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Matsuo

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

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H04R 1/02 (2006.01)
H04R 1/40 (2006.01)

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CPC ***H04R 3/005*** (2013.01); ***H04R 1/406*** (2013.01); *H04R 2201/403* (2013.01); *H04R 2430/20* (2013.01)

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USPC 381/94.1–94.3, 92, 317, 71.1, 71.11, 381/122, 107, 91; 704/226, 233, E21.004, 704/E21.002, 225
See application file for complete search history.

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

(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.

10 Claims, 10 Drawing Sheets

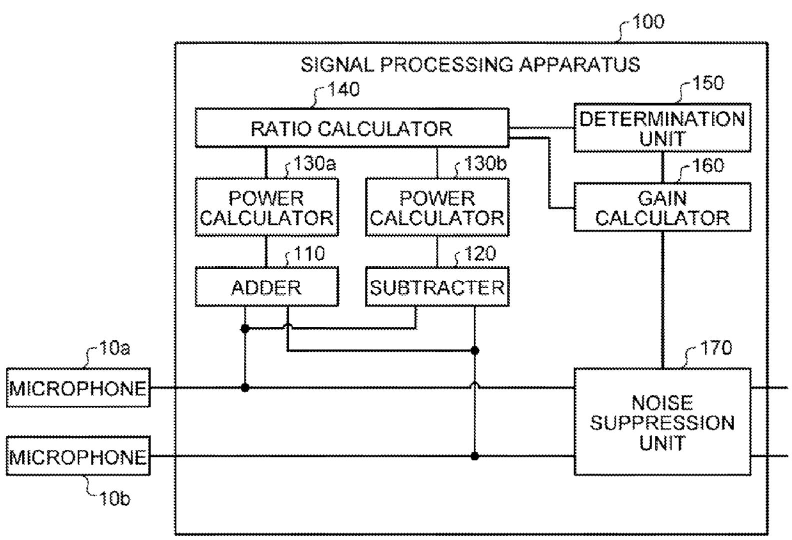

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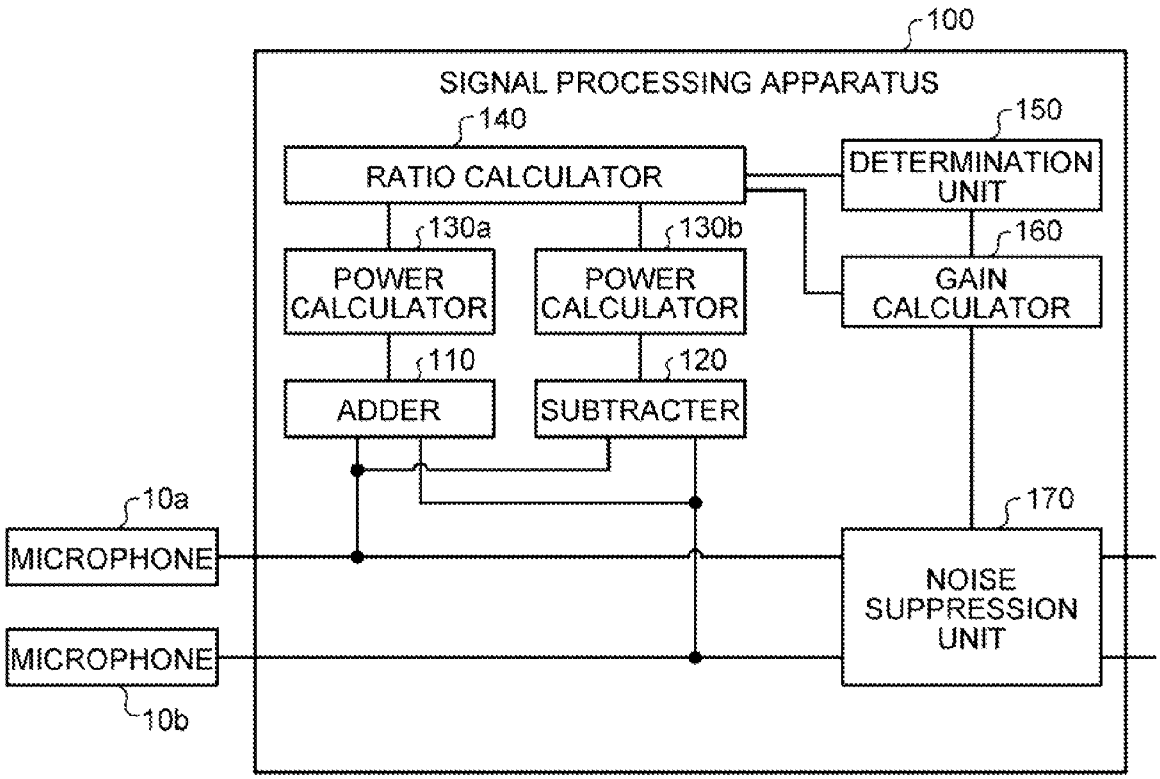
FIG.1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SIGNAL PROCESSING APPARATUS
140
RATIO CALCULATOR
150
DETERMINATION UNIT
130a
POWER CALCULATOR
130b
POWER CALCULATOR
160
GAIN CALCULATOR
110
ADDER
120
SUBTRACTER
10a
MICROPHONE
170
NOISE SUPPRESSION UNIT
MICROPHONE
10b

