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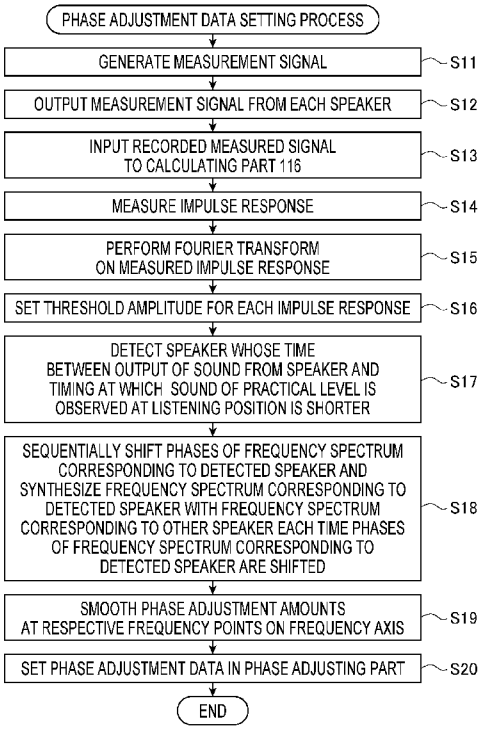

(57) Abstract: A phase control device including a measuring part for measuring impulse responses between a predetermined position and respective one of two speakers, a setting part for setting a threshold value of amplitude for each of the measured impulse responses, a detecting part for detecting, from among the two speakers, a speaker whose time, from the output of the sound, required for an amplitude of the corresponding impulse response to reach the threshold value is shorter, and a generating part for generating phase adjustment data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker.

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Description

Title of Invention: PHASE CONTROL DEVICE, ACOUSTIC DEVICE AND PHASE CONTROL METHOD

Technical Field

[0001] The present invention relates to a phase control device, an acoustic device and a phase control method.

Background Art

[0002] Generally, in a vehicle interior, speakers are provided at a plurality of positions. For example, a front right speaker on the right door and a front left speaker on the left door are provided at positions symmetrical to each other with respect to a central line of an in-vehicle space. However, if a listening position (e.g., a driver seat, a front passenger seat or a rear seat) is considered as a reference position, these speakers are not positioned symmetrically.

[0003] For example, in case the listener is sitting on the driver seat, a distance between the front right speaker and the listener and a distance between the front left speaker and the listener are not the same. In case of a right-hand drive car, the former distance is shorter than the latter distance. Therefore, if sounds are output from both speakers at the same time, in general, the sound output from the front right speaker reaches the ears of the listener sitting on the driver seat first and then the sound output from the front left speaker reaches the ears of the listener. Due to such difference in times required for playback sounds output from respective speakers to reach the ears of the listener caused by the differences in distances between the listening position and respective speakers, sound image localization bias by Haas effect occurs.

[0004] As a technique for suppressing the sound image localization bias, a time alignment is known. The time alignment is a technique for setting time delays to respective channels corresponding to respective speakers such that sounds output from respective speakers reach the listener (listening position) at the same time. Specific configuration of a device for performing the time alignment is disclosed for example in Japanese Patent Provisional Publication No. 2005-341534 (hereinafter referred to as "Patent Document 1").

[0005] The device disclosed in Patent Document 1 sequentially collects sounds which are sequentially output from respective speakers at a predetermined position set as a target for the time alignment, measures impulse responses between respective speakers and the predetermined position from the collected sounds, measures rise times of the impulse responses corresponding to respective speakers, and sets time delays to respective channels corresponding to respective speakers based on difference in the rise

times. The device disclosed in Patent Document 1 is configured to improve calculation accuracy of the delay times to be set to respective channels corresponding to respective speakers by applying a low pass filter to the impulse responses to remove high-frequency noise to thereby detect the rises of the impulse responses with high accuracy.

Summary of Invention

[0006] In special listening environments such as a vehicle interior, interferences between sounds output from respective speakers are likely to occur. Since such interferences causes dips in the frequency spectrum, it is difficult to suppress degradation in sound quality and decrease in sound pressure due to the dip only by the conventional technique (i.e., the time alignment) for localizing sound image.

[0007] The present invention is made in view of the above mentioned circumstances and thus the object of the present invention is to provide a phase control device, an acoustic device and a phase control method that are capable of suppressing the sound image localization bias at the listening position and further suppressing degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the listening position.

[0008] A phase control device according to an embodiment of the present invention includes a measuring part configured to measure impulse responses between a predetermined position and respective one of two speakers from signals of sounds collected at the predetermined position, the sounds being output from respective one of the two speakers at timings such that the sounds output from respective one of the two speakers do not interfere with each other at the predetermined position, a setting part configured to set a threshold value of amplitude for each of the measured impulse responses, a detecting part configured to detect, from among the two speakers, a speaker of which time, from the output of the sound, required for an amplitude of the corresponding impulse response to reach the threshold value is shorter, and a generating part configured to generate phase adjustment data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker to suppress destructive interference between sounds output from the two speakers at respective frequencies at the predetermined position.

[0009] According to the phase control device configured as described above, it becomes possible to obtain preferred phase adjustment data for suppressing the sound image localization bias at the predetermined position (e.g., a listening position) and further suppressing degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined position.

[0010] The generating part may sequentially shift phases of a frequency spectrum of the impulse response corresponding to the detected speaker for a predetermined angle and synthesize the impulse response corresponding to the detected speaker with a frequency spectrum of the impulse response corresponding to the other speaker each time the phases of the frequency spectrum of the impulse response corresponding to the detected speaker are shifted. The generating part may obtain the phase adjustment data by obtaining phase adjustment amounts at respective frequencies of the frequency spectrum of the impulse response corresponding to the detected speaker based on the result of the synthesis.

[0011] The generating part may further perform smoothing on the obtained phase adjustment amounts at respective frequencies on the frequency axis and obtain the smoothed phase adjustment amounts at respective frequencies as the phase adjustment data.

[0012] By performing the above process, the phase adjustment data in which sudden change in phase in the frequency domain is suppressed can be obtained. Therefore, when a sound signal of which phases are adjusted using the phase adjustment data is input to a speaker, harmonics due to the sudden phase change are suppressed and thus auditory allophones due to such harmonics can be suppressed.

[0013] The phase adjustment data is, for example, data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker such that, at respective frequencies, phases of a sound output by the detected speaker and phases of a sound output by the other speaker become substantially the same at the predetermined position.

[0014] The setting part may be configured to set, for each of the impulse responses, the threshold value in accordance with a maximum amplitude of the impulse response.

[0015] By setting the threshold value as described above, it becomes possible to detect a speaker whose phases of a signal to be input to are to be controlled with high accuracy.

[0016] An acoustic device according to an embodiment of the present invention includes the above-described phase control device and is configured to output a sound signal input from a sound source to the two speakers. The acoustic device includes an adjusting part configured to adjust, by using the phase adjustment data, phases at respective frequencies of the sound signal input from the sound source and to be input to the detected speaker.

[0017] According to the acoustic device configured as described above, it becomes possible to input, to the two speakers, respective sound signals that can suppress the sound image localization bias at the predetermined position (e.g., a listening position) and further suppress degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined

position.

[0018] An acoustic device according to another embodiment of the present invention is configured to input a sound signal input from a sound source to two speakers. The acoustic device includes an adjusting part configured to adjust phases at respective frequencies of the sound signal input from the sound source and to be input to a specific speaker among the two speakers such that, at respective frequencies, destructive interference between sounds output by respective one of the two speakers at a predetermined position is reduced. The specific speaker is a speaker, among the two speakers, of which time, from an output of a sound from the speaker, required for an amplitude of the sound observed at the predetermined position to reach a predetermined threshold value is shorter.

[0019] According to the acoustic device configured as described above, it becomes possible to input, to the two speakers, respective sound signals that can suppress the sound image localization bias at the predetermined position (e.g., a listening position) and further suppress degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined position.

[0020] A phase control method, according to an embodiment of the present invention, to be executed by a computer includes a measuring step of measuring impulse responses between a predetermined position and respective one of two speakers from signals of sounds collected at the predetermined position, the sounds being output from respective one of the two speakers at timings such that the sounds output from respective one of the two speakers do not interfere with each other at the predetermined position; a setting step of setting a threshold value of amplitude for each of the measured impulse responses; a detecting of detecting, from among the two speakers, a speaker of which time, from the output of the sound, required for an amplitude of the corresponding impulse response to reach the threshold value is shorter; and a generating step of generating phase adjustment data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker to suppress destructive interference between sounds output from the two speakers at respective frequencies at the predetermined position.

[0021] According to the phase control method described above, it becomes possible to obtain preferred phase adjustment data for suppressing the sound image localization bias at the predetermined position (e.g., a listening position) and further suppressing degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined position.

[0022] A phase control method, according to another embodiment of the present invention, to be executed by an acoustic device configured to input a sound signal input from a

sound source to two speakers includes an adjusting step of adjusting phases at respective frequencies of the sound signal input from the sound source and to be input to a specific speaker among the two speakers such that, at respective frequencies, destructive interference between sounds output by respective one of the two speakers at a predetermined position is reduced. The specific speaker is a speaker, among the two speakers, of which time, from an output of a sound from the speaker, required for an amplitude of the sound observed at the predetermined position to reach a predetermined threshold value is shorter.

[0023] According to the phase control method described above, it becomes possible to input, to the two speakers, respective sound signals that can suppress the sound image localization bias at the predetermined position (e.g., a listening position) and further suppress degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined position.

[0024] In the generating step, phases of a frequency spectrum of the impulse response corresponding to the detected speaker may be sequentially shifted for a predetermined angle and the impulse response corresponding to the detected speaker may be synthesized with a frequency spectrum of the impulse response corresponding to the other speaker each time the phases of the frequency spectrum of the impulse response corresponding to the detected speaker are shifted. The phase adjustment data may be obtained by obtaining phase adjustment amounts at respective frequencies of the frequency spectrum of the impulse response corresponding to the detected speaker based on the result of the synthesis.

[0025] In the generating step, smoothing may be performed on the obtained phase adjustment amounts at respective frequencies on the frequency axis and the smoothed phase adjustment amounts at respective frequencies may be obtained as the phase adjustment data.

[0026] By performing the above process, the phase adjustment data in which sudden change in phase in the frequency domain is suppressed can be obtained. Therefore, when a sound signal of which phases are adjusted using the phase adjustment data is input to a speaker, harmonics due to the sudden phase change are suppressed and thus auditory allophones due to such harmonics can be suppressed.

[0027] The phase adjustment data may be data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker such that, at respective frequencies, phases of a sound output by the detected speaker and phases of a sound output by the other speaker become substantially the same at the predetermined position.

[0028] In the setting step, for each of the impulse responses, the threshold value may be set

in accordance with a maximum amplitude of the impulse response.

[0029] By setting the threshold value as described above, it becomes possible to detect a speaker whose phases of a signal to be input to are to be controlled with high accuracy.

[0030] The above phase control method may further include, when inputting a sound signal input from a sound source to the two speakers, an adjusting step of adjusting, by using the phase adjustment data, phases at respective frequencies of the sound signal input from the sound source and to be input to the detected speaker.

[0031] According to the phase control method described above, it becomes possible to input, to the two speakers, respective sound signals that can suppress the sound image localization bias at the predetermined position (e.g., a listening position) and further suppress degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined position.

[0032] A phase control method, according to another embodiment of the present invention, to be executed by an acoustic device configured to input a sound signal input from a sound source to two speakers includes an adjusting step of adjusting phases at respective frequencies of the sound signal input from the sound source and to be input to a specific speaker among the two speakers such that, at respective frequencies, destructive interference between sounds output by respective one of the two speakers at a predetermined position is reduced. The specific speaker is a speaker, among the two speakers, whose time, from an output of a sound from the speaker, required for an amplitude of the sound observed at the predetermined position to reach a predetermined threshold value is shorter.

[0033] According to the phase control method described above, it becomes possible to input, to the two speakers, respective sound signals that can suppress the sound image localization bias at the predetermined position (e.g., a listening position) and further suppress degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from respective speakers at the predetermined position.

Brief Description of Drawings

[0034] [fig.1]Fig. 1 is a diagram illustrating a vehicle in which an acoustic system according to an embodiment of the present invention is installed.

[fig.2]Fig. 2 is a block chart showing a configuration of the acoustic system according to the embodiment of the present invention.

[fig.3]Fig. 3 is a flowchart showing a phase adjustment data setting process to be executed by the acoustic system according to the embodiment of the present invention.

[fig.4]Fig. 4 is a block chart showing a configuration of a calculating part in an

acoustic device included in the acoustic system according to the embodiment of the present invention.

[fig.5A]Fig. 5A is a diagram showing an impulse response between a front left speaker and a listening position (i.e., a driver seat).

[fig.5B]Fig. 5B is a diagram showing an impulse response between a front right speaker and the listening position.

[fig.6A]Fig. 6A is a diagram showing a frequency characteristic of amplitude obtained by performing Fourier transform on the impulse response between the front left speaker and the listening position.

[fig.6B]Fig. 6B is a diagram showing a frequency characteristic of amplitude obtained by performing Fourier transform on the impulse response between the front right speaker and the listening position.

[fig.7A]Fig. 7A is a diagram showing a frequency characteristic of phase obtained by performing Fourier transform on the impulse response between the front left speaker and the listening position.

[fig.7B]Fig. 7B is a diagram showing a frequency characteristic of phase obtained by performing Fourier transform on the impulse response between the front right speaker and the listening position.

[fig.8A]Fig. 8A is a diagram showing the impulse response between the front left speaker and the listening position in which the amplitude is converted into sound pressure level (in dB).

[fig.8B]Fig. 8B is a diagram showing the impulse response between the front right speaker and the listening position in which the amplitude is converted into sound pressure level (in dB).

[fig.9]Fig. 9 is a diagram showing a relationship between phase adjustment amount of a frequency spectrum corresponding to the front right speaker and amplitude of a synthesized frequency spectrum at 100Hz and 400Hz, respectively, the relationship being obtained as a result of a synthesizing process by a phase control part included in a calculating part according to the embodiment of the present invention.

[fig.10]Fig. 10 is a diagram showing phase adjustment amounts at respective frequency points when, at respective frequency points, phases of a sound from the front right speaker and phases of a sound from the front left speaker become the same (or substantially the same) at the listening position.

[fig.11]Fig. 11 is a diagram showing the phase adjustment amounts after a smoothing process by a smoothing part included in the calculating part according to the embodiment of the present invention.

[fig.12]Fig. 12 is a block chart showing a configuration of a phase adjusting part included in the acoustic device according to the embodiment of the present invention.

[fig.13A]Fig. 13A is a diagram showing a time characteristic of an audio signal measured at the listening position in an example without adjustment.

[fig.13B]Fig. 13B is a diagram showing a time characteristic of an audio signal measured at the listening position in an example with time alignment.

[fig.13C]Fig. 13C is a diagram showing a time characteristic of an audio signal observed at the listening position in an example with phase adjustment according to the embodiment.

[fig.14]Fig. 14 is a diagram showing a frequency characteristic of amplitude of the audio signal observed at the listening position in the example without adjustment and a frequency characteristic of amplitude of the audio signal observed at the listening position in the example with time alignment.

[fig.15]Fig. 15 is a diagram showing the frequency characteristic of amplitude of the audio signal observed at the listening position in the example without adjustment and a frequency characteristic of amplitude of the audio signal observed at the listening position in the example with phase adjustment.

