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(54) **SIGNAL PROCESSING METHOD, SIGNAL PROCESSING DEVICE, AND SIGNAL PROCESSING PROGRAM**

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(57) **ABSTRACT**

Provided is a signal processing method which reduces a plurality of echoes by receiving a plurality of reception signals and subtracting a pseudo echo generated by a plurality of adaptive filters which input the reception signals from a plurality of echoes generated by the reception signals. At least one of the reception signals is delayed to generate a delayed reception signal. The reception signal and the delayed reception signal are inputted to the adaptive filters to generate a pseudo echo. The frequency of inputting the reception signal and the delayed reception signal to the adaptive filters is controlled in accordance with the sensitivity of a localization change of the reception signals.

FIG. 1

FIG. 2

FIG. 3

(A)

$c_0(k)$

M

M

$(2i-1)M$ $2iM$ $(2i+1)M$

(B)

$c_0(k)$

M

M $M-2J$

$(2i-1)M$ $2iM$ $2iM+J$ $(2i+1)M-J$ $(2i+1)M$

(C)

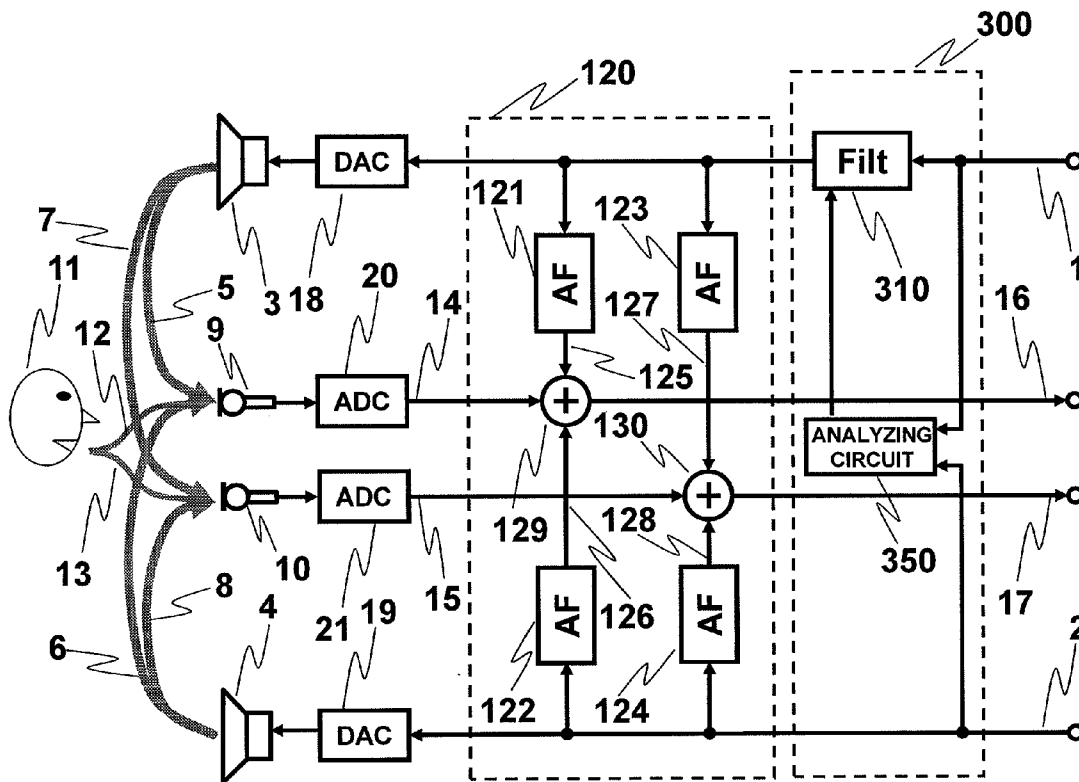$c_0(k)$

M

M $M-2J$

$(2i-1)M$ $2iM-J$ $2iM$ $2iM+J$ $(2i+1)M-J$ $(2i+1)M$

FIG. 4
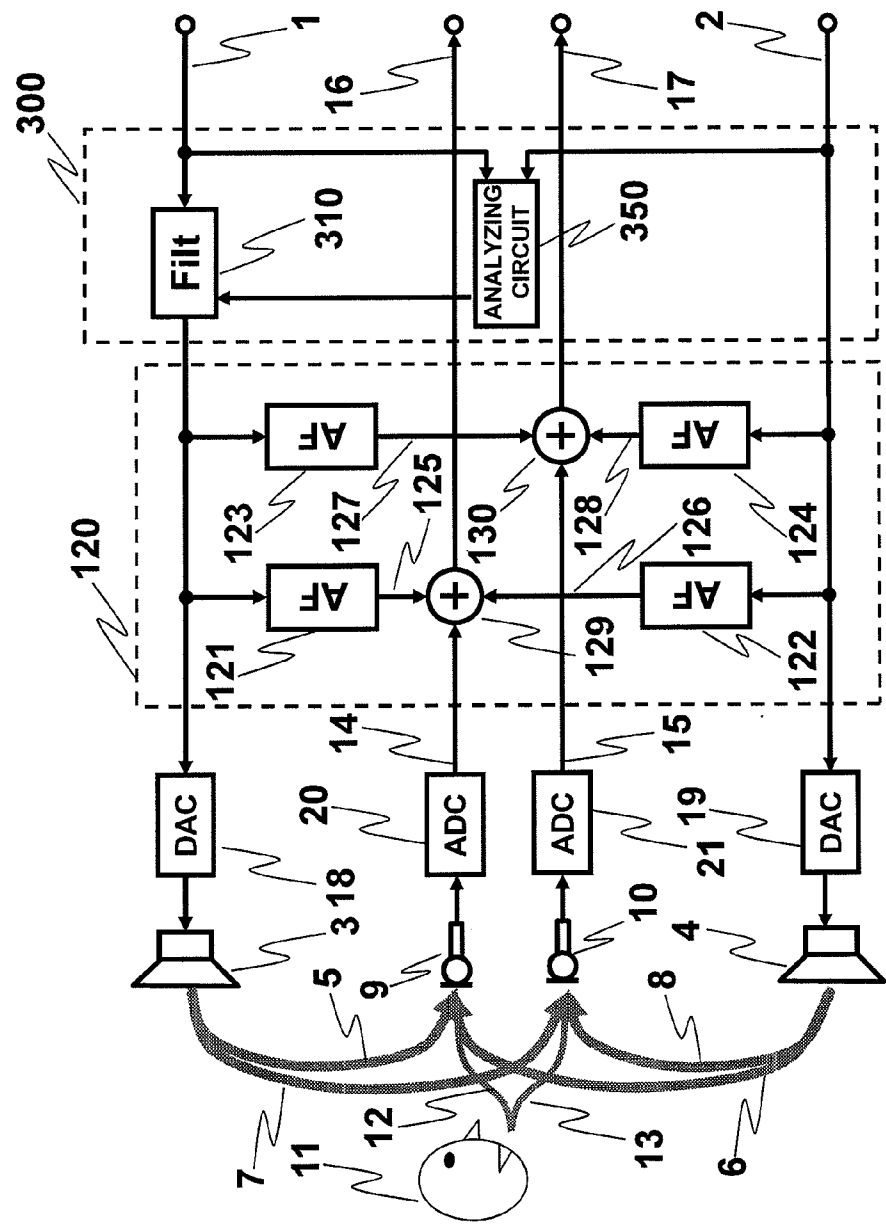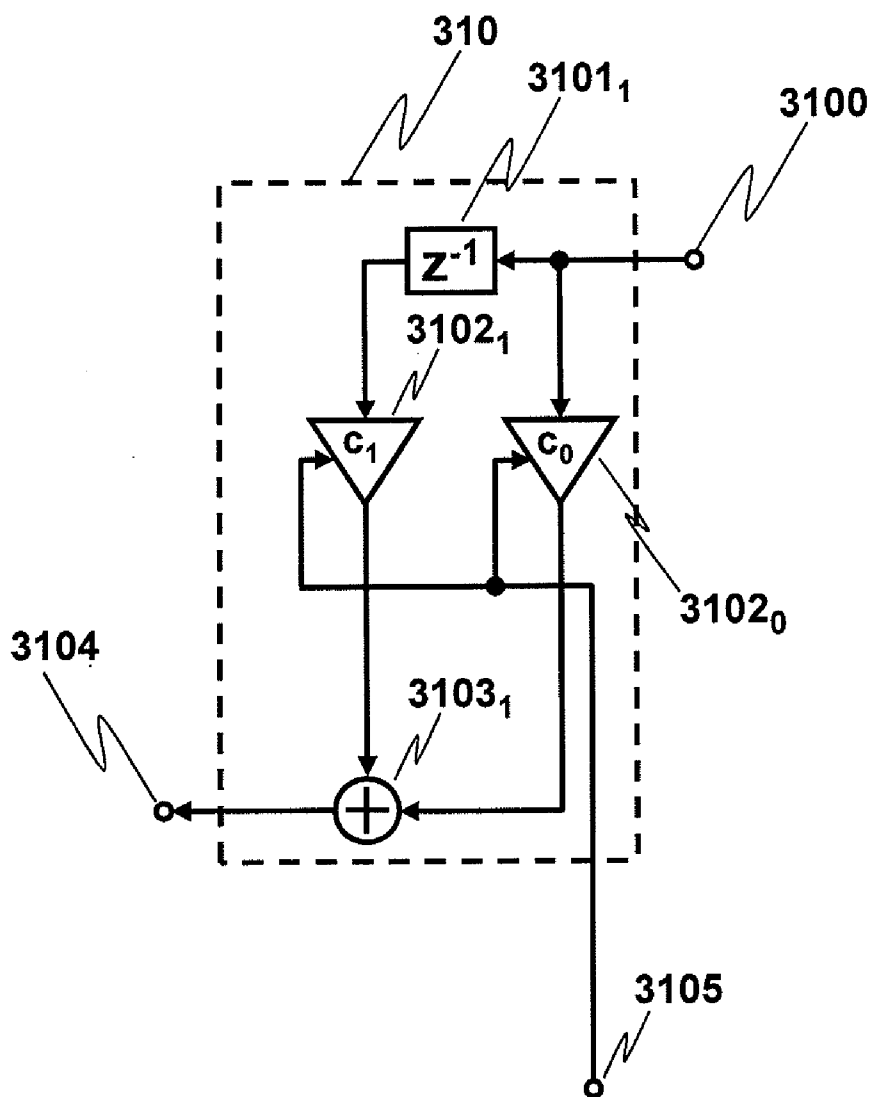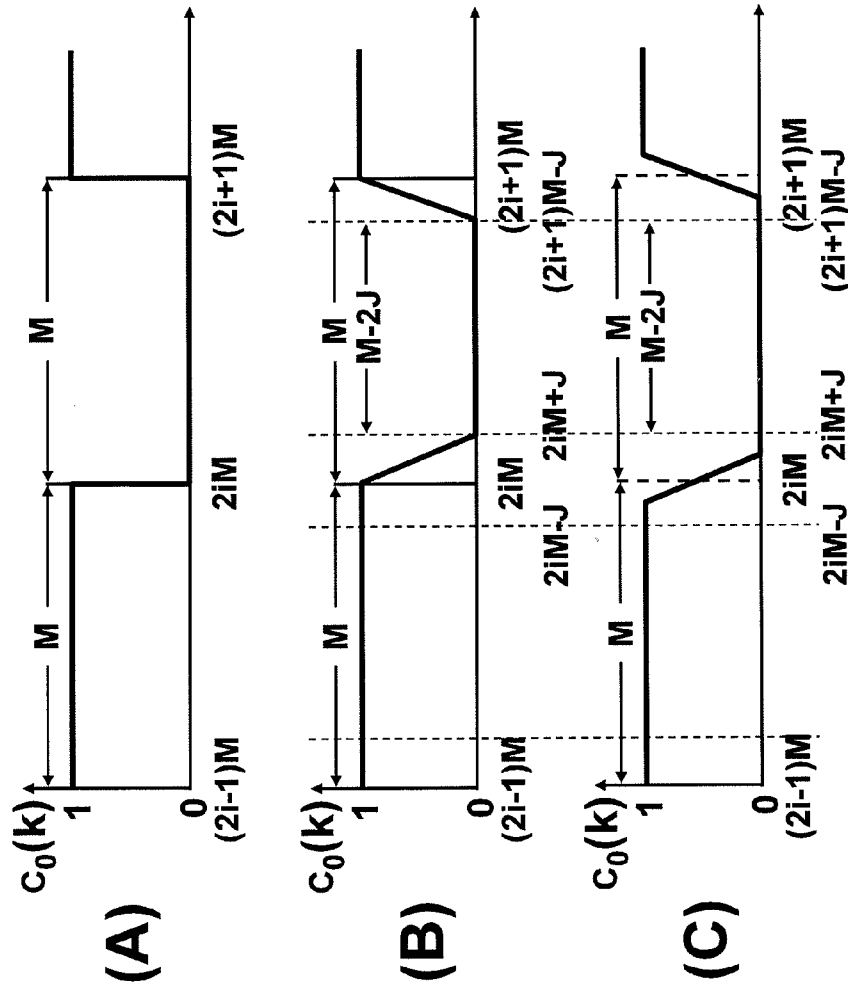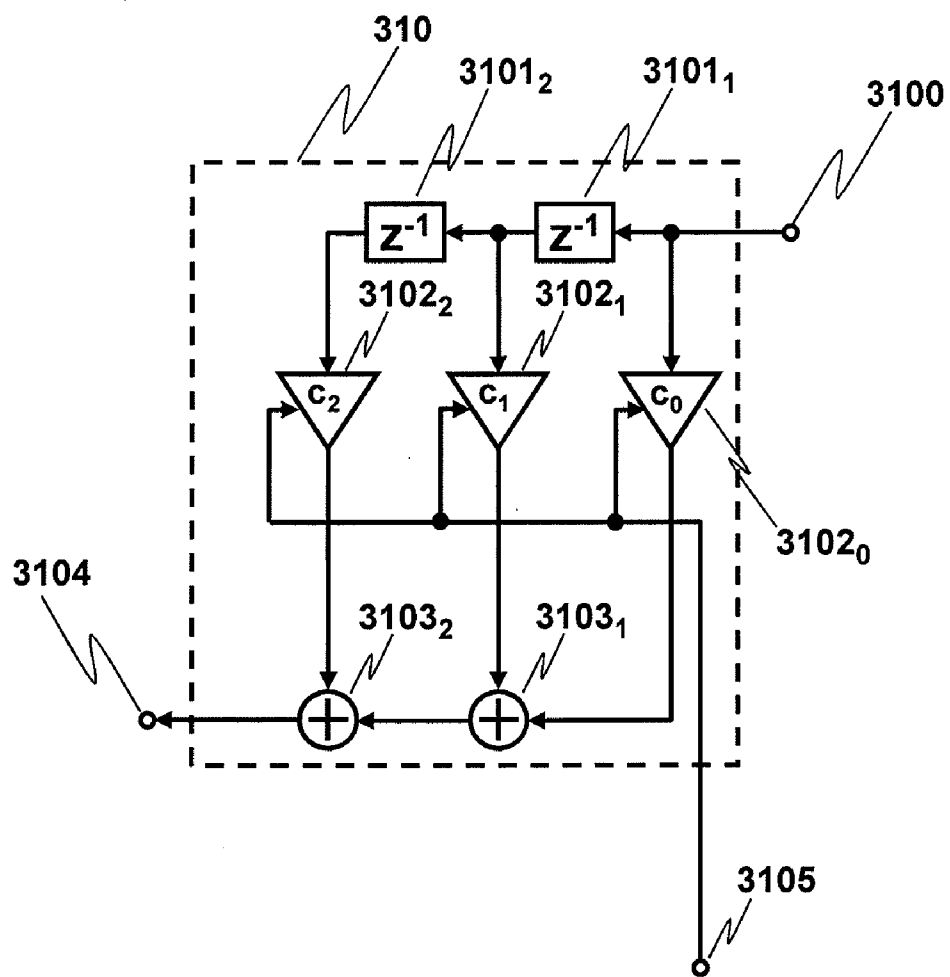
FIG. 5
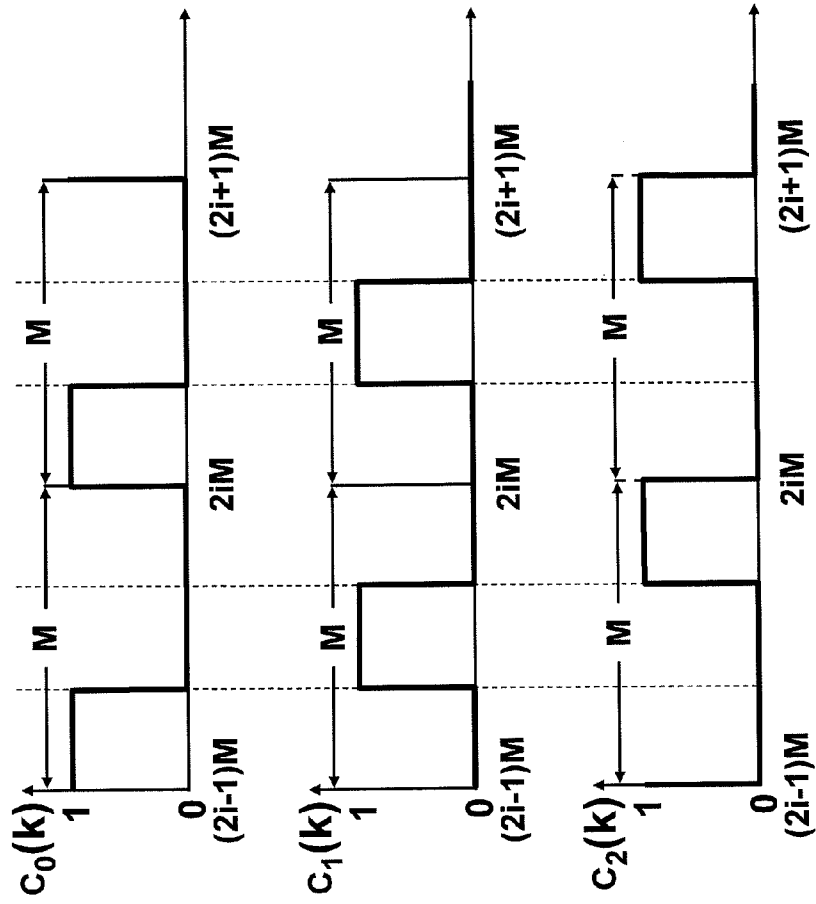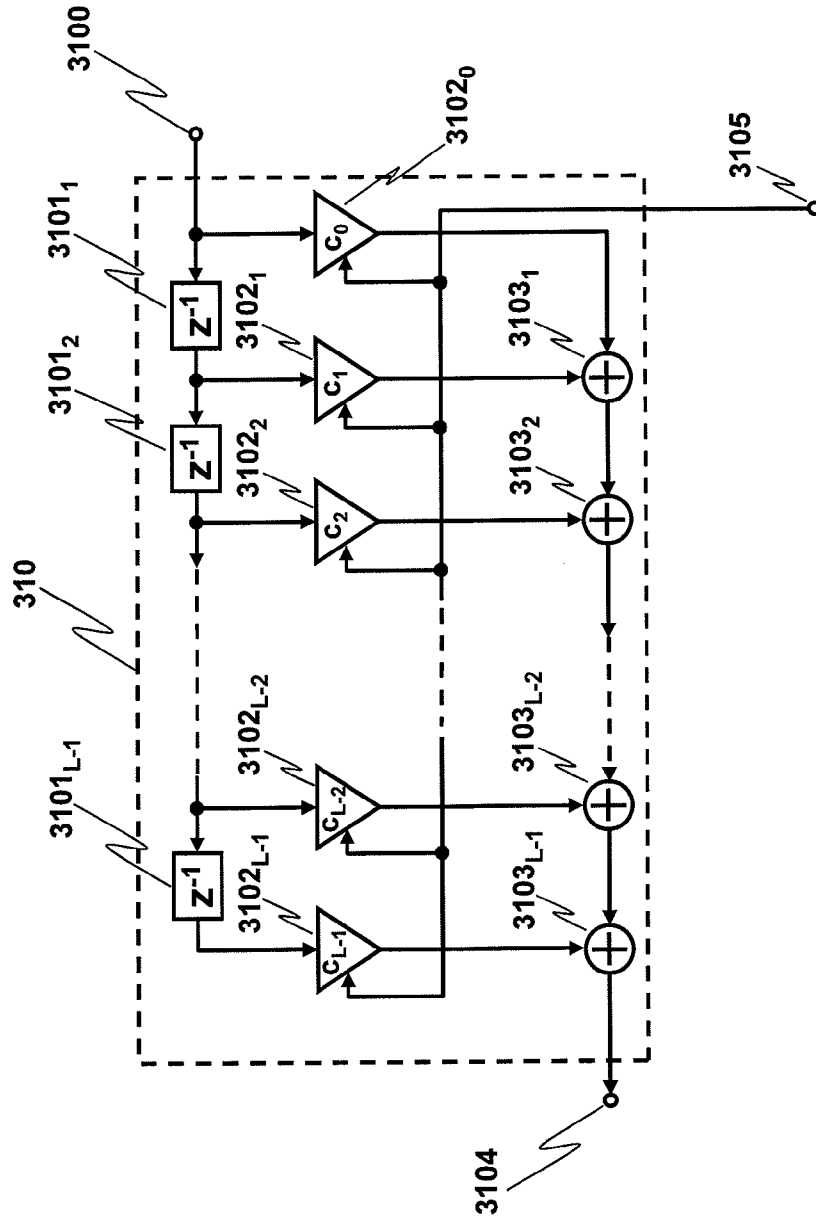
FIG. 6