FIG.2

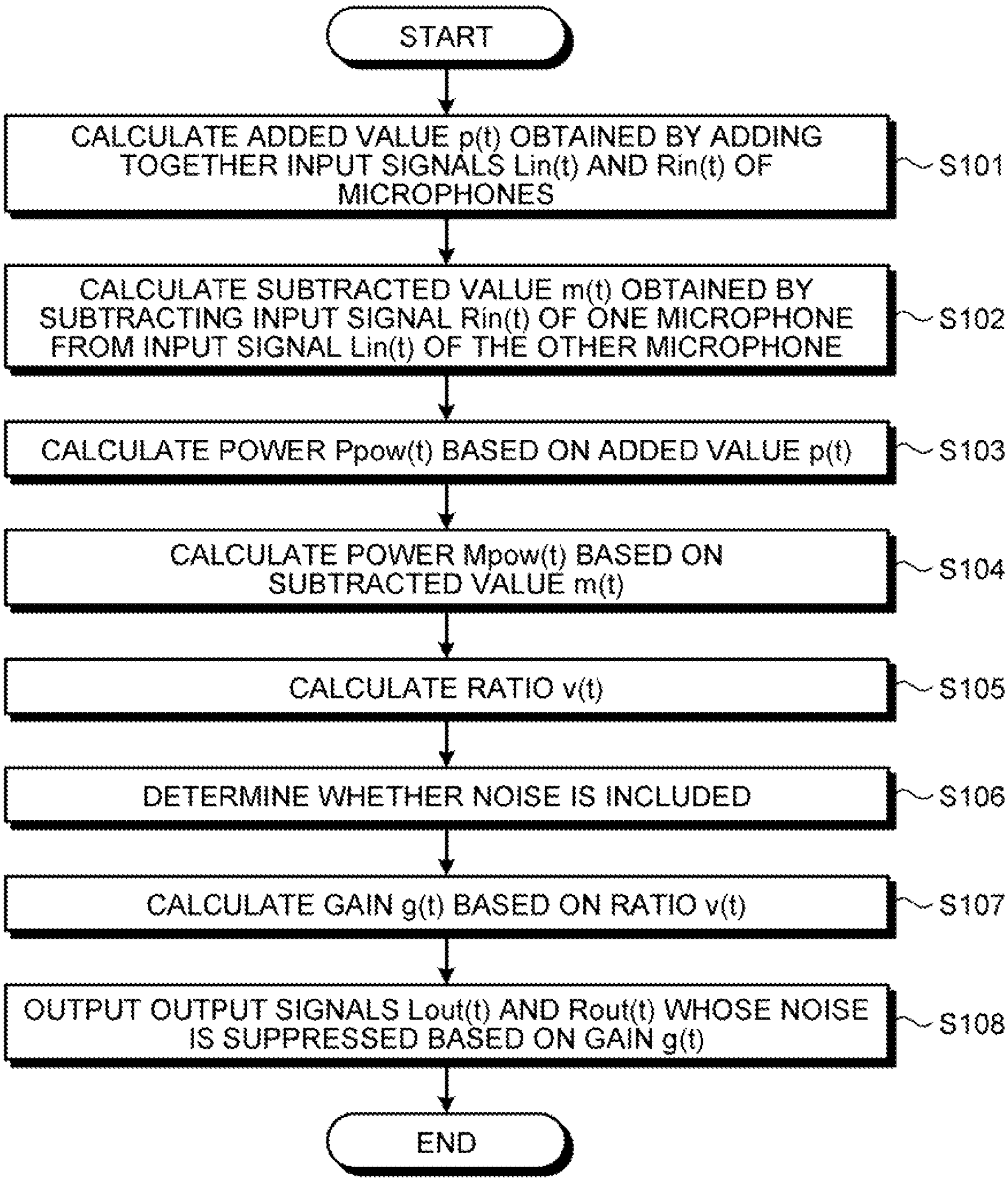

START
CALCULATE ADDED VALUE p(t) OBTAINED BY ADDING TOGETHER INPUT SIGNALS Lin(t) AND Rin(t) OF MICROPHONES
S101
CALCULATE SUBTRACTED VALUE m(t) OBTAINED BY SUBTRACTING INPUT SIGNAL Rin(t) OF ONE MICROPHONE FROM INPUT SIGNAL Lin(t) OF THE OTHER MICROPHONE
S102
CALCULATE POWER Ppow(t) BASED ON ADDED VALUE p(t)
S103
CALCULATE POWER Mpow(t) BASED ON SUBTRACTED VALUE m(t)
S104
CALCULATE RATIO v(t)
S105
DETERMINE WHETHER NOISE IS INCLUDED
S106
CALCULATE GAIN g(t) BASED ON RATIO v(t)
S107
OUTPUT OUTPUT SIGNALS Lout(t) AND Rout(t) WHOSE NOISE IS SUPPRESSED BASED ON GAIN g(t)
S108
END