Description of Embodiments

[0035] Hereinafter, an embodiment of the present invention will be described with reference to the accompanying drawings. In the following, an acoustic system is cited as an embodiment of the present invention.

[0036] Fig. 1 is a diagram illustrating a vehicle A in which an acoustic system 1 according to the embodiment of the present invention is installed. Fig. 2 is a block diagram showing a configuration of the acoustic system 1.

[0037] As shown in Figs. 1 and 2, the acoustic system 1 includes an acoustic device 10, speakers SP_{FR} and SP_{FL} and a microphone MIC.

[0038] The acoustic device 10 has a phase adjustment data generating function (in other words, a phase control device) for generating phase adjustment data for suppressing sound image localization which is likely to occur in vehicle interiors and for suppressing sound quality degradation and sound pressure decrease due to interferences between sounds output from respective speakers installed in the vehicle A.

[0039] It is noted that various processes in the acoustic device 10 are executed under cooperation between hardware (e.g., a CPU, an ASIC or a combination thereof being examples of a computer) and software provided in the acoustic device 10. At least an OS (Operating System) of the software provided in the acoustic device 10 is provided as an embedded system, and the other part of the software, such as a software module for executing the phase adjustment data generating process, may be provided as an application which can be distributed through a network or an application storable in a recording medium such as a memory card. That is, the phase adjustment data

generating function according to the present embodiment may be the function embedded in the acoustic device 10 in advance (e.g., before the shipment) or may be the function which can be added to the acoustic device 10 through a network or a recording medium.

[0040] As shown in Fig. 1, the speaker SP_{FR} is a front right speaker embedded in a right door (a door on the driver seat side) and the speaker SP_{FL} is a front left speaker embedded in a left door (a door on the front passenger seat side). Further speakers (e.g., rear speakers) may be provided to the vehicle A (i.e., three or more speakers may be provided to the vehicle A).

[0041] The acoustic device 10 has a controller 100, a display part 102, an operating part 104, a measurement signal generating part 106, a recording medium reproducing part 108, a phase adjusting part 110, an amplifier 112, a signal recording part 114 and a calculating part 116.

[0042] Fig. 3 is a flowchart showing a phase adjustment data setting process to be executed by the acoustic system 1. It is noted that various processes by the acoustic system 1 such as the phase adjustment data setting process shown in this flowchart are executed under control of the controller 100. Upon receiving a predetermined touching operation on the display part 102 or a predetermined operation on the operating part 104, the controller 100 starts executing the phase adjustment data setting process shown in this flowchart.

[0043] Once the execution of the phase adjustment data setting process shown in Fig. 3 is started, the measurement signal generating part 106 generates a predetermined measurement signal (step S11). The generated measurement signal is, for example, of M-sequence (Maximal length sequence) symbol. A length of the measurement signal is more than twice a length of the symbol. It is noted that the measurement signal may be signals of other types such as, for example, a TSP (Time Stretched Pulse) signal.

[0044] The measurement signal passes through the controller 100 and the phase adjusting part 110 by through-output and is sequentially output to respective speakers SP_{FR} and SP_{FL} via the amplifier 112 (step S12). A predetermined measurement sound is thereby sequentially output from respective speakers SP_{FR} and SP_{FL} with a predetermined time interval.

[0045] The microphone MIC is installed at a position where the sound image localization bias, sound quality degradation and sound pressure decrease are to be suppressed. In the present embodiment, the microphone MIC is installed at the driver seat in order to suppress the sound image localization bias, sound quality degradation and sound pressure decrease at a position of the listener (i.e., a listening position) sitting on the driver seat.

[0046] The microphone MIC collects the measurement sounds which are sequentially output

from respective speakers SP_{FR} and SP_{FL} at timings such that the measurement sounds output from respective speakers SP_{FR} and SP_{FL} do not interfere with each other at the microphone MIC. Signals of the measurement sounds collected by the microphone MIC (i.e., measured signals) are stored in the signal recording part 114 and are input from the signal recording part 114 to the calculating part 116 (step S13). It is noted that, if the calculating part 116 has the function to store the measured signals, the recording part 114 may be omitted and the measured signals output from the microphone MIC may be directly input to the calculating part 116.

[0047] Fig. 4 is a block chart showing a configuration of the calculating part 116. As shown in Fig. 4, the calculating part 116 includes measuring parts 116A and 116B.

[0048] The measuring parts 116A and 116B measure impulse responses (step S14).

[0049] Specifically, the measuring part 116A obtains a cross-correlation function between the measured signal of the measurement sound output by the speaker SP_{FL} (hereinafter referred to as a "measured signal L") and a reference measurement signal input from the controller 100 by calculation to calculate an impulse response of the measured signal L (in other words, an impulse response between the speaker SP_{FL} and the listening position; hereinafter referred to as an "impulse response L'").

[0050] Similarly, the measuring part 116B obtains a cross-correlation function between the measured signal of the measurement sound output by the speaker SP_{FR} (hereinafter referred to as a "measured signal R") and a reference measurement signal input from the controller 100 by calculation to calculate an impulse response of the measured signal R (in other words, an impulse response between the speaker SP_{FR} and the listening position; hereinafter referred to as an "impulse response R'").

[0051] It is noted that the reference measurement signal is a signal that is identical to the measurement signal generated by the measurement signal generating part 106 and that is time-synchronized with the measurement signal.

[0052] As described above, the measuring parts 116A and 116B operate as a measuring part for measuring impulse responses between a predetermined position and respective one of two speakers from signals of sounds collected at the predetermined position, the sounds being output from respective one of the two speakers at timings such that the sounds output from respective one of the two speakers do not interfere with each other at the predetermined position.

[0053] Fig. 5A shows an example of the impulse response L', and Fig. 5B shows an example of the impulse response R'. In Figs. 5A and 5B, the vertical axis represents amplitude and the horizontal axis represents time (unit: sec). In the examples shown in Figs. 5A and 5B, the sampling frequency is 44.1 kHz, the symbol length of the M-sequence symbol is 32,767, and the frequency range is 3 kHz. It is noted that the frequency range (i.e., a frequency range for the phase adjustment according to the

present invention) can be arbitrarily set within the range of up to Nyquist frequency. In the present embodiment, a range from the lower frequency limit of sound reproducible by the speakers SP_{FR} and SP_{FL} (60 Hz) up to twice an upper frequency of a frequency range that has great influence on the image localization (in other words, a frequency range within which shift in phase greatly influences the image localization) (i.e., 3 kHz) is set as the frequency range for the phase adjustment. However, the frequency range for the phase adjustment may be arbitrarily set in accordance with, for example, desired degree of effect and processing capacity of the system.

[0054] As shown in Figs. 5A and 5B, due to the special listening environment in the vehicle interior (i.e., an environment influenced by reflections with short propagation delay time differences due to structures in the vehicle interior, masking of sounds due to structures in the vehicle interior, interferences between sounds output from respective speakers and the like), a sound of low sound pressure level is observed at the listening position first (in other words, a sound pressure level of a rising part of the impulse response is low) and then a sound of high sound pressure level is observed at the listening position.

[0055] As shown in Fig. 4, the calculating part 116 includes Fourier transform parts 116C and 116D.

[0056] The Fourier transform part 116C performs Fourier transform on the impulse response L' input from the measuring part 116A and obtains a frequency spectrum of the impulse response L' (i.e., a frequency characteristic of amplitude and a frequency characteristic of phase of the impulse response L'; hereinafter referred to as "frequency spectrum L"") (step S15). The Fourier transform part 116D performs Fourier transform on the impulse response R' input from the measuring part 116B and obtains a frequency spectrum of the impulse response R' (i.e., a frequency characteristic of amplitude and a frequency characteristic of phase of the impulse response R'; hereinafter referred to as "frequency spectrum R"") (step S15).

[0057] Fig. 6A is a diagram showing the frequency characteristic of amplitude of the impulse response L' obtained by performing Fourier transform on the impulse response L', and Fig. 6B is a diagram showing the frequency characteristic of amplitude of the impulse response R' obtained by performing Fourier transform on the impulse response R'. In Figs. 6A and 6B, the vertical axis represents sound pressure level (unit: dB) and the horizontal axis represents frequency (unit: Hz).

[0058] Fig. 7A is a diagram showing the frequency characteristic of phase of the impulse response L' obtained by performing Fourier transform on the impulse response L', and Fig. 7B is a diagram showing the frequency characteristic of phase of the impulse response R' obtained by performing Fourier transform on the impulse response R'. In Figs. 7A and 7B, the vertical axis represents angle (unit: degree) and the horizontal

axis represents frequency (unit: Hz).

[0059] In the examples shown in Figs. 6A, 6B, 7A and 7B, the Fourier transform length is 8,192 samples. The number of frequency points is set to 4,097 points obtained by dividing a frequency range of 0 Hz to Nyquist frequency of 22.05 kHz with intervals of 5.38 Hz. The number of frequency points within a frequency range of 0 Hz to 3 kHz is 557 points. Since the reflections, masking and interferences occur in the vehicle interior, in the examples shown in Figs. 6A and 6B, amplitude varies widely depending on frequency and, in the examples shown in Figs. 7A and 7B, phase varies widely depending on frequency.

[0060] As shown in Fig. 4, the calculating part 116 includes a setting part 116E and a detecting part 116F.

[0061] The setting part 116E sets a threshold value of amplitude for each of the impulse responses L' and R' (step S16). The detecting part 116F detects, from among two speakers (in the present embodiment, the speaker SP_{FL} and the speaker SP_{FR}), a speaker whose time between an output of a sound from the speaker and a timing at which a sound of practical sound pressure level is observed at the listening position is shorter (step S17).

[0062] The following is the detailed description of Step S16. The setting part 116E converts the amplitude of the impulse response L' input from the measuring part 116A into sound pressure level (in dB), detects a maximum value of the sound pressure level (hereinafter referred to as a "maximum value MAX_L"), and sets a threshold value (hereinafter referred to as a "threshold value TH_L") in accordance with the detected maximum value MAX_L. The setting part 116E further converts the amplitude of the impulse response R' input from the measuring part 116B into sound pressure level (in dB), detects a maximum value of the sound pressure level (hereinafter referred to as a "maximum value MAX_R"), and sets a threshold value (hereinafter referred to as a "threshold value TH_R") in accordance with the detected maximum value MAX_R.

[0063] Fig. 8A is a graph of the impulse response L' input from the measuring part 116A in which the amplitude is converted into sound pressure level (in dB) and Fig. 8B is a graph of the impulse response R' input from the measuring part 116B in which the amplitude is converted into sound pressure level (in dB). In Figs. 8A and 8B, the vertical axis represents sound pressure level (unit: dB) and the horizontal axis represents time (unit: sec).

[0064] The setting part 116E detects the maximum value MAX_L of the amplitude of the impulse response L' input from the measuring part 116A and sets an amplitude being -30 dB from the detected maximum value MAX_L as the threshold value TH_L (see Fig. 8A). The setting part 116E further detects the maximum value MAX_R of the amplitude of the impulse response R' input from the measuring part 116B and sets an amplitude

being -30 dB from the detected maximum value MAX_R as the threshold value TH_R (see Fig. 8B).

[0065] As described above, the setting part 116E operates as a setting part for setting a threshold value of amplitude for each of the impulse responses between respective speakers and the predetermined position which are measured by the measuring parts 116A and 116B.

[0066] The following is the detailed description of Step S17. In the present embodiment, "a time between an output of a sound from the speaker and a timing at which a sound of practical sound pressure level is observed at the listening position" is "a time, from an output of a sound from a speaker, required for amplitude of an impulse response to reach the threshold value set at step S16 for the first time" (hereinafter referred to as a "reaching time"). The detecting part 116F detects a reaching time (hereinafter referred to as a "reaching time T_L"), required for the impulse response L' to reach the threshold value TH_L for the first time. The detecting part 116F further detects a reaching time (hereinafter referred to as a "reaching time T_R"), required for the impulse response R' to reach the threshold value TH_R for the first time.

[0067] Then, the detecting part 116F detects, among the two speakers, a speaker corresponding to the shorter reaching time. In the example shown in Fig. 8A, the reaching time T_L is 4.1 msec, and in the example shown in Fig. 8B, the reaching time T_R is 2.8 msec. Therefore, the detecting part 116F detects the speaker SP_{FR} as the speaker corresponding to the shorter reaching time.

[0068] As described above, the detecting part 116F operates as a detecting part for detecting, from among two speakers, a speaker corresponding to an impulse response of which a time, from an output of a sound from a speaker, required for amplitude to reach the threshold value is shorter.

[0069] It is noted that, if the maximum amplitude of an impulse response is set as the threshold value, the reaching time will vary widely due to noises. Therefore, in the present embodiment, an amplitude being -30 dB from the maximum amplitude of an impulse response is set as the threshold value. It becomes possible to detect a speaker corresponding to the shorter reaching time, that is, a speaker whose time between an output of a sound from the speaker and a timing at which a sound of practical sound pressure level is observed at the listening position is shorter, with high accuracy by setting the threshold value in accordance with the maximum value of amplitude of an impulse response. It is noted that -30 dB is a mere example. The threshold values may be set to arbitrary values provided the threshold values are not too close to the maximum value and a value around the beginning of the impulse response. In the examples shown in Figs. 8A and 8B, the value around the beginning of the impulse response is about -45 dB. Accordingly, the threshold values may be set to, for example,

a value between -40 dB to -20 dB.

[0070] Even if a distance from a speaker to a listening position is the same, if a position of the speaker and/or listening environment changes, the reaching time may change due to changes in influences of, for example, the reflection, masking and interference of sound. In the present embodiment, as shown in Figs. 8A and 8B, the reaching time of the impulse response R' corresponding to the speaker SP_{FR} being nearer to the listening position is shorter than the reaching time of the impulse response L' corresponding to the speaker SP_{FL} being farther from the listening position. However, if the positions of the speakers SP_{FL} and SP_{FR} and/or the listening environment are changed without changing the distances from the speakers SP_{FL} and SP_{FR} to the listening position, the reaching time of the impulse response R' may become longer than that the reaching time of the impulse response L'. For example, the reaching time of the impulse response R' may change to 5.0 msec but the reaching time of the impulse response L' may remain 4.1 msec (as shown in Fig. 8A). Even in this case, since the distance between the speaker SP_{FR} and the listening position is shorter than the distance between the speaker SP_{FL} and the listening position, the sound from the speaker SP_{FR} reaches the listening position faster than the sound from the speaker SP_{FL} (i.e., rise time of the impulse response R' is shorter than that of the impulse response L'). Therefore, in the conventional time alignment, the sound from the speaker SP_{FR} is delayed to suppress the image localization bias.

[0071] However, the sound pressure level of the rising part of the impulse response R' remains low for 5 msec. This low-level sound cannot be said to be a sound of practical sound pressure level that has substantial influence on the sound image localization. The impulse response R' reaches the practical sound pressure level at about 5.0 msec. On the other hand, as described above, the impulse response L' reaches the practical sound pressure level at about 4.1 msec.

[0072] That is, with the conventional time alignment, there are cases where a speaker whose time between an output of a sound from the speaker and a timing at which a sound of practical sound pressure level is observed at the listening position is longer is set as a speaker to which delay is to be given. Therefore, there are cases where, with the conventional time alignment, image localization bias is not sufficiently suppressed.

