

(51) International Patent Classification:
G10K 11/178 (2006.01)(21) International Application Number:
PCT/IB2010/052475(22) International Filing Date:
3 June 2010 (03.06.2010)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
09161866.0 3 June 2009 (03.06.2009) EP(71) Applicant (for all designated States except US): **NXIP B.V.** [NL/NL]; High Tech Campus, 60, NL-5656 AG Eindhoven (NL).

(72) Inventors; and

(75) Inventors/Applicants (for US only): **GAUTAMA, Temujin** [BE/BE]; c/o NXP B.V., Intellectual Property & Licensing Department, High Tech Campus 32, NL-5656 AE Eindhoven (NL). **DOCLO, Simon** [BE/BE]; c/o NXP B.V., Intellectual Property & Licensing Department, High Tech Campus 32, NL-5656 AE Eindhoven (NL).(74) Agents: **KROTT, Michel, Willy, François, Maria** et al.; c/o NXP B.V., Intellectual Property & Licensing Department, High Tech Campus 32, NL-5656 AE Eindhoven (NL).

(81) Designated States (unless otherwise indicated, for every kind of national protection available): AE, AG, AL, AM,

AO, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CL, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PE, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, ST, SV, SY, TH, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.

(84) Designated States (unless otherwise indicated, for every kind of regional protection available): ARIPO (BW, GH, GM, KE, LR, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AL, AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MK, MT, NL, NO, PL, PT, RO, SE, SI, SK, SM, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Declarations under Rule 4.17:

- as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii))
- as to the applicant's entitlement to claim the priority of the earlier application (Rule 4.17(iii))

Published:

- without international search report and to be republished upon receipt of that report (Rule 48.2(g))

(54) Title: HYBRID ACTIVE NOISE REDUCTION DEVICE FOR REDUCING ENVIRONMENTAL NOISE, METHOD FOR DETERMINING AN OPERATIONAL PARAMETER OF A HYBRID ACTIVE NOISE REDUCTION DEVICE, AND PROGRAM ELEMENT

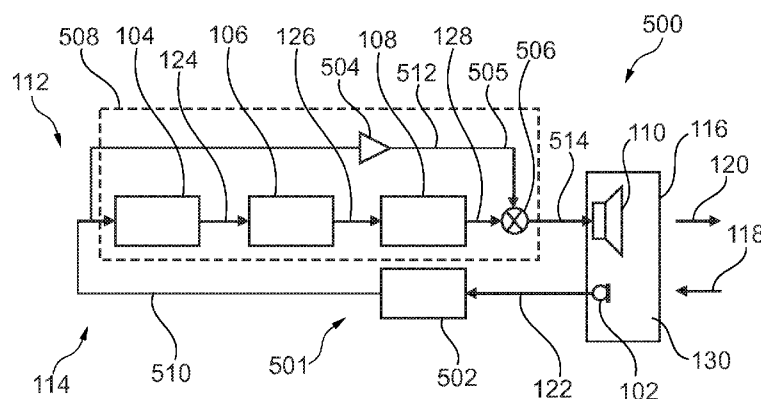


Fig. 5

(57) Abstract: A hybrid active noise reduction device for reducing environmental noise is provided, the device (500) comprising a microphone (102) for receiving a noise signal (118) and for transmitting a microphone signal (122), a filter unit (114) for receiving the microphone signal (122) and for transmitting a filtered microphone signal (514), the filter unit (114) comprising a low-latency filter part (501) and a high-latency filter part (112), the low-latency filter part (501) and the high-latency filter part (112) being arranged in series to one another, and a loudspeaker (110) for transmitting a noise reduction signal (120) based on the filtered microphone signal (514).

HYBRID ACTIVE NOISE REDUCTION DEVICE FOR REDUCING
ENVIRONMENTAL NOISE, METHOD FOR DETERMINING AN OPERATIONAL
PARAMETER OF A HYBRID ACTIVE NOISE REDUCTION DEVICE, AND
PROGRAM ELEMENT

FIELD OF THE INVENTION

The invention relates to a hybrid active noise reduction device for reducing environmental noise.

Further, the invention relates to a method for determining an operational parameter of a hybrid active noise reduction device.

Further, the invention relates to a program element.

BACKGROUND OF THE INVENTION

Active noise reduction (ANR) is a widely known principle, which may be applied in order to reduce environmental noise which a person may experience and thus to increase a level of comfort for the person in his noisy environment.

An ANR device may comprise at least one microphone, a filter unit, and a loudspeaker. The microphone receives a noise signal from the noisy environment of the person and transmits a microphone signal to the filter unit. The filter unit performs a filtering operation on the microphone signal and transmits a noise reduction signal to the loudspeaker for further broadcast to the environment. The noise reduction signal is in counter phase to the environmental noise such that the level of ambient noise perceived by the person is reduced, thereby enhancing the comfort of the person.

An ANR device may comprise a feedforward configuration or a feedback configuration, or a combination thereof, and may be implemented in a headset, in a hearing aid or in a car compartment.

In the feedforward configuration, the ANR device comprises a reference microphone which may be arranged in such a way that it mainly picks up the environmental noise from outside. The noise reduction signal emitted by the loudspeaker is a filtered version of the reference microphone signal. The filtering operation performed by

the filter unit may be optimized by arranging an error microphone in such a way that the error microphone records ambient noise which a person perceives. The filter unit may be optimized in that the sound level at the error microphone may be minimized. When implementing the device in a headset, the reference microphone is arranged on the outside of an earphone of the headset, whereas the error microphone and the loudspeaker are arranged on the inside of an earphone of the headset.

A feedback ANR device uses only an error microphone arranged in such a way that the error microphone records ambient noise which a person may perceive. The error microphone signal is filtered by the filter unit and is emitted by the loudspeaker. The filter unit may also be optimized by minimizing the sound level at the error microphone.

The filter unit of an ANR device may comprise an analog filter and/or a digital filter. An ANR device comprising an analog filter and a digital filter is known as a “hybrid” ANR device. The use of an analog filter may allow for using the instantaneous analog filter response, whereas the use of a digital filter may enable a more complex signal filtering capability. However, using a digital filter may introduce delays into the digital filter response which may result from analog-to-digital and digital-to-analog convertors being constructively indispensable in the signal processing path and being arranged upstream and downstream of the digital filter. In particular, these delays may force the digital filter response being zero in particular time intervals.