FIG.3

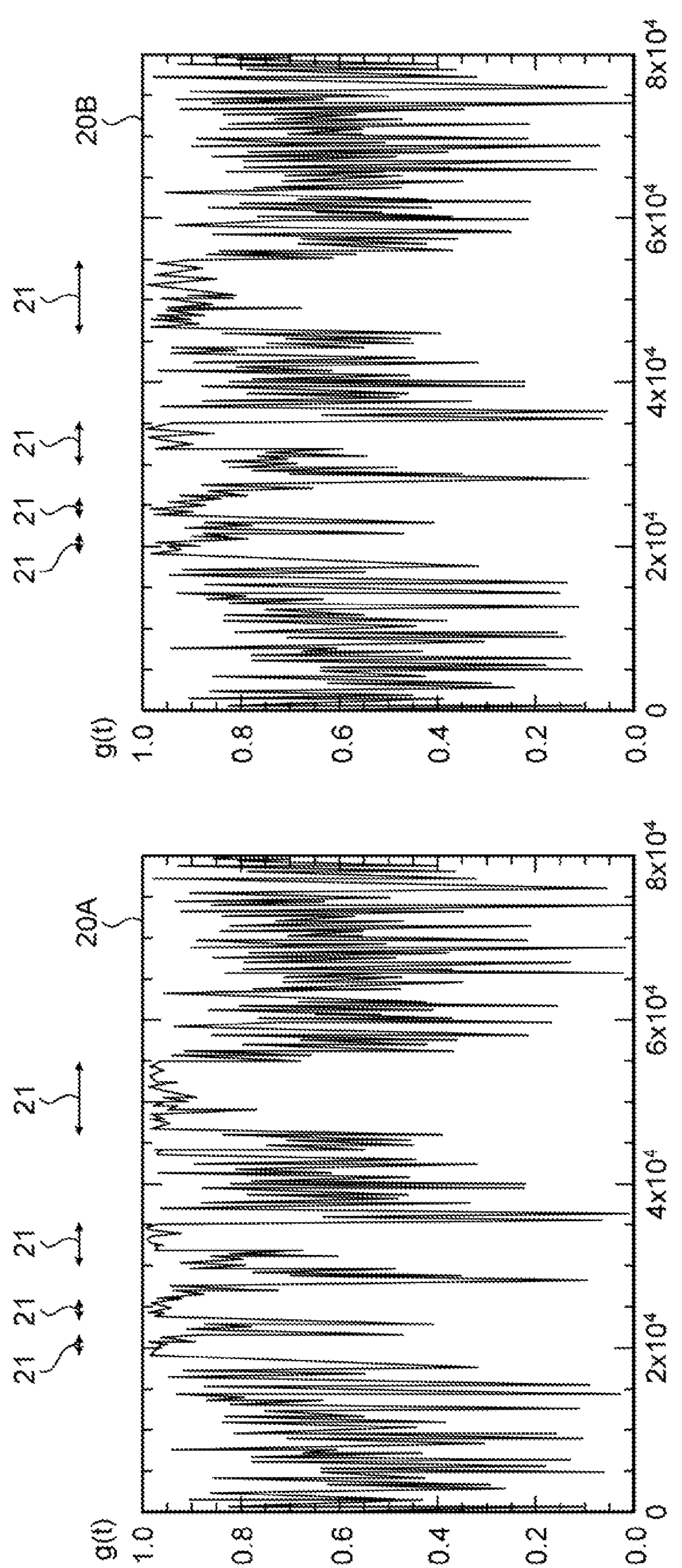

20A
20B
21
g(t)
1.0
0.8
0.6
0.4
0.2
0.0
0
2x10^4
4x10^4
6x10^4
8x10^4
SAMPLING NUMBER

FIG.4

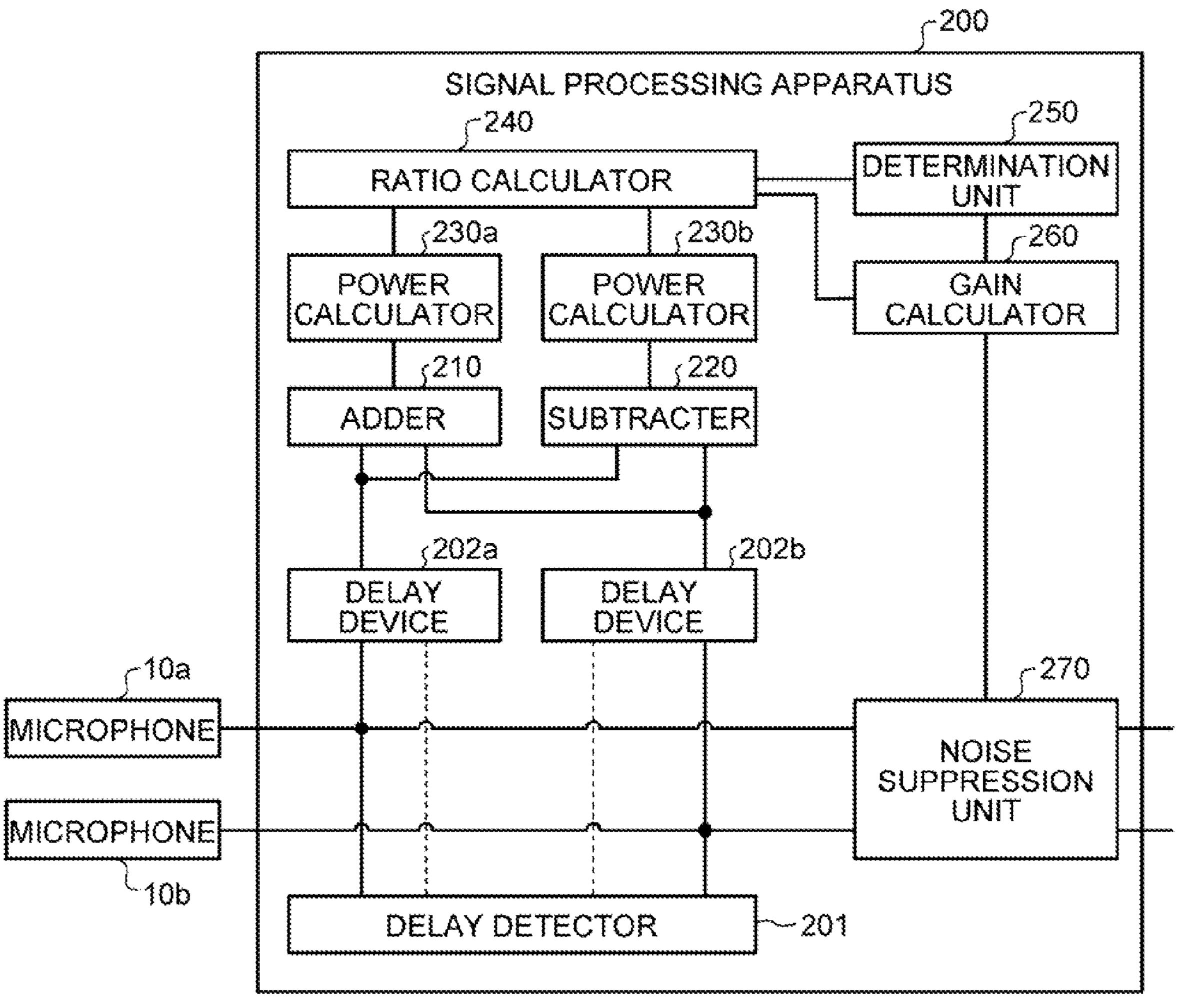

200
SIGNAL PROCESSING APPARATUS
240
RATIO CALCULATOR
250
DETERMINATION UNIT
230a
POWER CALCULATOR
230b
POWER CALCULATOR
260
GAIN CALCULATOR
210
ADDER
220
SUBTRACTER
202a
DELAY DEVICE
202b
DELAY DEVICE
10a
MICROPHONE
270
NOISE SUPPRESSION UNIT
MICROPHONE
10b
DELAY DETECTOR
201