[0073] In contrast, in the present embodiment, the detecting part 116F detects not a speaker of which rise time of a corresponding impulse response is shorter (in other words, a speaker being nearer to the listening position) but a speaker whose time between an output of a sound from the speaker and a timing at which a sound of practical sound pressure level that has substantial influence on the sound image localization is observed at the listening position is shorter as a speaker whose phases of a signal to be input to are to be controlled to suppress the image localization bias (hereinafter

referred to as a "target speaker"). Therefore, even in special listening environments such as in the vehicle interior, the image localization bias can be sufficiently suppressed.

[0074] As shown in Fig. 4, the calculating part 116 includes a phase control part 116G.

[0075] The phase control part 116G sequentially shifts phases of the frequency spectrum R" corresponding to the speaker SP_{FR}, which has been detected by the detecting part 116F as a speaker corresponding to the short reaching time, within a range of -180 degrees to +180 degrees at predetermined angle intervals and synthesizes the frequency spectrum R" with the frequency spectrum L" corresponding to the other speaker (i.e., the speaker SP_{FL}) each time the phases of the frequency spectrum R" are shifted (step S18). This synthesizing process is executed not for every frequency points but for, for example, a total of 557 frequency points within a range of 0 Hz to 3 kHz in order to reduce processing load.

[0076] Fig. 9 is a diagram showing results of the synthesizing process at, among the 557 frequency points, 100Hz (thick solid line) and 400Hz (fine solid line). In Fig. 9, the vertical axis represents sound pressure level (unit: dB) after the synthesizing process and the horizontal axis represents phase (unit: degree). The sound pressure level at the phase of 0 degree (in other words, the sound pressure level when the phase adjustment amount is zero) indicates sound pressure level when the frequency spectrum R" is synthesized with the frequency spectrum L" without changing the phases of the frequency spectrum R".

[0077] When phases of a sound signal to be input to the speaker SP_{FR} is shifted by an angle at which the sound pressure level indicated in Fig. 9 is the greatest, phases of a sound from the speaker SP_{FR} and phases of a sound from the speaker SP_{FL} become the same (or substantially the same) at the listening position (i.e., the strongest constructive interference between the sound from the speaker SP_{FR} and the sound from the speaker SP_{FL} occurs at the listening position). When the phases of the sound signal to be input to the speaker SP_{FR} is shifted by an angle at which the sound pressure level indicated in Fig. 9 is the smallest, the phases of the sound from the speaker SP_{FR} and the phases of the sound from the speaker SP_{FL} become the opposite at the listening position (i.e., the strongest destructive interference between the sound from the speaker SP_{FR} and the sound from the speaker SP_{FL} occurs at the listening position).

[0078] At 100 Hz, the phase of the sound from the speaker SP_{FR} and the phase of the sound from the speaker SP_{FL} become the same (or substantially the same) at the listening position when the phases of the sound signal to be input to the speaker SP_{FR} is shifted by about +110 degrees and become the opposite at the listening position when the phases of the sound signal to be input to the speaker SP_{FR} is shifted by about -70 degrees. At 400 Hz, the phase of the sound from the speaker SP_{FR} and the phase of the

sound from the speaker SP_{FL} become the same (or substantially the same) at the listening position when the phases of the sound signal to be input to the speaker SP_{FR} is shifted by about -180 degrees and become the opposite at the listening position when the phases of the sound signal to be input to the speaker SP_{FR} is shifted by about +10 degrees.

[0079] The phase control part 116G obtains, from the result of the synthesizing process for each of the frequency points, the phase adjustment amount for each of the frequency points of the frequency spectrum R" for making the phases of the sound from the speaker SP_{FR} and the phases of the sound from the speaker SP_{FL} the same (or substantially the same) at the listening position.

[0080] Fig. 10 is a diagram showing the phase adjustment amounts at respective frequency points of the frequency spectrum R" when, at respective frequency points within the range of 60 Hz to 3 kHz, the phases of the sound from the speaker SP_{FR} and the phases of the sound from the speaker SP_{FL} become the same (or substantially the same) at the listening position. In Fig. 10, the vertical axis represents the phase adjustment amount (unit: degree) and the horizontal axis represents frequency (unit: Hz).

[0081] As shown in Fig. 10, in the present embodiment, the phase adjustment amount varies widely depending on frequency due to variations in propagation delay times at respective frequencies in the vehicle interior.

[0082] As shown in Fig. 4, the calculating part 116 includes a smoothing part 116H.

[0083] The smoothing part 116H performs smoothing on the phase adjustment amounts at respective frequency points, input from the phase control part 116G, on the frequency axis (step S19).

[0084] A set of the phase adjustment amounts at respective frequency points after the smoothing process is phase adjustment data for adjusting phases at respective frequency points of the sound signal to be input to the speaker SP_{FR}. That is, the phase control part 116G and the smoothing part 116H operate as a generating part for generating the phase adjustment data for adjusting phases at respective frequencies of the sound signal to be input to a speaker detected by the detecting part 116F to suppress destructive interferences between sounds from two speakers at the listening position at respective frequency points.

[0085] Fig. 11 is a diagram showing the phase adjustment amounts after the smoothing process by the smoothing part 116H. In Fig. 11, the vertical axis represents the phase adjustment amount (unit: degree) and the horizontal axis represents frequency (unit: Hz). The smoothing part 116H performs smoothing on the phase adjustment amounts at respective frequency points input from the phase control part 116G by using an FIR (Finite Impulse Response) filter of eight taps.

[0086] As can be seen by comparing Figs. 10 and 11, due to the smoothing process by the

smoothing part 116H, sudden change in phase in the frequency domain can be suppressed. Therefore, by outputting a sound signal whose phases are adjusted using the phase adjustment data, harmonics due to the sudden phase change can be suppressed and thus auditory allophones due to such harmonics can be suppressed.

[0087] The controller 100 sets the phase adjustment data generated by the phase control part 116G and the smoothing part 116H to the phase adjusting part 110 (step S20). The phase adjusting part 110 operates as a phase adjusting part for adjusting phases at respective frequencies of the sound signal input from a sound source and to be input to a specific speaker (i.e., a speaker detected by the detecting part 116F) by using the phase adjustment data.

[0088] Next, operations for reproducing a sound signal input from a sound source by using the phase adjustment data which has been set to the phase adjusting part 110 will be described.

[0089] The recording medium reproducing part 108 reproduces sound signals S_R and S_L (hereinafter referred to as "audio signals S_R and S_L") input from a sound source such as a CD (Compact Disc) or a DVD (Digital Versatile Disc). The controller 100 outputs the audio signals S_R and S_L reproduced by the recording medium reproducing part 108 to the phase adjusting part 110.

[0090] The phase adjusting part 110 adjusts phases at respective frequency points of an audio signal to be input to the target speaker and outputs the audio signal whose phases have been adjusted. The phase adjusting part 110 further outputs an audio signal to be input to the other speaker by through output without adjusting its phases. The audio signals S_R and S_L output from the phase adjusting part 110 are output from respective speakers SP_{FR} and SP_{FL} via the amplifier 112 into the vehicle interior.

[0091] Fig. 12 is a block chart showing a configuration of the phase adjusting part 110. As shown in Fig. 12, the phase adjusting part 110 includes an FFT (Fast Fourier Transform) part 110A, a complex multiplication part 110B and an IFFT (Inverse Fast Fourier Transform) part 110C.

[0092] The FFT part 110A performs an overlapping process and a weighting by use of a window function on an audio signal input from the controller 100 (specifically, an audio signal to be input to the target speaker), converts the audio signal from the time domain to the frequency domain by the STFT (Short-Term Fourier Transform), and outputs a signal of a frequency spectrum consisting of a frequency characteristic of amplitude and a frequency characteristic of phase to the complex multiplication part 110B. The overlapping process and the window function are for reducing noise when the localization position is changed by a user operation (e.g., when the localization position is changed from the driver seat to the front passenger seat) by gradually changing the phase adjustment amount.

[0093] In the present embodiment, the FFT part 110A has a sampling frequency of 44.1 kHz, a Fourier transform length of 8,192 samples, an overlap length of 6,144 samples, and a window function being Hanning. The FFT part 110A obtains a frequency spectrum consisting of 4,097 points, the 4,097 points being obtained by dividing a frequency range from 0 Hz up to 22.05 kHz being Nyquist frequency at 5.38 Hz intervals, by performing STFT while shifting the time by 2,048 samples.

[0094] The phase adjustment data input from the controller 100 is set to the complex multiplication part 110B. The complex multiplication part 110B adjusts phases at respective frequency points of an audio signal to be input to a channel corresponding to the target speaker by performing complex multiplication on a signal of the frequency spectrum input from the FFT part 110A based on the phase adjustment data.

[0095] The IFFT part 110C converts a signal of the frequency spectrum, input from the complex multiplication part 110B and whose phases have been adjusted, from the frequency domain to the time domain by ISTFT (Inverse Short-Term Fourier Transform), performs weighting by a window function and an overlapping addition on the converted signal, and output the processed signal to the amplifier 112. An audio signal obtained after the overlapping addition is a signal whose phases have been adjusted for respective frequency points in accordance with the phase adjustment data such that phases of a sound output from the target speaker and phases of a sound output from the other speaker become the same (or substantially the same) at respective frequency points at the listening position

[0096] A specific example of the phase adjustment of an audio signal to be input to the channel corresponding to the target speaker will be described with reference to Figs. 13A-13C, 14 and 15. In the example, the audio signal is a monaural impulse signal with a frequency range of 0 Hz to 3 kHz.

[0097] Fig. 13A is a diagram showing a time characteristic of a sound observed at the listening position when the audio signal on which neither the conventional time alignment process (provision of time delay) nor the phase adjustment process according to the present embodiment are performed is output from respective speakers SP_{FR} and SP_{FL} at the same time. Fig. 13B is a diagram showing a time characteristic of a sound observed at the listening position when the conventional time alignment process is performed, that is, when a time delay is provided to the audio signal to be input to the speaker SP_{FR} such that sounds output from respective speakers SP_{FR} and SP_{FL} reach the listening position at the same time or substantially at the same time (i.e., such that rise times of impulse responses corresponding to respective speakers SP_{FR} and SP_{FL} become the same or substantially the same). Fig. 13C is a diagram showing a time characteristic of a sound observed at the listening position when the phase adjustment process according to the present embodiment is performed on the audio signal

to be input to the speaker SP_{FR}. In Figs. 13A-13C, the vertical axis represents amplitude and the horizontal axis represents time (unit: sec).

[0098] For the sake of convenience, the example shown in Fig. 13A will be hereinafter referred to as an "example without adjustment", the example shown in Fig. 13B will be hereinafter referred to as an "example with time alignment", and the example shown in Fig. 13C will be hereinafter referred to as an "example with phase adjustment". As can be seen from Figs. 13A-13C, amplitudes of the example with time alignment and the example with phase adjustment are greater than the amplitude of the example without adjustment. Furthermore, the amplitude of the example with phase adjustment is generally greater than the amplitude of the example with time alignment. This is because the destructive interference between the sound from the speaker SP_{FR} and the sound from the speaker SP_{FL} at the listening position (i.e., at the driver seat) is suppressed by performing the phase adjustment process according to the present embodiment on the audio signal.

[0099] Fig. 14 is a diagram showing a frequency characteristic of amplitude of the sound observed at the listening position in the example without adjustment (fine solid line) and a frequency characteristic of amplitude of the sound observed at the listening position in the example with time alignment (thick solid line). Fig. 15 is a diagram showing the frequency characteristic of amplitude of the sound observed at the listening position in the example without adjustment (fine solid line) and a frequency characteristic of amplitude of the sound observed at the listening position in the example with phase adjustment (thick solid line). In Figs. 14 and 15, the vertical axis represents sound pressure level (unit: dB) and the horizontal axis represents frequency (unit: Hz).

[0100] As shown in Fig. 14, the example with time alignment includes frequencies at which the sound pressure levels are greater than that of the example without adjustment and frequencies at which the sound pressure levels are lower than that of the example without adjustment. That is, in the example with time alignment, the destructive interference between the sound from the speaker SP_{FR} and the sound from the speaker SP_{FL} at the listening position is not sufficiently suppressed. Therefore, in the special listening environment in the vehicle interior, the image localization bias cannot be sufficiently suppressed by the time alignment process. Furthermore, since, in the example with time alignment, there are local drops in sound pressure levels in the frequency domain (i.e., occurrence of dips), degradation in sound quality and decrease in sound pressure due to the interferences between sounds output from the speakers SP_{FR} and SP_{FL} cannot be sufficiently suppressed.

[0101] In contrast, as shown in Fig. 15, in the example with phase adjustment according to the present embodiment, the sound pressure level is greater than that in the example

without adjustment across the entire frequency range (especially in the frequency range between 20 Hz to 750 Hz which is a range that has great influence on the image localization and in the frequency range between 750 Hz to 1.5 kHz which is a range that has relatively large influence on the image localization). That is, with the example with phase adjustment according to the present embodiment, the destructive interference between the sound from the speaker SP_{FR} and the sound from the speaker SP_{FL} at the listening position is suppressed as compared to the example with time alignment. Furthermore, in the example with phase adjustment according to the present embodiment, there are substantially no occurrence of dip in the frequency domain. Therefore, with the example with phase adjustment according to the present embodiment, the image localization bias and the degradation in sound quality and decrease in sound pressure at the listening position due to the interferences between sounds output from the speakers SP_{FR} and SP_{FL} can be suppressed as compared to the example with time alignment.

[0102] The foregoing is explanation of an exemplary embodiment of the present invention. The present invention is not limited to the above described embodiment, but can be varied in various ways within the scope of the invention. For example, embodiments of the present invention include appropriate combinations of embodiments explicitly described in this specification and/or embodiments that are obvious from the above described embodiment.

[0103] In the above-described embodiment, the acoustic device 10 has the phase adjustment data generating function (phase control device) for generating phase adjustment data. However, an acoustic device without the phase adjustment data generating function is also within the scope of the present invention. For example, such acoustic device has phase adjustment data generated in a manufacturing facility or the like and stored in the acoustic device in advance (e.g., before the shipment). This acoustic device is configured to suppress the destructive interference between sounds from two speakers at the listening position at respective frequency points by adjusting phases at respective frequency points of a sound signal input from a sound source and to be input to a specific speaker (i.e., a speaker, among two speakers, whose time from the output of a sound until a sound whose amplitude is greater than a predetermined threshold value is observed at the listening position is shorter) with use of the stored phase adjustment data.

[0104] In another embodiment, the phase adjusting part 110 may perform Fourier transform on the phase adjustment amounts at respective frequency points to generate FIR filter coefficients and may perform the phase adjustment for respective frequency points on the audio signal to be input to the target speaker by a process in the time domain with use of the FIR filter. The phase adjusting part 110 may also perform a dividing process on each frequency point in accordance with the phase adjustment amount for each

frequency point and may perform the phase adjustment on the audio signal to be input to the target speaker by using an all-pass filter which uses a second-order IIR (Infinite Impulse Response) filter or the like.

[0105] In the above-described embodiment, a process for the case where the listening position is the driver seat has been described. However, similar process may be performed for each seat. In this case, the controller 100 may have pieces of phase adjustment data for respective seats as pieces of preset data. A listener can switch the target seat where the image localization bias and the like are to be suppressed by selecting one of the pieces of preset data through operation of the operating part 104.