FIG. 7

FIG. 8

(A)

$c_0(k)$

1

0
(2i-1)M

M

2iM

M

(2i+1)M

(B)

$c_0(k)$

1

0
(2i-1)M

M

2iM

M

(2i+1)M

FIG. 9

$c_0(k)$

$c_1(k)$

$c_2(k)$

FIG. 10

FIG. 11

FIG. 12

FIG. 13

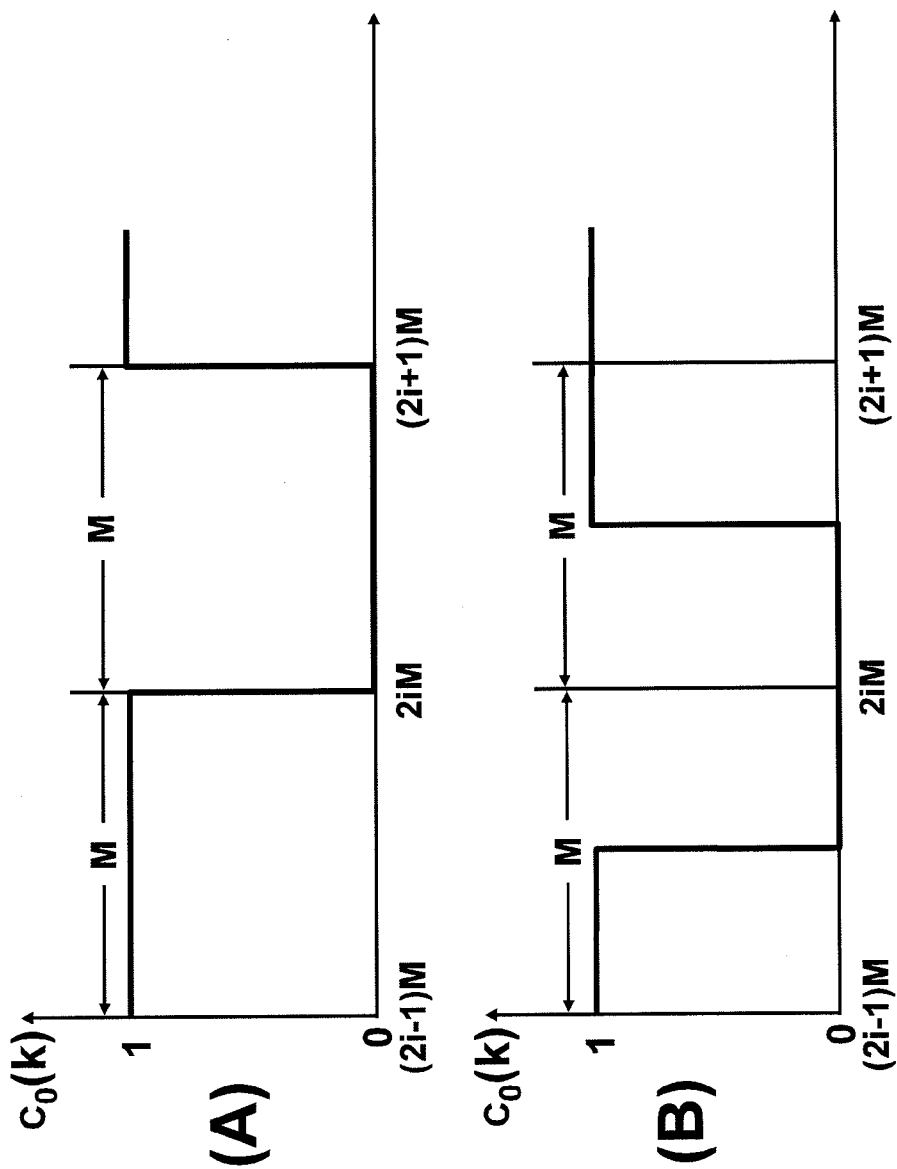**516**   **511**

512

515

513

514

510
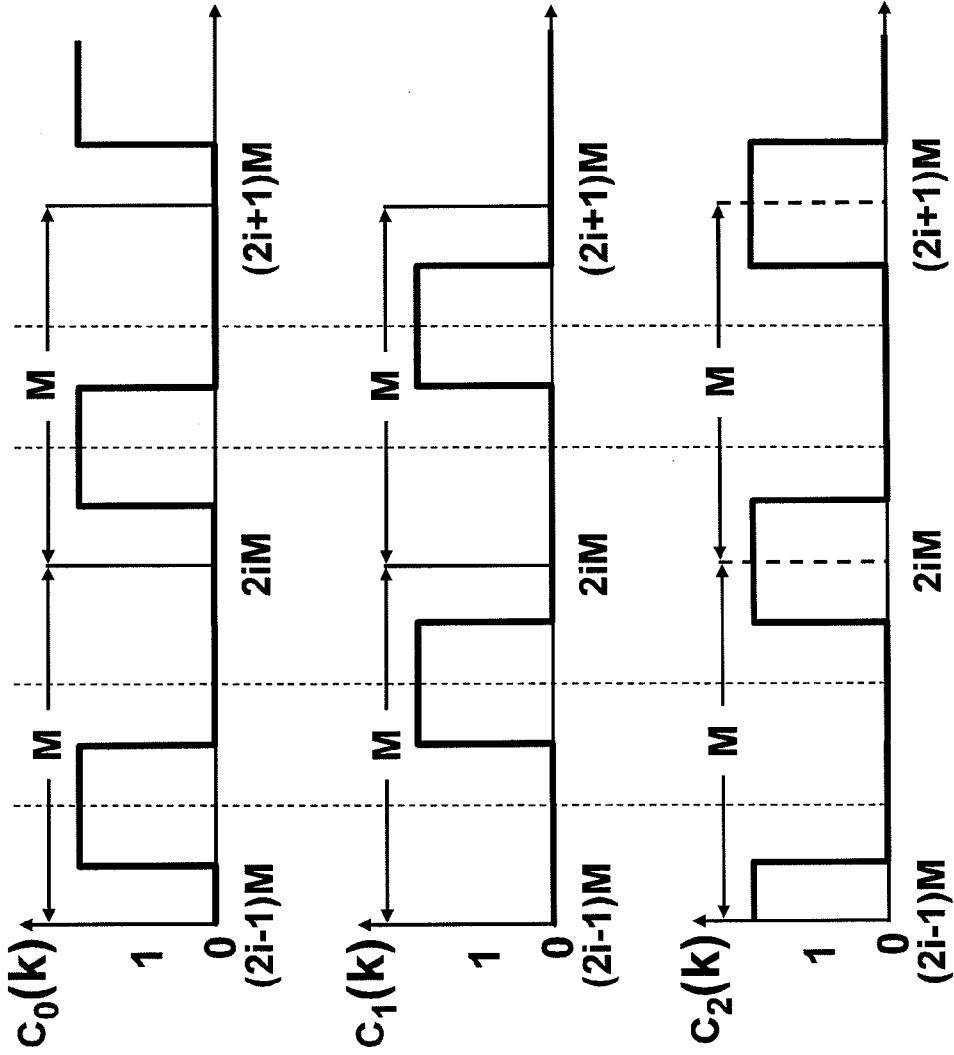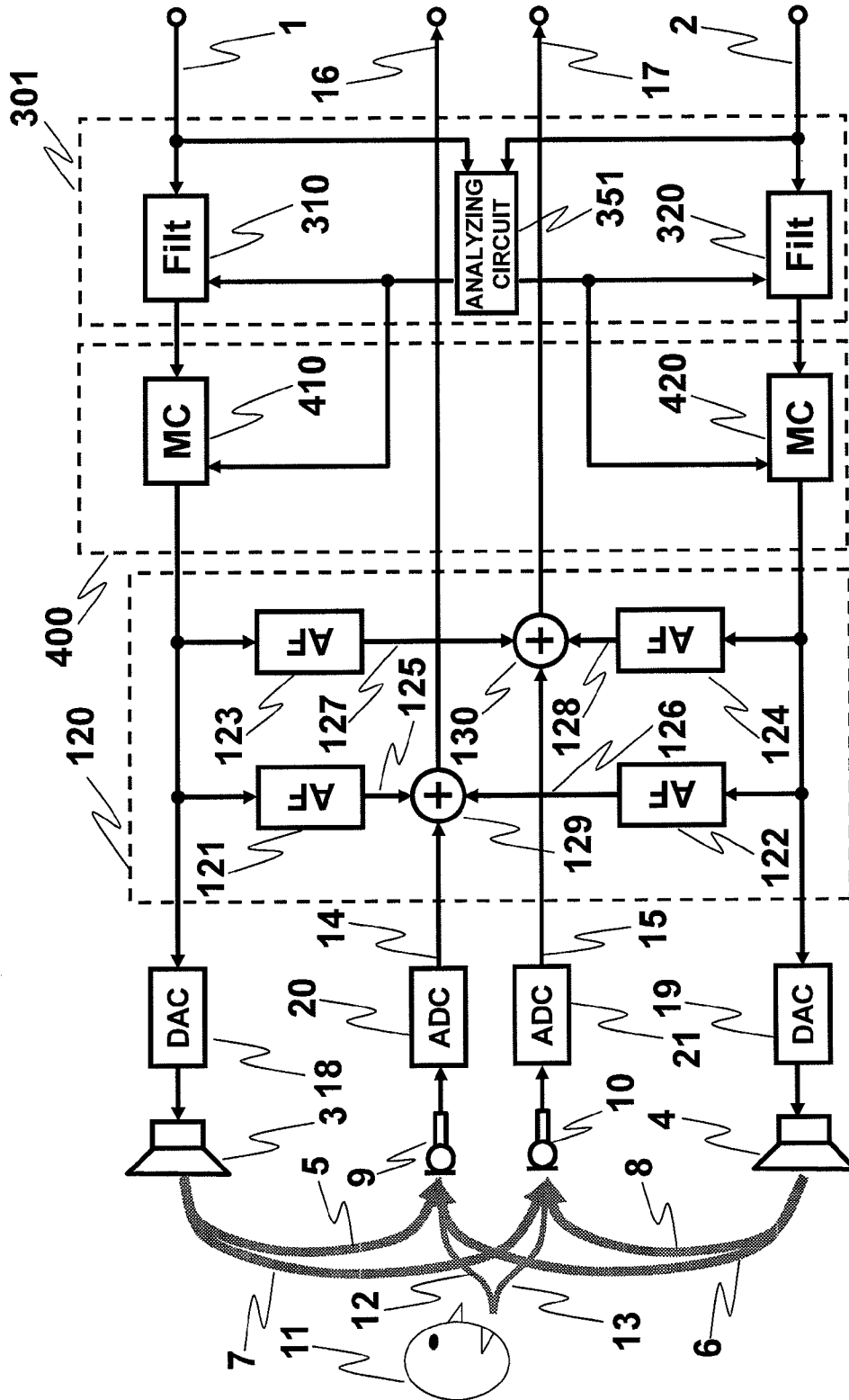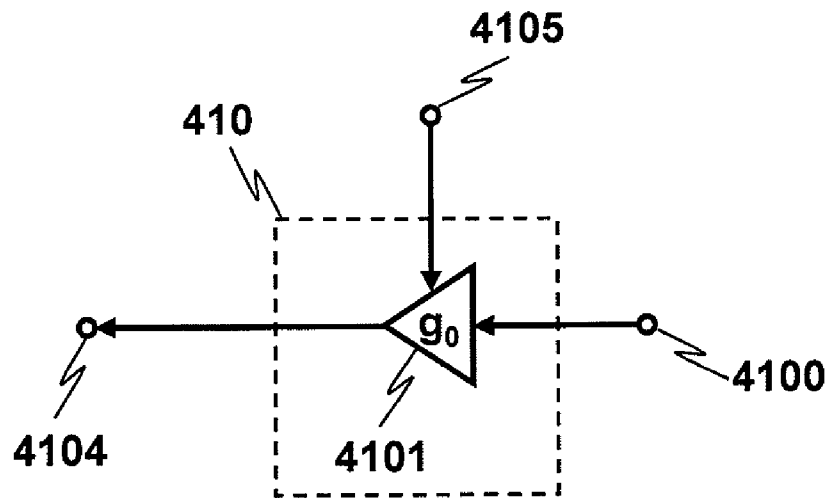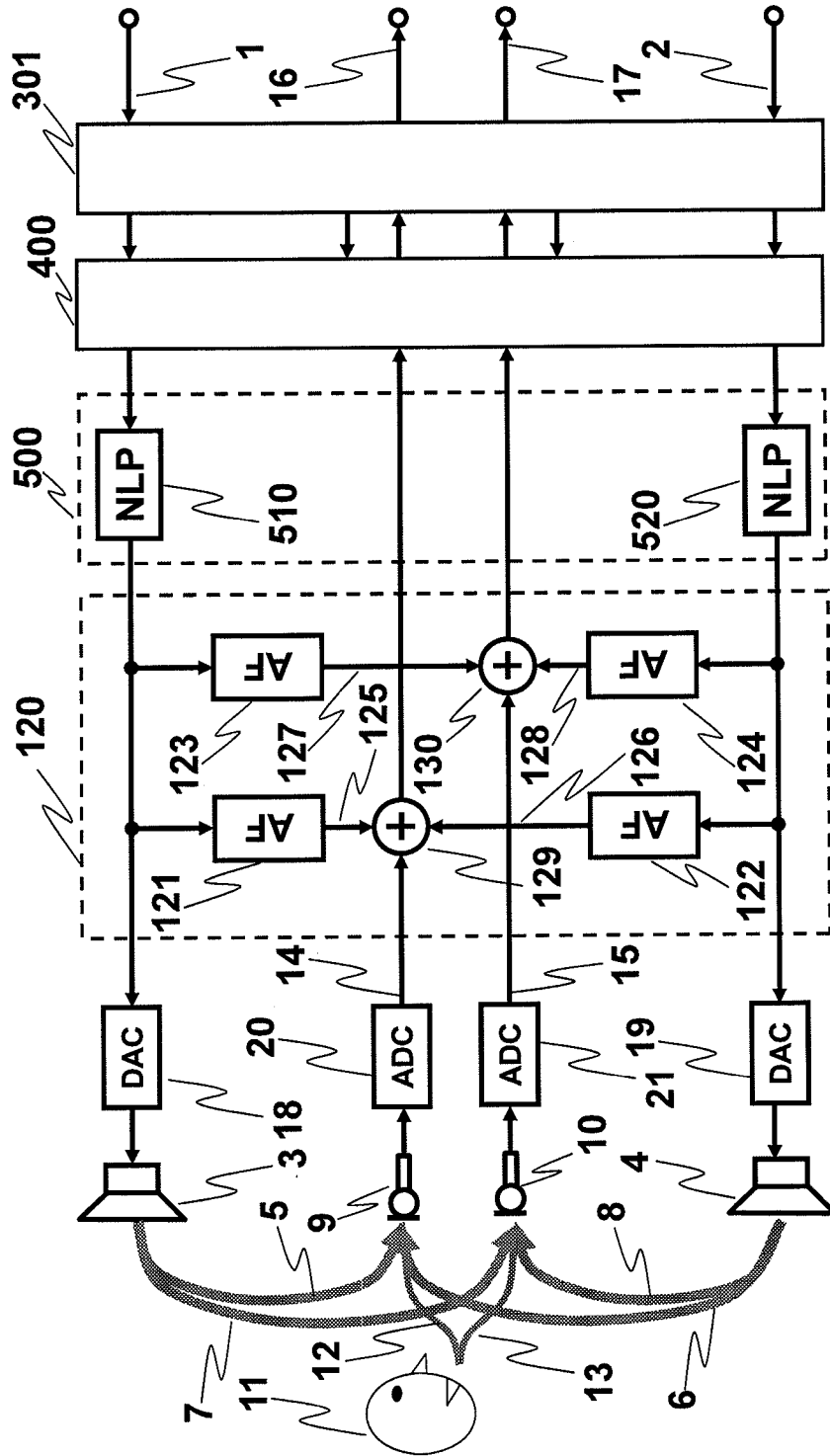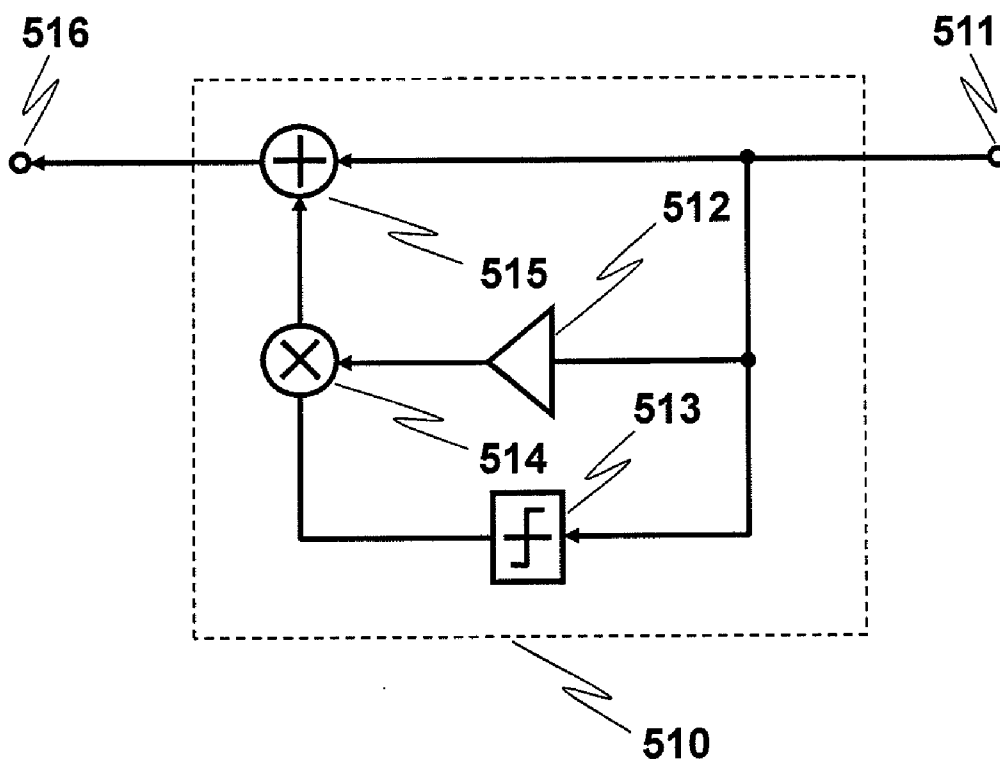
FIG. 14