FIG.5

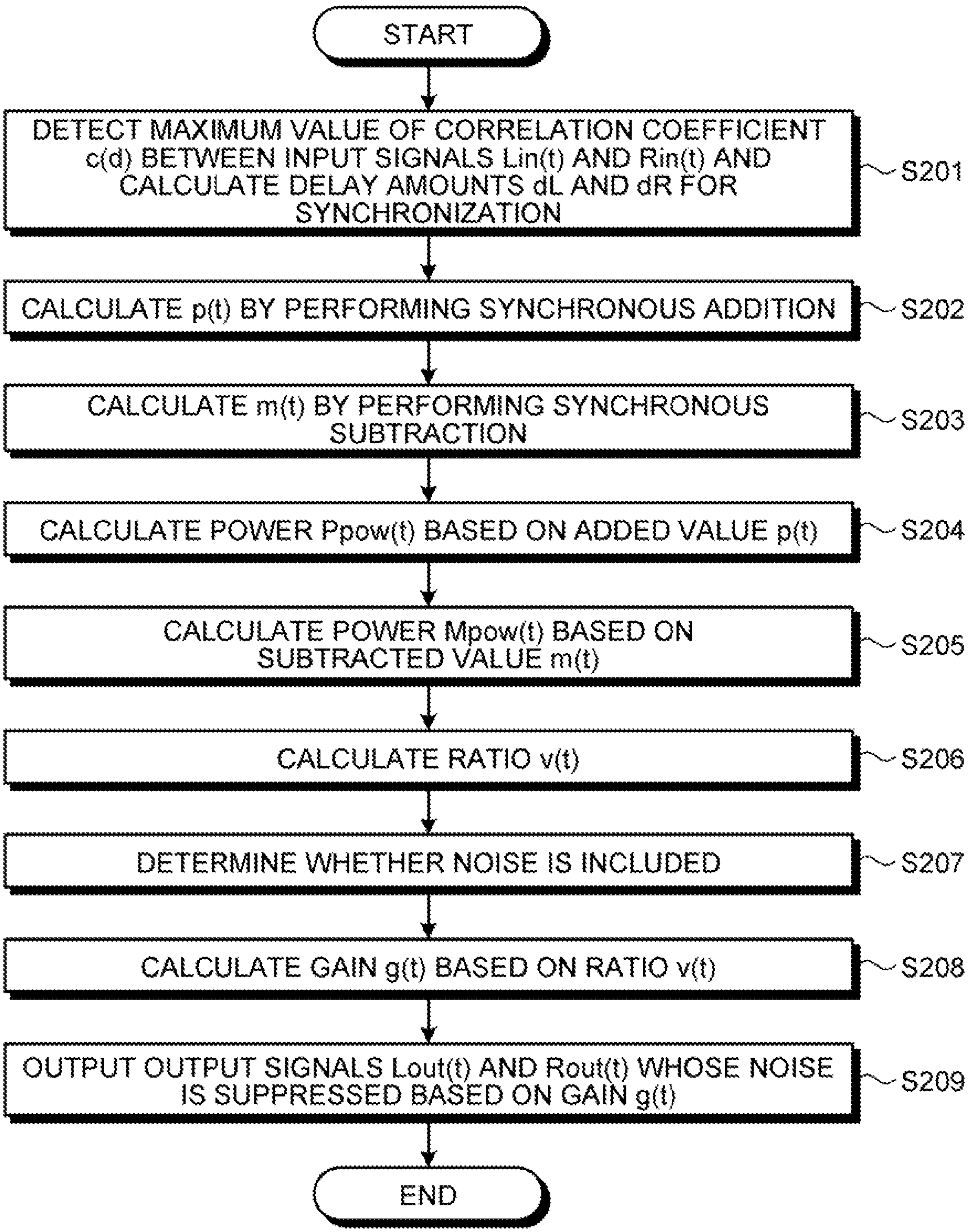
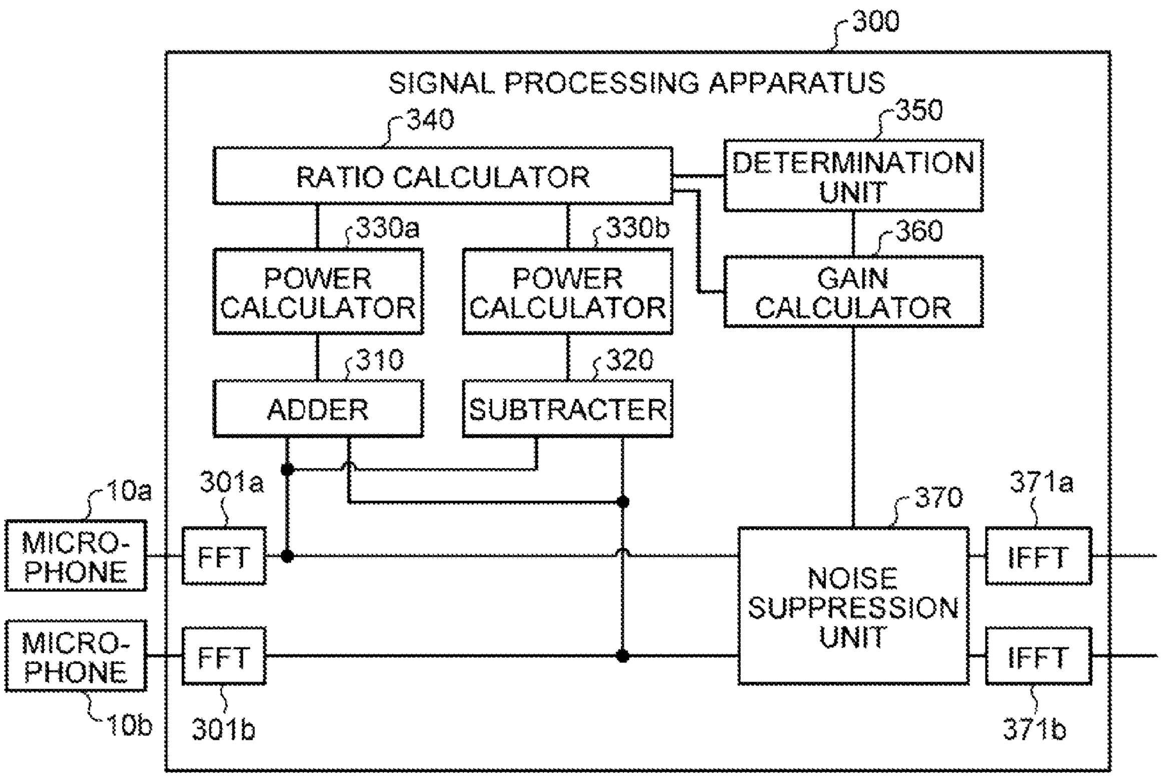

START
DETECT MAXIMUM VALUE OF CORRELATION COEFFICIENT c(d) BETWEEN INPUT SIGNALS Lin(t) AND Rin(t) AND CALCULATE DELAY AMOUNTS dL AND dR FOR SYNCHRONIZATION
S201
CALCULATE p(t) BY PERFORMING SYNCHRONOUS ADDITION
S202
CALCULATE m(t) BY PERFORMING SYNCHRONOUS SUBTRACTION
S203
CALCULATE POWER Ppow(t) BASED ON ADDED VALUE p(t)
S204
CALCULATE POWER Mpow(t) BASED ON SUBTRACTED VALUE m(t)
S205
CALCULATE RATIO v(t)
S206
DETERMINE WHETHER NOISE IS INCLUDED
S207
CALCULATE GAIN g(t) BASED ON RATIO v(t)
S208
OUTPUT OUTPUT SIGNALS Lout(t) AND Rout(t) WHOSE NOISE IS SUPPRESSED BASED ON GAIN g(t)
S209
END

