$ and calculates the delay amounts dL and dR for synchronization (step S201). The signal processing apparatus **200** calculates the $p(t)$ by performing synchronous addition (step S202) and calculates the $m(t)$ by performing synchronous subtraction (step S203).

The signal processing apparatus **200** calculates the power $P_{pow}(t)$ based on the added value $p(t)$ (step S204). The signal processing apparatus **200** calculates the power $M_{pow}(t)$ based on the subtracted value $m(t)$ (step S205).

The signal processing apparatus **200** calculates the ratio $v(t)$ (step S206) and determines whether noise is included (step S207). The signal processing apparatus **200** calculates the gain $g(t)$ based on the ratio $v(t)$ (step S208). The signal processing apparatus **200** outputs the output signals $L_{out}(t)$ and $R_{out}(t)$ whose noise is suppressed based on the gain $g(t)$ (step S209).

Next, effects of the signal processing apparatus **200** according to the second embodiment will be described. In the signal processing apparatus **200**, the delay detector **201** calculates the delay amounts dL and dR and the delay devices **202a** and **202b** synchronize the input signals. In the signal processing apparatus **200**, the determination unit **250** determines whether noise is included in the input signals $L_{in}(t)$ and $R_{in}(t)$ based on the added value $p(t)$ calculated by the adder **210** and the subtracted value $m(t)$ calculated by the subtracter **220**. Therefore, even if the timings when voice of a user is collected by the microphones **10a** and **10b** are different from each other, the input signals can be synchronized by adjusting the delay amounts. Thus, it is possible to accurately determine whether noise is included in a direction of a user whose voice is desired to be recorded, so that noise included in the input signals can be suppressed by using the determination result.

[c] Third Embodiment

Next, a signal processing apparatus according to the third embodiment will be described. FIG. 6 is a diagram illustrating a configuration of the signal processing apparatus according to the third embodiment. As illustrated in FIG. 6, a signal processing apparatus **300** is connected to the microphones **10a** and **10b**. The microphones **10a** and **10b** are spaced apart from each other by a predetermined distance. The description about the microphones **10a** and **10b** is the same as that in the first embodiment.

The signal processing apparatus **300** includes FFTs (Fast Fourier Transforms) **301a** and **301b**, an adder **310**, a subtracter **320**, and power calculators **330a** and **330b**. The signal processing apparatus **300** also includes a ratio calculator **340**, a determination unit **350**, a gain calculator **360**, a noise suppression unit **370**, and IFFTs (Inverse FFTs) **371a** and **371b**.

The FFT **301a** positions a Hanning window with 50% overlap on the input signal $L_{in}(t)$ acquired from the microphone **10a** and converts the input signal $L_{in}(t)$ into a signal on the frequency axis by fast Fourier transform. The input signal $L_{in}(t)$ converted into a signal on the frequency axis is represented as $LIN(i, f)$. Here, i is a number of an analysis frame on which FFT is performed and f indicates the frequency. The FFT **301a** outputs the $LIN(i, f)$ to the adder **310**, the subtracter **320**, and the noise suppression unit **370**.

The FFT **301b** positions a Hanning window with 50% overlap on the input signal $R_{in}(t)$ acquired from the microphone **10b** and converts the input signal $R_{in}(t)$ into a signal on the frequency axis by fast Fourier transform. The input signal $R_{in}(t)$ converted into a signal on the frequency axis is represented as $RIN(i, f)$. The FFT **301b** outputs the $RIN(i, f)$ to the adder **310**, the subtracter **320**, and the noise suppression unit **370**. The FFTs **301a** and **301b** are an example of a frequency converter.

The adder **310** is a processing unit which calculates an added value $p(i, f)$ by adding the input signal $LIN(i, f)$ and the input signal $RIN(i, f)$ together. The added value $p(i, f)$ is

represented by the formula (14). The adder **310** outputs the added value $p(i, f)$ to the power calculator **330a**.

$$p(i, f) = LIN(i, f) + RIN(i, f) \quad (14)$$

The subtracter **320** calculates a subtracted value $m(i, f)$ by subtracting the input signal $RIN(i, f)$ from the input signal $LIN(i, f)$. The subtracted value $m(i, f)$ is represented by the formula (15). The subtracter **320** outputs the subtracted value $m(i, f)$ to the power calculator **330b**.