US 5,440,642 B1 discloses a digitally controlled hybrid ANR device in a feedback configuration which comprises a microphone, a filter unit, and a loudspeaker. A microphone signal is filtered by the filter unit, and a filtered signal is transmitted to the loudspeaker. The filter unit comprises an analog filter which is digitally controlled using a digital signal processor comprising a digital filter. The analog filter and the digital filter are arranged in parallel to one another.

OBJECT AND SUMMARY OF THE INVENTION

It is an object of the invention to provide a hybrid active noise reduction device, a method for determining an operational parameter of a hybrid active noise reduction device, and a program element offering an improved noise reduction capability.

In order to achieve the object defined above, a hybrid active noise reduction device for reducing environmental noise, a method for determining an operational parameter of a hybrid active noise reduction device, and a program element are provided.

According to an embodiment of the invention, a hybrid active noise reduction device for reducing environmental noise is provided, the device comprising a microphone for receiving a noise signal and for transmitting a microphone signal, a filter unit for receiving the microphone signal and for transmitting a filtered microphone signal, the filter unit comprising a low-latency filter part and a high-latency filter part, the low-latency filter part and the high-latency filter part being arranged in series to one another, and a loudspeaker for transmitting a noise reduction signal based on the filtered microphone signal. The low-latency filter part may be an analog filter or it may be a digital filter, such as a series connection of delta-sigma modulators and an N-bit digital filter. The high-latency filter part may be a digital filter with traditional analog-to-digital and digital-to-analog converters, with non-negligible conversion latencies. Note that this is a redefinition of the term 'hybrid', since it is usually used to refer to a combination of an analog filter part and a digital filter part.

The latency of a filter is the time delay between the input and the output of the filter. One way of measuring latency is to look for a peak in the cross-correlation between the input and the output of the filter. A filter is considered to have a low latency if the latency is smaller than the time for one input signal sample, e.g. if the input signal is sampled at 48KHz, and the latency of the filter is less than $1/48000$ seconds, then the latency is considered to be low. A filter is considered to have a high latency if the latency is greater than or equal to the time for one input signal sample, e.g. if the input signal is sampled at 48KHz, and the latency of the filter is greater than or equal to $1/48000$ seconds, then the latency is considered to be high.

According to another embodiment of the invention, a headset is provided, the headset comprising a hybrid active noise reduction device for reducing environmental noise.

According to another embodiment of the invention, a method for determining an operational parameter of a hybrid active noise reduction device is provided, the active noise reduction device comprising a microphone for receiving a noise signal and for transmitting a microphone signal, a filter unit for receiving the microphone signal and for transmitting a filtered microphone signal, the filter unit comprising a low-latency filter part and a high-latency filter part, and a loudspeaker for transmitting a noise reduction signal based on the filtered microphone signal, the method comprising selecting a first operational parameter of the device and determining a second operational parameter of the

device based on the first operational parameter.

According to another embodiment of the invention, a program element is provided, which, when being executed by a processor, is adapted to control or carry out a method for determining an operational parameter of a hybrid active noise reduction device.

According to the above embodiments, a hybrid active noise reduction device is provided which offers the possibility for effectively reducing environmental noise such that a level of comfort of a person using the device is significantly enhanced. In particular, the device is adapted to significantly reduce environmental noise.

The hybrid active noise reduction device comprises a filter unit which comprises a low-latency filter part and a high-latency filter part being arranged in series to one another such that a filtering operation of the microphone signal may be enhanced in that two filtering operations are performed in series to one another. Thus the output signal of the filter unit may combine a (nearly) instantaneous filter response of the low-latency filter part and the complex signal modeling capability of the high-latency digital filter part.

The low latency filter part may be an analog filter part comprising an analog filter, for example, a first order low-pass filter.

The high latency filter part may be a digital filter part comprising a digital filter, for example, a finite impulse response filter.

Further, the device comprises a constructively easy design, since a simple series circuit may be used for arranging the low-latency filter part and the high-latency filter part of the filter unit.

The noise reduction signal may be identical or be based on the filtered microphone signal.

The device may comprise a feedback configuration with the microphone being an error microphone and the filtered microphone signal being fed to the loudspeaker. The noise reduction signal transmitted by the loudspeaker may be recorded by the microphone together with a noise signal. This microphone signal may be used as an error signal for optimizing the noise reduction capability of the device. The error signal may be identical or be based on a combination, particularly a sum, of an ambient noise signal received by the microphone and the noise reduction signal provided by the loudspeaker. In particular, the noise reduction signal transmitted by the loudspeaker may be convolved with a transfer function describing a signal path from the loudspeaker to the microphone.

In particular, the error microphone and the loudspeaker may be implemented inside or form part of an inner part of an earphone of a hybrid active noise reduction headset. The microphone may record ambient noise which a user perceives when wearing the headset.

The device may also form part of a hearing aid or a car compartment or any other device application in which the principle of active noise reduction may be usefully applied.

Respecting the method for determining an operational parameter of a hybrid active noise reduction device, an optimization of an operation of the device may be performed such that the noise reduction capability of the device may be increased. A first operational parameter of the device is selected, and a second operational parameter of the device is determined based on the first operational parameter. The first operational parameter and/or the second operational parameter may comprise a device characteristic, in particular a characteristic of a device component. Such a component of the device may be the low-latency filter part and the high-latency filter part. Determining the second operational parameter based on the first operational parameter may comprise fixing the first operational parameter or keeping the first operational parameter constant and determining the second operational parameter. In particular, determining the second operational parameter may comprise computational or mathematical techniques.

In particular, the method may be performed in an operational state or a resting state of the device.

The method may be applied to any hybrid active noise reduction device or device application comprising such a hybrid active noise reduction device. In particular, the hybrid active noise reduction device may be of a feedback type or a feedforward type, or of a combination thereof. In particular, the low-latency filter part and the high-latency filter part may be in series to one another or in parallel to one another.

Respecting the program element, the program element may be identical or be part of e.g. a software routine, a source code or an executable code. In particular, the program element may be stored in a computer readable medium (e.g. volatile memory, non-volatile memory, a CD, a DVD, a USB stick, a floppy disk or a harddisk).

Next, further embodiments of the hybrid active noise reduction device for reducing environmental noise will be explained. However, these embodiments also apply to the headset, the method, and the program element.