FIG.6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SIGNAL PROCESSING APPARATUS
340
RATIO CALCULATOR
350
DETERMINATION UNIT
330a
POWER CALCULATOR
330b
POWER CALCULATOR
360
GAIN CALCULATOR
310
ADDER
320
SUBTRACTER
10a
MICRO-PHONE
301a
FFT
370
NOISE SUPPRESSION UNIT
371a
IFFT
MICRO-PHONE
10b
FFT
301b
IFFT
371b

FIG.7

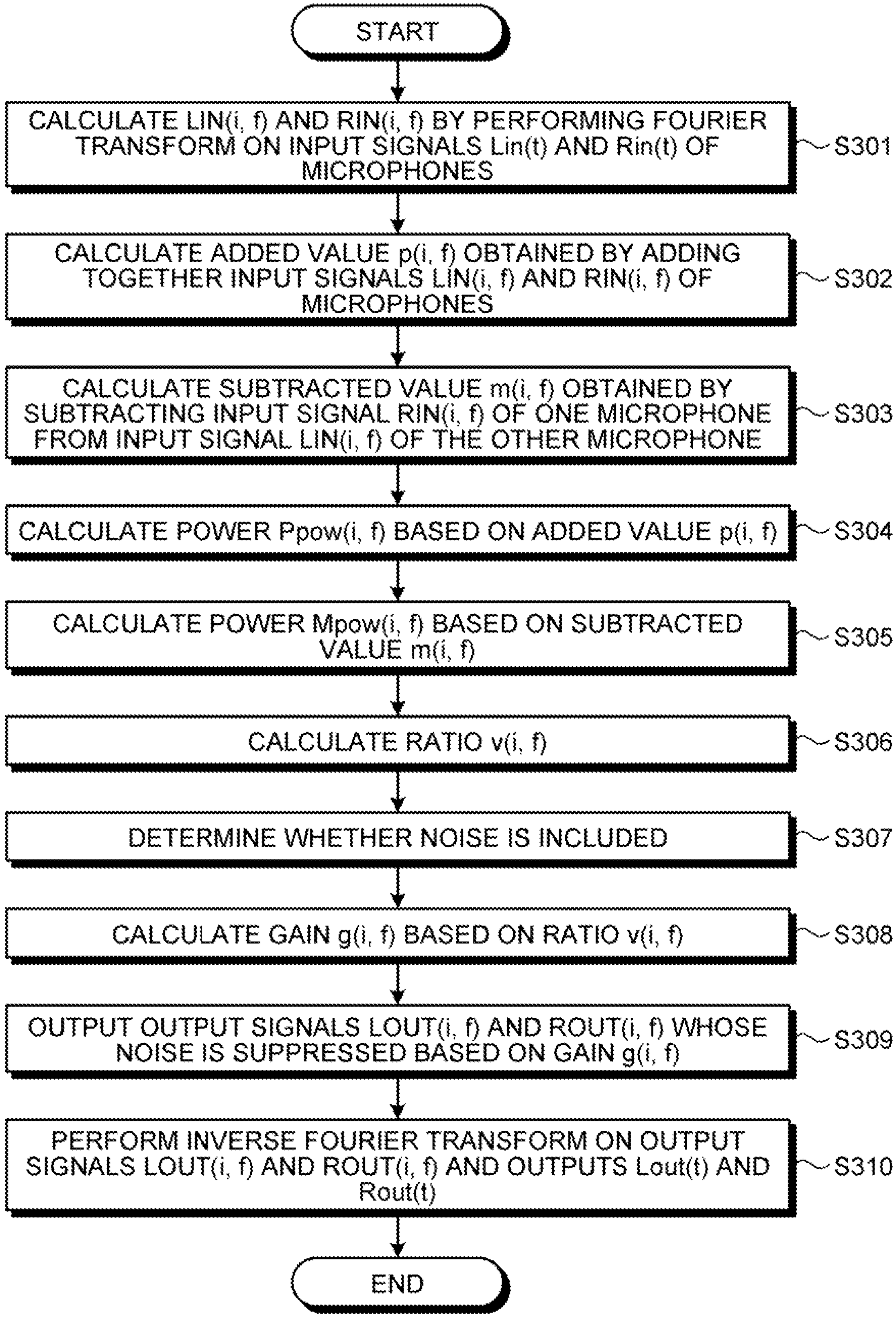

START
CALCULATE LIN(i, f) AND RIN(i, f) BY PERFORMING FOURIER TRANSFORM ON INPUT SIGNALS Lin(t) AND Rin(t) OF MICROPHONES
S301
CALCULATE ADDED VALUE p(i, f) OBTAINED BY ADDING TOGETHER INPUT SIGNALS LIN(i, f) AND RIN(i, f) OF MICROPHONES
S302
CALCULATE SUBTRACTED VALUE m(i, f) OBTAINED BY SUBTRACTING INPUT SIGNAL RIN(i, f) OF ONE MICROPHONE FROM INPUT SIGNAL LIN(i, f) OF THE OTHER MICROPHONE
S303
CALCULATE POWER Ppow(i, f) BASED ON ADDED VALUE p(i, f)
S304
CALCULATE POWER Mpow(i, f) BASED ON SUBTRACTED VALUE m(i, f)
S305
CALCULATE RATIO v(i, f)
S306
DETERMINE WHETHER NOISE IS INCLUDED
S307
CALCULATE GAIN g(i, f) BASED ON RATIO v(i, f)
S308
OUTPUT OUTPUT SIGNALS LOUT(i, f) AND ROUT(i, f) WHOSE NOISE IS SUPPRESSED BASED ON GAIN g(i, f)
S309
PERFORM INVERSE FOURIER TRANSFORM ON OUTPUT SIGNALS LOUT(i, f) AND ROUT(i, f) AND OUTPUTS Lout(t) AND Rout(t)
S310
END