$$m(i, f) = LIN(i, f) - RIN(i, f) \quad (15)$$

The power calculator **330a** calculates a power $P_{pow}(i, f)$ by summing up squared values of the added value $p(i, f)$ of different frame numbers. The power $P_{pow}(i, f)$ is represented by the formula (16). The power calculator **330a** sequentially changes the value of j from 1 to n and calculates the power $P_{pow}(i, f)$. The value of n is a natural number. For example, the value of n is 128. The power calculator **330a** outputs the power $P_{pow}(i, f)$ to the ratio calculator **340**.

$$P_{pow}(i, f) = \sum p(i-j, f) \quad (16)$$

The power calculator **330b** calculates a power $M_{pow}(i, f)$ by summing up squared values of the subtracted value $m(i, f)$ of different frame numbers. The power $M_{pow}(i, f)$ is represented by the formula (17). The power calculator **330b** sequentially changes the value of j from 1 to n and calculates the power $M_{pow}(i, f)$. The value of n is a natural number. For example, the value of n is 128. The power calculator **330b** outputs the power $M_{pow}(i, f)$ to the ratio calculator **340**.

$$M_{pow}(i, f) = \sum m(i-j, f) \quad (17)$$

The ratio calculator **340** calculates a ratio $v(i, f)$ of the power $P_{pow}(i, f)$ and the power $M_{pow}(i, f)$. For example, the ratio calculator **340** calculates the ratio $v(i, f)$ by the formula (18). The ratio calculator **340** outputs the ratio $v(i, f)$ to the determination unit **350** and the gain calculator **360**.

$$v(i, f) = M_{pow}(i, f) / P_{pow}(i, f) \quad (18)$$

The determination unit **350** determines whether noise is included in the input signals $LIN(i, f)$ and $RIN(i, f)$ based on the ratio $v(i, f)$. When the value of the ratio $v(i, f)$ is greater than or equal to a predetermined threshold value, the determination unit **350** determines that noise is included in the input signals $LIN(i, f)$ and $RIN(i, f)$. The determination unit **350** outputs a determination result to the gain calculator **360**.

The gain calculator **360** calculates a gain based on the ratio $v(i, f)$. The gain calculator **360** first calculates $g'(i, f)$ by the formula (19) and thereafter calculates a gain $g(i, f)$ by the formula (20).

$$g'(i, f) = 1 - v(i, f) \quad (19)$$

$$g(i, f) = \max(g'(i, f), \alpha) \quad (20)$$

In the formula (20), α is a lower limit value greater than or equal to 0 and smaller than or equal to 1. The gain calculator **360** calculates $g'(i, f)$ and thereafter calculates the larger value of $g'(i, f)$ and α as the gain $g(i, f)$. The gain calculator **360** outputs the gain $g(i, f)$ to the noise suppression unit **370**.

The gain calculator **360** may calculate the gain $g(i, f)$ when noise is included in the input signals $LIN(i, f)$ and $RIN(i, f)$ based on the determination result of the determination unit **350**. On the other hand, when noise is not included in the input signals $LIN(i, f)$ and $RIN(i, f)$, the gain calculator **360** may set 1 to the gain $g(i, f)$ and output the gain $g(i, f)$ to the noise suppression unit **370**.

The noise suppression unit **370** suppresses noise by multiplying the input signals $LIN(i, f)$ and $RIN(i, f)$ by the gain $g(i, f)$. For example, the noise suppression unit **370** generates

output signals LOUT(i, f) and ROUT (i, f) from the input signals LIN(i, f) and RIN(i, f) and outputs the generated signals to the IFFTs **371a** and **371b**. For example, the output signals LOUT(i, f) and ROUT (i, f) are represented by the formulas (21) and (22).

$$LOUT(i, f) = g(i, f)LIN(i, f) \quad (21)$$

$$ROUT(i, f) = g(i, f)RIN(i, f) \quad (22)$$

The IFFT **371a** is a processing unit which generates a signal Lin(t) on the time axis by performing an inverse Fourier transform and 50% overlap addition on the LOUT(i, f). The IFFT **371a** outputs the output signal Lin(t) to an external apparatus.

The IFFT **371b** is a processing unit which generates a signal Rin(t) on the time axis by performing an inverse Fourier transform and 50% overlap addition on the ROUT(i, f). The IFFT **371b** outputs the output signal Rin(t) to an external apparatus.

Next, a processing procedure of the signal processing apparatus **300** according to the third embodiment will be described. FIG. 7 is a flowchart illustrating the processing procedure of the signal processing apparatus according to the third embodiment. For example, the process illustrated in FIG. 7 is performed when input signals are received from the microphones **10a** and **10b**.