The device may further comprise a bypass line being arranged in parallel to the

high-latency filter part. Thus the bypass line and the signal path comprising the high-latency filter part may form a parallel circuit being in series to the low-latency filter part. A first branch of the parallel circuit may be identical to or comprise the bypass line, and a second branch of the parallel circuit may be identical to or comprise the high-latency filter part. The output signal of the filter unit and thus the noise reduction signal may comprise a combination of a signal being filtered by the low-latency filter part and being bypassed and a signal being filtered by the low-latency filter part and the high-latency filter part. In particular, the output signal of the filter unit may comprise portions which result from the low-latency filtering operation and the combined low-latency and high-latency filtering operation. In particular, the filter response of the low-latency filter part may comprise a (nearly) instantaneous filter response which may compensate for delayed portions of the filter response of the high-latency filter part which may be equal to or almost zero over time. Thus the device may comprise a yet more enhanced noise reduction capability, since a more improved signal processing may be performed by the filter unit. In particular, the noise reduction signal transmitted by the loudspeaker may be instantaneous due to a bypassing of a signal being already or subsequently to be filtered by the low-latency filter part. Further, the noise reduction signal may be effectively shaped by the digital filtering operation performed by the high-latency filter part. In particular, delays resulting from the high-latency filter part, particularly resulting from an analog-to-digital signal conversion and a digital-to-analog signal conversion may be compensated for by the (nearly) instantaneous filter response of the low-latency filter part.

In particular, the device may be used for any application in which device delays due to signal conversions from the analog to the digital domain and vice versa may be present.

The bypass line may comprise a gain unit. Thus the bypassed signal may be amplified, in order to offer a further modification capability of the output signal of the filter unit, particularly of the noise reduction signal, before transmitting the filtered signal to the loudspeaker. Therefore the noise reduction capability of the device may be further enhanced. In particular, the gain unit may comprise a controllable gain unit or a fixed gain unit.

The device may further comprise a signal adding unit for adding a bypassed signal and a (high-latency) digitally filtered signal. The signal adding unit may enable a combination or mixture of the bypassed (and amplified) signal and the (high-latency) digitally filtered signal in that the two branches of the formed parallel circuit may be

added.

The signal adding unit may be adapted to sum the bypassed signal and the (high-latency) digitally filtered signal in an analog way. This embodiment of the signal adding unit may represent a very easy and effective way for performing an adding operation of two signals.

The low-latency filter part may be arranged downstream with respect to the digital filter part. Thus the device may be used in a headset offering voice communications and/or speech capabilities, since it may be desirable that the microphone signal may not be filtered by the low-latency filter part before inputting it to the high-latency filter part. In particular, this arrangement of the low-latency filter part and the high-latency filter part may lead to a low degree of instrumental noise of the device which may comprise self-noise of the microphone and electrical noise of the device. In particular, a further amplifier unit may be arranged downstream with respect to the high-latency filter part for further amplifying the digitally filtered signal.

The low-latency filter part may be arranged upstream with respect to the high-latency filter part, so that the output of the low latency filter part supplies the input of the high latency filter part. Then, the output of the low latency filter part is available to the high latency filter part, and this can be used for optimizing the filters. This arrangement of the low-latency filter part and the high-latency filter part may also be used in device applications which may not offer voice communications or speech capabilities. In particular, an amplifier unit may be arranged downstream with respect to or may be implemented in the microphone for amplifying the recorded noise signal, since the instrumental noise of the device may comprise a relatively high level.

The high-latency filter part may be a digital filter part that may comprise an analog-to-digital convertor for converting an analog signal into a digital signal, a digital filter, and a digital-to-analog convertor for converting a digital signal into an analog signal. This embodiment of the high-latency filter part offers the technical possibility for digitally filtering a recorded analog microphone signal. In particular, the analog-to-digital convertor and the digital-to-analog convertor may introduce signal conversion delays into the signal transmission of the device, thereby reducing the noise reduction capability of the device. In particular, the series arrangement of the low-latency filter part and the high-latency filter part in combination with the bypass line may compensate for the conversion delays in the device signal path. The analog-to-digital convertor and the digital-to-analog convertor may

be identical or form part of a codec (coder-encoder). The codec may comprise a bypass capability, which may be with a controllable gain, thus comprising the bypass line. The digital filter may be identical or form part of a digital signal processor (DSP).

The device may comprise a further microphone for receiving a further noise signal and for transmitting a further microphone signal, wherein the low-latency filter part and/or the high-latency filter part may operate based on the further microphone signal. Thus the device may comprise a feedforward configuration with the low-latency filter part and the high-latency filter part being arranged in series to one another. The microphone may be a reference microphone recording or receiving noise. The further microphone may be an error microphone which may record the further noise, which a person may perceive. The filter unit may filter the reference microphone signal and transmit the filtered reference microphone signal, in particular the noise reduction signal, to the loudspeaker. An error microphone signal recorded by the error microphone may be used for optimizing the noise reduction capability of the device. The error signal may be identical or be based on a combination, particularly a sum, of an ambient noise signal received by the microphone and the noise reduction signal provided by the loudspeaker. In particular, the noise reduction signal transmitted by the loudspeaker may be convolved with a transfer function describing a signal path from the loudspeaker to the microphone. In particular, the error microphone may be arranged in series to an analog-to-digital convertor for further converting the analog error microphone signal into the digital domain for further signal processing. In particular, the error microphone and the loudspeaker may be arranged on the inside of an earphone of a headset, whereas the reference microphone may be arranged on the outside of an earphone of a headset.

The high-latency filter part may comprise a digital filter, wherein the digital filter may be a finite impulse response filter.

The low-latency filter part may comprise an analog filter, wherein the analog filter may be a first order low pass filter. A cut-off frequency of the first order low pass filter may be of about 700 Hz, particularly of about 500 Hz, further particularly of about 300 Hz. In particular, the analog filter may also be a higher order low pass filter, a high pass filter, a band pass filter, or a combination thereof, each of which comprising suitable cut-off frequencies.

The low-latency filter part may comprise a digital filter. It may be implemented as a series connection of sigma-delta modulators and an N-bit digital filter.

Next, further embodiments of the method for determining on operational parameter of a hybrid active noise reduction device may be explained. However, these embodiments also apply to the device, the headset, and the program element.