FIG.8

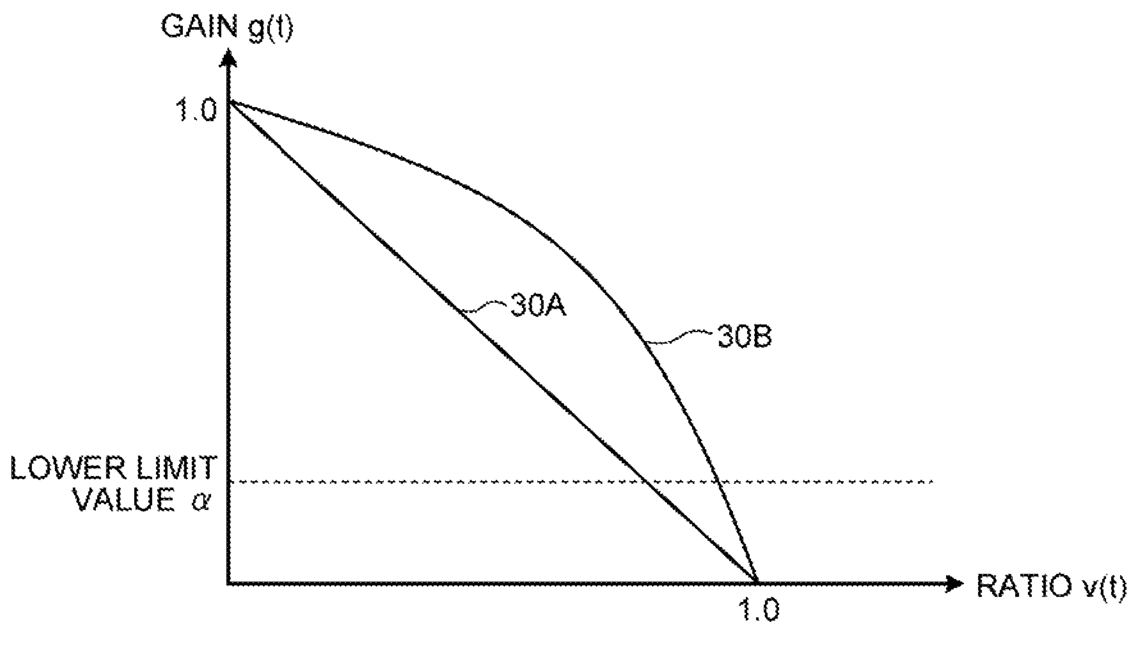


FIG.9

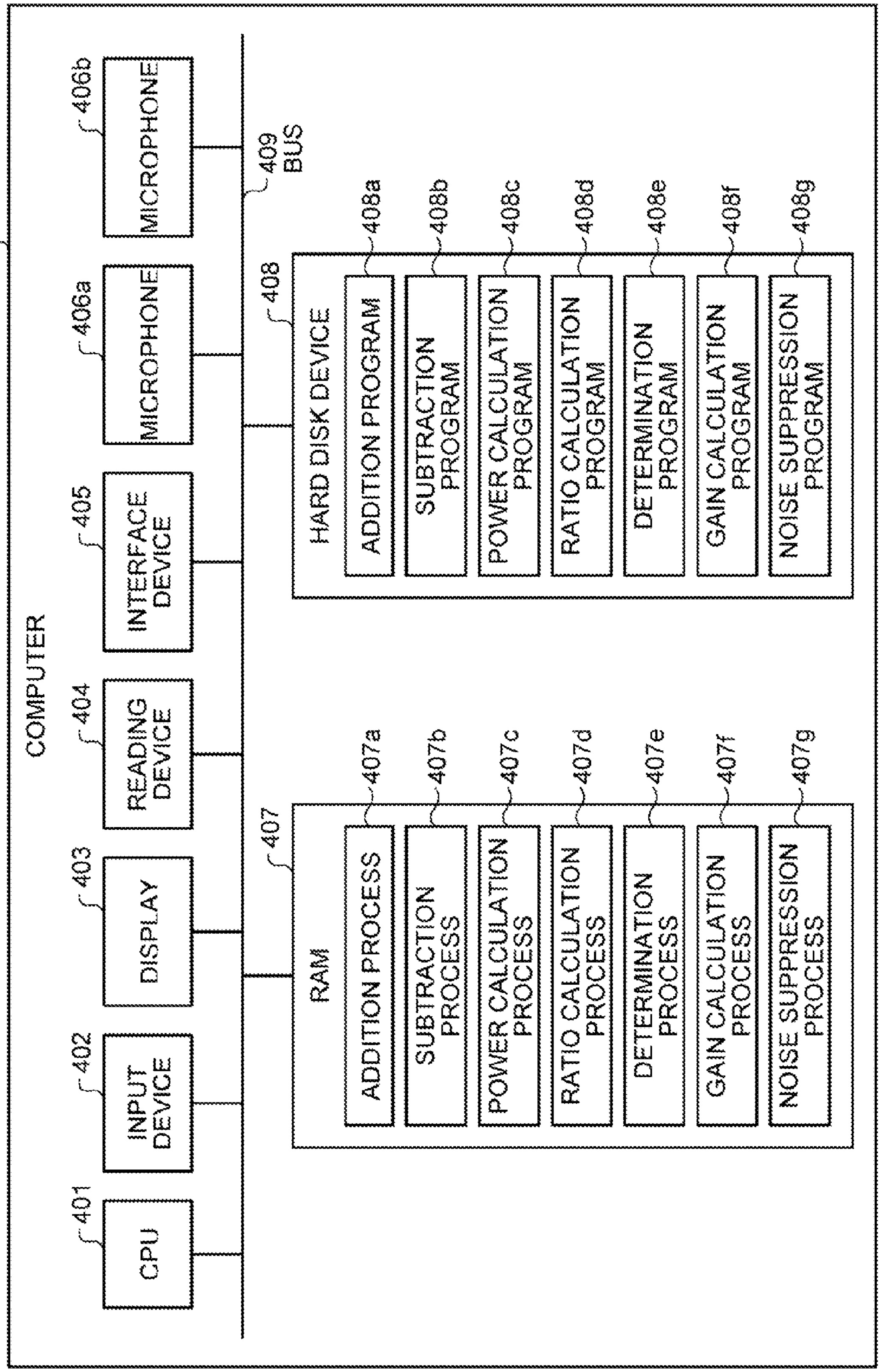
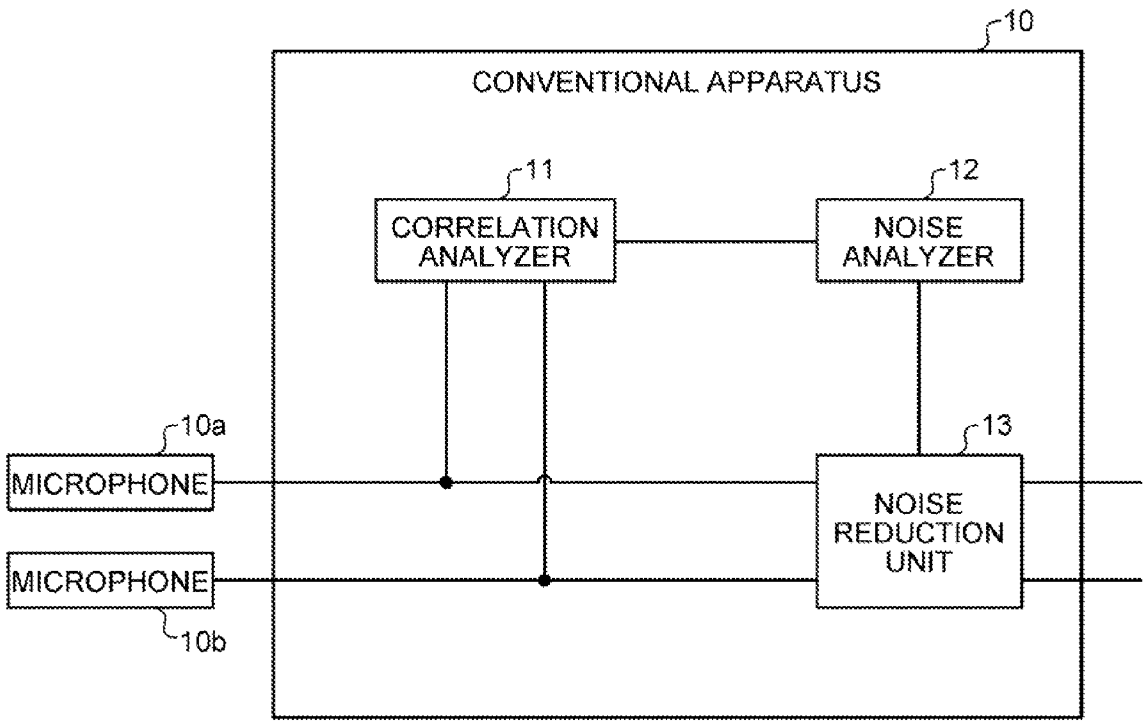