As illustrated in FIG. 7, the signal processing apparatus **300** calculates the LIN(i, f) and the RIN(i, f) by performing a Fourier transform on the input signals Lin(t) and Rin(t) of the microphones (step S301). The signal processing apparatus **300** calculates the added value p(i, f) obtained by adding together the input signals LIN(i, f) and RIN(i, f) of the microphones (step S302).

The signal processing apparatus **300** calculates the subtracted value m(i, f) obtained by subtracting the input signal RIN(i, f) of one microphone from the input signal LIN(i, f) of the other microphone (step S303). The signal processing apparatus **300** calculates the power Ppow(i, f) based on the added value p(i, f) (step S304).

The signal processing apparatus **300** calculates the power Mpow(i, f) based on the subtracted value m(i, f) (step S305). The signal processing apparatus **300** calculates the ratio v(i, f) (step S306) and determines whether noise is included (step S307).

The signal processing apparatus **300** calculates the gain g(i, f) based on the ratio v(i, f) (step S308). The signal processing apparatus **300** outputs the output signals LOUT(i, f) and ROUT(i, f) whose noise is suppressed based on the gain g(i, f) (step S309). The signal processing apparatus **300** performs an inverse Fourier transform on the output signals LOUT(i, f) and ROUT(i, f) and outputs Lout(t) and Rout(t) (step S310).

Next, effects of the signal processing apparatus **300** according to the third embodiment will be described. The signal processing apparatus **300** performs a Fourier transform on the input signals Lin(t) and Rin(t) and determines whether there is noise at each frequency. Then, the signal processing apparatus **300** calculates the gain g(i, f) at each frequency and multiplies the input signals LIN(i, f) and RIN(i, f) at each frequency by the gain g(i, f). Therefore, noise at each frequency can be suppressed.

[d] Fourth Embodiment

The processes of the aforementioned signal processing apparatuses **100** to **300** are an example. Hereinafter, another process of the signal processing apparatuses of the present invention will be described as a fourth embodiment. The process according to the fourth embodiment will be described with reference to the configuration diagram in FIG. 1.

Another process of the gain calculator **160** will be described. Although the gain calculator **160** calculates the gain g(t) by using the formula (7) and the formula (8), it is not limited to this. The gain calculator **160** may calculate the gain g(t) by using a decision table indicating a relationship between the gain g(t) and the ratio v(t) for each distance between the microphones **10a** and **10b**.

FIG. 8 is a diagram for explaining the other process of the gain calculator. For example, the gain calculator **160** defines that when the distance between the microphones **10a** and **10b** is 4.2 cm, the relationship between the gain g(t) and the ratio v(t) is a relationship changing as illustrated by a straight line **30A** in FIG. 8. The relationship illustrated by the straight line **30A** corresponds to the relationship of the formula (7).

On the other hand, the gain calculator **160** defines that when the distance between the microphones **10a** and **10b** is smaller than 4.2 cm, the relationship changes as illustrated by a curved line **30B** in FIG. 8. In other words, the gain calculator **160** sets the change rate of the gain g(t) larger than that indicated by the straight line **30A** when the value of the ratio v(t) is about 1 to 0.8. The gain calculator **160** performs the process as described above, so that it is possible to accurately suppress noise corresponding to the change of characteristics of the input signals caused by the difference of the distance between the microphones.

The gain calculator **160** may acquire information of the distance between the microphones **10a** and **10b** in any manner. A user may input the distance into the signal processing apparatus **100** by using an input device.

Another process of the power calculator **130a** will be described. The power calculator **130a** obtains the power Ppow(t) based on the formula (4). However, instead of the power Ppow(t), an average Pamp(t) of the absolute values may be calculated. For example, the power calculator **130a** obtains the average of the absolute values by the formula (23).

$$P_{amp}(t) = \sum |p(t-j)| \quad (23)$$

Similarly, the power calculator **130b** may calculate an average Mamp(t) of the absolute values instead of the power Mpow(t). For example, the power calculator **130b** obtains the average of the absolute values by the formula (24).

$$M_{amp}(t) = \sum |m(t-j)| \quad (24)$$

It is known that when the average of the absolute values is obtained, the processing load of the computer is smaller than when the power is obtained. Therefore, when it is configured so that the power calculators **130a** and **130b** calculate the average of the absolute values instead of the power, the cost of the signal processing apparatus **100** can be reduced.