[0106] In the above embodiment, a case where two front speakers are provided to a vehicle has been described. However, if, for example, there are two rear speakers in addition to the two front speakers, similar process as that of the above-described embodiment may be performed for the rear speakers.

Claims

[Claim 1] A phase control device comprising:
a measuring part configured to measure impulse responses between a predetermined position and respective one of two speakers from signals of sounds collected at the predetermined position, the sounds being output from respective one of the two speakers at timings such that the sounds output from respective one of the two speakers do not interfere with each other at the predetermined position;
a setting part configured to set a threshold value of amplitude for each of the measured impulse responses;
a detecting part configured to detect, from among the two speakers, a speaker whose time, from the output of the sound, required for an amplitude of the corresponding impulse response to reach the threshold value is shorter; and
a generating part configured to generate phase adjustment data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker to suppress destructive interference between sounds output from the two speakers at respective frequencies at the predetermined position.

[Claim 2] The phase control device of claim 1,
wherein the generating part:
sequentially shifts phases of a frequency spectrum of the impulse response corresponding to the detected speaker for a predetermined angle and synthesizes the impulse response corresponding to the detected speaker with a frequency spectrum of the impulse response corresponding to the other speaker each time the phases of the frequency spectrum of the impulse response corresponding to the detected speaker are shifted; and
obtains the phase adjustment data by obtaining phase adjustment amounts at respective frequencies of the frequency spectrum of the impulse response corresponding to the detected speaker based on the result of the synthesis.

[Claim 3] The phase control device of claim 2,
wherein the generating part performs smoothing on the obtained phase adjustment amounts at respective frequencies on the frequency axis and obtains the smoothed phase adjustment amounts at respective frequencies as the phase adjustment data.

[Claim 4] The phase control device of any one of claims 1 to 3, wherein the phase adjustment data is data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker such that, at respective frequencies, phases of a sound output by the detected speaker and phases of a sound output by the other speaker become substantially the same at the predetermined position.

[Claim 5] The phase control device of any one of claims 1 to 4, wherein the setting part sets, for each of the impulse responses, the threshold value in accordance with a maximum amplitude of the impulse response.

[Claim 6] An acoustic device comprising the phase control device of any one of claims 1-5 and configured to output a sound signal input from a sound source to the two speakers,
the acoustic device comprising an adjusting part configured to adjust, by using the phase adjustment data, phases at respective frequencies of the sound signal input from the sound source and to be input to the detected speaker.

[Claim 7] An acoustic device configured to input sound signals input from a sound source to respective one of the two speakers,
the acoustic device comprising an adjusting part configured to adjust phases at respective frequencies of the sound signal input from the sound source and to be input to a specific speaker among the two speakers such that, at respective frequencies, destructive interference between sounds output by respective one of the two speakers at a predetermined position is reduced,
wherein the specific speaker is a speaker, among the two speakers, whose time, from an output of a sound from the speaker, required for an amplitude of the sound observed at the predetermined position to reach a predetermined threshold value is shorter.

[Claim 8] A phase control method to be executed by a computer, including:
a measuring step of measuring impulse responses between a predetermined position and respective one of two speakers from signals of sounds collected at the predetermined position, the sounds being output from respective one of the two speakers at timings such that the sounds output from respective one of the two speakers do not interfere with each other at the predetermined position;
a setting step of setting a threshold value of amplitude for each of the measured impulse responses;

a detecting of detecting, from among the two speakers, a speaker whose time, from the output of the sound, required for an amplitude of the corresponding impulse response to reach the threshold value is shorter; and

a generating step of generating phase adjustment data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker to suppress destructive interference between sounds output from the two speakers at respective frequencies at the predetermined position.

[Claim 9] The phase control method of claim 8,

wherein, in the generating step:

phases of a frequency spectrum of the impulse response corresponding to the detected speaker are sequentially shifted for a predetermined angle and the impulse response corresponding to the detected speaker is synthesized with a frequency spectrum of the impulse response corresponding to the other speaker each time the phases of the frequency spectrum of the impulse response corresponding to the detected speaker are shifted; and

the phase adjustment data is obtained by obtaining phase adjustment amounts at respective frequencies of the frequency spectrum of the impulse response corresponding to the detected speaker based on the result of the synthesis.

[Claim 10] The phase control method of claim 9,

wherein, in the generating step, smoothing is performed on the obtained phase adjustment amounts at respective frequencies on the frequency axis and the smoothed phase adjustment amounts at respective frequencies is obtained as the phase adjustment data.

[Claim 11] The phase control method of any one of claims 8 to 10,

wherein the phase adjustment data is data for adjusting phases at respective frequencies of a sound signal to be input to the detected speaker such that, at respective frequencies, phases of a sound output by the detected speaker and phases of a sound output by the other speaker become substantially the same at the predetermined position.

[Claim 12] The phase control method of any one of claims 8 to 11,

wherein, in the setting step, for each of the impulse responses, the threshold value is set in accordance with a maximum amplitude of the impulse response.

[Claim 13] The phase control method of any one of claims 8 to 12,

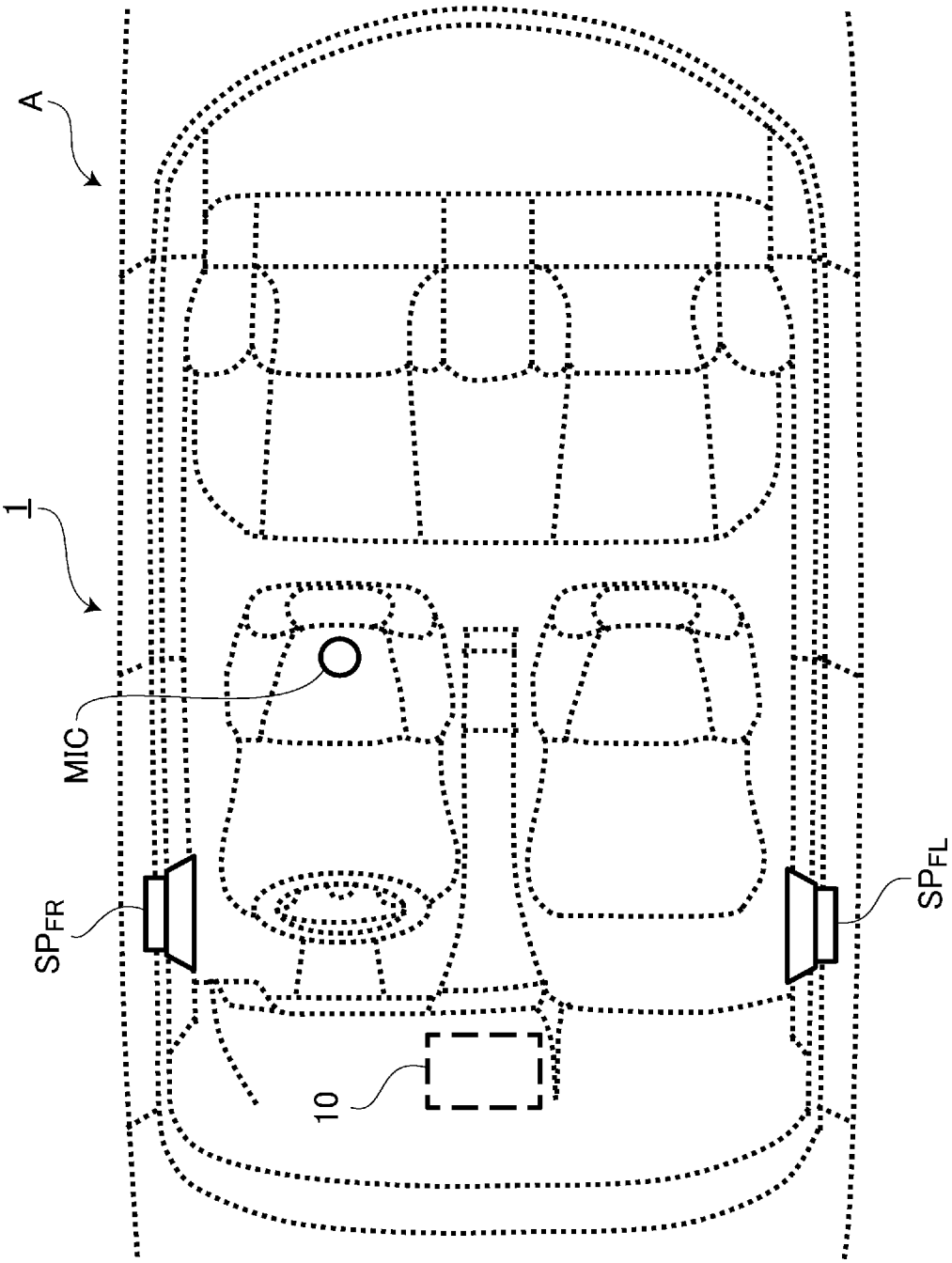
further including, when inputting a sound signal input from a sound source to the two speakers, an adjusting step of adjusting, by using the phase adjustment data, phases at respective frequencies of the sound signal input from the sound source and to be input to the detected speaker.

[Claim 14] A phase control method to be executed by an acoustic device configured to input a sound signal input from a sound source to two speakers,

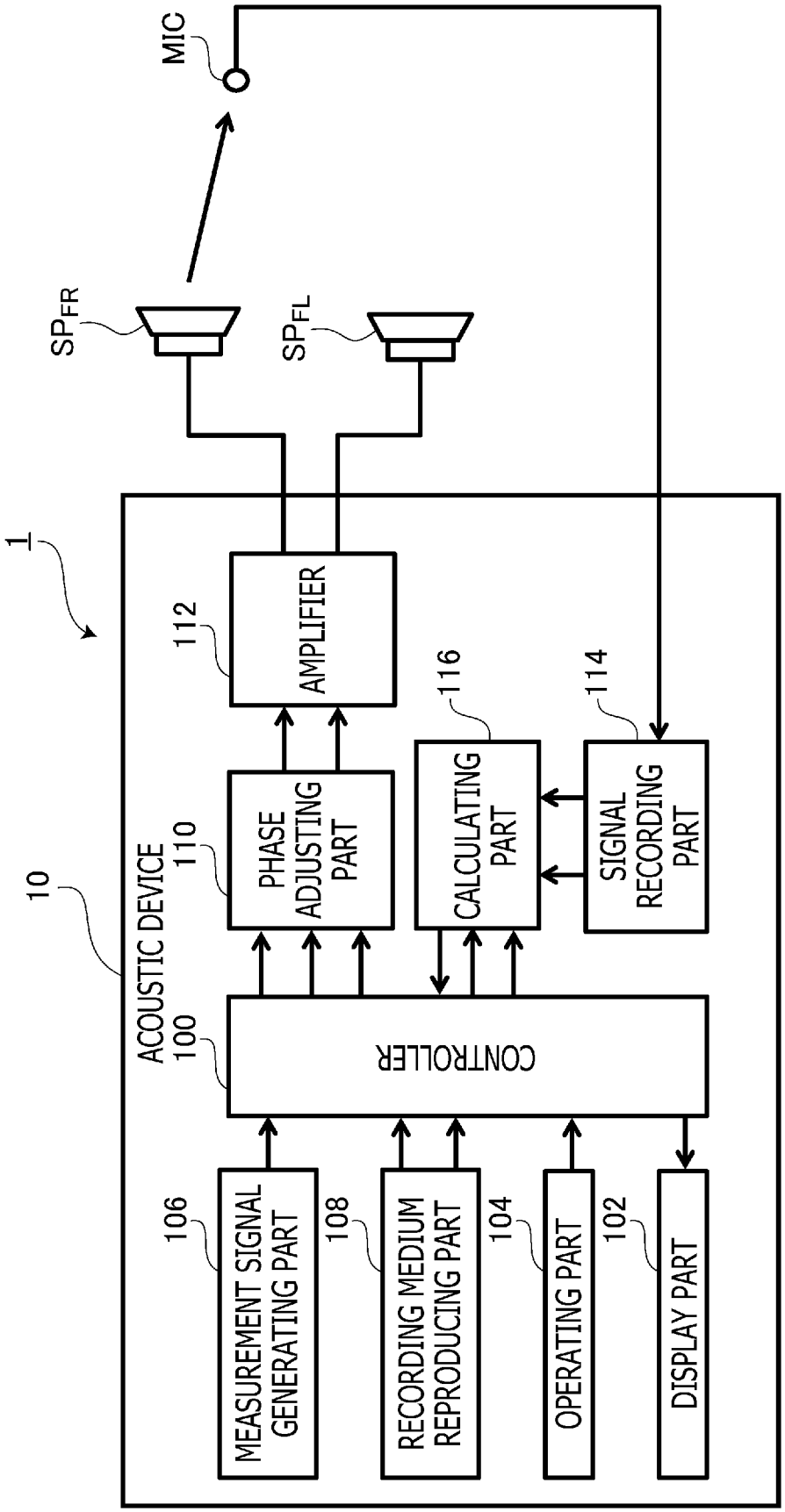
the method including an adjusting step of adjusting phases at respective frequencies of the sound signal input from the sound source and to be input to a specific speaker among the two speakers such that, at respective frequencies, destructive interference between sounds output by respective one of the two speakers at a predetermined position is reduced,

wherein the specific speaker is a speaker, among the two speakers, whose time, from an output of a sound from the speaker, required for an amplitude of the sound observed at the predetermined position to reach a predetermined threshold value is shorter.