400
COMPUTER
401
CPU
402
INPUT DEVICE
403
DISPLAY
404
READING DEVICE
405
INTERFACE DEVICE
406a
MICROPHONE
406b
MICROPHONE
409
BUS
407
RAM
407a
ADDITION PROCESS
407b
SUBTRACTION PROCESS
407c
POWER CALCULATION PROCESS
407d
RATIO CALCULATION PROCESS
407e
DETERMINATION PROCESS
407f
GAIN CALCULATION PROCESS
407g
NOISE SUPPRESSION PROCESS
408
HARD DISK DEVICE
408a
ADDITION PROGRAM
408b
SUBTRACTION PROGRAM
408c
POWER CALCULATION PROGRAM
408d
RATIO CALCULATION PROGRAM
408e
DETERMINATION PROGRAM
408f
GAIN CALCULATION PROGRAM
408g
NOISE SUPPRESSION PROGRAM

FIG.10


10
CONVENTIONAL APPARATUS
11
CORRELATION ANALYZER
12
NOISE ANALYZER
10a
MICROPHONE
13
NOISE REDUCTION UNIT
MICROPHONE
10b

SIGNAL PROCESSING APPARATUS AND SIGNAL PROCESSING METHOD

CROSS-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 **10***a* and **10***b*. The microphones **10***a* and **10***b* 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 **10***a* 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 **10***b* 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**.

$$r(t)=\Sigma L\mathrm{in}(t-i)R\mathrm{in}(t-i)/((\Sigma L\mathrm{in}(t-i)^2)^{1/2}(\Sigma R\mathrm{in}(t-i)^2)^{1/2}) \quad (1)$$

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

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 connected to microphones **10**a and **10**b. The microphones **10**a and **10**b 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 **130**a and **130**b. 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 **10**a 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 **10**b 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 **130**a.

$$p(t)=L\mathrm{in}(t)+R\mathrm{in}(t) \tag{2}$$

The subtracter **120** calculates an 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 **130**b.

$$m(t)=L\mathrm{in}(t)-R\mathrm{in}(t) \tag{3}$$

The power calculator **130**a 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 **130**a 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 **130**a outputs the power Ppow(t) to the ratio calculator **140**.

$$P\mathrm{pow}(t)=\Sigma p(t-j)^2 \tag{4}$$

The power calculator **130**b 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 **130**b 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 **130**b outputs the power Mpow(t) to the ratio calculator **140**.

$$M\mathrm{pow}(t)=\Sigma m(t-j)^2 \tag{5}$$

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).

$$v(t)=M\mathrm{pow}(t)/P\mathrm{pow}(t) \tag{6}$$

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

$$g'(t)=1-v(t) \tag{7}$$

$$g(t)=\max(g'(t),\alpha) \tag{8}$$

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).

$$L\mathrm{out}(t)=g(t)L\mathrm{in}(t) \tag{9}$$

$$R\mathrm{out}(t)=g(t)R\mathrm{in}(t) \tag{10}$$

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 **10**a and **10**b.

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 **10**a and **10**b (step S**101**). The signal processing apparatus **100** calculates the subtracted value m(t) obtained by subtracting the input signal Rin(t) of one microphone **10**b from the input signal Lin(t) of the other microphone **10**a (step S**102**).

The signal processing apparatus **100** calculates the power Ppow(t) based on the added value p(t) (step S**103**). The signal processing apparatus **100** calculates the power Mpow(t) based on the subtracted value m(t) (step S**104**).

The signal processing apparatus **100** calculates the ratio v(t) (step S**105**) and determines whether noise is included (step S**106**). The signal processing apparatus **100** calculates the gain g(t) based on the ratio v(t) (step S**107**). 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 S**108**).

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 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 **20**A in FIG. **3** indicates a relationship between the sampling number and the gain g(t) of the conventional apparatus. Signals **20**B 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 **20**A and **20**B are sections including voice. When comparing the signals **20**A and **20**B, in the signals **20**B, 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 Ppow(t) and the Mpow(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 **10***a* and **10***b*. The microphones **10***a* and **10***b* are spaced apart from each other by a predetermined distance. The description about the microphones **10***a* and **10***b* is the same as that in the first embodiment.