FIG. 15

FIG. 16

FIG. 17

COMPUTER
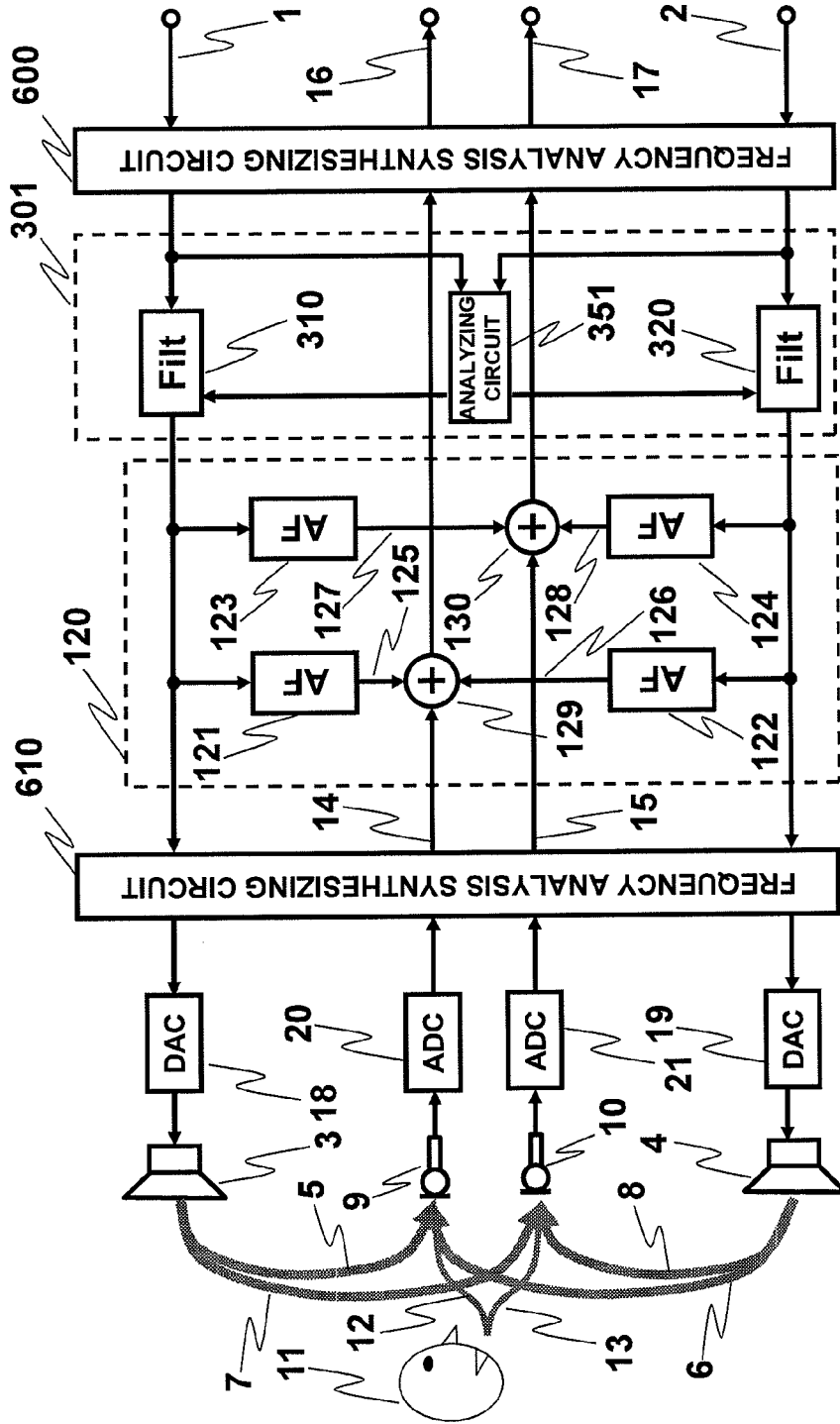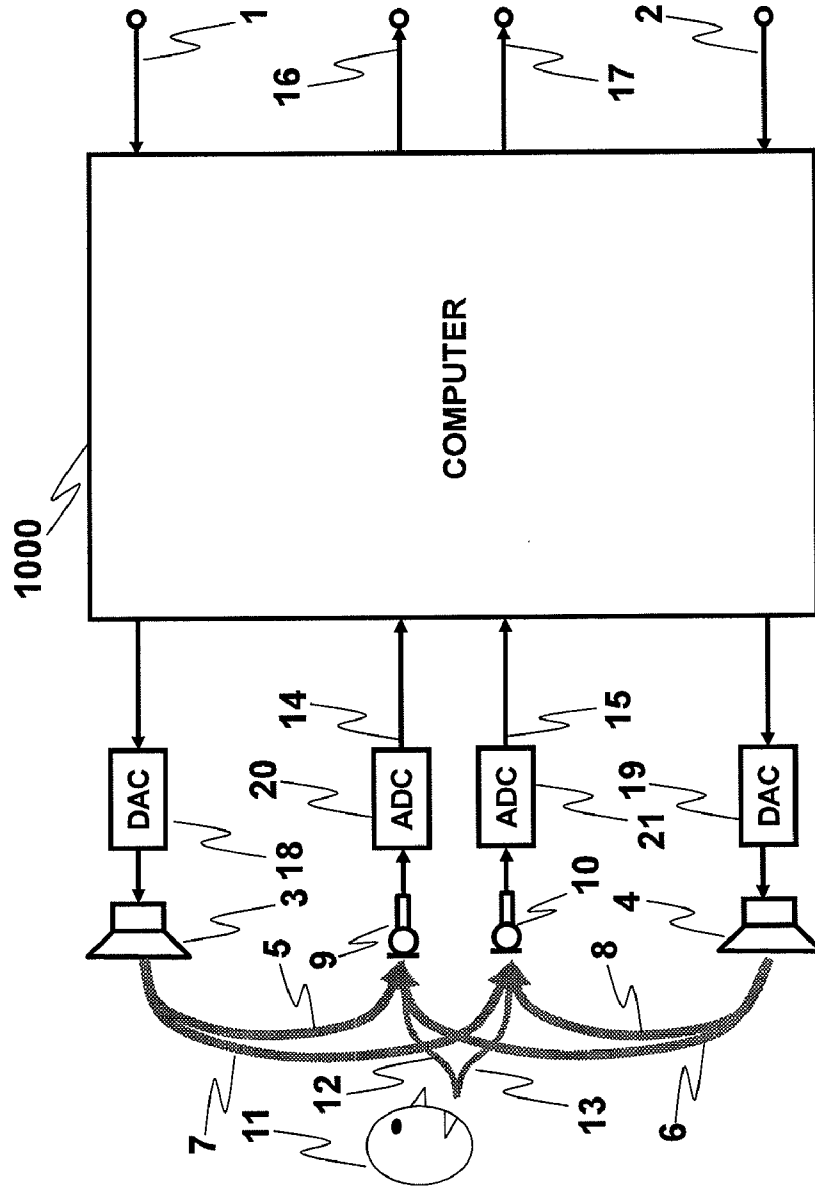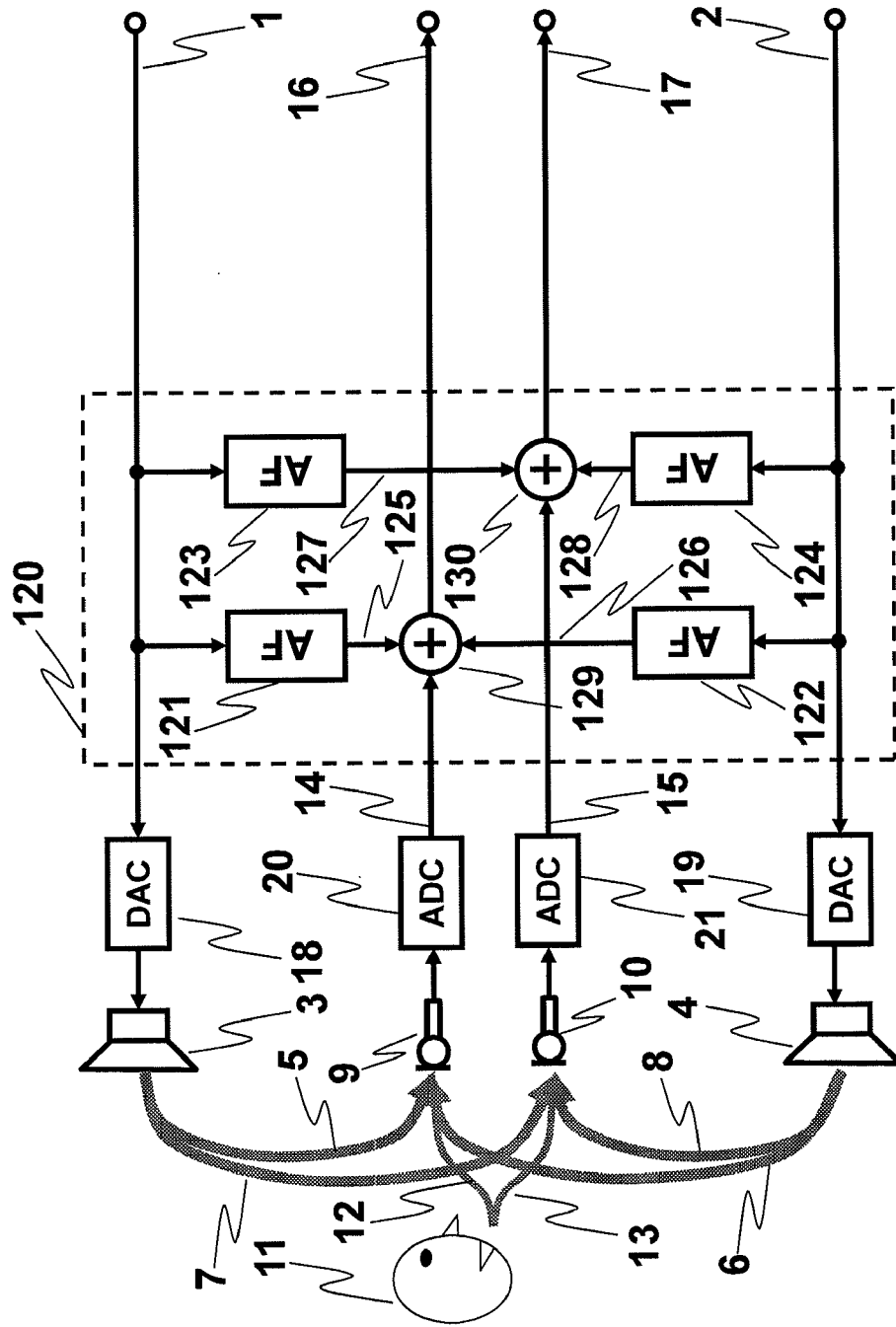
FIG. 18

SIGNAL PROCESSING METHOD, SIGNAL PROCESSING DEVICE, AND SIGNAL PROCESSING PROGRAM

TECHNICAL FIELD

[0001] The present invention relates to a signal processing method, a signal processing apparatus, and a signal processing program.

BACKGROUND ART

[0002] As a signal processing apparatus for canceling echoes caused by received signals propagating through spatial acoustic paths in a system using plural received signals and a single or plural transmission signals, the apparatus of a linear coupled type is disclosed in Non-Patent Literature 1. A block diagram of a multichannel signal processing apparatus of a linear coupled type in the case that the number of channels is two, namely, in the case of the signal processing apparatus having a stereo signal as a target, is shown in FIG. 18. According to Patent Literature 1, the apparatus of the linear coupled type has a problem that the coefficients have an indefinite number, namely a problem that adaptive filter coefficients converge to indefinite values other than values equal to the features of the echo paths (correct solutions). The filter coefficient value that has converged depends upon a cross correlation between the filter input signals, and thus, when the cross correlation is changed due to movement etc. of a far-end talker, the correct coefficient value also varies. A variation in the correct coefficient value caused by a change in the cross correlation means that the echo canceling capability is degraded even with no variation in the echo paths. Thus, the residual echoes are perceived, and thus, a speech quality is degraded.