The device may further comprise a bypass line in parallel to the high-latency filter part, wherein the bypass line may comprise a gain unit, wherein the method may comprise selecting the first operational parameter and determining a second operational parameter and a third operational parameter of the device based on the first operational parameter. Thus a more improved optimization of the operation of the device is provided, since more operational parameters of the device and thus device characteristics may be determined. In particular, the analog filter part may be arranged in or be part of the bypass line.

The first operational parameter may be an operational parameter of the low-latency filter part, the second operational parameter may be an operational parameter of the high-latency filter part, and the third operational parameter may be an operational parameter of the gain unit. Thus the operation of the device may be optimized by selecting the operational parameter of the low-latency filter part which may be known from a constructive design of the low-latency filter part. The operational parameters of a parallel circuit comprising the bypass line with the gain unit and the high-latency filter part may thus be determined.

Determining the second operational parameter based on the first operational parameter may comprise minimizing a cost function of the device with respect to the second parameter. This measure represents a very effective and easily performable way for determining minima of the cost function for the second operational parameter, yielding the optimal second operational parameter of the device for operating the device.

Determining the second operational parameter and the third operational parameter based on the first operational parameter may comprise minimizing a cost function of the device with respect to the second operational parameter and the third operational parameter. Upon simultaneously or jointly determining the second and third operational parameters of the device a very effective and easily performable optimization of the device operation may be achieved.

The aspects defined above and further aspects of the present invention are apparent from the examples of embodiment to be described hereinafter and are explained with reference to the examples of embodiment. The invention will be described in more detail

hereinafter with reference to examples of embodiment but to which the invention is not limited.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be described in more detail hereinafter with reference to examples of embodiment, but to which the invention is not limited.

Fig. 1 shows a known digital ANR device in a feedback configuration.

Fig. 2 shows a frequency dependency of an active performance of the device in Fig. 1 for different delays.

Fig. 3 shows a sample index dependency of an impulse response of a filter unit of the device in Fig. 1 for zero delay.

Fig. 4 shows a sample index dependency of an impulse response for the filter unit of the device in Fig. 1 for a delay of 15 samples.

Fig. 5 shows a hybrid ANR device in a feedback configuration according to an exemplary embodiment of the invention.

Fig. 6 shows a frequency dependency of an active performance of the device in Fig. 5 for different delays.

Fig. 7 shows a known digital ANR device in a feedforward configuration.

Fig. 8 shows a frequency dependency of an active performance of the device in Fig. 7 for different delays.

Fig. 9 shows a sample index dependency of an impulse response of a filter unit of the device in Fig. 7 for different delays.

Fig. 10 shows a hybrid ANR device in a feedforward configuration according to another exemplary embodiment of the invention.

Fig. 11 shows a known hybrid ANR device in a feedforward configuration.

Fig. 12 shows a frequency dependency of an active performance of the device in Fig. 11 for different delays.

Fig. 13 shows a sample index dependency of an impulse response of the filter unit of the device in Fig. 11 for different delays.

DESCRIPTION OF EMBODIMENTS

The illustration in the drawing is schematically. It is noted that in different figures, similar or identical elements are provided with the same reference signs or with reference

signs, which are different from the corresponding reference signs only within the first digits.

Hybrid ANR devices comprise a filter unit consisting of a low-latency filter part and high-latency filter part and thus combine the (nearly) instantaneous filtering capability of a low-latency filter part (analog or digital) and the complex signal tuning capability of a high-latency (digital) filter part.

In the following, active noise reduction using a feedback configuration is explained.

Fig. 1 shows a prior art digital ANR device 100 in a feedback configuration. The device 100 comprises a microphone 102, an analog-to-digital convertor (ADC) 104, an ANR digital filter 106, a digital-to-analog convertor (DAC) 108, and a loudspeaker 110. The ADC 104, the digital filter 106, and the DAC 108 may form a digital filter part 112 of a filter unit 114. The device 100 may also comprise amplifiers (not shown) which operate in their linear region.

This feedback device 100 may be implemented into a headset such that the microphone 102 and the loudspeaker 110 are both arranged on the inside of an earphone 116 of the headset. The microphone 102 may then be called an “error microphone” and receives a noise signal which a user of the headset may perceive. This noise signal may differ from a noise signal a user may perceive without the headset.

For convenience, the analog-to-digital and digital-to-analog conversions are in the following assumed to be ideal, i.e., instantaneous and without loss of information. Thus, an analog signal may be represented by a digital signal equivalent, and actual, i.e. non-ideal, conversions may then be conveniently represented by time delays Δ_{ADC} and Δ_{DAC} .

In operation of the device 100, the microphone 102 picks up a signal that comprises an acoustical summation of an ambient noise signal 118 which a user perceives and a noise reduction signal 120 emitted by the loudspeaker 110. An error microphone signal 122 is converted into the digital domain, thereby introducing a delay Δ_{ADC} and thus yielding a digital microphone signal 124. This error signal 124 is then filtered by the digital filter 106, and a digitally filtered signal 126 of the digital filter 106 is converted by the DAC 108 to the analog domain, thereby introducing a delay Δ_{DAC} to the converted signal 128 and yielding the noise reduction signal 120. The transfer function of the loudspeaker 110 to the microphone 102 describing the acoustical path 130 from the loudspeaker 110 to the microphone 102 is assumed to be linear. The error microphone 102 hence records a sum of the ambient noise signal 118 and the noise reduction signal 120 convolved with the transfer

function of the acoustical path 130. The error microphone signal 122 may thus be regarded as an error signal of the device 100.

As shown in Fig. 2, the device 100 shows a poor ANR performance.

In general, the ANR performance of the device 100 may be expressed in terms of the “active performance” which is evaluated on the microphone 102 and may be defined as a difference in a power spectral density (PSD) with the ANR device 100 being not operational and with the ANR device 100 being operational. $G(\omega)$ thus reads

$$G(\omega) = 10 \log_{10} \Phi_{\epsilon}(\omega) - 10 \log_{10} \Phi_{\alpha}(\omega),$$

with $\Phi_{\epsilon}(\omega)$ denoting the PSD of the (residual) error microphone signal 122 comprising an ANR processing and $\Phi_{\alpha}(\omega)$ denoting the PSD of the ambient noise signal 118 at the error microphone 102 without an ANR processing. Thus a negative active performance indicates a noise reduction, with a lower value of the active performance indicating an increased noise reduction. Although it is generally assumed the digital ANR devices 100 being expected to be superior to analog ANR devices, the actual performance of the digital ANR device 100 is drastically deteriorated owing to the presence of the time delays Δ_{ADC} and Δ_{DAC} introduced by the ADC 104 and the DAC 108.