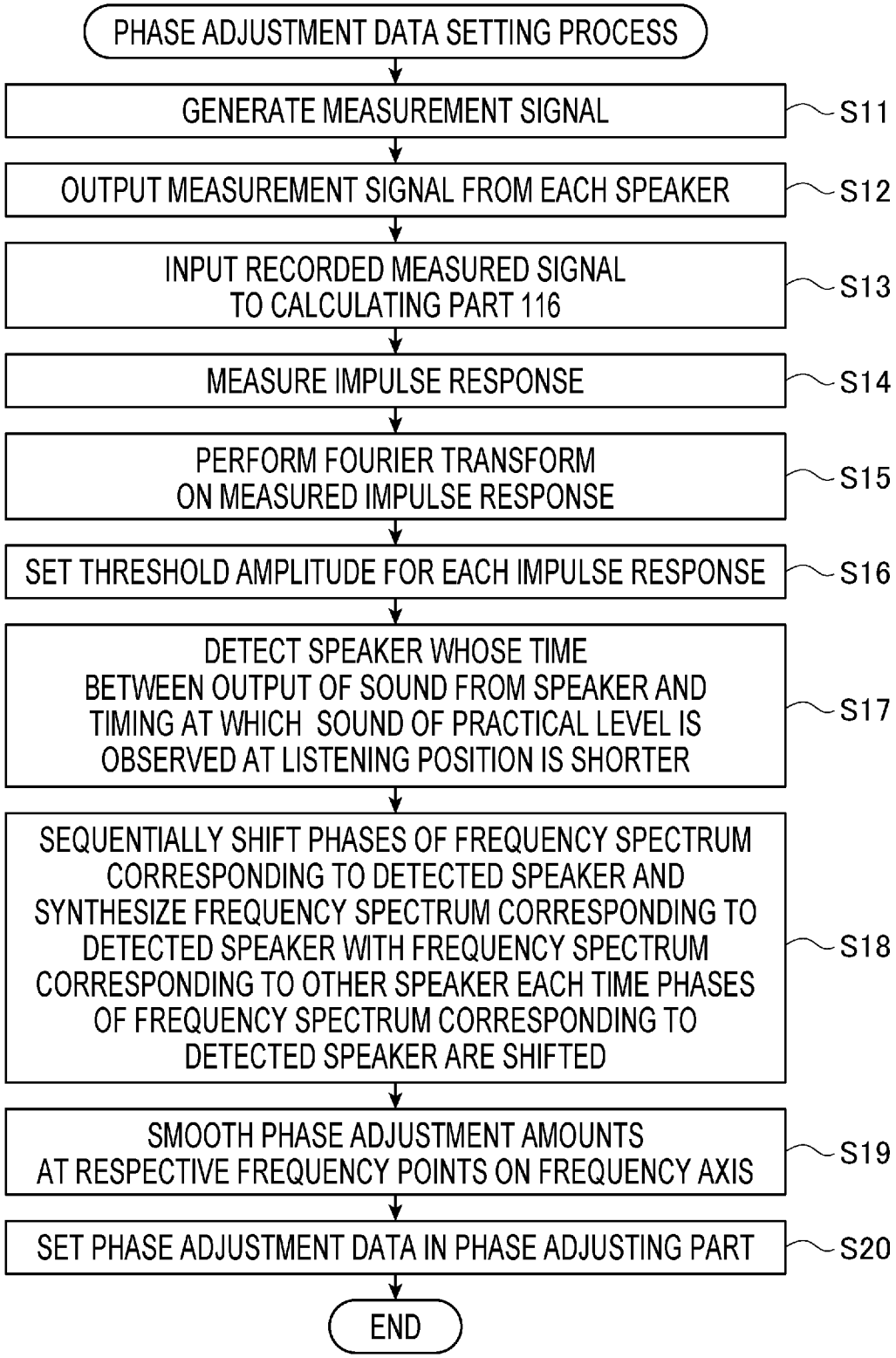
[Fig. 1]



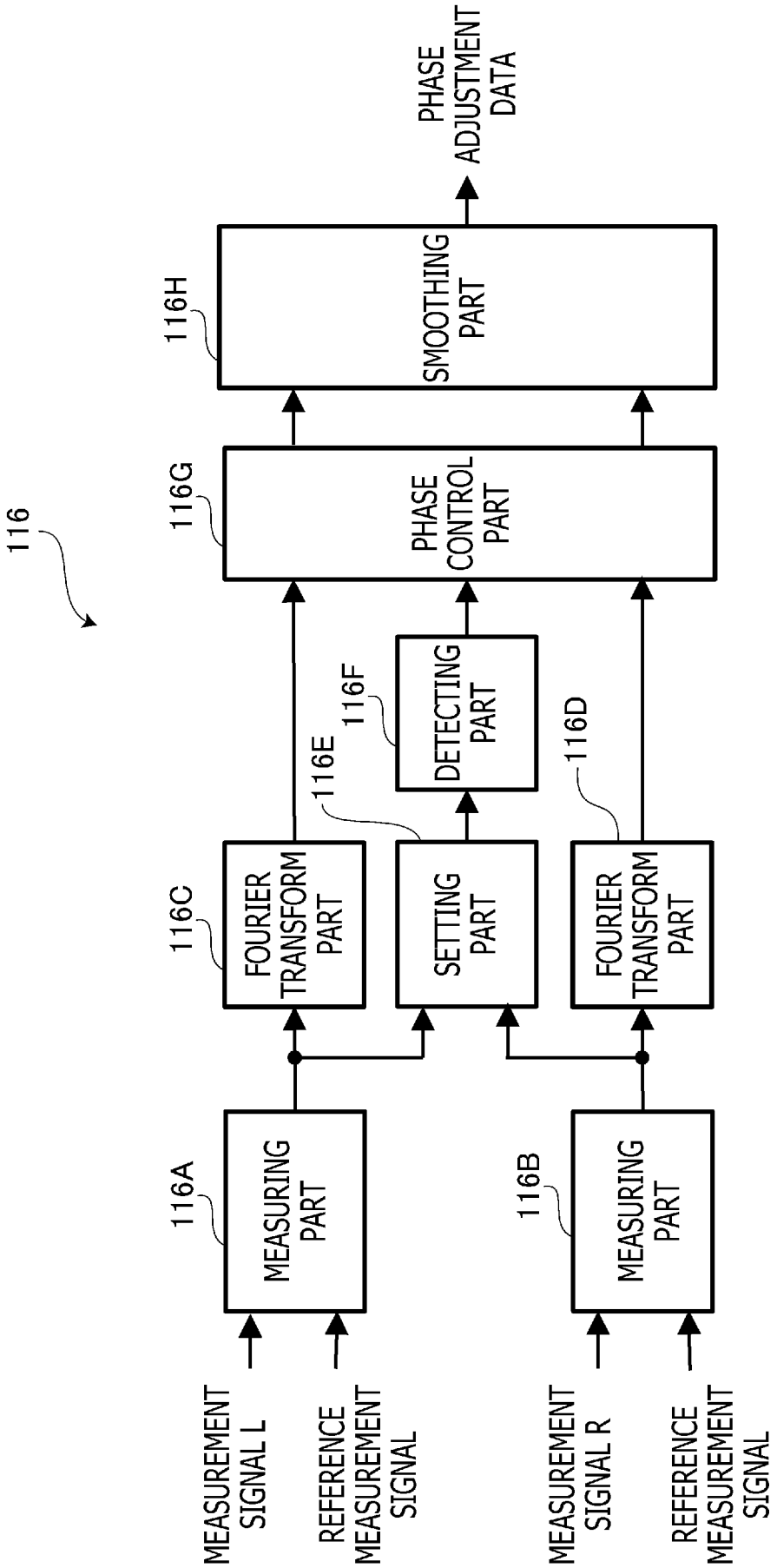
[Fig. 2]


MIC
SPFR
SPFL
1
10
ACOUSTIC DEVICE
100
CONTROLLER
112
AMPLIFIER
110
PHASE ADJUSTING PART
116
CALCULATING PART
114
SIGNAL RECORDING PART
106
MEASUREMENT SIGNAL GENERATING PART
108
RECORDING MEDIUM REPRODUCING PART
104
OPERATING PART
102
DISPLAY PART

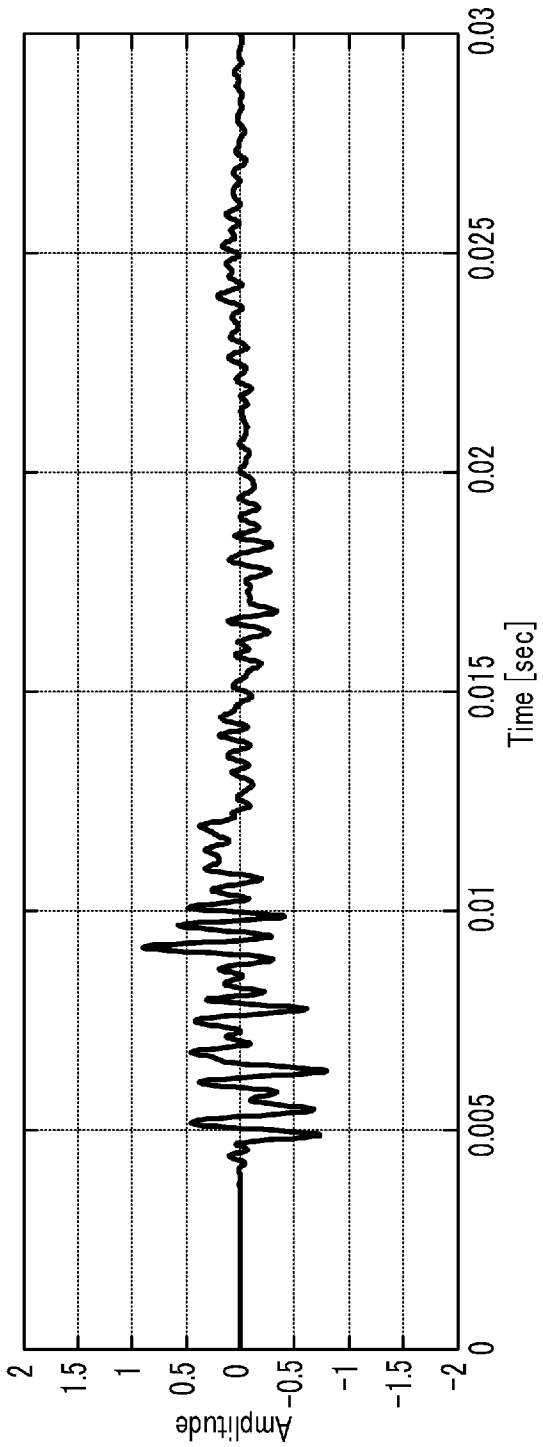
[Fig. 3]


PHASE ADJUSTMENT DATA SETTING PROCESS
GENERATE MEASUREMENT SIGNAL
S11
OUTPUT MEASUREMENT SIGNAL FROM EACH SPEAKER
S12
INPUT RECORDED MEASURED SIGNAL TO CALCULATING PART 116
S13
MEASURE IMPULSE RESPONSE
S14
PERFORM FOURIER TRANSFORM ON MEASURED IMPULSE RESPONSE
S15
SET THRESHOLD AMPLITUDE FOR EACH IMPULSE RESPONSE
S16
DETECT SPEAKER WHOSE TIME BETWEEN OUTPUT OF SOUND FROM SPEAKER AND TIMING AT WHICH SOUND OF PRACTICAL LEVEL IS OBSERVED AT LISTENING POSITION IS SHORTER
S17
SEQUENTIALLY SHIFT PHASES OF FREQUENCY SPECTRUM CORRESPONDING TO DETECTED SPEAKER AND SYNTHESIZE FREQUENCY SPECTRUM CORRESPONDING TO DETECTED SPEAKER WITH FREQUENCY SPECTRUM CORRESPONDING TO OTHER SPEAKER EACH TIME PHASES OF FREQUENCY SPECTRUM CORRESPONDING TO DETECTED SPEAKER ARE SHIFTED
S18
SMOOTH PHASE ADJUSTMENT AMOUNTS AT RESPECTIVE FREQUENCY POINTS ON FREQUENCY AXIS
S19
SET PHASE ADJUSTMENT DATA IN PHASE ADJUSTING PART
S20
END

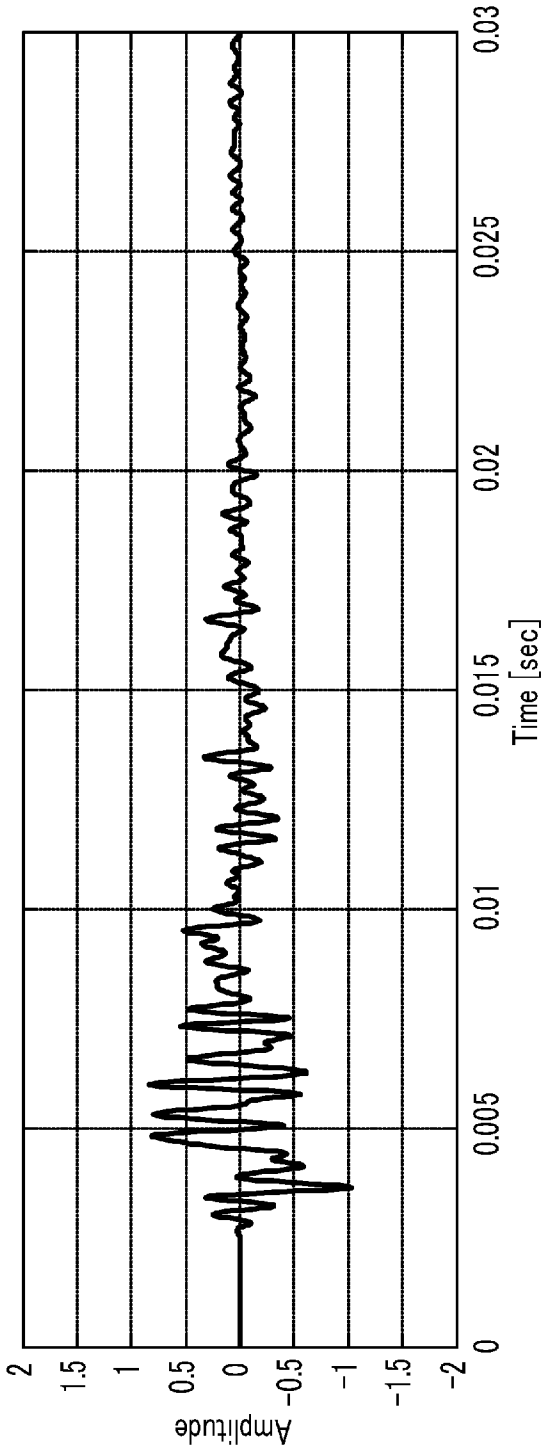
[Fig. 4]


116
PHASE ADJUSTMENT DATA
116H
SMOOTHING PART
116G
PHASE CONTROL PART
116F
DETECTING PART
116E
116D
116C
FOURIER TRANSFORM PART
SETTING PART
FOURIER TRANSFORM PART
116A
MEASURING PART
116B
MEASURING PART
MEASUREMENT SIGNAL L
REFERENCE MEASUREMENT SIGNAL
MEASUREMENT SIGNAL R
REFERENCE MEASUREMENT SIGNAL

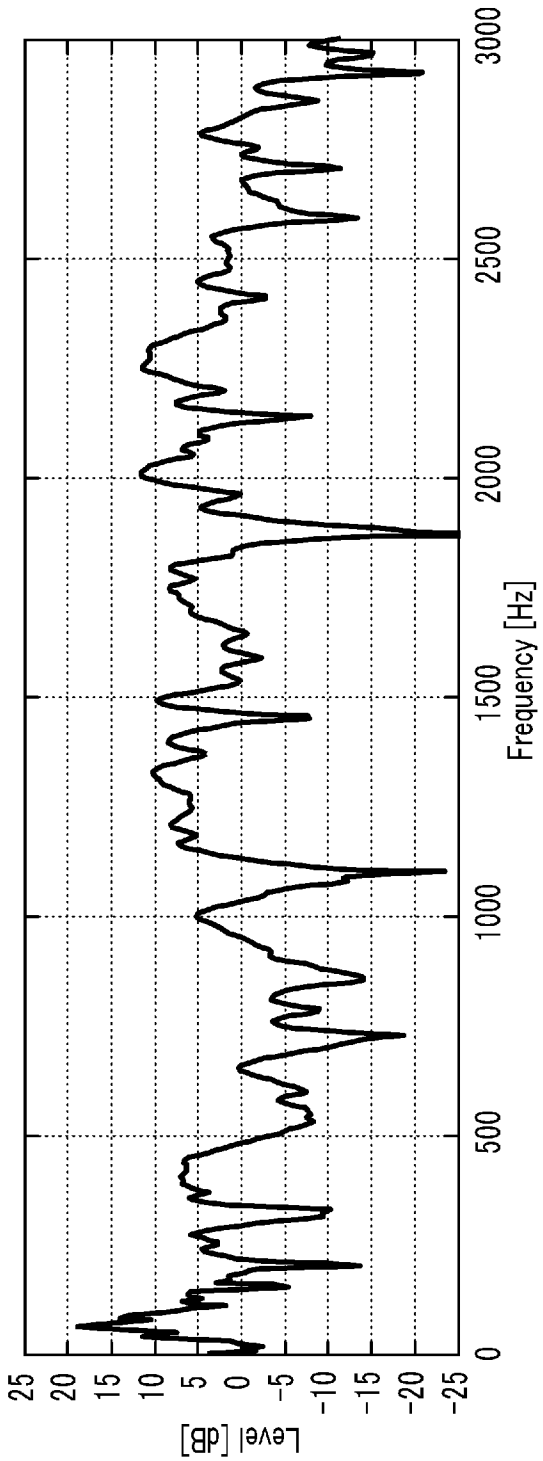
[Fig. 5A]