The signal processing apparatus **200** includes a delay detector **201** and delay devices **202***a* and **202***b*. The signal processing apparatus **200** also includes an adder **210**, a subtracter **220**, and power calculators **230***a* and **230***b*. The signal 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 dmax by which the value of correlation coefficient c(d) is the maximum.

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

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

When the value of the dmax 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 dmax. Then, the delay detector **201** outputs the dL to the delay device **202***a* and outputs the dR to the delay device **202***b*.

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

The delay device **202***a* performs a delay process of the input signal Lin(t) from the microphone **10***a*. The delay device **202***a* 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 **202***b* performs a delay process of the input signal Rin(t) from the microphone **10***b*. The delay device **202***b* 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 **230***a*.

The subtracter **220** calculates an 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 **230***b*.

The descriptions about the power calculators **230***a* and **230***b*, 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 **130***a* and **130***b*, 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 **10***a* and **10***b*.

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 S**201**). The signal processing apparatus **200** calculates the p(t) by performing synchronous addition (step S**202**) and calculates the m(t) by performing synchronous subtraction (step S**203**).

The signal processing apparatus **200** calculates the power Ppow(t) based on the added value p(t) (step S**204**). The signal processing apparatus **200** calculates the power Mpow(t) based on the subtracted value m(t) (step S**205**).

The signal processing apparatus **200** calculates the ratio v(t) (step S**206**) and determines whether noise is included (step S**207**). The signal processing apparatus **200** calculates the gain g(t) based on the ratio v(t) (step S**208**). The signal processing apparatus **200** outputs the output signals Lout(t) and Rout(t) whose noise is suppressed based on the gain g(t) (step S**209**).

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 **202***a* and **202***b* synchronize the input signals. In the signal processing apparatus **200**, the determination unit **250** 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 **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 **10***a* and **10***b* 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 **10***a* and **10***b*. The microphones **10***a* and **10***b* are spaced apart from each other by a predetermined distance. The description about the microphones **10***a* and **10***b* is the same as that in the first embodiment.

The signal processing apparatus **300** includes FFTs (Fast Fourier Transforms) **301***a* and **301***b*, an adder **310**, a subtracter **320**, and power calculators **330***a* and **330***b*. 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) **371***a* and **371***b*.

The FFT **301***a* positions a Hanning window with 50% overlap on the input signal Lin(t) acquired from the microphone **10***a* and converts the input signal Lin(t) into a signal on the frequency axis by fast Fourier transform. The input signal Lin(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 **301***a* outputs the LIN(i, f) to the adder **310**, the subtracter **320**, and the noise suppression unit **370**.

The FFT **301***b* positions a Hanning window with 50% overlap on the input signal Rin(t) acquired from the microphone **10***b* and converts the input signal Rin(t) into a signal on the frequency axis by fast Fourier transform. The input signal Rin(t) converted into a signal on the frequency axis is represented as RIN(i, f). The FFT **301***b* outputs the RIN(i, f) to the adder **310**, the subtracter **320**, and the noise suppression unit **370**. The FFTs **301***a* and **301***b* 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 **330***a*.

$$p(i,f)=LIN(i,f)+RIN(i,f) \tag{14}$$

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 **330***b*.

$$m(i,f)=LIN(i,f)-RIN(i,f) \tag{15}$$

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

$$Ppow(i,f)=\Sigma p(i-j,f) \tag{16}$$

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

$$Mpow(i,f)=\Sigma m(i-j,f) \tag{17}$$

The ratio calculator **340** calculates a ratio v(i, f) of the power Ppow(i, f) and the power Mpow(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**.

$$v(i,f)=Mpow(i,f)/Ppow(i,f) \tag{18}$$

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).

$$g'(i,f)=1-v(i,f) \tag{19}$$

$$g(i,f)=\max(g'(i,f),\alpha) \tag{20}$$

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 **371**a and **371**b. 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 **371**a 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 **371**a outputs the output signal Lin(t) to an external apparatus.

The IFFT **371**b 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 **371**b 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 **10**a and **10**b.

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 S**301**). 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 S**302**).

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 S**303**). The signal processing apparatus **300** calculates the power Ppow(i, f) based on the added value p(i, f) (step S**304**).

The signal processing apparatus **300** calculates the power Mpow(i, f) based on the subtracted value m(i, f) (step S**305**). The signal processing apparatus **300** calculates the ratio v(i, f) (step S**306**) and determines whether noise is included (step S**307**).

The signal processing apparatus **300** calculates the gain g(i, f) based on the ratio v(i, f) (step S**308**). 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 S**309**). 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 S**310**).

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 **10**a and **10**b.

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 **10**a and **10**b 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 **30**A in FIG. **8**. The relationship illustrated by the straight line **30**A corresponds to the relationship of the formula (7).

On the other hand, the gain calculator **160** defines that when the distance between the microphones **10**a and **10**b is smaller than 4.2 cm, the relationship changes as illustrated by a curved line **30**B 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 **30**A 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 **10**a and **10**b 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 **130**a will be described. The power calculator **130**a 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 **130**a obtains the average of the absolute values by the formula (23).

$$Pamp(t)=\Sigma|p(t-j)| \quad (23)$$

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

$$Mamp(t)=\Sigma|m(t-j)| \quad (24)$$

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 **130**a and **130**b 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 **10**c and **10**d not illustrated in the drawings in addition to the pair of microphones **10**a and **10**b. For example, the signal processing apparatus **100** calculates the ratio v(t) from the microphones **10**a and **10**b by the same process as that in the first embodiment. The v(t) calculated from the microphones **10**a and **10**b is defined as v**1**(t).

Also, the ratio v(t) is calculated from the microphones **10**c and **10**d by the same process as that in the first embodiment. The v(t) calculated from the microphones **10**c and **10**d is defined as v**2**(t). When both the v**1**(t) and v**2**(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 first embodiment. The g'(t) may be obtained by subtracting the average value of the v**1**(**1**) and v**2**(**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 **10***a* to **10***d*, 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 **406***a* and **406***b*. 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 **408***a*, a subtraction program **408***b*, a power calculation program **408***c*, a ratio calculation program **408***d*, a determination program **408***e*, a gain calculation program **408***f*, and a noise suppression program **408***g*. The CPU **401** reads the programs **408***a* to **408***g* and develops the programs in the RAM **407**.

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

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

The programs **408***a* to **408***g* 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 **408***a* to **408***g* from these media and execute the programs. The programs of 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

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;

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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