Active performance $G(\omega)$ curves 202-212 are shown for delay values

$\Delta = \Delta_{ADC} + \Delta_{DAC}$ between zero and ten samples in steps of two samples. Signal sampling at 48 kHz using a digital finite impulse response (FIR) filter of 512 taps is used. Further, the signal recording is performed in the diffuse noise condition. It can be observed that the performance $G(\omega)$ degrades considerably upon presence of a delay Δ : In the delay-free case indicated by the curve 202, the ANR performance $G(\omega)$ shows a deep and broad region of noise reduction down to a maximum of about 18 dB, while for a delay of ten samples indicated by the curve 212, the performance $G(\omega)$ is rather poor with a value of less than 6 dB.

This deterioration of the active performance of the device 100 resulting from the delay Δ may be explained with respect to Fig. 3 and 4 which show the sample dependency of an impulse response of the digital filter 100 having zero delay Δ and a delay of $\Delta = 15$

samples. In both figures, only the first 100 taps of the (high-latency) digital filtering operation are shown.

Respecting a curve 302 (zero delay) in Fig. 3, the first 15 filter taps comprise most of the energy (highest filter coefficients) and are the decisive filter coefficients for the digital filter operation. As seen from a curve 402 in Fig. 4, introducing a delay of $\Delta = 15$ taps yields the first delay coefficients being forced to zero, since the digital filter is causal. Thus, owing to the linearity of the device 100, the conversion delay Δ may be interpreted as a delay of the digital filter 106 or a series of zeros at the beginning of the digital filter 106. If this important part of the digital filter 106 is forced to zero, the ANR performance, particularly $G(\omega)$, may be substantially degraded, since the digital filter 106 cannot compensate for the acausality of the filter.

In order to compensate for the delay of the digital ANR device 100, a hybrid feedback ANR device according to an exemplary embodiment of the invention may be used, which is shown in Fig. 5.

The device 500 may be used in any ANR application. However, the device 500 is most usefully integrated in applications with conversion delays Δ degrading the performance of the device 500. In particular, the device 500 may be part of an ANR headset of a feedback type.

The hybrid device 500 comprises an error microphone 102, a low-latency filter part 501 comprising a low-latency filter 502 (analog or digital), a subsequent high-latency (digital) filter part 112 being formed by an ADC 104, a digital filter 106, and a DAC 108 and being in series to the low-latency filter part 501, and a loudspeaker 110. The ADC 104 and the DAC 108 are provided before and after the digital filter 106 seen in a signal transmitting direction. A gain unit 504 is provided in parallel to the digital filter 106, thus forming a signal bypass line 505. Further, the device 500 comprises a signal adding unit 506 for electrically combining or mixing the signals of the two branches of a parallel circuit 508 of the filter unit 114 before transmitting a combined filtered microphone signal to the loudspeaker 110. The parallel circuit 508 is defined by the gain unit 504 and the high-latency filter part 112. The low-latency filter 502 may be designed as a first-order low pass filter.

Thus, the device 500 uses a combination of a low-latency filter 502 and high-latency (digital) filter 106 to span the acausal part of the desired impulse response of the

filter unit 114. Thus the acausal part of the high-latency digital filter 106 can be modelled by low-latency filter response. The two filters 502, 106 are combined in such a way that they can be conveniently implemented using a bypass of the digital filter 106 which is present in many ADC 104 and DAC 108, reducing the required circuitry of the hybrid device 500. Thus, the device 500 shows an increased robustness compared to the device 100.

In operation of the device 500, an error signal 122 corresponding to the error microphone signal 122 is filtered by the low-latency filter 502. The filtered microphone signal 510 is then fed into the parallel circuit 508 comprising the instantaneous gain unit 504 and the digital filter 106 having the conversion delays Δ_{ADC} and Δ_{DAC} . An amplified signal 512 and a digitally filtered and converted signal 128 are then summed to form the combined filtered microphone signal 514 which, in turn, is sent to the loudspeaker 110.

The device 500 may comprise the following modifications:

The low-latency filter 502 may also be provided after the high-latency filter 106 seen in a signal transmitting direction. Thus the low-latency filtering operation may take place before inputting a signal to the ADC 104 or after outputting of the signal of the DAC 108. The arrangement of the low-latency filter 502 and high-latency filter 106 may be chosen according to the application of the device 500.

When performing the low-latency filtering operation before the signal input to the ADC 104 (shown in Fig. 5), a microphone amplifier may be needed for boosting the signal considerably, in order to obtain a reasonable headroom on the high-latency filter 106, particularly such that there is a sufficient accuracy for fixed-point representation of the signal, and, as a consequence, instrumental noise, i.e. self-noise of the microphone 102 and electrical noise, might become too large.

When performing the low-latency filtering operation after the DAC 108, instrumental noise may be of the same order of magnitude as in a normal application such as voice communications. However, it may be possible that the noise reduction signal may need to be boosted beyond a value being possible in the DAC 108, therefore necessitating an additional amplifier stage.

If the ANR device 500 is implemented in a system which comprises voice communications or speech capture, it may be desirable that the microphone signal remains unfiltered before inputting into the high-latency filter 106, yielding the configuration of the

device 500 comprising the low-latency filter 502 after the DAC 108 when seen in a signal transmitting direction.

Again referring to Fig. 4, the impulse response of the device 500 is given as the superposition by a curve 404 and the impulse response of the high-latency filter (not shown) in a range of 15 samples to 100 samples. Here, a delay of 15 taps is assumed. The acausal part of the impulse response of the high-latency filter 106 is given by the impulse response of the low-latency filter 502, here a first-order analog low pass filter with cut-off frequency at 700 Hz. Thus, the overall impulse response of the filter unit 114 of the device 500 is given as a concatenation of the curves 404, 302.

Therefore the analog bypass functionality of the ADC 104 and DAC 108 allows for a convenient implementation of the branches of the parallel circuit 508. By setting the global output gain of the parallel circuit 508, the gain unit 504 may be controlled with the high-latency filter 106 which may be inversely scaled in the digital domain.