[0003] In order to solve this problem, the signal processing method in which a single adaptive filter per channel estimates echoes caused by signals propagating from a single sound source in a plurality of paths by generating pseudo echoes (echo replicas) with adaptive filters corresponding one to one to mixed signals with one received signal as an input is disclosed in Patent Literature 1. A problem that the coefficients have an indefinite number does not exist in this signal processing method because one adaptive filter cancels the echo to be generated on one channel. As a result, the adaptive filter coefficients converge to optimum values that are uniquely determined. However, the Non-patent Literature 2 discloses evaluation results proving that the fact that the echo canceling capability is degraded when the parameters determined by the environment in use such as the locations of microphones receiving the taker's voice are not within a certain range. Hence, in order to use the cancellation apparatus in a wide variety of environments, a multichannel echo canceller of a linear coupled type has to be used.

[0004] Based upon this premise, Patent Literature 2 discloses the method capable of uniquely determining the adaptive filter coefficients by delaying the received signal of the multichannel echo canceller of the linear coupled type, thereby to generate the delayed signal, and continuously and mutually alternating this as a new received signal with the received signal. In this signal processing method, the number of conditionals used to calculate the adaptive filter coefficients is increased because of the introduction of the delayed received signals, whereby a problem that the solution, being the adaptive filter coefficient, becomes indefinite does not

occur. As a result, the adaptive filter coefficients converge to optimum values that are uniquely determined. However, with the case of the method proposed in the Patent Literature 2, a movement of the acoustic image may often be perceived when the received signal and the delayed received signal are switched. The movement of the acoustic image is perceived as an unnatural sound because it seems as if the acoustic image had moved even though it does not move as a matter of fact, and hence, a subjective sound quality of the received signals is degraded. In order to solve this, the method of correcting the amplitudes of signals in both channels when the received signal and the delayed received signal are switched is disclosed in Patent Literature 3.

[0005] On the other hand, the method capable of uniquely determining the adaptive filter coefficients by applying a nonlinear processing to the received signals in both channels instead of switching the received signal and the delayed received signal is disclosed in Non-patent Literature 3. However, Non-patent Literature 4 makes it clear that the methods disclosed in the Patent Literature 2, the Patent Literature 3, and the Non-patent Literature 3 provide a slow convergence rate, respectively, as compared with the multichannel echo canceller of the linear coupled type. It is shown in the Non-patent Literature 4 that the method disclosed in the Non-patent Literature 3 provides a yet slower convergence rate as compared with each of the method disclosed in the Patent Literature 2 and the method disclosed in the Patent Literature 3.

CITATION LIST

Patent Literature

[0006] PTL 1: JP-P1992-284732A
[0007] PTL 2: JP-P1999-004183A
[0008] PTL 3: JP-P2000-078061A

Non-Patent Literature

[0009] NON-PTL 1: The Technical Report of the institute of Electronics, Information and Communication Engineers (IEICE) of Japan, Vol. 84, No. 330, pp. 7-14, CS-84-178
[0010] NON-PTL 2: IEEE Proceedings of International Conference on Acoustics, Speech and Signal Processing, Vol. 2, 1994, pp. 245-248
[0011] NON-PTL 3: IEEE Proceedings of International Conference on Acoustics, Speech and Signal Processing, Vol. 1, 1997, pp. 303-306
[0012] NON-PTL 4: IEEE Proceedings of International Conference on Acoustics, Speech and Signal Processing, Vol. 6, 1998, pp. 3677-3680

SUMMARY OF INVENTION

Technical Problem

[0013] Each of the methods disclosed in the Patent Literature 3 and the Non-patent Literature 3 is slow in the convergence rate as compared with the signal processing apparatus of the linear coupled type. Further, the method disclosed in the Patent Literature 3 has a problem that making the convergence rate fast often causes a movement of the acoustic image localization to be perceived, and hence the subjective sound quality of the received signals is degraded. Thus, the method disclosed in the Patent Literature 3 is not able to simultaneously accomplish a short convergence time and a high subjective sound quality.

[0014] Thereupon, the present invention has been accomplished in consideration of the above-mentioned problems, and an object thereof is to provide a signal processing method, a signal processing apparatus, and a signal processing program with an excellent subjective sound quality of the received signals and a short convergence time (a fast convergence rate), wherein the coefficient values of the adaptive filters converge to correct values that are uniquely determined by impulse responses of the echo paths.

Solution to Problem

[0015] The present invention is a signal processing method of receiving a plurality of received signals, and subtracting pseudo echoes generated by a plurality of adaptive filters having said plurality of received signals as an input, respectively, from a plurality of echoes to be generated from said plurality of received signals, thereby to reduce said plurality of echoes, comprising: generating delayed received signals by delaying at least one received signal, out of said plurality of received signals; generating pseudo echoes by inputting said received signals and said delayed received signals into said adaptive filters; and controlling a frequency of inputting said received signals and said delayed received signals into said adaptive filters based upon a perceptual sensitivity to a change in localization of said plurality of received signals.

[0016] In additions, the present invention is a signal processing apparatus for receiving a plurality of received signals, and subtracting pseudo echoes generated by a plurality of adaptive filters having said plurality of received signals as an input, respectively, from a plurality of echoes to be generated from said plurality of received signals, thereby to reduce said plurality of echoes, comprising: a linear processing circuit for generating delayed received signals by delaying at least one received signal, out of said plurality of received signals; an adaptive filter for generating pseudo echoes by receiving said received signals and said delayed received signals, a plurality of subtracters each generating echo-reduced signals by subtracting said pseudo echoes from a plurality of mixed signals; and an analyzing circuit for obtaining a perceptual sensitivity to a change in localization of said plurality of received signals, and wherein said signal processing apparatus controlling a frequency of inputting said received signals and said delayed received signals into said adaptive filters based upon said perceptual sensitivity, and controlling coefficients of said plurality of adaptive filters so that outputs of said plurality of subtracters are minimized.

[0017] In additions, the present invention is a signal processing program for causing a computer to execute a receiving process of receiving a plurality of received signals, and an echo reducing process of reducing a plurality of echoes that are generated by said plurality of received signals, said signal processing program comprising: a delayed received signal generating process of generating delayed received signals by delaying at least one received signal, out of said plurality of received signals; a pseudo echo generating process of generating pseudo echoes by inputting said received signals and said delayed received signals into said adaptive filters; and a pseudo echo subtracting process of subtracting said pseudo echoes from said plurality of received signals, respectively, wherein a frequency of inputting said received signals and said delayed received signals into said adaptive filters is con-

trolled based upon a perceptual sensitivity to a change in localization of said plurality of received signals.

Advantageous Effect of Invention

[0018] The signal processing method, the signal processing apparatus, and the signal processing program of the present invention generate delayed received signals by delaying at least one received signal, and activate the adaptive filters with the foregoing received signals and the foregoing delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the received signal and the delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0019] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

BRIEF DESCRIPTION OF DRAWINGS

[0020] FIG. 1 is a block diagram illustrating a best mode of the signal processing apparatus of the present invention.

[0021] FIG. 2 is a block diagram illustrating an exemplary configuration of a filter **310**.

[0022] FIG. 3 is a view illustrating an example of a variation in a coefficient $c_0(k)$ in the filter **310**.

[0023] FIG. 4 is block diagram illustrating a second exemplary configuration of the filters **310** and **320**.

[0024] FIG. 5 is a view illustrating an example of a variation in the coefficient $c_0(k)$, a coefficient $c_1(k)$, and a coefficient $c_2(k)$ in the filter **310**.

[0025] FIG. 6 is block diagram illustrating a third exemplary configuration of the filter **310**.

[0026] FIG. 7 is a block diagram illustrating a second embodiment of the present invention.

[0027] FIG. 8 is a view illustrating an example of a variation in a coefficient $c_0(k)$ in the filters **310** and **320**.

[0028] FIG. 9 is a view illustrating an example of a variation in the coefficient $c_0(k)$, the coefficient $c_1(k)$, and the coefficient $c_2(k)$ in the filter **320**.

[0029] FIG. 10 is a block diagram illustrating a third embodiment of the present invention.

[0030] FIG. 11 is a block diagram illustrating an exemplary configuration of an amplitude processing circuit **410**.

[0031] FIG. 12 is a block diagram illustrating a fourth embodiment of the present invention.

[0032] FIG. 13 is a block diagram illustrating an exemplary configuration of a non-linear amplitude processing circuit **510**.

[0033] FIG. 14 is a block diagram illustrating a fifth embodiment of the present invention.

[0034] FIG. 15 is a block diagram illustrating an exemplary configuration of a linear amplitude processing circuit **530**.

[0035] FIG. 16 is a block diagram illustrating a sixth embodiment of the present invention.

[0036] FIG. 17 is a block diagram illustrating a seventh embodiment of the present invention.

[0037] FIG. 18 is a block diagram illustrating the signal processing apparatus of the linear coupled type.

DESCRIPTION OF EMBODIMENTS

[0038] The embodiments of the present invention will be explained in details by using FIG. 1 to FIG. 18. It is now assumed to use a two-channel acoustic echo canceller having a first received signal and a second received signal that cancels the acoustic echoes caused by received signals propagating from loudspeakers to microphones via spatial acoustic paths.

[0039] The signal processing apparatus of the present invention with two received signals and two transmission signals, according to the best mode, is shown in FIG. 1. A difference with the apparatus of the linear coupled type disclosed in the Non-patent literature 1 lies in a point that the received signal 1 to be supplied to adaptive filters 122 and 123 are processed by a delay processing circuit 300.

[0040] The received signals 1 and 2 are supplied to the delay processing circuit 300. The delay processing circuit 300 generates the delayed received signals by delaying the received signal 1, and transmits them to adaptive filters 121 and 123 and a digital-analogue (DA) converter 18, respectively. The delay processing circuit 300 transmits the received signal 2 to adaptive filters 122 and 124 and a digital-analogue (DA) converter 19, respectively. The DA converters 18 and 19 convert the digital delayed received signals or the digital received signals into analogue delayed received signals or analogue received signals, respectively, and transmit them to loudspeakers 3 and 4, respectively. The loudspeakers 3 and 4, upon receipt of the delayed received signals and the received signals, radiate them into the space. Microphones 9 and 10, upon receipt of a talker 11's voice, and crosstalks (echoes) of the delayed received signals radiated from the loudspeakers 3 and 4 into the spaces, transmits them to analogue-digital (AD) converters 20 and 21 as mixed signals 14 and 15, respectively. The AD converters 20 and 21 convert the mixed signals 14 and 15 from the analogue signals into the digital signals, and transmit them to subtracters 129 and 130. On the other hand, the adaptive filters 121 and 123 having received the foregoing delayed received signals and the adaptive filters 122 and 124 having received the foregoing received signals generate the pseudo echoes (echo replicas), and transmit them to the subtracters 129 and 130, respectively. The subtracters 129 and 130 subtract the pseudo echoes generated by the adaptive filters 121 and 122, and the pseudo echoes generated by the adaptive filters 123 and 124 from the foregoing mixed signals 14 and 15, respectively, and output them as output signals 16 and 17. With the subtraction by the subtracters 129 and 130, the echoes are canceled, and as a result, the output signals 16 and 17 are converted into the signals including no echo.

[0041] As algorithm for updating the coefficients of the adaptive filters 121, 122, 123, and 124, the Least Mean Square (LMS) algorithm is disclosed in Non-patent Literature 4 ("Adaptive Signal Processing", 1985, pp. 99-113, Prentice-Hall Inc., USA) and the Normalized Least Mean Square (NLMS) algorithm is disclosed in Non-patent Literature 5 ("Adaptive Filters", 1985, pp. 49-56, Kulwer Academic Publishers, USA).

[0042] As algorithm of the adaptive filter, the Sequential Regression Algorithm (SRA) described in the Non-patent

Literature 4, and the RLS algorithm described in the Non-patent Literature 5 may be also used.

[0043] The delayed signals are intermittently generated in the delay processing circuit 300, and the signal obtained by delaying the received signal 1, and the signal equal to the received signal 1, that is, the not-delayed signal coexist in the delayed received signal. The cross correlation between the signals to be supplied to the adaptive filters 121 and 123, and the signals to be supplied to the adaptive filters 122 and 124 differs from each other in these two kinds of the statuses (in a status where the received signals have been delayed and in a status where the received signals have not been delayed). For this, the two kinds of the conditionals (more than the conditionals of the apparatus of the linear coupled type) responding to two kinds of the statuses can be gained, and the coefficients of the adaptive filters 121, 123, 122, and 124 converge to the correct values.

[0044] The relative delay amount (relative delay) of the delayed received signal to the received signal can be set to an integer multiple of a sampling period. In this case, the minimum value is equalized to the sampling period. The larger the amount of the relative delay, the larger the cross correlation between the signals to be supplied to the adaptive filters 121 and 123 and the signals to be supplied to the adaptive filters 122 and 124, hence, the convergence time is shortened. However, the movement amount of the acoustic images at the moment that the delayed received signals are reproduced in the loudspeakers 3 and 4 becomes larger, and thus, the subjective sound quality is degraded. For this, the large relative delay is appropriately selected within an allowable range of the movement amount of the acoustic images that are perceived.

[0045] The relative delay can be also set to a non-integer multiple of the sampling period. In this case, the more suitable selection is enabled because a fine adjustment is enabled in a selection of the allowable movement amount of the acoustic images and the relative delay that is as large as possible.

[0046] The relative delay does not need to be restricted to one kind, and a plurality of values can be also used alternately. For example, the three statuses (two kinds of the relative delays), that is, a status in which the relative delay is a zero multiple of the sampling period (no delay), a status in which the relative delay is a one multiple of the sampling period (a one-sample delay), and a status in which the relative delay is twice as large as the sampling period (a two-sample delay) can be switched. In this case, the number of the foregoing conditionals is three times as large as that of the apparatus of the linear coupled type, and hence, the foregoing adaptive filters can be caused to converge in a far faster rate. As the number of the relative delay amounts that are utilized is increased, the foregoing adaptive filters converge at a faster rate.

[0047] The delay processing circuit 300 includes a filter 310, and an analyzing circuit 350. The filter 310 generates the delayed received signal by delaying the received signal 1. Further, the filter 310 outputs the received signal 1 as it stands without delaying it in some cases. A frequency at which the output of the filter 310 varies between the delayed received signal and the received signal 1 is controlled with a clock signal that is supplied from the analyzing circuit 350. The adapter filterers 121 and 123, to which this switching signal is supplied, converge more quickly as this frequency becomes higher. The reason is that with the switching, a cross correlation between the received signal 1 and the received signal 2 is

4

changed, and hence a combination as well of the conditionals representing a relation between the received signals **1** and **2**, and the coefficients of the adaptive filters **121** and **123** is changed. On the other hand, the switching allows the acoustic images that are localized by the loudspeakers **3** and **4** to move toward the loudspeaker **4**. Hence, a high switching frequency causes the movement of the acoustic image localization to be easily perceived subjectively, which leads to a degradation in the subjective sound quality. That is, there is a trade-off between the convergence rate and the subjective sound quality in terms of the foregoing switching frequency.

[0048] The analyzing circuit **350**, upon receipt of the received signal **1** and the received signal **2**, calculates the perceptual sensitivity to the movement of the acoustic images that are localized by these received signals. The analyzing circuit **350** generates the clock signal corresponding to the obtained perceptual sensitivity, and supplies it to the filter **310**. The filter **310** decides a generating frequency of the delayed received signals based upon the supplied clock signal.

[0049] A high perceptual sensitivity to the acoustic image movement by the received signal **1** and the received signal **2** means that the movement of the acoustic images is easily perceived. That is, even the slight movement of the acoustic images is perceived. To the contrary, a low perceptual sensitivity means that the movement of the acoustic images is hardly perceived. For example, when the received signal **1** and the received signal **2** resemble each other, it follows that the amplitudes and the phases of both resemble each other. In such a case, the acoustic images are localized in a position extremely close to the center between the loudspeaker **3** for reproducing the received signal **1** and the loudspeaker **4** for reproducing the received signal **2**. The movement of the acoustic images is easily perceived when the acoustic images are located midway between the frontal loudspeakers **3** and **4**. To the contrary, when the acoustic images are localized at a position far away from the center of the loudspeakers **3** and **4**, namely, are localized at a position close to the loudspeaker **3** or the loudspeaker **4**, it is not easy to perceive the movement of the acoustic images. Thereupon, the analyzing circuit **350** generates the clock signal in such a manner that a switching frequency of the received signal and the delayed receive signal becomes low when the perceptual sensitivity is high, and a switching frequency of the received signal and the delayed receive signal becomes high when the perceptual sensitivity is low, and supplies it to the filter **310**.