Amplitude
2
1.5
1
0.5
0
−0.5
−1
−1.5
−2
0
0.005
0.01
0.015
0.02
0.025
0.03
Time [sec]

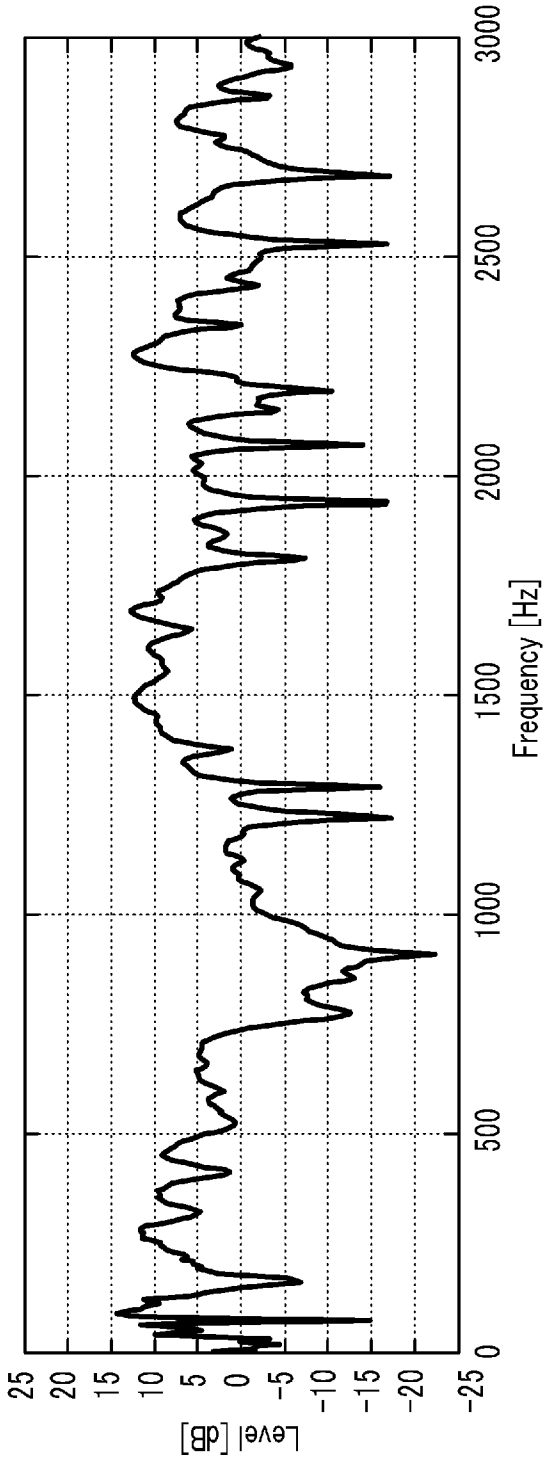
[Fig. 5B]


Amplitude
2
1.5
1
0.5
0
−0.5
−1
−1.5
−2
0
0.005
0.01
0.015
0.02
0.025
0.03
Time [sec]

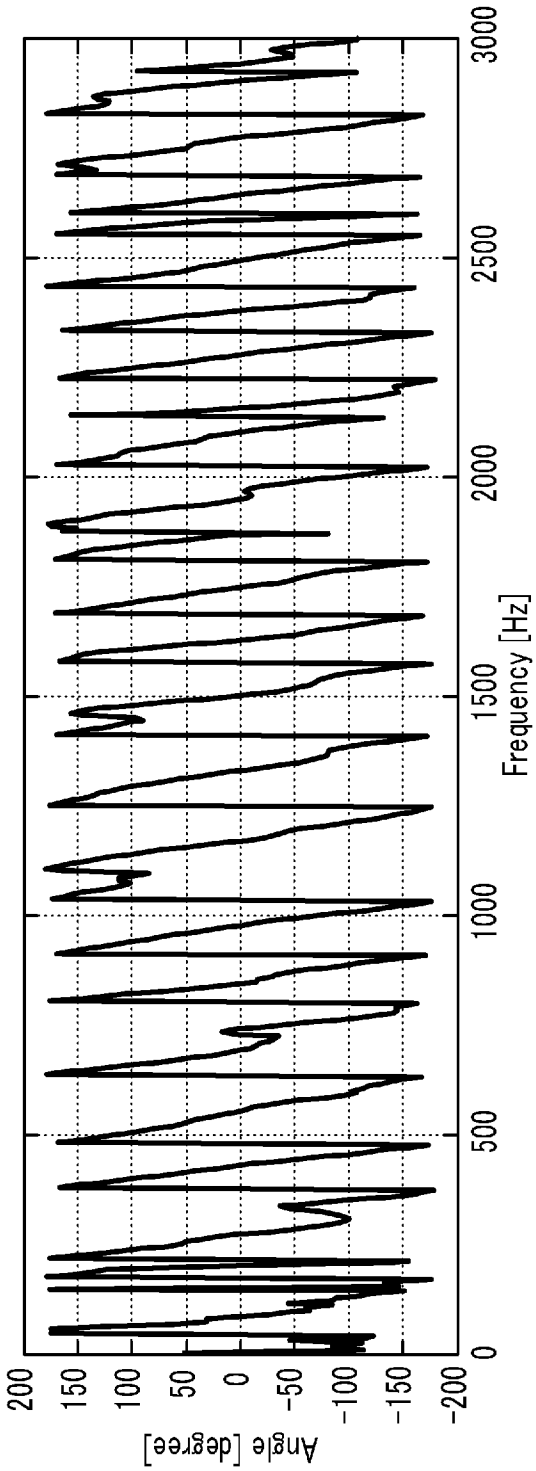
[Fig. 6A]


Level [dB]
25
20
15
10
5
0
-5
-10
-15
-20
-25
0
500
1000
1500
2000
2500
3000
Frequency [Hz]

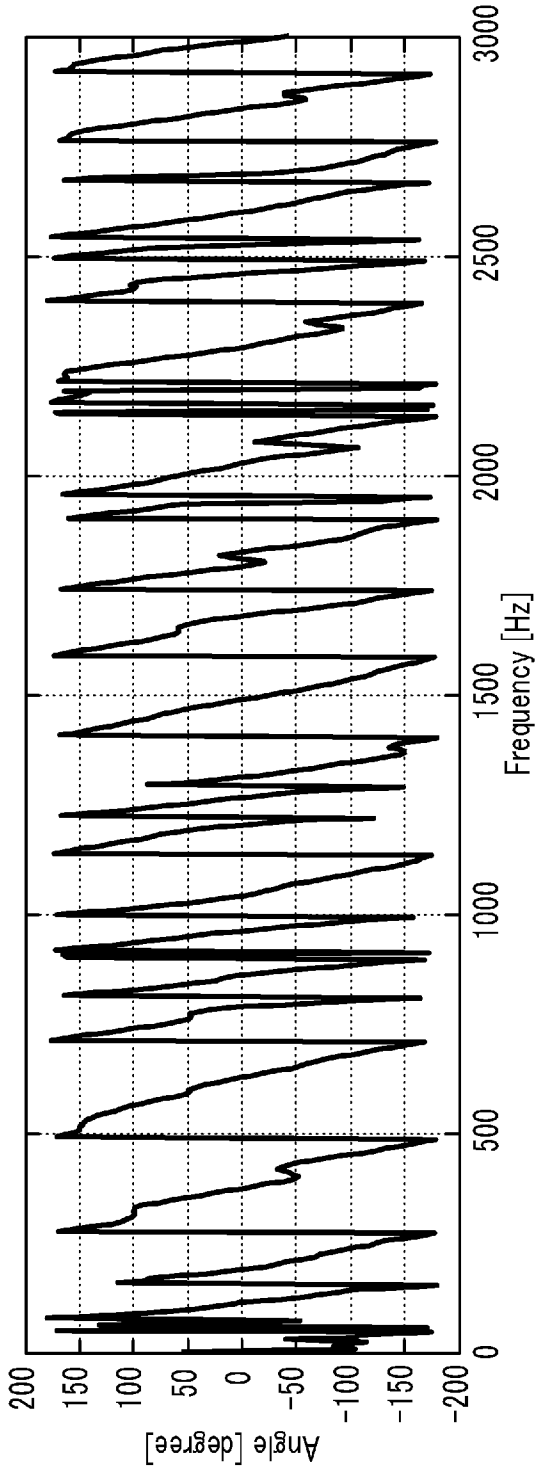
[Fig. 6B]


Level [dB]
25
20
15
10
5
0
-5
-10
-15
-20
-25
0
500
1000
1500
2000
2500
3000
Frequency [Hz]

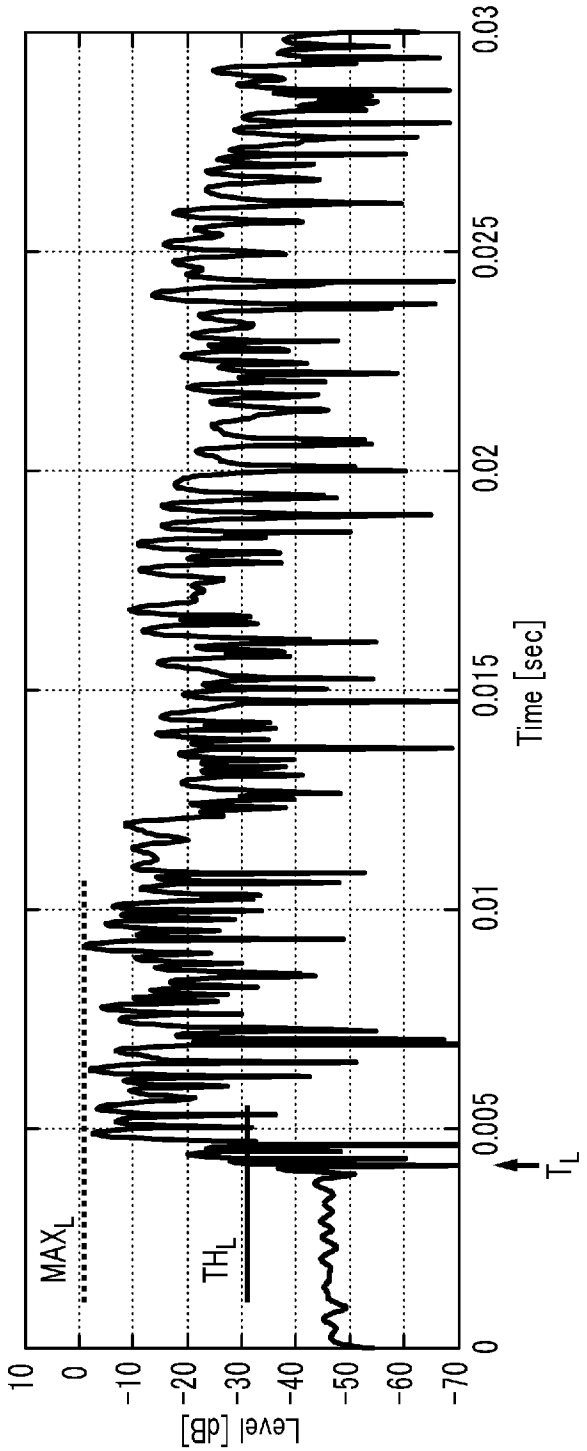
[Fig. 7A]


Angle [degree]
200
150
100
50
0
-50
-100
-150
-200
0
500
1000
1500
2000
2500
3000
Frequency [Hz]

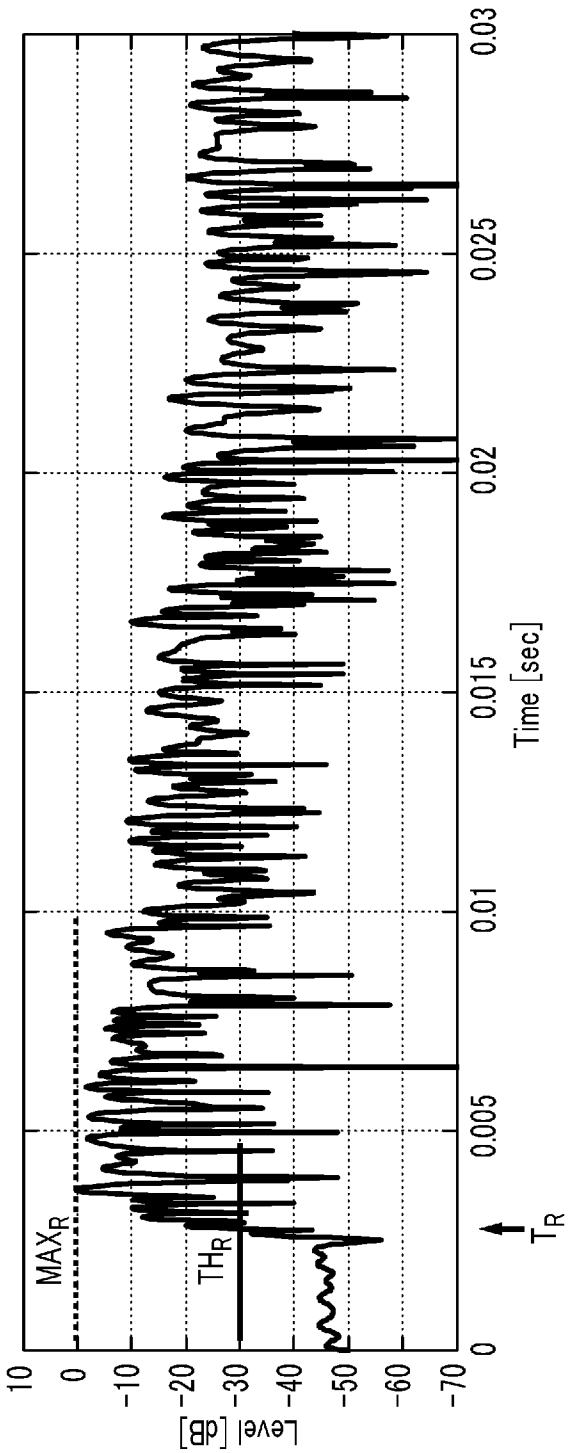
[Fig. 7B]


Angle [degree]
200
150
100
50
0
-50
-100
-150
-200
Frequency [Hz]
0
500
1000
1500
2000
2500
3000