Fig. 6 shows the active performance $G(\omega)$ of the hybrid feedback ANR device 500 for a conversion delay Δ of zero, two and ten samples (curves 602 and 604, respectively). The length of the high-latency digital filter 106 is set to 512 with a sampling rate of 48 kHz. Further, the low-latency filter 502 is designed as a first-order analog low pass filter with a cut-off frequency of 700 Hz. For convenience, a curve 606 corresponds the curve 202 and indicates the active performance of the digital feedback device 100 in the case of zero delay.

Similarly to the delay-free digital feedback device 100 (curve 202 in Fig. 2), the noise reduction is prominent in a wide frequency area with a deep noise reduction between about 50 Hz and about 2 kHz. A maximal noise reduction value is about 18 dB. However, in contrast to the ANR device 100, the active performance $G(\omega)$ is maintained for a delay of 10 samples, since no significant changes in $G(\omega)$ for different delays are visible. In general, the maximal delay Δ for which the active performance $G(\omega)$ is robust depends on a design of the device 500 and particularly on the low-latency filter 502 used. In this exemplary case, the active performance is maintained for delays below 10 samples using a first-order analog filter 502. When using a higher-order low-latency filter 502, the hybrid device 500 is made robust to longer delays Δ .

In the following, a method for determining an operational parameter of the device 500 according to an exemplary embodiment of the invention will be explained. The method is adapted to optimize an operation of the device 500.

If an operational parameter of the low-latency filter 502 is fixed, the hybrid device 500 may be optimised in terms of optimizing an operational parameter of components of the parallel circuit 508. In particular, the gain unit 504 and the filter coefficients of the high-latency filter 106 may be jointly optimised without the need of a model of the low-latency filter 502, thereby omitting an introduction of additional errors into the optimal performance parameters of the device 500. In order to select the operational parameter of the device 500, a cost function of the device may be minimized with respect to the operational parameters of the gain unit 504 and the high-latency filter 106, with the operational parameter of the low-latency filter 502 being fixed.

In the following, a hybrid ANR device in a feedforward configuration will be explained.

Fig. 7 shows a known digital ANR device 700 in a feedforward configuration. The device 700 is part of a headset and comprises a reference microphone 102, an ADC 104, a digital filter 106, a DAC 108, and a loudspeaker 110. Further, the device comprises an error microphone 702, arranged together with the loudspeaker 110 on the inside of an earphone 116. A further ADC 704 is provided and is arranged upstream with respect to the error microphone 702. The reference microphone 102 may be arranged on the outside of the earphone 116 such that the reference microphone 102 receives the actual noise of the environment. As the error microphone 702 is arranged on the inside of the earphone 116 of the headset, the error microphone 702 receives the noise a user of the headset may perceive.

In operation of the device 700, the reference microphone 102 records a noise signal 118 and outputs a reference microphone signal 122 which is converted by the ADC 104, yielding a delayed signal 124. This signal 124 is filtered by the digital filter 106, and the digitally filtered signal 126 is converted by the DAC 108, yielding a delayed signal 128. The noise reduction signal 120 is played via the loudspeaker 110. The error microphone 702 receives an error signal 708 which is a combination of the ambient noise signal 706 and the noise reduction signal 120. The error signal 708 is then converted by the ADC 704 and forms a digital error signal 710 to be used for operating the digital filter part 112.

The frequency dependency of the ANR active performance $G(\omega)$ of the device 700 is shown in Fig. 8 for different delays Δ . Again, $G(\omega)$ reads

$$G(\omega) = 10 \log_{10} \Phi_a(\omega) - 10 \log_{10} \Phi_e(\omega),$$

with $\Phi_e(\omega)$ denoting the PSD of the error microphone signal 708 and $\Phi_a(\omega)$ denoting the PSD of the ambient noise signal 706 at the error microphone 702.

Curves 802-808 correspond to a delay Δ of zero samples, eight samples, 16 samples, and 26 samples using a sampling rate of 48 kHz. For convenience, a reference ANR performance for an analog, instantaneous first-order low pass filter indicated is indicated by a curve 810, the parameters of which have been optimised in a similar manner. Upon increasing the delay Δ the active performance considerably degrades. In case of the (unrealistic) delay-free conversion indicated by the curve 802, the ANR performance $G(\omega)$ is high. However, upon increasing the delay Δ (curves 802-808), the gain becomes less negative meaning a lower noise reduction.

Fig. 9 shows the corresponding sample index dependency of the impulse response of the digital filter 106 for the delay Δ being zero samples, eight samples, 16 samples, and 26 samples (curves 902-908) for only the first 200 samples, in order to understand the performance degradation of the device 700. Due to the delay Δ the first Δ samples of the impulse response curves 902-908 are equal to zero. Upon increasing the delay Δ the acausal part of the impulse response corresponding to the zero values at low sample indices becomes longer. Thus the corresponding part of the optimal delay-free impulse response (curve 902) cannot be achieved by a causal digital filter 106, and the ANR performance of the device 700 is degraded.

In the following, a method of optimizing the operation of the device 700 will be explained. Under the assumption of the digital filter 106 being a L -taps finite impulse response (FIR) filter 106, minimizing the sound level at the error microphone 702 corresponds to minimizing a least-square cost function of the device 700, namely

$$J(\mathbf{w}_{FIR}) = \int_{\Omega} |D_a(\omega) + S_a(\omega)X_a(\omega)\mathbf{w}_{FIR}^T \mathbf{g}_{FIR}(\omega) \exp(-j\Delta\omega)|^2 d\omega,$$

with $\mathbf{w}_{FIR}$ denoting the digital FIR filter 106, $D_a(\omega)$ denoting the ambient noise 706 perceived by the user, $S_a(\omega)$ denoting the acoustical path 130 assuming a linear model,

and $X_{\alpha}(\omega)$ denoting the reference microphone signal 122 received by the reference microphone 102. In this context, the function $g_{FIR}(\omega)$ equals

$[1 \exp(-i\omega) \dots \exp(-i(L-1)\omega)]^T$. Integration is performed over a frequency range Ω of interest.

Minimization of the cost function results in optimal operational parameters of the digital filter 106, given the acoustical path 130 (without conversion delay Δ) and the delay Δ .