By the way, the signal processing apparatus **100** may suppress noise by using microphones **10c** and **10d** not illustrated in the drawings in addition to the pair of microphones **10a** and **10b**. For example, the signal processing apparatus **100** calculates the ratio v(t) from the microphones **10a** and **10b** by the same process as that in the first embodiment. The v(t) calculated from the microphones **10a** and **10b** is defined as v1(t).

Also, the ratio v(t) is calculated from the microphones **10c** and **10d** by the same process as that in the first embodiment. The v(t) calculated from the microphones **10c** and **10d** is defined as v2(t). When both the v1(t) and v2(t) are greater than a predetermined threshold value, the signal processing apparatus **100** may adjust the input signals Lin(t) and Rin(t) by the gain g(t).

For example, the signal processing apparatus **100** may obtain g'(t) by the formula (25) or the formula (26). The processes after calculating the g'(t) is the same as that in the

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first embodiment. The $g'(t)$ may be obtained by subtracting the average value of the $v1(1)$ and $v2(2)$ from 1.

$$g'(t)=1-v1(t) \quad (25)$$

$$g'(t)=1-v2(t) \quad (26)$$

In this way, the gain $g(t)$ is calculated by using the microphones **10a** to **10d**, so that it is possible to accurately suppress noise of the input signals even when there are a plurality of users.

The processing units included in the signal processing apparatuses **100** to **300** described in the first to the fourth embodiments correspond to integrated devices such as, for example, ASIC (Application Specific Integrated Circuit) and FPGA (Field Programmable Gate Array). Also, the processing units correspond to electronic circuits such as CPU and MPU (Micro Processing Unit).

Next, an example of a computer that executes a signal processing program which realizes the same function as that of the signal processing apparatuses **100** to **300** described in the embodiments will be described. FIG. 9 is a diagram illustrating an example of the computer that executes the signal processing program.

As illustrated in FIG. 9, a computer **400** includes a CPU **401** that performs various calculation processes, an input device **402** that receives input of data from a user, and a display **403**. The computer **400** also includes a reading device **404** that reads a program and the like from a storage medium and an interface device **405** that transmits and receives data to and from another computer through a network. The computer **400** also includes microphones **406a** and **406b**. The computer **400** also includes a RAM **407** that temporarily stores various information and a hard disk device **408**. The devices **401** to **408** are connected to a bus **409**.

The hard disk device **408** includes, for example, an addition program **408a**, a subtraction program **408b**, a power calculation program **408c**, a ratio calculation program **408d**, a determination program **408e**, a gain calculation program **408f**, and a noise suppression program **408g**. The CPU **401** reads the programs **408a** to **408g** and develops the programs in the RAM **407**.

The addition program **408a** functions as an addition process **407a**. The subtraction program **408b** functions as a subtraction process **407b**. The power calculation program **408c** functions as a power calculation process **407c**. The ratio calculation program **408d** functions as a ratio calculation process **407d**. The determination program **408e** functions as a determination process **407e**. The gain calculation program **408f** functions as a gain calculation process **407f**. The noise suppression program **408g** functions as a noise suppression process **407g**.

For example, the addition process **407a** corresponds to the adder **110**. The subtraction process **407b** corresponds to the subtracter **120**. The power calculation process **407c** corresponds to the power calculators **130a** and **130b**. The ratio calculation process **407d** corresponds to the ratio calculator **140**. The determination process **407e** corresponds to the determination unit **150**. The gain calculation process **407f** corresponds to the gain calculator **160**. The noise suppression process **407g** corresponds to the noise suppression unit **170**.

The programs **408a** to **408g** need not necessarily be stored in the hard disk device **408** from the beginning. For example, the programs are stored in a "portable physical medium" such as a flexible disk (FD), a CD-ROM, a DVD disk, a magneto-optical disk, and an IC card which are inserted in the computer **400**. The computer **400** may read the programs **408a** to **408g** from these media and execute the programs. The programs of

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the signal processing apparatuses **200** and **300** described in the second to the fourth embodiments are executed by the computer **400** in the same manner as the programs of the signal processing apparatus **100**.

According to the disclosed signal processing apparatus, there is an effect that noise included in the input signal can be suppressed.

All examples and conditional language recited herein are intended for pedagogical purposes of aiding the reader in understanding the invention and the concepts contributed by the inventor to further the art, and are not to be construed as limitations to such specifically recited examples and conditions, nor does the organization of such examples in the specification relate to a showing of the superiority and inferiority of the invention. Although the embodiments of the present invention have been described in detail, it should be understood that the various changes, substitutions, and alterations could be made hereto without departing from the spirit and scope of the invention.