[0050] An example of an index of the perceptual sensitivity, a similarity degree that is typified by the cross correlation between the received signal **1** and the received signal **2** may be used. The cross correlation, as a rule, can be expressed by a product of sample values at each time of two signals. So as to obtain an averaged cross relation for a non-stationary signal, the quotient of the foregoing products divided by a total number of the samples accumulated for a constant time can be also calculated. In addition, so as to avoid dependence of the obtained quotient upon an absolute value of an input signal power, a normalized cross correlation obtained by normalizing the foregoing accumulation of the products in terms of a product of a sum of squared sample values of the received signal **1** and a sum of squared sample values of the received signal **2** each of which corresponds to an identical time can be also used. As a special version of the cross correlation, the phase correlation using the phases of the received signals and the amplitude correlation using the amplitudes of the received

signals, an averaged value thereof, and a normalized value thereof can be also used. In addition, it is also possible that the analyzing circuit **350** is configured not to operate the correlation calculation when the signal power or the amplitude is extremely low. Excluding small signal samples that are susceptible to an undesirable influence due to added noise makes it possible to increase a tolerance to noise.

[0051] An example of another index of the perceptual sensitivity, powers of the received signal **1** and the received signal **2** may be used. When the powers of the received signal **1** and the received signal **2** are small, the movement of the acoustic images is hardly perceived even though the acoustic images are moved. The reason is that the small power makes it difficult to listen to the received signals themselves. To the contrary, when the powers of the received signal **1** and the received signal **2** are large, the movement of the acoustic images is easily perceived. Thus, the analyzing circuit **350** generates the clock signal such that a switching frequency between the received signal and the delayed receive signal becomes low when the powers of the received signal **1** and the received signal **2** are large, and a switching frequency between the received signal and the delayed receive signal becomes high when the powers of the received signal **1** and the received signal **2** are small, and supplies it to the filter **310**. Either one power can be used as an index of the powers of the received signal **1** and received signal **2** because it can be presumed that a correlation between both is high. Further, an averaged value of the powers of the received signals **1** and **2** can be also used. In either case, not only an instantaneous value but also an averaged value for a constant time can be also used. In addition, when the signal power or the amplitude is extremely low, it is also possible that the analyzing circuit **350** is configured not to operate the correlation calculation. Excluding small signal samples that are susceptible to an undesirable influence due to added noise makes it possible to increase a tolerance to noise.

[0052] FIG. **2** is a block diagram illustrating an exemplary configuration of the filter **310**. The filter is configured as a two-tap FIR filter having c_0 and c_1 as the coefficient. The received signal **1** of FIG. **1** is supplied to an input terminal **3100** of FIG. **2**. The signal to be obtained in an output terminal **3104** of FIG. **2** is the delayed received signal.

[0053] The signal supplied to the input terminal **3100** is transmitted to a delay element **3101₁** and a coefficient multiplier **3102₀**. The coefficient multiplier **3102₀** multiplies the inputted received signal sample by a coefficient value c_0 and transmits its product to an adder **3103₁**. The delay element **3101₁** delays the received signal sample by one sample, and transmits it to a coefficient multiplier **3102₁**.

[0054] The coefficient multiplier **3102₁** multiplies the inputted received signal sample by a coefficient value c_1 and transmits its product to an adder **3103₁**. The adder **3103₁** adds the output of the coefficient multiplier **3102₀** and the output of the coefficient multiplier **3102₁**, and outputs its sum as the delayed received signal to the output terminal **3104**.

[0055] The clock signal, which is supplied to an input terminal **3105** from the analyzing circuit **350** of FIG. **1**, is transmitted to the coefficient multiplier **3102₀**, a coefficient multiplier **3102₁**, and a coefficient multiplier **3102₂**. Based upon the clock signal supplied from the input terminal **3105**, each of the coefficient multiplier **3102₀**, the coefficient multiplier **3102₁**, and the coefficient multiplier **3102₂** varies its coefficient value.

[0056] The coefficient c_0 of the coefficient multiplier 3102_0 and the coefficient c_1 of the coefficient multiplier 3102_1 vary with a time. So as to clearly express this, c_0 and c_1 are denoted as $c_0(k)$ and $c_1(k)$, respectively. $c_1(k)$ is given according to the following numerical equation.

$$c_1(k) = 1 - c_0(k)$$ <Numerical equation 1>

[0057] One example of $c_0(k)$ is shown in FIG. 3(A). i is assumed to be an arbitrary natural number. $c_0(k)$ periodically has 1 and 0 every M (integer) samples. Further, as apparent from the numerical equation 1, $c_1(k)$ varies as represented in the figure that is obtained by reversing FIG. 3(A) up and down. That is, $c_0(k)$ and $c_1(k)$ are exclusive to each other, and either $c_0(k)$ or $c_1(k)$ is zero that is inputted into the adder 3103. Hence, the output of the adder 3103 becomes equal to either the received signal or the delayed received signal, which is equivalent to switching the received signal or the delayed received signal every M samples. Additionally, while the maximum value of $c_0(k)$ can be set to an arbitrary value, the output needs to be scaled by compensating a change in the amplitudes at that moment so that an output identical to the output that is gained when the maximum value of $c_0(k)$ is 1 is yielded.

[0058] In FIG. 3(B) that differs from FIG. 3(A), $c_0(k)$ is set so that it does not vary abruptly, but smoothly varies with a long lapse of a time at the moment of varying between a zero value and a non-zero value. A smooth variation in the value yields an effect that the acoustic images that are generated at the moment of mutually switching the received signal and the delayed received signal move smoothly, and the acoustic image movement is hardly perceived. Further, there is also an effect of avoiding perception of the click sound at the moment of the foregoing switching. This is effective in improving the subjective sound quality.

[0059] Upon comparing FIG. 3(B) with FIG. 3(C), the time of $c_0(k)=1$ and the time of $c_0(k)=0$ differ from each other. The convergence of the adaptive filter coefficients to the corrective values can be accomplished for a shortest time when each of the time of $c_0(k)=1$ and the time of $c_0(k)=0$ becomes maximum because the cross correlation between the signal to be supplied to the adaptive filters 121 and 123 and the signal to be supplied to the adaptive filters 122 and 124 most largely differs from that of the apparatus of the linear coupled type when $c_0(k)=0$. In other words, the shorter the section in which the foregoing smooth variation in the value occurs, the shorter the convergence time. On the other hand, it is felt that the movement of the acoustic images is abrupt all the more as the section in which the foregoing smooth variation in the value occurs becomes shorter. Hence, the section in which the foregoing smooth variation in the value occurs is set to have an appropriate length by taking into consideration both of the perception of the acoustic image movement and the convergence time. While FIGS. 3(B) and (C) show an example where a variation of $c_0(k)$ from $c_0(k)=1$ to $c_0(k)=0$ (or the contrary hereto) is proportional to a time, an arbitrary smooth curved line or straight line for connecting $c_0(k)=1$ and $c_0(k)=0$ can be used.

[0060] FIG. 4 is a block diagram illustrating a second exemplary configuration of the filter 310. The filter is configured as a three-tap FIR filter having c_0, c_1, and c_2 as the coefficient. The received signal 1 of FIG. 1 is supplied to an input terminal 3100 of FIG. 4. The signal to be obtained in an output terminal 3104 of FIG. 4 is the delayed received signal.

[0061] The signal supplied to the input terminal 3100 is transmitted to a delay element 3101_1 and a coefficient multiplier 3102_0.

[0062] The coefficient multiplier 3102_0 multiplies the inputted received signal sample by a coefficient value c_0 and transmits its product to an adder 3103_1. The delay element 3101_1 delays the received signal sample by one sample, and transmits it to a coefficient multiplier 3102_1 and a delay element 3101_2.

[0063] The coefficient multiplier 3102_1 multiplies the output of the delay element 3101_1 by a coefficient value c_1 and transmits its product to an adder 3103_1. The adder 3103_1 adds the output of the coefficient multiplier 3102_0 and the output of the coefficient multiplier 3102_1, and outputs its sum to an adder 3103_2. The delay element 3101_2 delays the output of the delay element 3101_1 by one sample, and transmits it to a coefficient multiplier 3102_2.

[0064] The coefficient multiplier 3102_2 multiplies the output of the delay element 3101_2 by a coefficient value c_1 and transmits its product to an adder 3103_2. The adder 3103_2 adds the output of the adder 3103_1 and the output of the coefficient multiplier 3102_2, and outputs its sum as the delayed received signal to an output terminal 3104.

[0065] The clock signal, which is supplied to the input terminal 3105 from the analyzing circuit 350 of FIG. 1, is transmitted to the coefficient multiplier 3102_0, the coefficient multiplier 3102_1, and the coefficient multiplier 3102_2. Based upon the clock signal supplied from the input terminal 3105, each of the coefficient multiplier 3102_0, the coefficient multiplier 3102_1, and the coefficient multiplier 3102_2 varies its coefficient value.

[0066] An example of the coefficient $c_0(k)$ of the coefficient multiplier 3102_0, the coefficient $c_1(k)$ of the coefficient multiplier 3102_1, and the coefficient $c_2(k)$ of the coefficient multiplier 3102_2 is shown in FIG. 5. The coefficient $c_0(k)$, the coefficient $c_1(k)$, and the coefficient $c_2(k)$ have 1 exclusively to each other, thereby allowing the received signals subjected to the delay, which correspond to respective coefficient multipliers, to be gained as the delayed received signals in the output terminal 3104. Like FIGS. 3(B) and (C) corresponding to FIG. 3(A), $c_0(k)$, $c_1(k)$, and $c_2(k)$ shown in FIG. 5 can be set so that they smoothly vary at the moment of varying between a zero value and a non-zero value. A smooth variation in the value yields an effect that the acoustic images that are generated at the moment of mutually switching the received signal and the delayed received signal move smoothly, and the acoustic image movement is hardly perceived. Further, there is also an effect of avoiding perception of the click sound at the moment of the foregoing switching. This is effective in improving the subjective sound quality.

[0067] FIG. 6 is a block diagram illustrating a third exemplary configuration of the filter 310. The filter is configured as an L-tap FIR filter having $c_0, c_1, \ldots, c_{L-1}$ as the coefficient. The received signal 1 of FIG. 1 is supplied to an input terminal 3100 of FIG. 6. The signal to be obtained in an output terminal 3104 of FIG. 6 is the delayed received signal.

[0068] The signal supplied to the input terminal 3100 is transmitted to a delay element 3101_1 and a coefficient multiplier 3102_0.

[0069] The coefficient multiplier 3102_0 multiplies the inputted received signal sample by the coefficient value c_0 and transmits its product to an adder 3103_1. The delay element

3101_1 delays the received signal sample by one sample, and transmits it to a coefficient multiplier 3102_1 and a delay element 3101_2.

[0070] The coefficient multiplier 3102_1 multiplies the output of the delay element 3101_1 by a coefficient value c_1 and transmits its product to an adder 3103_1. The adder 3103_1 adds the output of the coefficient multiplier 3102_0 and the output of the coefficient multiplier 3102_1, and transmits its sum to an adder 3103_2. The delay element 3101_2 delays the output of the delay element 3101_1 by one sample, and transmits it to a coefficient multiplier 3102_2. Hereinafter, this processing is repeated up to a coefficient multiplier 3102_{L-2}.

[0071] A coefficient multiplier 3102_{L-1} multiplies the output of a delay element 3101_{L-1} by a coefficient value c_{L-1} and transmits its product to an adder 3103_{L-1}. The adder 3103_{L-1} adds the output of an adder 3103_{L-2} and the output of the coefficient multiplier 3102_{L-1}, and outputs its sum as the delayed received signal to an output terminal 3104.

[0072] The clock signal, which is supplied to an input terminal 3105 from the analyzing circuit 350 of FIG. 1, is transmitted to the coefficient multiplier 3102_0, the coefficient multiplier 3102_1, . . . , and the coefficient multiplier 3102_{L-1}. Based upon the clock signal supplied from the input terminal 3105, each of the coefficient multiplier 3102_0, the coefficient multiplier 3102_1, . . . , and the coefficient multiplier 3102_{L-1} varies its coefficient value.

[0073] It may be considered that the coefficient $c_0(k)$ of the coefficient multiplier 3102_0, the coefficient $c_1(k)$ of the coefficient multiplier 3102_1, . . . , and the coefficient $c_{L-1}(k)$ of the coefficient multiplier 3102_{L-1} correspond to respective taps of the filter 310 connected in parallel. In other words, the coefficient $c_0(k)$, the coefficient $c_1(k)$, . . . , and the coefficient $c_{L-1}(k)$ have a non-zero value exclusively, and when one coefficient is non-zero, the other coefficients become zero. As explained by using FIG. 3(A) and FIG. 5, $c_0(k)$, $c_1(k)$, . . . , and $c_{L-1}(k)$ have non-zero exclusively to each other, thereby allowing the received signals subjected to the delay, which correspond to respective coefficient multipliers, to be gained as the delayed received signals in the output terminal 3104. Like FIGS. 3(B) and (C) corresponding to FIG. 3(A), $c_0(k)$, $c_1(k)$, . . . , and $c_{L-1}(k)$ can be also set so that they smoothly vary at the moment of varying between a zero value and a non-zero value. A smooth variation in the value yields an effect that the acoustic images that are generated at the moment of mutually switching the received signal and the delayed received signal move smoothly, and the acoustic image movement is hardly perceived. Further, there is also an effect of avoiding perception of the click sound at the moment of the foregoing switching. This is effective in improving the subjective sound quality.

[0074] While the explanation was made so far on the assumption that each of the delay amounts of the delay elements 3101_1, 3101_2, . . . , and 3101_{L-1} was equal to a one-sampling period, the delay amount may be an integer multiple of the sampling period. Further, respective delay elements may be configured to give different delay amounts, respectively. Not limiting the delay amount of each delay element to the one-sampling period makes it possible to efficiently set the delays of the received signals to different various values, respectively.

[0075] Further, while the explanation was made so far on the assumption that the filter 310 had a configuration of the FIR filter, the filter may have the other structures such as a combination of a variable delay circuit and a switch, and a

combination of a variable delay circuit and a variable weighting mixing circuit so long as they have a configuration capable of switching and outputting the received signal and delayed receive signal with a time. Generating a plurality of the delayed received signals by giving different delays to the received signal with a plurality of the variable delay circuits, switching a plurality of these delayed received signals and received signal with the switch in a certain case, and appropriately mixing them with the variable weighting mixing circuit in another case makes it possible to realize a function similar to that of a time-varying-coefficient FIR filter.

[0076] As explained above in details, the best mode of the present invention generates the delayed received signals by delaying at least one received signal, and activates the adaptive filters with the foregoing received signals and the foregoing delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the received signal and the delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0077] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

[0078] The signal processing apparatus of the present invention with two received signals and two transmission signals, according to the second embodiment, is shown in FIG. 7. A difference with the best mode explained by using FIG. 1 and FIG. 6 lies in a point of including a delay processing circuit 301 instead of the delay processing circuit 300. Hereinafter, this difference will be explained in details.

[0079] The delay processing circuit 301 generates the delayed received signals by delaying the received signal 1 and the received signal 2 and transmits them to the adaptive filters 121 and 123 and the digital-analogue (DA) converter 18 as well as the adaptive filters 122 and 124 and the DA converter 19, respectively.

[0080] The delay processing circuit 301 includes filters 310 and 320 and a analyzing circuit 351. The filters 310 and 320 generate the delayed received signals by delaying the received signal 1 and the received signal 2. Further, the filters 310 and 320 output the received signal 1 and the received signal 2 as they stand without delaying them in some cases. A frequency at which the output of the filter 310 varies between the delayed received signal and the received signal 1 or the output of the filter 320 varies between the delayed received signal and the received signal 2 is controlled by the clock signal that is supplied from the analyzing circuit 351. The adapter filterers 121 and 123 as well as the adapter filterers 122 and 124, to which this switching signal is supplied, converge more quickly as this frequency becomes higher. The reason is that a cross correlation between the received signal 1 and the received signal 2 is changed by this switching, and hence a combination of the conditionals representing a relation between the received signals 1 and 2, and the coefficients of the adaptive filters 121 and 123 as well as a combination of

the conditionals representing a relation between the received signals **1** and **2**, and the coefficients of the adaptive filters **122** and **124** are also changed. On the other hand, the acoustic images that are localized by the loudspeakers **3** and **4** move toward the loudspeaker **4** with the switching by the filter **310** and move toward the loudspeaker **3** with the switching by the filter **320**. Hence, a high switching frequency causes the movement of the acoustic image localization to be easily perceived subjectively, which leads to a degradation in the subjective sound quality. That is, there is a trade-off between the convergence rate and the subjective sound quality in terms of the foregoing switching frequency.

[0081] The analyzing circuit **351**, upon receipt of the received signal **1** and the received signal **2**, calculates the perceptual sensitivity to the movement of the acoustic images that are localized by these received signals. The analyzing circuit **351** generates the clock signals corresponding to the obtained perceptual sensitivity, and supplies them to the filters **310** and **320**, respectively. The filters **310** and **320** decide a generating frequency of the delayed received signals based upon the supplied clock signals, respectively. Further, the shift of the phase of the clock signal to be supplied to the filters **310** and that of the phase of the clock signal to be supplied to the filters **320** differ from each other. This phase shift will be explained later by using FIG. **8**.