Fig. 10 shows a hybrid ANR device 1000 according to another exemplary embodiment of the invention. The device 1000 comprises a reference microphone 102, a low-latency filter 502, a parallel circuit 508 comprising an ADC 104, a digital filter 106, and a DAC 108 in one branch line and a gain unit 504 in a further bypass branch line 505, a signal adding unit 506, and a loudspeaker 110. A sequence of arrangement of these device components is seen in a signal transmitting direction. The device 1000 comprises an error microphone 702 and a further ADC 704.

When implementing the device 1000 in a headset, the reference microphone 102 is arranged on the outside of an earphone 116 of the headset, and the error microphone 702 and the loudspeaker 110 are arranged on the inside of an earphone 116 of the headset.

In operation of the device 1000, a noise signal 118 is received by the reference microphone 102 and transmitted to the low-latency filter 502. After a low-latency filtering operation the filtered microphone signal 510 is fed into the parallel circuit 508 such that the filtered microphone signal 510 is on the one hand converted into the digital domain, filtered by the digital filter 126 and converted back to the analog domain. Simultaneously, the filtered microphone signal 510 is bypassed over the gain unit 504. The amplified microphone signal 512 and the digitally filtered microphone signal 128 are summed by the signal adding unit 506 to form a filtered microphone signal 514. A noise reduction signal 120 is emitted by the loudspeaker 110. The error microphone 702 receives an ambient noise signal 706 and the noise reduction signal 120. An error signal 708 being transmitted by the error microphone 702 is fed to a further ADC 704. A converted signal 710 may be used for parameter optimization of the device 1000.

A method for determining a first operational parameter of the device 1000 according to another exemplary embodiment of the invention is based on minimizing a cost function describing characteristics of the device 1000. The cost function associated to the characteristics of the device 1000 reads

$$J(\mathbf{w}_{FIR}, \mathbf{p}_\alpha) = \int_{\Omega} |D_\alpha(\omega) + S_\alpha(\omega)X_\alpha(\omega)H_\alpha(\omega)(\mathbf{w}_{FIR}^T \mathbf{g}_{FIR}(\omega) \exp(-i\Delta\omega) + K)|^2 d\omega$$

with the notations given above, $\mathbf{p}_\alpha$ denoting a vector describing the operational parameters of the low-latency filter 502, $H_\alpha(\omega)$ denoting the transfer function of the instantaneous filter 502, and K describing the gain unit 504. As already stated, the kernel function $\mathbf{g}_{FIR}(\omega)$ equals

$$\mathbf{g}_{FIR}(\omega) = \begin{bmatrix} 1 \\ \exp(-i\omega) \\ \dots \\ \exp(-i(L-1)\omega) \end{bmatrix}$$

with L being the number of taps of the high-latency digital filter 106 (mathematical notation $\mathbf{w}_{FIR}$).

Optimizing the cost function may be performed jointly, computing the optimal operational parameters of the gain unit 504 (parameter K) and of the digital filter 106 (parameter $\mathbf{w}_{FIR}$) based on the low-latency filter 502 (parameter $\mathbf{p}_\alpha$).

Minimisation of the cost function with the transfer $H_\alpha(\omega)$ being assumed to be known yields optimal values for the parameter vector $\mathbf{w}_H = [K; \mathbf{w}_{FIR}]$ which is a concatenation of the gain K and the digital filter coefficients $\mathbf{w}_{FIR}$, namely

$$\mathbf{w}_H = -\mathbf{Q}_R^{-1} \int_{\Omega} R \{ \mathbf{g}(\omega) X_\alpha(\omega) S_\alpha(\omega) H_\alpha(\omega) D_\alpha^*(\omega) \} d\omega$$

with $R\{\dots\}$ denoting a real part of the function $\{\dots\}$.

Further, $\mathbf{Q}_R$ and $\mathbf{g}(\omega)$ are defined by the following formulas

$$\mathbf{Q}_R = \int_{\Omega} \mathbf{g}(\omega) \mathbf{g}(\omega)^H |X_\alpha(\omega)|^2 |S_\alpha(\omega)|^2 |H_\alpha(\omega)|^2 d\omega$$

And

$$\mathbf{g}(\omega) = \begin{bmatrix} 1 \\ g_{FIR}(\omega) \exp(-i\Delta\omega) \end{bmatrix}$$

respectively.

In the following, a method for determining an operational parameter of a hybrid ANR device according to another exemplary embodiment of the invention will be described.

Fig. 11 shows a known hybrid ANR device 1100 comprising a feedforward configuration. The device 1100 is based on the device 700 with further comprising a parallel circuit 508 being part of the filter unit 114. A first branch of the parallel circuit 508 corresponds to the arrangement of the ADC 104, the digital filter 106, and the DAC 108. A second branch of the parallel circuit 508 comprises a low-latency filter 502 and a gain unit 504 being provided downstream with respect to the low-latency filter 502. Output signals of the two branches are summed by the signal adding unit 506. The error microphone 702 may be replaced by a temporary measurement microphone.

In operation of the device 1100, the reference microphone 102 records a noise signal 118 and feeds a reference microphone signal 122 to the parallel circuit 508. The signal 122 is filtered by the low-latency filter 106 in the digital domain and meanwhile filtered by the low-latency filter 502 and amplified by the gain unit 504. After signal summation a filtered microphone signal 514 is provided to the loudspeaker 110 which outputs a noise reduction signal 120. Further, the error microphone 702 records a combination of an ambient noise signal 706 and the noise reduction signal 120 and outputs an error signal 708 to the further ADC 104.

The device 1100 combines the (nearly) instantaneous filter response of the low-latency filter 502 with the filter accuracy and complexity of the high-latency digital filter 106. Since the low-latency filter 502 is able to perform a nearly instantaneous filtering operation, the acausal part of the impulse response of the high-latency digital filter 106 may be crudely modelled such that the ANR performance of the device 1100 is largely robust with respect to the presence of delays Δ_{ADC} and Δ_{DAC} of the ADC 104 and the DAC 108, respectively.

Fig. 12 shows a frequency dependency of the active ANR performance $G(\omega)$ of the hybrid device 1100 for different delay values Δ being zero samples, eight samples, 16 samples, and 26 samples indicated by curves 1202-1208. For convenience, the active performance of an analog first-order low pass filter is indicated by a curve 1210. It can be observed that the ANR performance is largely maintained for increasing values of the

delay Δ , with only a slight degradation occurring in a frequency region between about 300 Hz and about 1000 Hz.