What is claimed is:

1. A signal processing apparatus comprising:

an adder that acquires a plurality of input signals from a plurality of microphones and calculates an added value obtained by adding the input signals together;

a subtracter that acquires a plurality of input signals from the plurality of microphones and calculates a subtracted value obtained by subtracting one input signal from the other input signal;

a ratio calculator that calculates a ratio of the subtracted value to the added value;

a gain calculator that calculates a gain based on a value of the ratio and information indicating a relationship between the gain and the value of the ratio which varies according to a distance between the microphones;

a determination unit that determines noise being included in the input signals when a difference between the added value and the subtracted value is less than a threshold value; and

a noise suppression unit that suppresses noise of the input signals based on the gain calculated by the gain calculator.

2. The signal processing apparatus according to claim 1, wherein the determination unit determines that noise is included in the input signals when a value of the ratio calculated by the ratio calculator is greater than or equal to a threshold value.

3. The signal processing apparatus according to claim 1, further comprising:

a delay detector that acquires input signals from the plurality of microphones, changes a delay amount of each of the input signals, and detects delay amounts by which a correlation coefficient of the input signals is a maximum; and

a delay unit that delays each of the input signals based on the delay amounts detected by the delay detector, wherein the adder calculates the added value based on the input signals delayed by the delay unit, and the subtracter calculates the subtracted value based on the input signals delayed by the delay unit.

4. The signal processing apparatus according to claim 1, further comprising

a frequency converter that acquires input signals from the plurality of microphones and converts each of the input signals into a signal on a frequency axis, wherein the adder calculates the added value based on each signal on the frequency axis, and

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the subtracter calculates the subtracted value based on each signal on the frequency axis.

5. The signal processing apparatus according to claim 1, wherein the ratio calculator calculates a ratio of a value obtained by squaring the subtracted value to a value obtained by squaring the added value.

6. The signal processing apparatus according to claim 1, wherein the ratio calculator calculates a ratio of an absolute value of the subtracted value to an absolute value of the added value.

7. The signal processing apparatus according to claim 1, wherein

the adder calculates a first added value obtained by adding together input signals acquired from a first microphone and a second microphone and a second added value obtained by adding together input signals acquired from a third microphone and a fourth microphone,

the subtracter calculates a first subtracted value obtained by subtracting an input signal acquired from the second microphone from an input signal acquired from the first microphone and a second subtracted value obtained by subtracting an input signal acquired from the fourth microphone from an input signal acquired from the third microphone, and

the gain calculator calculates a gain based on the first added value, the second added value, the first subtracted value, and the second subtracted value.

8. The signal processing apparatus according to claim 1, wherein the gain calculator calculates the gain based on a formula $g(t)=1-v(t)$, $g(t)$ indicating the gain, $v(t)$ indicating ratio when a distance between the microphones is 4.2 cm, and calculates the gain based on a decision table which sets a change rate of the gain different from the change rate of the gain calculated by the formula when the distance between the microphones is smaller than 4.2 cm.

9. A signal processing method comprising:

acquiring a plurality of input signals from a plurality of microphones and calculating an added value obtained by adding the input signals together;

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acquiring a plurality of input signals from the plurality of microphones and calculating a subtracted value obtained by subtracting one input signal from the other input signal;

calculating a ratio of the subtracted value to the added value;

calculating a gain based on a value of the ratio and information indicating a relationship between the gain and the value of the ratio which varies according to a distance between the microphones;

determining noise being included in the input signals when a difference between the added value and the subtracted value is less than a threshold value; and

suppressing noise of the input signals based on the gain calculated at the calculating the gain.

10. A non-transitory computer-readable recording medium having stored therein a program for causing a computer to execute a signal processing procedure comprising:

acquiring a plurality of input signals from a plurality of microphones and calculating an added value obtained by adding the input signals together;

acquiring a plurality of input signals from the plurality of microphones and calculating a subtracted value obtained by subtracting one input signal from the other input signal;

calculating a ratio of the subtracted value to the added value;

calculating a gain based on a value of the ratio and information indicating a relationship between the gain and the value of the ratio which varies according to a distance between the microphones;

determining noise being included in the input signals when a difference between the added value and the subtracted value is less than a threshold value; and

suppressing noise of the input signals based on the gain calculated at the calculating the gain.

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