[0082] A configuration of the filer **320** is completely identical to that of the filter **310** explained by using FIG. **2**, FIG. **4**, and FIG. **6**. Further, the filter **320**, similarly to the filter **310**, may have the other structures such as a combination of a variable delay circuit and a switch, and a combination of a variable delay circuit and a variable weighting mixing circuit so long as they have a configuration capable of switching and outputting the received signal and delayed receive signal with a time. Generating a plurality of the delayed received signals by giving different delays to the received signal with a plurality of the variable delay circuits, switching a plurality of these delayed received signals and received signal with the switch in a certain case, and appropriately mixing them with the variable weighting mixing circuit in another case makes it possible to realize a function similar to that of a time-varying-coefficient FIR filter.

[0083] In FIG. **8**, an example of a variation in $c_0(k)$ in the filter **310** and the filter **320** is shown on the assumption that the filters are configured as a two-tap FIR filter. When $c_0(k)$ of the filter **310** is varied according to FIG. **8**(A), $c_0(k)$ of the filter **320** is varied according to FIG. **8**(B). When $c_0(k)$ of the filter **310** and the filter **320** is varied according to FIGS. **8**(A) and (B), at least a moment that one outputs the received signal and the other outputs the delayed received signal exists. In an example of FIG. **8**, the output of the filter **310** is the received signal and the output of the filter **320** is the delayed received signal at M/2 sample just before 2iM. This status is defined as status **1**. Further, contrarily, the output of the filter **310** is the delayed received signal and the output of the filter **320** is the received signal at M/2 sample just before (2i+1)M. This status is defined as status **2**. The outputs of both of the filters **310** and **320** are the received signals at M/2 sample just after (2i−1)M, and the outputs of both of the filters **310** and **320** are the delayed received signals at M/2 sample just after 2iM. Such a status in which the output of the filter **310** and the output of the filter **320** are identical to the received signals or the delayed received signals all alike is defined as status **3**. The cross correlation between the signal to be supplied to the adaptive filters **121** and **123**, and the signal to be supplied to the

adaptive filters **122** and **124** in the status **3** is equal to that in the case of the apparatus of the linear coupled type. That is, the cross correlation between the signal to be supplied to the adaptive filters **121** and **123**, and the signal to be supplied to the adaptive filters **122** and **124** in the status **3** is equal to that in the case of not utilizing the delayed received signal. Switching this status and the status **1**, and updating the adaptive filter coefficients so that two kinds of the cross correlation statuses are simultaneously satisfied enables the adaptive filter coefficients to converge to the correct coefficients. In addition, combining the status **2**, and updating the adaptive filter coefficients so that "three kinds of the cross correlation statuses" of the status **1**, the status **2**, and the status **3** are simultaneously satisfied enables the adaptive filter coefficients to converge to the correct values at a faster rate as compared with the case of utilizing two kinds of the cross correlation statuses.

[0084] Particularly, when the maximum value of the relative delay of the output signal of the filter **310** to the output signal of the filter **320** is equal to the maximum value of the relative delay of the output signal of the filter **320** to the output signal of the filter **310**, a shift amount of the acoustic image localization to the left caused by a switching to the delayed received signal and a shift amount to the right are equalized to each other, and the acoustic images are perceived as if the acoustic images had fluctuated left-right symmetrically with a time. For example, in the above-mentioned status **1** and status **2**, the relative delay of the output signal of the filter **310** to the output signal of the filter **320**, which is 1, and the relative delay of the output signal of the filter **320** to the output signal of the filter **310**, which is 1, are equal to each other. A degradation in the subjective sound quality is smaller with such a left-right symmetrical fluctuation in the acoustic images localization as compared with the asymmetric movement to either the left or the right because the left-right symmetrical fluctuation is perceived as a blur of the acoustic images.

[0085] The phase of $c_0(k)$ in FIG. **8**(A) and that of $c_0(k)$ in FIG. **8**(B) differ from each other by M/2 sample. This shift of the phase could be a value other than M/2 sample. Adjusting this shift of the phase allows the theoretical convergence time to become shortest when the above-mentioned three kinds of the cross correlation statuses appear equally. Further, a period of a variation in $c_0(k)$ does not need to be always equal to M/2 sample, and an arbitrary value can be selected. The clock signal having this phase shift is generated by the analyzing circuit **351** as already explained.

[0086] With the case that the filter **320** is configured as a three-tap FIR filter shown in FIG. **4**, the coefficient $c_0(k)$ of the coefficient multiplier **3102**$_0$, the coefficient $c_1(k)$ of the coefficient multiplier **3102**$_1$, the coefficient $c_2(k)$ of the coefficient multiplier **3102**$_2$ of the filter **320** will be explained, in contrast to that of the filter **310**. An example of the coefficient $c_0(k)$ of the coefficient multiplier **3102**$_0$, the coefficient $c_1(k)$ of the coefficient multiplier **3102**$_1$, and the coefficient $c_2(k)$ of the coefficient multiplier **3102**$_2$ of the filter **320** corresponding to FIG. **5**, is shown in FIG. **9**. With a relation between FIG. **5** and FIG. **9**, similarly to a relation between FIG. **8**(A) and FIG. **8**(B), the varying point (phase) of the corresponding coefficient value is shifted. Appropriately setting this shift of the phases makes it possible to change the convergence time. Further, as explained in an example of the filter **310**, the coefficient $c_0(k)$ of the coefficient multiplier **3102**$_0$, the coefficient $c_1(k)$ of the coefficient multiplier **3102**$_1$, and the coef-

ficient $c_2(k)$ of the coefficient multiplier 3102_2 can be set so that a variation from non-zero to zero (or the contrary hereto) is proportional to a time in a certain case, and can be set so that they have an arbitrary smooth curved line or straight line for connecting non-zero and zero in another case.

[0087] Also with the case that the filter 320 is configured as an L-tap FIR filter shown in FIG. 6, the filter 310 and the filter 320 differ from each other in the phase of $c_0(k)$ as explained by using FIGS. 8(A) and (B), FIG. 5, and FIG. 9. Appropriately setting this shift of the phases makes it possible to change the convergence time. Further, as explained in an example of the filter 310, the coefficient $c_0(k)$ of the coefficient multiplier 3102_0, the coefficient $c_1(k)$ of the coefficient multiplier 3102_1, and the coefficient $c_{L-1}(k)$ of the coefficient multiplier 3102_{L-1} can be set so that a variation from non-zero to zero (or the contrary hereto) is proportional to a time in a certain case, and can be set so that they have an arbitrary smooth curved line or straight line for connecting non-zero and zero in another case.

[0088] In addition, similarly to the case of the two-tap FIR filter and the three-tap FIR filter, the foregoing coefficient values can be controlled so that the maximum value of the relative delay of the output signal of the filter 310 to the output signal of the filter 320 is equal to the maximum value of the relative delay of the output signal of the filter 320 to the output signal of the filter 310. This condition, more generally, makes it possible to express that the maximum values of the relative delays of the delayed signals to the received signals in the channel that are reproduced by the left and right loudspeakers located remotest from a center are equalized to each other. This condition is equivalent to a difference of the maximum value of the relative delay between the left channel and the right channels being zero. When the foregoing left and right loudspeakers are located asymmetrically with respect to the center, the foregoing difference of the maximum value of the relative delay must be zero in a status of taking into consideration a bias of the acoustic images due to its asymmetry.

[0089] As explained above in details, the second embodiment of the present invention generates the delayed received signals by delaying two received signals or more, and activates the adaptive filters with the foregoing received signals and the foregoing delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the received signal and the delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0090] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

[0091] Further, using two delayed received signals or more makes it possible to furthermore increase the number of the foregoing conditionals and to shorten the convergence time of the solutions to the optimum values. In addition, generating the delayed received signals so that a difference between the left and right channels of the maximum value of a relative

delay of the delayed signal in a channel, which is reproduced by each of the left and right loudspeakers located remotest from a center, to the received signal is zero in a status of taking into consideration a bias of the acoustic images due to the left-right asymmetry in the arrangement of the foregoing left and right loudspeakers enables the shift amounts of the acoustic image localization to the left and the right caused by the delayed received signals to be equalized with each other, and a degradation in the subjective sound quality to be made small.

[0092] The signal processing apparatus of the present invention with two received signals and two transmission signals, according to the third embodiment, is shown in FIG. 10. A difference with the second embodiment explained by using FIG. 7 to FIG. 9 lies in a point that the output signals of a delay processing circuit 301 are processed by an amplitude correcting circuit 400, and then, supplied to the adaptive filters 121, 123, 122, and 124.

[0093] The amplitude correcting circuit 400 generates amplitude-corrected delayed received signals by correcting the amplitudes of the delayed received signals, being outputs of the delay processing circuit 301, and transmits them to the adaptive filters 121 and 123 and the digital-analogue (DA) converter 18 as well as the adaptive filters 122 and 124 and the DA converter 19, respectively.

[0094] The amplitude correction of the delayed received signals in the amplitude correcting circuit 400 is performed when the output of the delay processing circuit 301 is equal to the delayed received signal obtained by delaying the received signal 1 or the received signal 2. The amplitude correction makes it possible to correct a correlation of the amplitude of the signal between a plurality of channels, and to cancel the shift of the acoustic image localization generated at the moment of using the delayed received signals instead of the received signals. It is also possible to equalize the total power after the correction to the total power before the correction by making the correction for all channels at the moment of the amplitude correction. Maintaining the total power of all channels at a constant level can eliminate the subjective feeling of disorder at the moment that the amplitude-corrected signal and the amplitude-not-corrected signal are switched.

[0095] The amplitude correcting circuit 400 includes amplitude processing circuits 410 and 420. The amplitude processing circuit 410 corrects the amplitude of the delayed received signal generated by delaying the received signal 1, thereby to generate the amplitude-corrected delayed received signal. The amplitude processing circuit 420 corrects the amplitude of the delayed received signal generated by delaying the received signal 2, thereby to generate the amplitude-corrected delayed received signal. Each of the amplitude processing circuits 410 and 420 can assume a completely identical configuration. The clock signals are supplied to the amplitude processing circuits 410 and 420 from the analyzing circuit 351 that is included in the delay processing circuit 301. These clock signals are used for applying the amplitude correction according to the timing in which the delayed signals are generated in the delay processing circuit 301.

[0096] FIG. 11 is a block diagram illustrating an exemplary configuration of the amplitude processing circuit 410. The amplitude processing circuit 410 is configured as a multiplier 4101 having g_0 as a coefficient. The delayed received signal obtained by delaying the received signal 1 is supplied to an input terminal 4100 of FIG. 11. The multiplier 4101 increase the signal supplied to an input terminal 4100 by a factor of g_0,

and transmits it to an output terminal **4104**. The signal to be obtained in the output terminal **4104** of FIG. **11** is the signal obtained by increasing the delayed received signal supplied to the input terminal **4100** by a factor of g_0.

[0097] The amplitude processing circuit **420** can assume a configuration that is obtained by using g_1 instead of g_0 as the coefficient of the multiplier **4101** in FIG. **11**, being a block diagram illustrating an exemplary configuration of the amplitude processing circuit **410**. g_0 and g_1 have 1 when the received signal **1** and the received signal **2** are supplied to the amplitude processing circuit **410** and the amplitude processing circuit **420**, respectively, and otherwise, have a value other than 1 (g_0-bar and g_1-bar). Such a value for compensating the shift of the acoustic image localization caused by the delayed received signal is set to g_0-bar and g_1-bar. Further, the setting can be also made so that the total power after the correction is equalized to the total power before the correction. Maintaining the total power of all channels at a constant level can eliminate the subjective feeling of disorder at the moment that the amplitude-corrected signal and the amplitude-not-corrected signal are switched.

[0098] The amplitude processing circuit **410** and the amplitude processing circuit **420** operate complementarily. That is, the movement of the acoustic images is corrected by g_0-bar and g_1-bar. The principle of correcting the movement of the acoustic images caused by a change in the delay amount by the amplitude correction is disclosed in Non-patent Literature 6 ("Medical Research Council Special Report", No. 166, 1932, pp. 1-32), Non-patent Literature 7 ("Journal of Acoustical Society of America", Vol. 32, 1960, pp. 685-692), and Non-patent Literature 8 ("Journal of Acoustical Society of America", Vol. 94, 1993, pp. 98-110).

[0099] In an example of FIG. **10**, when the acoustic images of the acoustic signals to be reproduced for a talker **11** by loudspeakers **3** and **4** move toward the loudspeaker **3**, in order to correcting this and to return the acoustic images to an original status, the amplitudes of the signals to be radiated from the loudspeaker **4** in the acoustic space are increased, and simultaneously, the amplitudes of the signals to be radiated from the loudspeaker **3** in the acoustic space are decreased.

[0100] According to the Non-patent Literature 8, in order to move the acoustic image by the amplitude correction, with the total power of the received signal **1** and the received signal **2** maintained constant, the following numerical equation 2 must hold between respective powers P_1[dB] and P_2[dB].

$$P_1 + P_2 = C \qquad \text{<Numerical equation 2>}$$

[0101] where C is a positive constant. Hence, when the powers of the received signal **1** and the received signal **2** before the amplitude correction are P_1–bar[dB] and P_2–bar[dB], respectively, the following numerical equation 3 must hold for the powers P_1[dB] and P_2[dB] of the received signal **1** and the received signal **2** after amplitude correction.

$$P_1 = P_1\text{-bar} - \Delta P/2$$

$$P_2 = P_2\text{-bar} - \Delta P/2 \qquad \text{<Numerical equation 3>}$$

[0102] where ΔP is a power correction amount. For this reason, the values of the coefficients g_0-bar and g_1-bar of the multiplier **4101** can be determined with the following numerical equation from the numerical equation 3.

$$g_0\text{-bar} = 10^{-\Delta Pi/40}$$

$$g_1\text{-bar} = 10^{\Delta Pi/40} \qquad \text{<Numerical equation 4>}$$

[0103] where ΔP_i is a power compensation coefficient required to compensate the received signals delayed by i samples.

[0104] Additionally, in the explanation made so far, according to FIG. **10**, the configuration was explained of generating the delayed received signals by processing the received signals with the delay processing circuit **301**, generating the amplitude-corrected delayed received signals by correcting the amplitudes of the delayed received signals with the amplitude correcting circuit **400**, and supplying them to the adaptive filters **121**, **123**, **122**, and **124**. On the other hand, it is also possible to assume the configuration in which the order of the processing of the received signals is exchanged, namely the configuration of generating the amplitude-corrected received signals by correcting the amplitudes of the received signals with the amplitude correcting circuit **400**, generating the amplitude-corrected delayed received signals by processing the amplitude-corrected received signals with the delay processing circuit **301**, and supplying them to the adaptive filters **121**, **123**, **122**, and **124**. Configurations and operations of the delay processing circuit **301** and the amplitude correcting circuit **400** at that moment have been already explained, so its explanation is omitted herein.

[0105] As explained above in details, the third embodiment of the present invention generates the delayed received signals by delaying two received signals or more, and activates the adaptive filters with the foregoing received signals and the foregoing delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the received signal and the delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0106] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

[0107] Further, using two delayed received signals or more makes it possible to furthermore increase the number of the foregoing conditionals and to shorten the convergence time of the solutions to the optimum values. In addition, generating the delayed received signals so that a difference between the left and right channels of the maximum value of a relative delay of the delayed signal in a channel, which is reproduced by each of the left and right loudspeakers located remotest from a center, to the received signal is zero in a status of taking into consideration a bias of the acoustic images due to the left-right asymmetry in the arrangement of the foregoing left and right loudspeakers enables the shift amounts of the acoustic image localization to the left and the right caused by the delayed received signals to be equalized with each other, and a degradation in the subjective sound quality to be made small.

[0108] Further, a degradation in the sound quality of the audible received signals directly supplied to the loudspeaker is suppressed so that the excellent sound quality can be maintained because the acoustic image movement caused by the

introduction of the delayed received signals is offset by the process of correcting the amplitudes of the inputted signals.

[0109] The signal processing apparatus of the present invention with two received signals and two transmission signals, according to the fourth embodiment, is shown in FIG. 12. A difference with the third embodiment explained by using FIG. 10 and FIG. 11 lies in a point that the output signals of the amplitude correcting circuit 400 are processed by a non-linear processing circuit 500, and then supplied to the adaptive filters 121, 123, 122, and 124.

[0110] The non-linear processing circuit 500 generates non-linear amplitude-corrected delayed received signals by non-linearly processing the amplitude-corrected delayed received signals, being outputs of the amplitude correcting circuit 400, and transmits them to the adaptive filters 121 and 123 and the digital-analogue (DA) converter 18 as well as the adaptive filters 122 and 124 and the DA converter 19, respectively. The non-linear amplitude-corrected delayed received signals are smaller in the cross correlation between a plurality of the channels than the amplitude-corrected delayed received signals. Hence, the convergence of the adaptive filters 121, 123, 122, and 124 can be made yet faster.

[0111] The non-linear processing circuit 500 includes non-linear amplitude processing circuits 510 and 520. The non-linear amplitude processing circuit 510 non-linearly processes the amplitude of the amplitude-corrected delayed received signal obtained by delaying the received signal 1 and then correcting the amplitude thereof, thereby to generate the non-linear amplitude-corrected delayed received signal. The non-linear amplitude processing circuit 520 non-linearly processes the amplitude of the amplitude-corrected delayed received signal obtained by delaying the received signal 2 and then correcting the amplitude thereof, thereby to generate the non-linear amplitude-corrected delayed received signal. Each of the non-linear amplitude processing circuits 510 and 520 can assume a completely identical configuration.