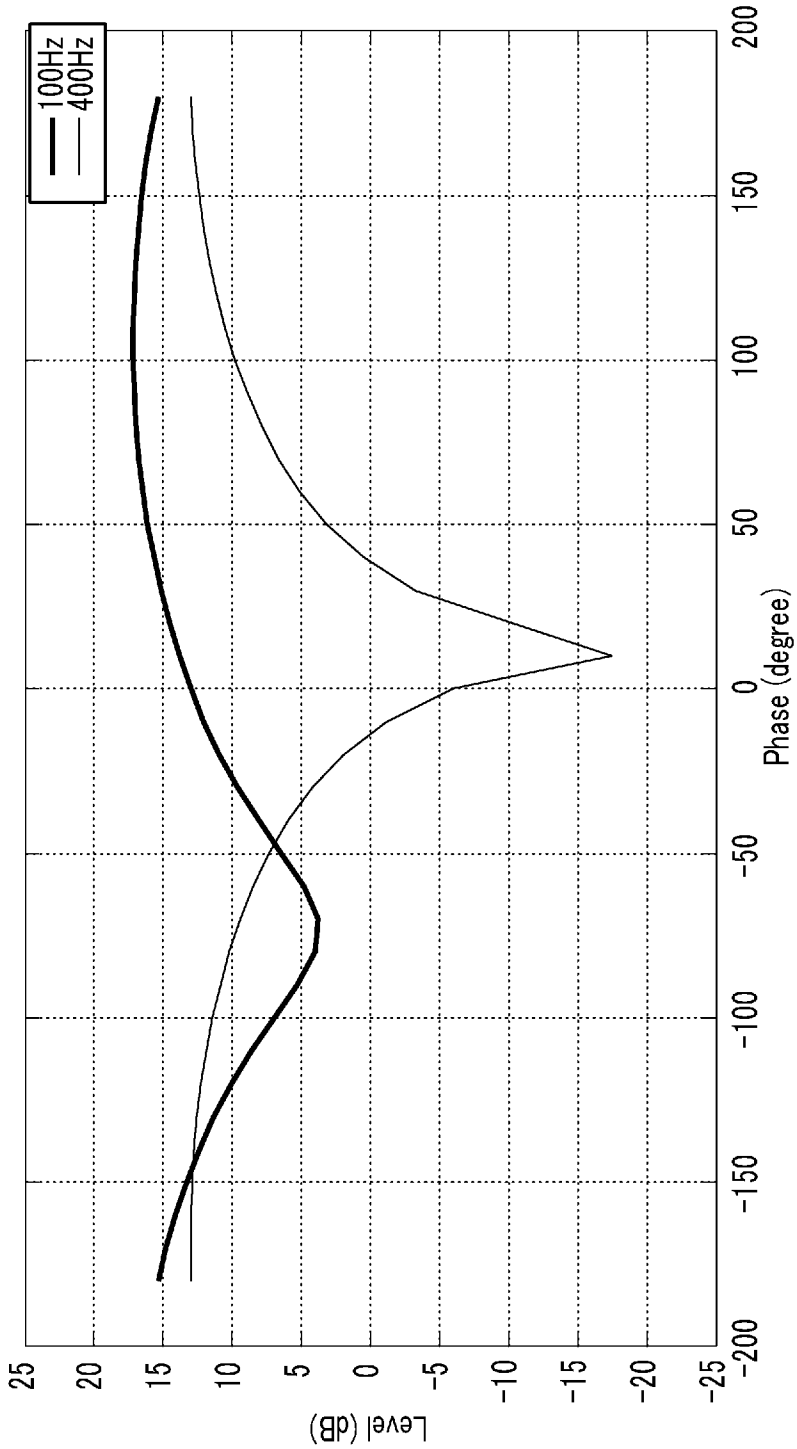
[Fig. 8A]


Level [dB]
10
0
-10
-20
-30
-40
-50
-60
-70
0
0.005
0.01
0.015
0.02
0.025
0.03
Time [sec]
MAX_L
TH_L
T_L

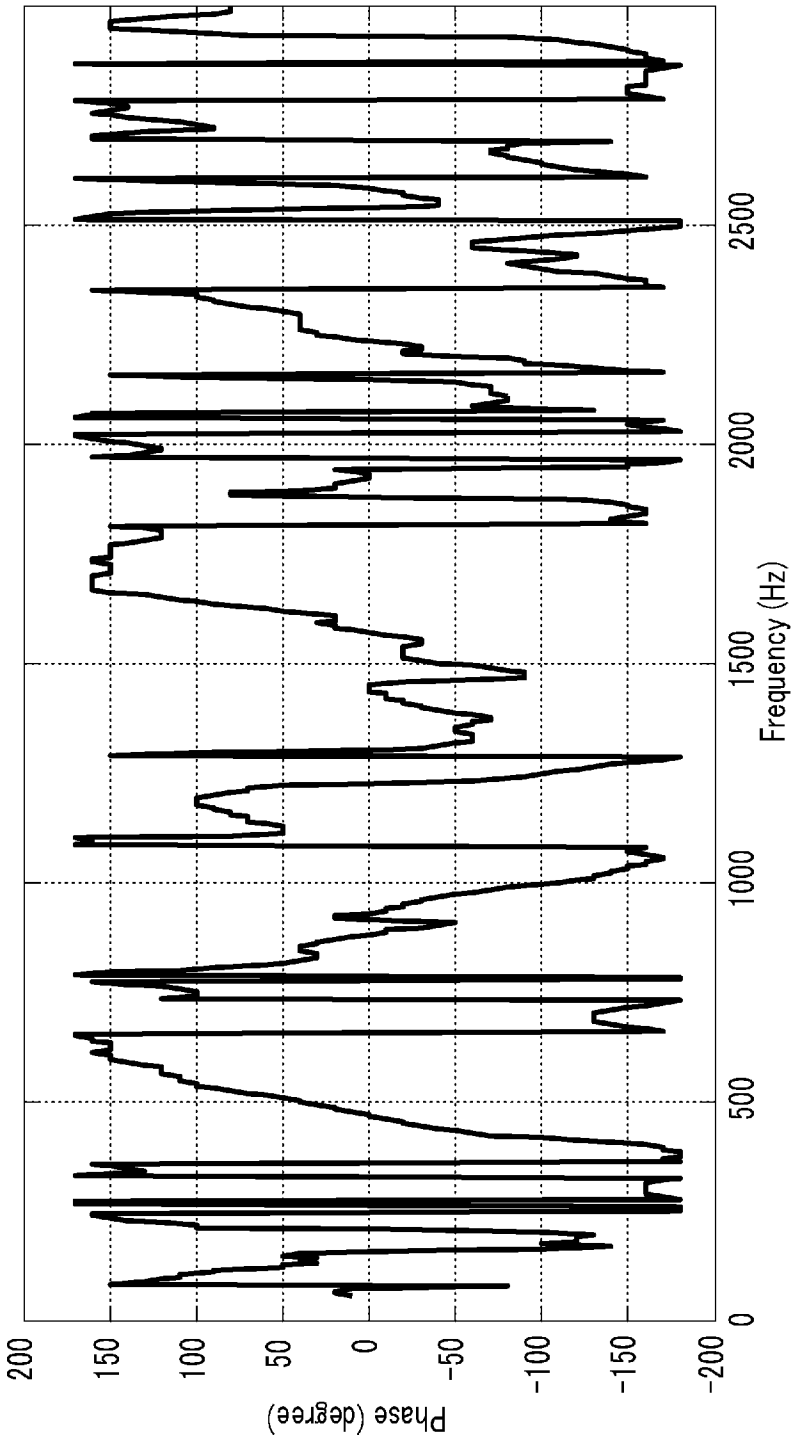
[Fig. 8B]


MAX_R
TH_R
T_R
Level [dB]
10
0
-10
-20
-30
-40
-50
-60
-70
0
0.005
0.01
0.015
0.02
0.025
0.03
Time [sec]

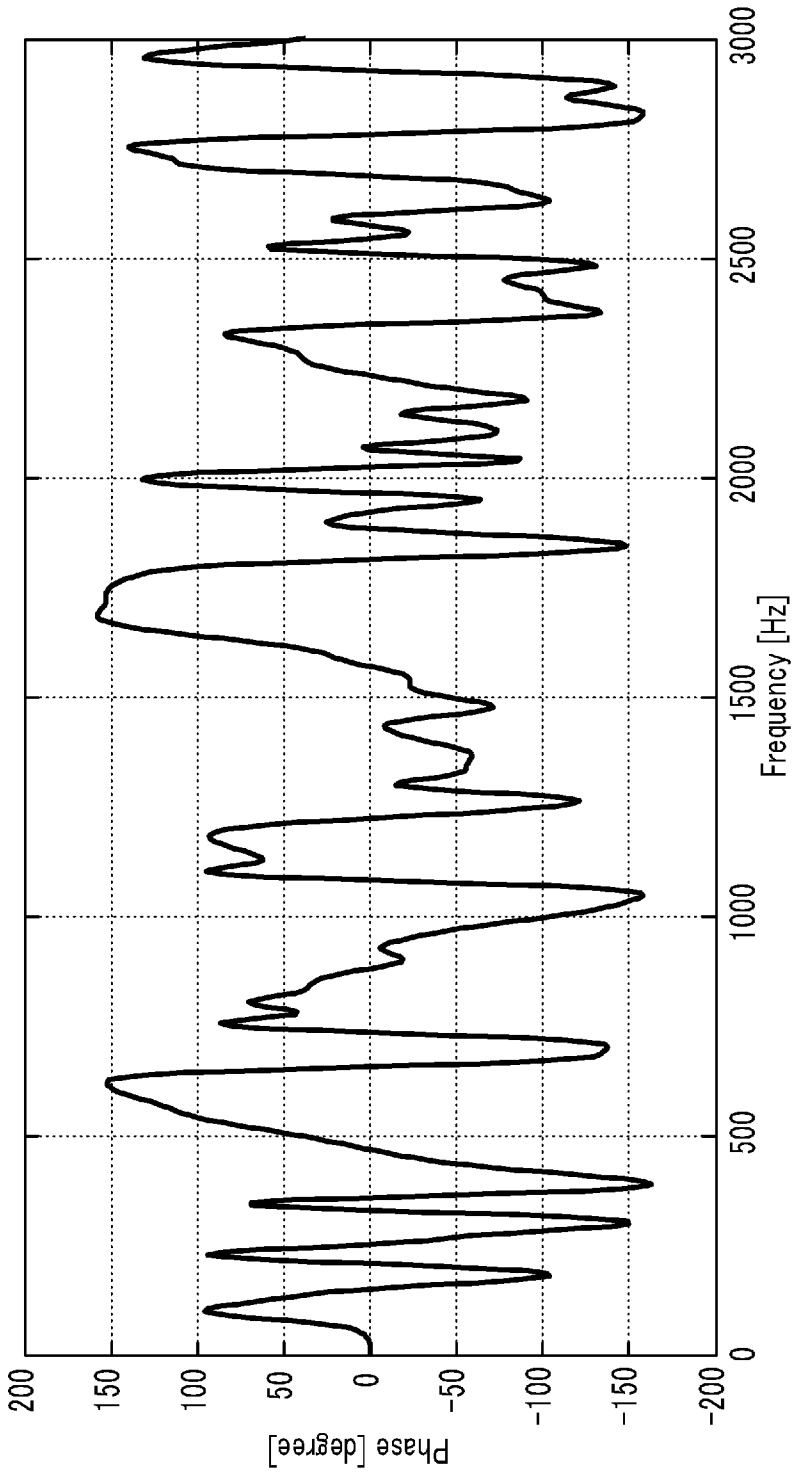
[Fig. 9]


100Hz
400Hz
Level (dB)
25
20
15
10
5
0
-5
-10
-15
-20
-25
Phase (degree)
200
150
100
50
0
-50
-100
-150
-200

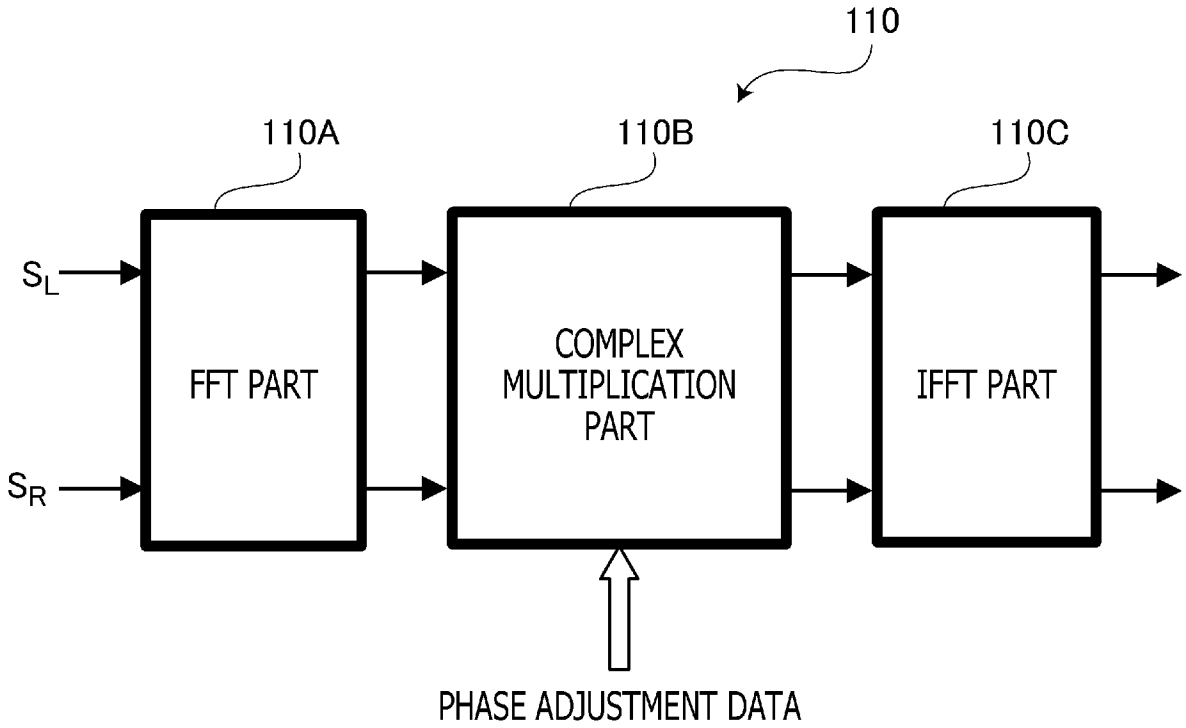
[Fig. 10]


Phase (degree)
200
150
100
50
0
-50
-100
-150
-200
0
500
1000
1500
2000
2500
Frequency (Hz)

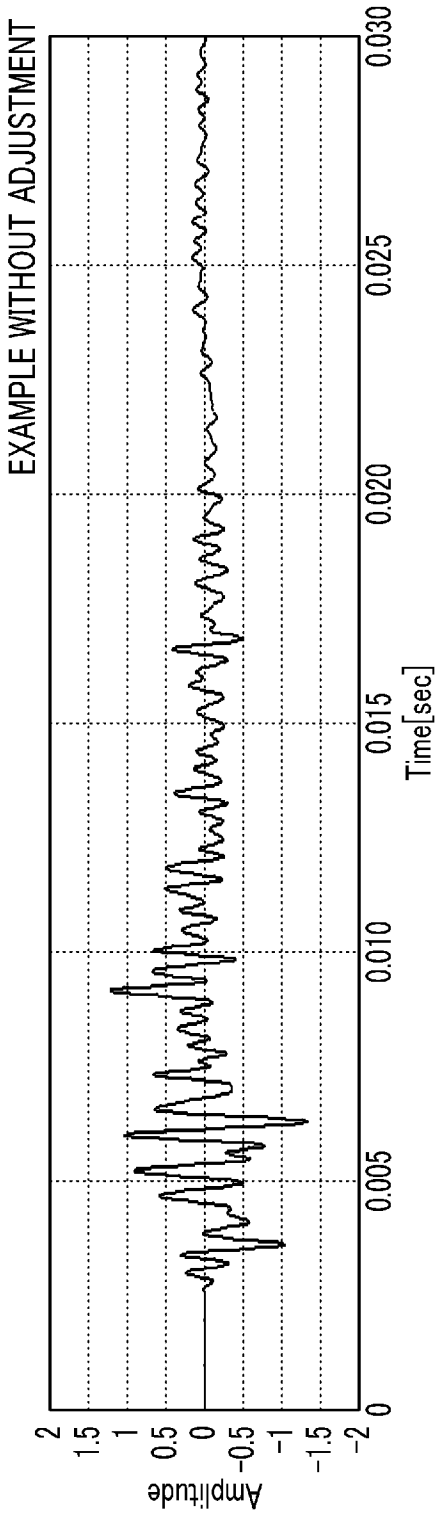
[Fig. 11]


Phase [degree]
200
150
100
50
0
-50
-100
-150
-200
Frequency [Hz]
0
500
1000
1500
2000
2500
3000

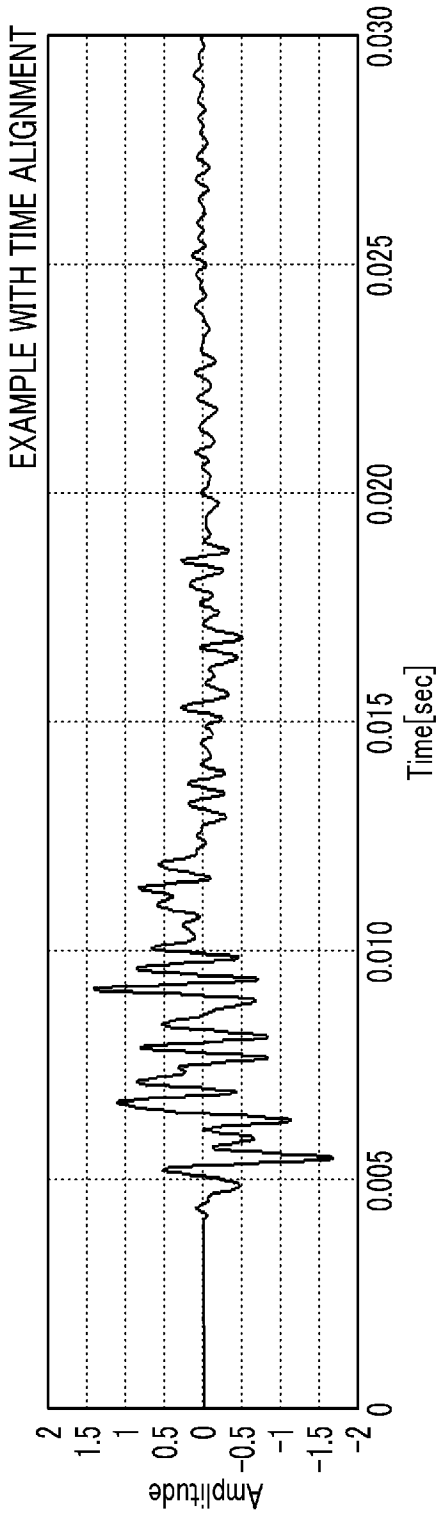
[Fig. 12]


110
110A
110B
110C
S_L
S_R
FFT PART
COMPLEX MULTIPLICATION PART
IFFT PART
PHASE ADJUSTMENT DATA

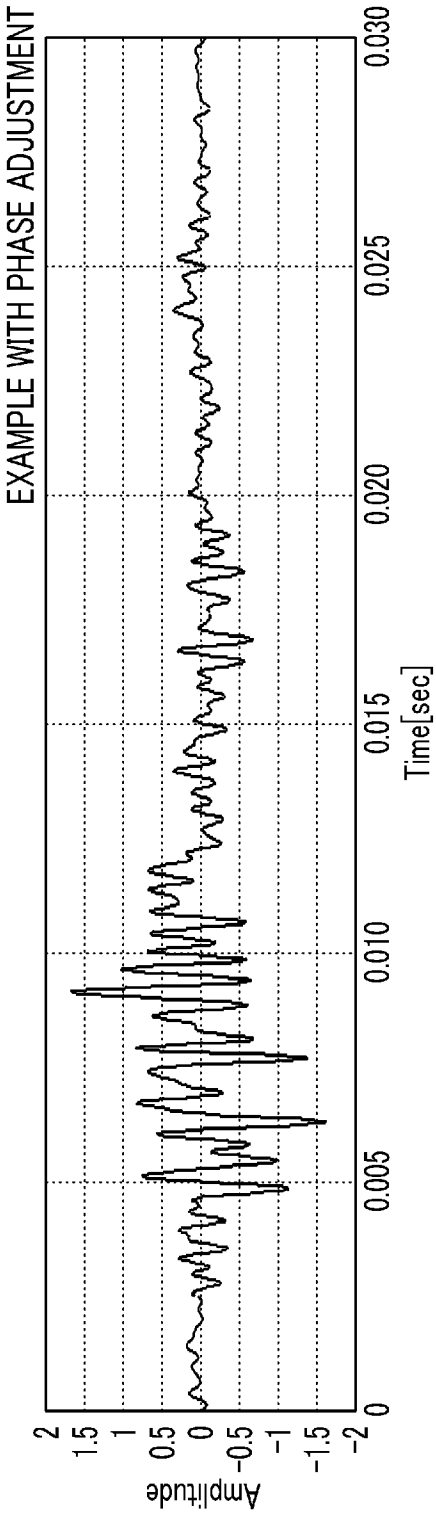
[Fig. 13A]


EXAMPLE WITHOUT ADJUSTMENT
Amplitude
2
1.5
1
0.5
0
−0.5
−1
−1.5
−2
0
0.005
0.010
0.015
0.020
0.025
0.030
Time[sec]

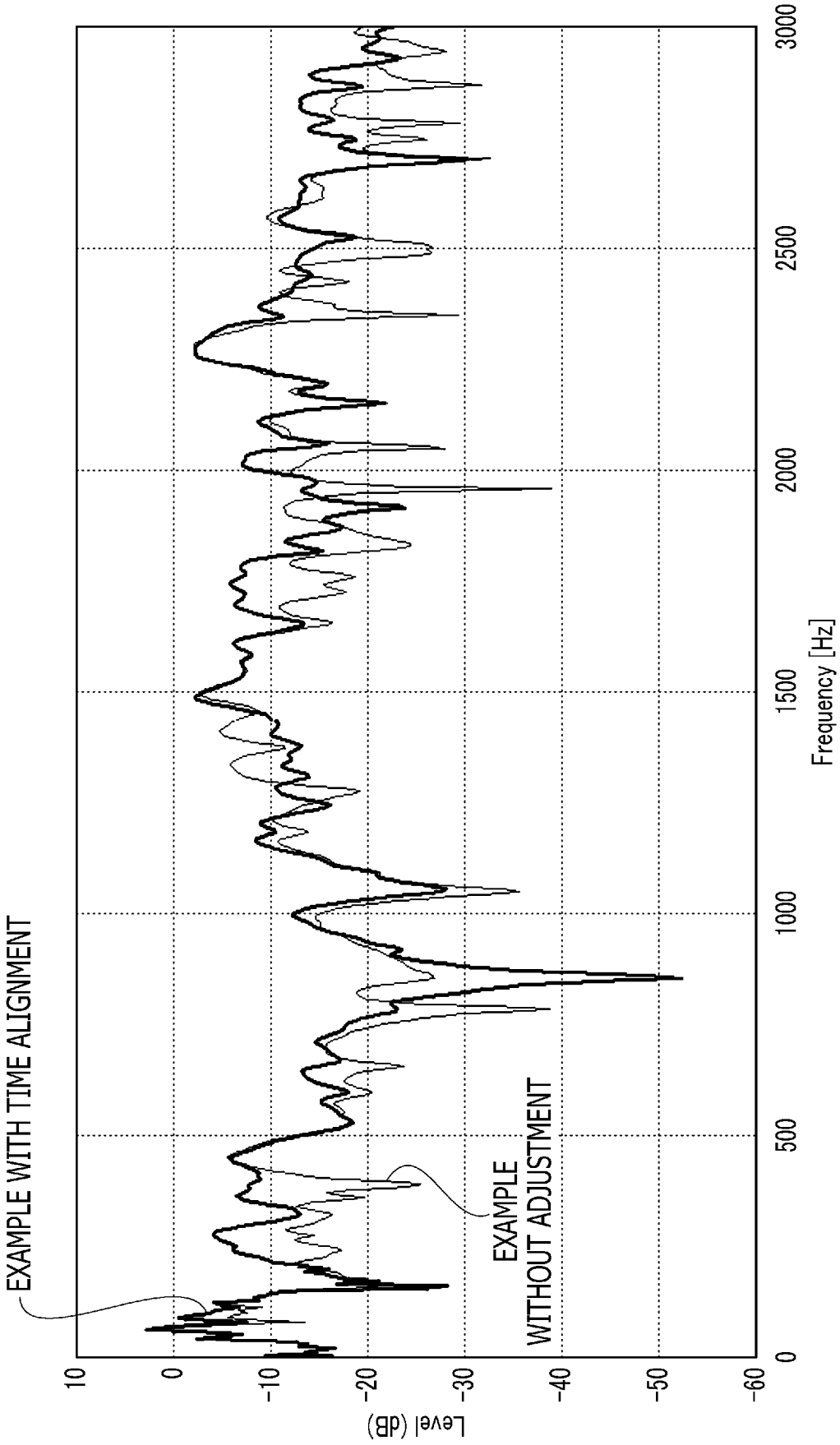
[Fig. 13B]


EXAMPLE WITH TIME ALIGNMENT
Amplitude
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.005
0.010
0.015
0.020
0.025
0.030
Time[sec]

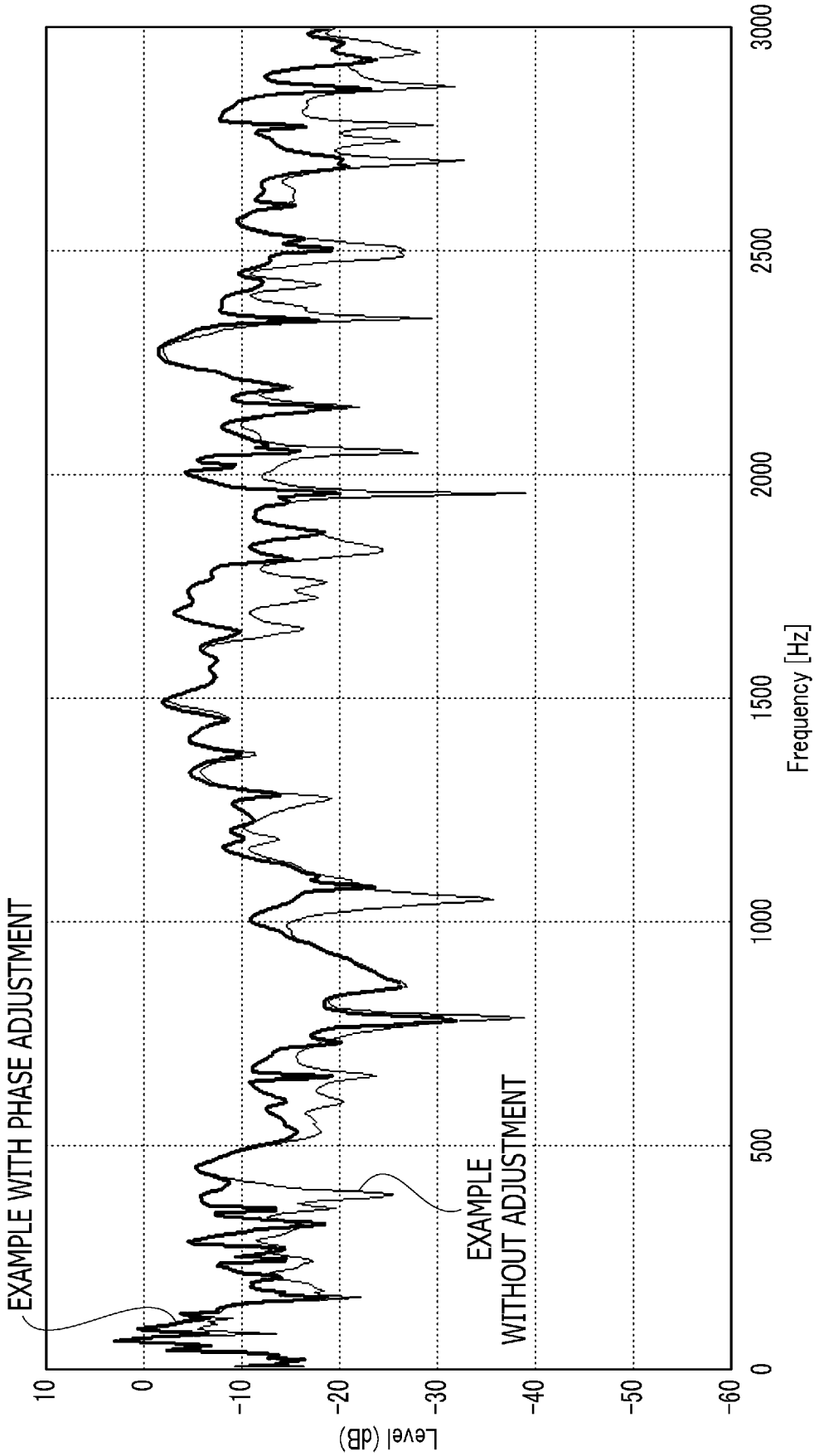
[Fig. 13C]


EXAMPLE WITH PHASE ADJUSTMENT
Amplitude
2
1.5
1
0.5
0
−0.5
−1
−1.5
−2
0
0.005
0.010
0.015
0.020
0.025
0.030
Time[sec]

[Fig. 14]


EXAMPLE WITH TIME ALIGNMENT
EXAMPLE WITHOUT ADJUSTMENT
Level (dB)
10
0
-10
-20
-30
-40
-50
-60
0
500
1000
1500
2000
2500
3000
Frequency [Hz]

[Fig. 15]


EXAMPLE WITH PHASE ADJUSTMENT
EXAMPLE WITHOUT ADJUSTMENT
Level (dB)
10
0
-10
-20
-30
-40
-50
-60
0
500
1000
1500
2000
2500
3000
Frequency [Hz]

INTERNATIONAL SEARCH REPORT

International application No
PCT/JP2020/005720

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04R3/04 H04S7/00
ADD. H04R3/12

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
H04R H04S

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 2016/134985 A1 (HASHIMOTO TAKESHI [JP] ET AL) 12 May 2016 (2016-05-12) figures 1, 9A, 9B, 10A, 10B paragraphs [0007], [0026], [0031], [0033], [0035], [0037] -----	1-14
Y	US 2015/195666 A1 (MASSEY HOWARD [US] ET AL) 9 July 2015 (2015-07-09) figures 1, 2 paragraphs [0006], [0017], [0018] -----	1-14

☐ Further documents are listed in the continuation of Box C.

☒ See patent family annex.

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
2 April 2020	14/04/2020

Name and mailing address of the ISA/	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Fax: (+31-70) 340-3016	Mendoza Lopez, Jorge

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No
PCT/JP2020/005720

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
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		EP 3016412 A1	04-05-2016
		JP 6216553 B2	18-10-2017
		JP 2015012366 A	19-01-2015
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