[0112] FIG. 13 is a block diagram illustrating an exemplary configuration of the non-linear amplitude processing circuit 510. The non-linear amplitude processing circuit 510 is configured of a coefficient multiplier 512, a polarity determining circuit 513, a multiplier 514, and an adder 515. The amplitude-corrected delayed received signals, being outputs of the amplitude correcting circuit 400 of FIG. 12, are supplied to an input terminal 511. The amplitude-corrected delayed received signals are transmitted to the coefficient multiplier 512, the polarity determining circuit 513, and the adder 515. The coefficient multiplier 512 increases its input signal by a factor of α, and outputs it. The polarity determining circuit 513 outputs 1 when the polarity of the signal supplied to the input is positive, and outputs 0 when it is negative. The multiplier 514, to which the output of the coefficient multiplier 512 and the output of the polarity determining circuit 513 are supplied, transmits a product of both to the adder 515. The amplitude-corrected delayed received signals are supplied to another input terminal of the adder 515 as they stand. That is, it follows that the output of the adder 515 for a signal sample x(k) in an input terminal 511 is $(1+\alpha)$x(k) when the polarity of the input signal is positive, and is x(k) when it is negative. This signal becomes the output signal of the non-linear amplitude processing circuits 510. That is, the non-linear amplitude processing circuits 510 constitutes a half-wave rectifier circuit. The non-linear amplitude processing circuits 520 can assume a configuration completely identical to that of the non-linear amplitude processing circuits 510.

[0113] Additionally, in the explanation made so far, according to FIG. 12, the configuration was explained of generating the delayed received signals by processing the received signals with the delay processing circuit 301, generating the amplitude-corrected delayed received signals by correcting the amplitudes of the delayed received signals with the amplitude correcting circuit 400, generating the non-linear amplitude-corrected delayed received signals by processing the amplitude-corrected delayed received signals with the non-linear amplitude processing circuit 500, and supplying them to the adaptive filters 121, 123, 122, and 124. On the other hand, it is also possible to assume the configuration in which the order of the processing of the received signals is exchanged, namely the configuration of, after processing the received signals in the order of the amplitude correction, the delay, and the non-linear processing, or in the order of the non-linear processing, the delay, and the amplitude correction, supplying them to the adaptive filters 121, 123, 122, and 124. Configurations and operations of the delay processing circuit 301, the amplitude correcting circuit 400, and the non-linear processing circuit 500 at that moment have been already explained, so its explanation is omitted herein.

[0114] As explained above in details, the fourth embodiment of the present invention generates the delayed received signals by delaying two received signals or more, generates the amplitude-corrected delayed received signals by correcting the amplitudes of the delayed received signals, generates the non-linear amplitude-corrected delayed received signals by non-linearly processing the amplitude-corrected delayed received signals, and activates the adaptive filters with the foregoing received signals and the foregoing non-linear amplitude-corrected delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the received signal and the non-linear amplitude-corrected delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0115] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

[0116] Further, using a plurality of the delayed received signals makes it possible to furthermore increase the number of the foregoing conditionals and to shorten the convergence time of the solutions to the optimum values. In addition, generating the delayed received signals so that a difference between the left and right channels of the maximum value of a relative delay of the delayed signal in a channel, which is reproduced by each of the left and the right loudspeakers located remotest from a center, to the received signal, is zero in a status of taking into consideration a bias of the acoustic images due to the left-right asymmetry in the arrangement of the foregoing left and right loudspeakers enables the shift amounts of the acoustic image localization to the left and the

right caused by the delayed received signals to be equalized with each other, and a degradation in the subjective sound quality to be made small.

[0117] Further, a degradation in the sound quality of the audible received signals directly supplied to the loudspeaker is suppressed so that the excellent sound quality can be maintained because the acoustic image movement caused by the introduction of the delayed received signals is offset by the process of correcting the amplitudes of the inputted signals. In addition, the convergence time can be shortened all the more by a synergistic effect of the non-linear processing and the introduction of the delayed received signals.

[0118] The signal processing apparatus of the present invention with two received signals and two transmission signals, according to the fifth embodiment, is shown in FIG. 14. A difference with the fourth embodiment explained by using FIG. 12 and FIG. 13 lies in a point that the non-linear processing circuit 500 is replaced with a non-linear processing circuit 501.

[0119] The non-linear processing circuit 501 includes non-linear amplitude processing circuits 530 and 540. The non-linear amplitude processing circuit 530 non-linearly processes the amplitude-corrected delayed received signal obtained by delaying the received signal 1 and then correcting the amplitude thereof by using the amplitude-corrected delayed received signal obtained by delaying the received signal 2 and then correcting the amplitude thereof, thereby to generate the non-linear amplitude-corrected delayed received signals. The non-linear amplitude processing circuit 540 non-linearly processes the amplitude-corrected delayed received signal obtained by delaying the received signal 2 and then correcting the amplitude thereof by using the amplitude-corrected delayed received signal obtained by delaying the received signal 1 and then correcting the amplitude thereof, thereby to generate the non-linear amplitude-corrected delayed received signals. Each of the non-linear amplitude processing circuits 530 and 540 can assume a completely identical configuration.

[0120] FIG. 15 is a block diagram illustrating an exemplary configuration of the non-linear amplitude processing circuit 530. The non-linear amplitude processing circuit 530 is configured of a coefficient multiplier 512, a polarity determining circuit 513, a multiplier 514, and an adder 515. The amplitude-corrected delayed received signal obtained by delaying the received signal 1 and then correcting the amplitude thereof, out of the outputs of the amplitude correcting circuit 400 of FIG. 14, is supplied to an input terminal 531. The amplitude-corrected delayed received signal obtained by delaying the received signal 2 and then correcting the amplitude thereof, out of the outputs of the amplitude correcting circuit 400 of FIG. 14, is supplied to an input terminal 537. The amplitude-corrected delayed received signal generated from the received signal 1 is transmitted to the polarity determining circuit 513 and the adder 515. The amplitude-corrected delayed received signal generated from the received signal 2 is transmitted to the coefficient multiplier 512. The coefficient multiplier 512 increases its input signal by a factor of α, and outputs it. The polarity determining circuit 513 outputs 1 when the polarity of the signal supplied to the input is positive, and outputs 0 when it is negative. The multiplier 514, to which the output of the coefficient multiplier 512 and the output of the polarity determining circuit 513 are supplied, transmits a product of both to the adder 515. The amplitude-corrected delayed received signal generated from

the received signal 1 is supplied to another input terminal of the adder 15 as it stands. That is, it follows that the output of the adder 515 for a signal sample $x_1(k)$ in an input terminal 531 and for a signal sample $x_2(k)$ in an input terminal 537 is $x_1(k)+\alpha x_2(k)$ when the polarity of the input signal is positive, and is $x_1(k)$ when it is negative. This signal becomes the output signal of the non-linear amplitude processing circuits 530. The non-linear amplitude processing circuits 530 has the configuration in which the input of the coefficient multiplier 512 in the non-linear amplitude processing circuits 510 has been changed from the amplitude-corrected delayed received signal generated from the received signal 1 to the amplitude-corrected delayed received signal generated from the received signal 2. The non-linear amplitude processing circuits 540 can assume a configuration completely identical to that of the non-linear amplitude processing circuits 530. In this configuration, the variation from the signal not subjected to the non-linear processing becomes large and an effect of reducing the correlation between the channels becomes large because the signal generated from the received signal of another channel is used for the non-linear processing.

[0121] Additionally, in the explanation made so far, according to FIG. 14, the configuration was explained of generating the delayed received signals by processing the received signals with the delay processing circuit 301, generating the amplitude-corrected delayed received signals by correcting the amplitudes of the delayed received signals with the amplitude correcting circuit 400, generating the non-linear amplitude-corrected delayed received signals by processing the amplitude-corrected delayed received signals with the non-linear amplitude processing circuit 501, and supplying them to the adaptive filters 121, 123, 122, and 124. On the other hand, it is also possible to assume the configuration in which the order of the processing of the received signals is exchanged, namely the configuration of, after processing the received signals in the order of the amplitude correction, the delay, and the non-linear processing, or in the order of the non-linear processing, the delay, and the amplitude correction, supplying them to the adaptive filters 121, 123, 122, and 124. Configurations and operations of the delay processing circuit 301, the amplitude correcting circuit 400, and the non-linear processing circuit 501 at that moment have been already explained, so its explanation is omitted herein.

[0122] As explained above in details, the fifth embodiment of the present invention generates the delayed received signals by delaying two received signals or more, generates the amplitude-corrected delayed received signals by correcting the amplitudes of the delayed received signals, generates the non-linear amplitude-corrected delayed received signals by non-linearly processing the amplitude-corrected delayed received signals, and activates the adaptive filters with the foregoing received signals and the foregoing non-linear amplitude-corrected delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the received signal and the non-linear amplitude-corrected delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0123] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the

foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

[0124] Further, using a plurality of the delayed received signals makes it possible to furthermore increase the number of the foregoing conditionals and to shorten the convergence time of the solutions to the optimum values. In addition, generating the delayed received signals so that a difference between the left and right channels of the maximum value of a relative delay of the delayed signal in a channel, which is reproduced by each of the left and right loudspeakers located remotest from a center, to the received signal is zero in a status of taking into consideration a bias of the acoustic images due to the left-right asymmetry in the arrangement of the foregoing left and right loudspeakers enables the shift amounts of the acoustic image localization to the left and the right caused by the delayed received signals to be equalized with each other, and a gradation in the subjective sound quality to be made small. Further, a degradation in the sound quality of the audible received signals directly supplied to the loudspeaker is suppressed so that the excellent sound quality can be maintained because the acoustic image movement caused by the introduction of the delayed received signals is offset by the process of correcting the amplitudes of the inputted signals. In addition, the convergence time can be shortened all the more by a synergistic effect of the non-linear processing using the received signals of a plurality of channels and the introduction of the delayed received signals.

[0125] The signal processing apparatus of the present invention with two received signals and two transmission signals, according to the sixth embodiment, is shown in FIG. 16. A difference with the second embodiment explained by using FIG. 7 to FIG. 9 lies in a point that a frequency analysis synthesizing circuit 600 is provided upstream of the delay processing circuit 301, and that a frequency analysis synthesizing circuit 610 is provided upstream of the DA converters 18 and 19 as well as downstream of the AD converters 20 and 21. Hence, all of the delay processing circuit 301, the adapter filters 121, 122, 123, and 124, and the subtracters 129 and 130 are operative in response to band-divided narrow band signals. The frequency analysis synthesizing circuit 600 band-divided the received signals 1 and 2, and transmits them to the delay processing circuit 301. The frequency analysis synthesizing circuit 600 also band-synthesizes the outputs of the subtracters 129 and 130, and constitutes all-band output signals 16 and 17. The frequency analysis synthesizing circuit 610 band-synthesizes the outputs of the delay processing circuit 301, and transmits them to the DA converters 18 and 19. The frequency analysis synthesizing circuit 610 also band-divides the outputs of the AD converters 20 and 21, and transmits them to the subtracters 129 and 130. The delay processing circuit 301 adds the delays to the band-divided signals, and outputs them as band-divided delayed received signals. The sixth embodiment enables the optimum delays to be given to the band-divided signals, respectively. Hence, it leads to an increase in a degree of freedom at the moment of selecting the relative delay that is as large as possible within the allowable movement amount of the acoustic images, and an improvement in the subjective sound quality, which was explained by using FIG. 1.

[0126] The frequency analysis function of the frequency analysis synthesizing circuits 600 and 610 can be realized by applying a frequency conversion for the input signal sample divided into the frames. As an example of the frequency conversion, a Fourier transform, a cosine transform, a KL (Karhunen Loeve) transform, etc. are known. The technology related to a specific arithmetic operation of these transforms, and its properties are disclosed in Non-patent Literature 9 (DIGITAL CODING OF WAVEFORMS, PRINCIPLES AND APPLICATIONS TO SPEECH AND VIDEO, PREN-TICE-HALL, 1990). Further, it is publicly known that other conversions such as a Hadamard transform, a Haar transform, and a wavelet transform can be used.

[0127] The foregoing frequency analysis function can be also realized by applying the foregoing transforms for a result obtained by weighting the input signal samples of the above frame with a window function W. As such a window function, the window functions such as a Hamming window, a Hanning (Hann) window, a Kaiser window, and a Blackman window are known. Further, more complicated window functions also can be used. The technology related to these window functions is disclosed in Non-patent Literature 10 and Non-patent Literature 11. In addition, the windowing as well by partially overlapping two continuous frames or more is widely carried out. In this case, the foregoing frequency transforms are used for the signal subjected to the overlap windowing. The technology related to the blocking involving the overlap and the conversion is disclosed in the Non-patent Literature 10 (DIGITAL SIGNAL PROCESSING, PRENTICE-HALL, 1975).

[0128] In addition, the frequency analysis function of the frequency analysis synthesizing circuits 600 and 610 may be configured of a band-division filter bank. The band-division filter bank is configured of a plurality of band-pass filters. An interval of each frequency band of the band-division filter bank could be equal in a certain case, and unequal in another case. Carrying out the band division at an unequal interval makes it possible to lower/raise a time resolution, that is, the time resolution can be lowered by carrying out the division into narrows bands with regard to a low-frequency area, and the time resolution can be raised by carrying out the division into wide bands with regard to a high-frequency area. As a typified example of the unequal-interval division, there exists an octave division in which the band gradually halves toward the low-frequency area, a critical band division that corresponds to an auditory feature of a human being, or the like. After dividing into the frequency bands having an equal interval, a hybrid filter bank may be used for furthermore carrying out the band division only with regard to a low-frequency area in order to enhance the frequency resolution of the frequency bands in a low-frequency area. The technology related to the band-division filter bank and its design method is disclosed in the Non-patent Literature 11 (MULTIRATE SYSTEMS AND FILTER BANKS, PRENTICE-HALL, 1993).

[0129] The frequency synthesis function of the frequency analysis synthesizing circuits 600 and 610 is configured of an inverse conversion corresponding to the frequency conversion for realizing the frequency analysis function of the frequency analysis synthesizing circuits 600 and 610. When the frequency analysis function of the frequency analysis synthesizing circuits 600 and 610 includes the weighting by a window function W, the frequency-synthesized signals are multiplied by the window function W. When the frequency analysis function of the frequency analysis synthesizing cir-

cuits **600** and **610** is configured of the band-division filter bank, the frequency synthesis function of the frequency analysis synthesizing circuits **600** and **610** is configured of a band-synthesis filter bank. The technology related to the band-synthesis filter bank and its design method is disclosed in the Non-patent Literature 11.

[0130] Additionally, it is self-evident that a processing similar to the processing so far explained can be performed for the band-divided signals by combining the frequency analysis synthesizing circuits **600** and **610**, and any of the best mode and the third embodiment to the fifth embodiment of the present invention.

[0131] As explained above in details, the sixth embodiment of the present invention generates the band-divided received signals by frequency-analyzing two received signals or more, generates the band-divided delayed received signals by delaying the above band-divided received signals, and activates the adaptive filters with the foregoing band-divided received signals and the foregoing band-divided delayed received signals taken as an input, respectively. The number of conditionals at the moment of obtaining the adaptive filter coefficients is increased because both of the band-divided received signal and the band-divided delayed received signal are used, and thus, the problem that the solutions become indefinite does not occur. Hence, the adaptive filter coefficients converge to the optimum values that are uniquely determined.

[0132] Further, a frequency of inputting the foregoing received signals and the foregoing delayed received signals into the foregoing adaptive filters is controlled based upon a perceptual sensitivity to a change in the localization of the foregoing plurality of received signals. This enables the foregoing received signals and the foregoing delayed received signals to be inputted into the foregoing adaptive filters according to a status of the signals at a frequency with which the subjective perception is avoided, and a degradation in the subjective sound quality to be made small.

[0133] Further, using a plurality of the delayed received signals makes it possible to furthermore increase the number of the foregoing conditionals and to shorten the convergence time of the solutions to the optimum values. In addition, generating the delayed received signals so that a difference between the left and right channels of the maximum value of a relative delay of the delayed signal in a channel, which is reproduced by each of the left and right loudspeakers located remotest from a center, to the received signal is zero in a status of taking into consideration a bias of the acoustic images due to the left-right asymmetry in the arrangement of the foregoing left and right loudspeakers enables the shift amounts of the acoustic image localization to the left and the right caused by the delayed received signals to be equalized with each other, and a degradation in the subjective sound quality to be made small.

[0134] Further, the sixth embodiment enables the optimum delays to be given to the band-divided signals, respectively, which leads to an increase in a degree of freedom at the moment of selecting the relative delay that is as large as possible within the allowable movement amount of the acoustic images, and an improvement in the subjective sound quality.

[0135] In the best mode and the second to sixth embodiments above, while the echo cancellation, with a multi-channel teleconference system as a target, was discussed, a similar discussion holds also in a single-channel multipoint telecon-

ferencing system, being another application field of the signal processing. Normally, the single-channel multipoint teleconferencing system performs the process of suitably adding an attenuation and a time delay to the voice of a talker received by a single microphone such that the talker is localized in a desired position between a plurality of loudspeakers to be used at the receive side. The signals processed in such a manner, of which the number is equivalent to that of the number of the loudspeakers at the receive side, are generated. When the number of the loudspeakers at the receive side is equal to 2, the two kinds of the signals, to which the above-mentioned attenuation and delay have been added in the embodiments shown in FIG. **1**, FIG. **7**, FIG. **10**, FIG. **12**, FIG. **14**, and FIG. **16**, correspond to the first received signal **1** and the second received signal **2**. Hence, the embodiments of the present invention can be applied without any change.

[0136] While the case of using the first received signal **1** and the second received signal **2**, and the first mixed signal **14** and the second mixed signal **15** was exemplified herein for explanation, the present invention is applicable to the general case that plural received signals and a single or plural transmission signals exist. Further, even though the description was performed with an example of the acoustic echoes in which the received signal propagated from the loudspeakers to the microphone via the spatial acoustic paths and the acoustic echoes received by the microphone were cancelled, the present invention is applicable to an application for canceling the echoes other than the acoustic echoes, for example, the echoes caused by the crosstalk etc. in a transmission line. Infinite impulse response adaptive filters may be used instead of the finite impulse response adaptive filters. Further, subband adaptive filters or transform-domain adaptive filters may be used.

[0137] Continuously, the seventh embodiment of the present invention will be explained in details by making a reference to FIG. **17**. The seventh embodiment of the present invention includes a computer **1000** that is operative under a program control. The computer **1000** is operative based upon a program for performing the processing related to any of the above-mentioned best mode and second embodiment to sixth embodiment for the received signals received from input terminals **1** and **2**, and outputting the signals of which the echoes have been canceled as output signals **16** and **17**.

[0138] The first example is characterized in that a signal processing method of receiving a plurality of received signals, and subtracting pseudo echoes generated by a plurality of adaptive filters having said plurality of received signals as an input, respectively, from a plurality of echoes to be generated from said plurality of received signals, thereby to reduce said plurality of echoes, comprising: generating delayed received signals by delaying at least one received signal, out of said plurality of received signals; generating pseudo echoes by inputting said received signals and said delayed received signals into said adaptive filters; and controlling a frequency of inputting said received signals and said delayed received signals into said adaptive filters based upon a perceptual sensitivity to a change in localization of said plurality of received signals.

[0139] The second example in the above-mentioned example is characterized in that at least one signal, out of said delayed received signals, is an amplitude-corrected delayed received signal subjected to an amplitude correction.

[0140] The third example in the above-mentioned examples is characterized in that at least one signal, out of the

signals to be inputted into said plurality of adaptive filters, is a non-linearly processed signal subjected to a non-linear processing.

[0141] The fourth example in the above-mentioned examples is characterized in that the signal processing method comprising decomposing said received signal into a plurality of frequency components, and generating the delayed received signals by delaying the received signal for every above plurality of frequency components.

[0142] The fifth example in the above-mentioned examples is characterized in that the perceptual sensitivity to a change in localization is obtained based upon a similarity degree of the received signal.

[0143] The sixth example in the above-mentioned examples is characterized in that the perceptual sensitivity to a change in localization is obtained based upon a power of the received signal.

[0144] The seventh example in the above-mentioned examples is characterized in that the signal processing method comprising generating the delayed received signals so that relative delays of said delayed received signals to said received signals have a plurality of values that vary with a time.

[0145] The eighth example in the above-mentioned examples is characterized in that the relative delay is an integer multiple of a sampling period.

[0146] The ninth example in the above-mentioned examples is characterized in that the delayed received signals are generated by processing the received signals with a filter having a plurality of time-varying coefficients with alternately a zero value or a non-zero value.

[0147] The tenth example in the above-mentioned examples is characterized that the time-varying coefficients have a zero value exclusively to each other.

[0148] The eleventh example in the above-mentioned examples is characterized in that the time-varying coefficients have a non-zero value exclusively to each other.

[0149] The twelfth example is characterized in that a signal processing apparatus for receiving a plurality of received signals, and subtracting pseudo echoes generated by a plurality of adaptive filters having said plurality of received signals as an input, respectively, from a plurality of echoes to be generated from said plurality of received signals, thereby to reduce said plurality of echoes, comprising: a linear processing circuit for generating delayed received signals by delaying at least one received signal, out of said plurality of received signals; an adaptive filter for generating pseudo echoes by receiving said received signals and said delayed received signals, a plurality of subtracters each generating echo-reduced signals by subtracting said pseudo echoes from a plurality of mixed signals; and an analyzing circuit for obtaining a perceptual sensitivity to a change in localization of said plurality of received signals, and wherein said signal processing apparatus controlling a frequency of inputting said received signals and said delayed received signals into said adaptive filters based upon said perceptual sensitivity, and controlling coefficients of said plurality of adaptive filters so that outputs of said plurality of subtracters are minimized.

[0150] The thirteenth example in the above-mentioned example is characterized in that the signal processing apparatus comprising an amplitude correcting circuit for generating amplitude-corrected delayed received signals by amplitude-correcting at least one signal, out of said delayed received signals.

[0151] The fourteenth example in the above-mentioned examples is characterized in that the signal processing apparatus comprising a non-linear processing circuit for generating non-linearly processed signals by non-linearly processing at least one signal, out of the signals to be inputted into said plurality of adaptive filters.

[0152] The fifteenth example in the above-mentioned examples is characterized in that the signal processing apparatus comprising: a frequency analyzing circuit for decomposing said received signal into a plurality of frequency components; and a linear processing circuit for generating the delayed received signals by delaying the received signal for every said plurality of frequency components.

[0153] The sixteenth example in the above-mentioned examples is characterized in that the analyzing circuit obtains said perceptual sensitivity to a change in localization based upon a similarity degree of the received signal.

[0154] The seventeenth example in the above-mentioned examples is characterized in that the analyzing circuit obtains said perceptual sensitivity to a change in localization based upon a power of the received signal.

[0155] The eighteenth example in the above-mentioned examples is characterized in that the linear processing circuit performs a processing such that relative delays of said delayed received signals to said received signals have a plurality of values that vary with a time.

[0156] The nineteenth example in the above-mentioned examples is characterized in that the linear processing circuit performs a processing such that said relative delay is an integer multiple of a sampling period.

[0157] The twentieth example in the above-mentioned examples is characterized in that the linear processing circuit comprises a filter having a plurality of time-varying coefficients with alternately a zero value or a non-zero value.

[0158] The twenty-first example in the above-mentioned examples is characterized in that the time-varying coefficients have a zero value exclusively to each other.

[0159] The twenty-second example in the above-mentioned examples is characterized in that the time-varying coefficients have a non-zero value exclusively to each other.

[0160] The twenty-third example is characterized in that a signal processing program for causing a computer to execute a receiving process of receiving a plurality of received signals, and an echo reducing process of reducing a plurality of echoes that are generated by said plurality of received signals, said signal processing program comprising: a delayed received signal generating process of generating delayed received signals by delaying at least one received signal, out of said plurality of received signals; a pseudo echo generating process of generating pseudo echoes by inputting said received signals and said delayed received signals into said adaptive filters; and a pseudo echo subtracting process of subtracting said pseudo echoes from said plurality of received signals, respectively, wherein a frequency of inputting said received signals and said delayed received signals into said adaptive filters is controlled based upon a perceptual sensitivity to a change in localization of said plurality of received signals.

[0161] The twenty-fourth example in the above-mentioned example is characterized in that at least one signal, out of said delayed received signals, is an amplitude-corrected delayed received signal subjected to an amplitude correction.

[0162] The twenty-fifth example in the above-mentioned examples is characterized in that at least one signal, out of the

signals to be inputted into said plurality of adaptive filters, is a non-linearly processed signal subjected to a non-linear processing.

[0163] The twenty-sixth example in the above-mentioned examples is characterized in that the signal processing program comprising decomposing said received signal into a plurality of frequency components and generating the delayed received signals by delaying the received signal for every above plurality of frequency components.

[0164] The twenty-seventh example in the above-mentioned examples is characterized in that the perceptual sensitivity to a change in localization is obtained based upon a similarity degree of the received signal.

[0165] The twenty-eight example in the above-mentioned examples is characterized in that the perceptual sensitivity to a change in localization is obtained based upon a power of the received signal.

[0166] The twenty-nine example in the above-mentioned examples is characterized in that the delayed received signals are generated so that relative delays of said delayed received signals to said received delays have a plurality of values that vary with a time.

[0167] The thirty example in the above-mentioned examples is characterized in that the relative delay is an integer multiple of a sampling period.

[0168] The thirty-first example in the above-mentioned examples is characterized in that the delayed received signals are generated by processing the received signals with a filter having a plurality of time-varying coefficients with alternately a zero value or a non-zero value.

[0169] The thirty-second example in the above-mentioned examples is characterized in that the time-varying coefficients have a zero value exclusively to each other.

[0170] The thirty-third example in the above-mentioned examples is characterized in that the time-varying coefficients have a non-zero value exclusively to each other.

[0171] Above, although the present invention has been particularly described with reference to the preferred embodiments and examples thereof, it should be readily apparent to those of ordinary skill in the art that the present invention is not always limited to the above-mentioned embodiment and examples, and changes and modifications in the form and details may be made without departing from the spirit and scope of the invention.

[0172] This application is based upon and claims the benefit of priority from Japanese patent application No. 2008-247273, filed on Sep. 26, 2008, the disclosure of which is incorporated herein in its entirety by reference.

REFERENCE SIGNS LIST

[0173]   **1** and **2** received signals
[0174]   **3** and **4** loudspeakers
[0175]   **5, 6, 7,** and **8** echoes
[0176]   **9** and **10** microphones
[0177]   **11** talker
[0178]   **12** and **13** transmission signals
[0179]   **14** and **15** mixed signals
[0180]   **16** and **17** output signals of a signal processing apparatus
[0181]   **18** and **19** digital-analogue converters
[0182]   **20** and **21** analogue-digital converters
[0183]   **121, 122, 123,** and **124** adaptive filters
[0184]   **125, 126, 127,** and **128** pseudo echoes
[0185]   **129** and **130** subtracters

[0186]   **300** and **301** delay processing circuits
[0187]   **310** and **320** filters
[0188]   **330** and **430** clock changing circuits
[0189]   **350** and **351** analyzing circuits
[0190]   **400** amplitude correcting circuit
[0191]   **410** and **420** amplitude processing circuits
[0192]   **500** and **501** non-linear processing circuits
[0193]   **510, 520, 530,** and **540** non-linear amplitude processing circuits
[0194]   **511, 531, 3100, 3105, 4100,** and **4105** input terminals
[0195]   **513** polarity determining circuit
[0196]   **514** multiplier
[0197]   **515** and **3103** adders
[0198]   **516, 536, 3104,** and **4104** output terminals
[0199]   **600** and **610** frequency analysis synthesizing circuits
[0200]   **1000** computer
[0201]   **3101** delay element
[0202]   **3102** and **4101** coefficient multipliers

1. A signal processing method of receiving a plurality of received signals, and subtracting pseudo echoes generated by a plurality of adaptive filters having said plurality of received signals as an input, respectively, from a plurality of echoes to be generated from said plurality of received signals, thereby to reduce said plurality of echoes, comprising:

generating delayed received signals by delaying at least one received signal, out of said plurality of received signals;

generating pseudo echoes by inputting said received signals and said delayed received signals into said adaptive filters; and

controlling a frequency of inputting said received signals and said delayed received signals into said adaptive filters based upon a perceptual sensitivity to a change in localization of said plurality of received signals.

2. A signal processing method according to claim **1**, wherein at least one signal, out of said delayed received signals, is an amplitude-corrected delayed received signal subjected to an amplitude correction.

3. A signal processing method according to claim **1**, wherein at least one signal, out of the signals to be inputted into said plurality of adaptive filters, is a non-linearly processed signal subjected to a non-linear processing.

4. A signal processing method according to claim **1**, comprising decomposing said received signal into a plurality of frequency components; and generating the delayed received signals by delaying the received signal for every above plurality of frequency components.

5. A signal processing method according to claim **1**, wherein said perceptual sensitivity to a change in localization is obtained based upon a similarity degree of the received signal.

6. A signal processing method according to claim **1**, wherein said perceptual sensitivity to a change in localization is obtained based upon a power of the received signal.

7. A signal processing method according to claim **1**, comprising generating the delayed received signals so that relative delays of said delayed received signals to said received signals have a plurality of values that vary with a time.

8. A signal processing method according to claim **7**, wherein said relative delay is an integer multiple of a sampling period.

9. A signal processing method according to claim **1**, wherein said delayed received signals are generated by processing the received signals with a filter having a plurality of time-varying coefficients with alternately a zero value or a non-zero value.

10. A signal processing method according to claim **9**, wherein said plurality of time-varying coefficients have a zero value exclusively to each other.

11. A signal processing method according to claim **9**, wherein said plurality of time-varying coefficients have a non-zero value exclusively to each other.

12. A signal processing apparatus for receiving a plurality of received signals, and subtracting pseudo echoes generated by a plurality of adaptive filters having said plurality of received signals as an input, respectively, from a plurality of echoes to be generated from said plurality of received signals, thereby to reduce said plurality of echoes, comprising:

a linear processing circuit that generates delayed received signals by delaying at least one received signal, out of said plurality of received signals;

an adaptive filter that generates pseudo echoes by receiving said received signals and said delayed received signals,

a plurality of subtracters, each of said subtracters that generates echo-reduced signals by subtracting said pseudo echoes from a plurality of mixed signals; and

an analyzing circuit that obtains a perceptual sensitivity to a change in localization of said plurality of received signals, and

wherein said signal processing apparatus controls a frequency of inputting said received signals and said delayed received signals into said adaptive filters based upon said perceptual sensitivity, and controls coefficients of said plurality of adaptive filters so that outputs of said plurality of subtracters are minimized.

13. A signal processing apparatus according to claim **12**, comprising an amplitude correcting circuit that generates amplitude-corrected delayed received signals by amplitude-correcting at least one signal, out of said delayed received signals.

14. A signal processing apparatus according to claim **12**, comprising a non-linear processing circuit that generates non-linearly processed signals by non-linearly processing at least one signal, out of the signals to be inputted into said plurality of adaptive filters.

15. A signal processing apparatus according to claim **12**, comprising:

a frequency analyzing circuit that decomposes said received signal into a plurality of frequency components; and

a linear processing circuit that generates the delayed received signals by delaying the received signal for every said plurality of frequency components.

16. A signal processing apparatus according to claim **12**, wherein said analyzing circuit obtains said perceptual sensitivity to a change in localization based upon a similarity degree of the received signal.

17. A signal processing apparatus according to claim **12**, wherein said analyzing circuit obtains said perceptual sensitivity to a change in localization based upon a power of the received signal.

18. A signal processing apparatus according to claim **12**, wherein said linear processing circuit performs a processing

such that relative delays of said delayed received signals to said received signals have a plurality of values that vary with a time.

19. A signal processing apparatus according to claim **18**, wherein said linear processing circuit performs a processing such that said relative delay is an integer multiple of a sampling period.

20. A signal processing apparatus according to claim **12**, wherein said linear processing circuit comprises a filter having a plurality of time-varying coefficients with alternately a zero value or a non-zero value.

21. A signal processing apparatus according to claim **20**, wherein said plurality of time-varying coefficients have a zero value exclusively to each other.

22. A signal processing apparatus according to claim **20**, wherein said plurality of time-varying coefficients have a non-zero value exclusively to each other.

23. A non-transitory computer readable storage medium storing a signal processing program for causing a computer to execute a receiving process of receiving a plurality of received signals, and an echo reducing process of reducing a plurality of echoes that are generated by said plurality of received signals, said signal processing program comprising:

a delayed received signal generating process of generating delayed received signals by delaying at least one received signal, out of said plurality of received signals;

a pseudo echo generating process of generating pseudo echoes by inputting said received signals and said delayed received signals into said adaptive filters; and

a pseudo echo subtracting process of subtracting said pseudo echoes from said plurality of received signals, respectively, wherein a frequency of inputting said received signals and said delayed received signals into said adaptive filters is controlled based upon a perceptual sensitivity to a change in localization of said plurality of received signals.

24. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, wherein at least one signal, out of said delayed received signals, is an amplitude-corrected delayed received signal subjected to an amplitude correction.

25. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, wherein at least one signal, out of the signals to be inputted into said plurality of adaptive filters, is a non-linearly processed signal subjected to a non-linear processing.

26. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, comprising decomposing said received signal into a plurality of frequency components and generating the delayed received signals by delaying the received signal for every above plurality of frequency components.

27. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, wherein said perceptual sensitivity to a change in localization is obtained based upon a similarity degree of the received signal.

28. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, wherein said perceptual sensitivity to a change in localization is obtained based upon a power of the received signal.

29. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, wherein said delayed received signals are generated so that relative delays of said delayed received signals to said received delays have a plurality of values that vary with a time.

30. A non-transitory computer readable storage medium storing a signal processing program according to claim **29**, wherein said relative delay is an integer multiple of a sampling period.

31. A non-transitory computer readable storage medium storing a signal processing program according to claim **23**, wherein said delayed received signals are generated by pro-

cessing the received signals with a filter having a plurality of time-varying coefficients with alternately a zero value or a non-zero value.

32. A non-transitory computer readable storage medium storing a signal processing program according to claim **31**, wherein said plurality of time-varying coefficients have a zero value exclusively to each other.

33. A non-transitory computer readable storage medium storing a signal processing program according to claim **31**, wherein said plurality of time-varying coefficients have a non-zero value exclusively to each other.

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