Fig. 13 shows the corresponding sample index dependency of the impulse response of the filter unit 114 (sum of the filter response of the low-latency filter 502 with gain unit 504 and the filter response of the high-latency filter 106) for the delays Δ being zero samples, eight samples, 16 samples, and 26 samples (curves 1302-1308). The acausal part of the impulse response is represented by a thick portion of the curves 1302-1308 and may be clearly modelled by exponentially decreasing functions. These functions are the impulse responses of the low-latency first-order low pass filter 502.

The method mentioned above is adapted for optimizing the operation of the device 1100. Two operational parameters of the device 1100 are jointly computed using minimization of a cost function of the device 1100 with respect to the two operational parameters to be determined, namely the cost function

$$J(\mathbf{w}_{FIR}, p_a) = \int_{\Omega} |D_a(\omega) + S_a(\omega)X_a(\omega)(\mathbf{w}_{FIR}^T \mathbf{g}_{FIR}(\omega) \exp(-l\Delta\omega) + K H_a(\omega))|^2 d\omega$$

with the denotations given above. Assuming an analog first-order low pass filter 502, the transfer function of the analog filter 502 is given by the equation $H_a(\omega) = \frac{1}{1+i\omega\tau}$, with τ being the time constant of the analog filter 502.

Minimisation of the cost function with the transfer $H_a(\omega)$ being assumed to be known yields optimal values for the parameter vector $\mathbf{w}_H = [K; \mathbf{w}_{FIR}]$ which is a concatenation of the gain K and the digital filter coefficients $\mathbf{w}_{FIR}$, namely

$$\mathbf{w}_H = -\mathbf{Q}_R^{-1} \int_{\Omega} R \{ \mathbf{g}(\omega) X_a(\omega) S_a(\omega) D_a^*(\omega) \} d\omega$$

with $R\{\dots\}$ denoting a real part of the function $\{\dots\}$.

Further, $\mathbf{Q}_R$ and $\mathbf{g}(\omega)$ are defined by the following formulas

$$\mathbf{Q}_R = \int_{\Omega} \mathbf{g}(\omega) \mathbf{g}(\omega)^H |X_a(\omega)|^2 |S_a(\omega)|^2 d\omega$$

and

$$g(\omega) = \left[\frac{H_a(\omega)}{g_{FIR}(\omega) \exp(-i\Delta\omega)} \right]$$

respectively.

For an easier implementation, it may be convenient to fix τ to a predetermined value and only optimise the gain K in combination with the digital filter coefficients w_{FIR} . Thus, given the reference microphone signal 122, the ambient noise signal 706 recorded on the error microphone 702, an estimate of the acoustical path 130, the length of the codec delay Δ , and the impulse response of the low-latency filter 502, it is possible to determine the optimal gain parameters K of the analog gain unit 504 and the optimal FIR filter w_{FIR} of the digital filter 106 of the hybrid device 1100. It is also possible to use further algorithms for computing the parameters such as an adaptive least mean square based algorithm.

If the error microphone 702 is absent and replaced by a temporary measurement microphone, optimisation of the parameters may then be performed offline, wherein the filter coefficients and the gain parameter may remain fixed during normal operation of the device 1100.

Finally, it should be noted that the above-mentioned embodiments illustrate rather than limit the invention, and that those skilled in the art will be capable of designing many alternative embodiments without departing from the scope of the invention as defined by the appended claims. In the claims, any reference signs placed in parentheses shall not be construed as limiting the claims. The word "comprising" and "comprises", and the like, does not exclude the presence of elements or steps other than those listed in any claim or the specification as a whole. The singular reference of an element does not exclude the plural reference of such elements and vice-versa. In a device claim enumerating several means, several of these means may be embodied by one and the same item of software or hardware. The mere fact that certain measures are recited in mutually different dependent claims does not indicate that a combination of these measures cannot be used to advantage.

It should be noted that the term "comprising" does not exclude other elements or steps and "a" or "an" does not exclude a plurality. Also elements described in association with different embodiments may be combined. It should also be noted that reference signs in the claims should not be construed as limiting the scope of the claims.

CLAIMS:

1. A hybrid active noise reduction device for reducing environmental noise, the device comprising:
 - a microphone for receiving a noise signal and for transmitting a microphone signal;
 - a filter unit for receiving the microphone signal and for transmitting a filtered microphone signal, the filter unit comprising a low-latency filter part and a high-latency filter part, the low-latency filter part and the high-latency filter part being arranged in series to one another; and
 - a loudspeaker for transmitting a noise reduction signal based on the filtered microphone signal.
2. The device of claim 1, further comprising a bypass line being arranged in parallel to the high-latency filter part.
3. The device of claim 2, further comprising a signal adding unit for adding a bypassed signal and a high-latency digitally filtered signal.
4. The device of claim 2, wherein the bypass line comprises a gain unit.
5. The device of claim 2, wherein the signal adding unit is adapted to sum the bypassed signal and the high-latency digitally filtered signal.
6. The device of claim 1, wherein the low latency filter part (501) is arranged upstream with respect to the high latency filter part.
7. The device of claim 1,

further comprising a further microphone for receiving a further noise signal and for transmitting a further microphone signal, wherein the low-latency filter part and/or the high-latency filter part operate based on the further microphone signal.

8. The device of claim 1,
wherein the high-latency filter part is a digital filter part comprising a digital filter.

9. The device of claim 1,
wherein the low-latency filter part is an analog filter part comprising an analog filter.

10. A method for determining an operational parameter of a hybrid active noise reduction device, the active noise reduction device comprising a microphone for receiving a noise signal and for transmitting a microphone signal, a filter unit for receiving the microphone signal and for transmitting a filtered microphone signal, the filter unit comprising an low-latency filter part and a high-latency filter part, and a loudspeaker for transmitting a noise reduction signal based on the filtered microphone signal,
the method comprising selecting a first operational parameter of the device and determining a second operational parameter of the device based on the first operational parameter.

11. The method of claim 10,
wherein the device further comprises a bypass line, the bypass line being arranged in parallel to the high-latency filter part and comprising a gain unit, wherein the method comprises selecting the first operational parameter and determining a second operational parameter and a third operational parameter of the device based on the first operational parameter.

12. The method of claim 11,
wherein the first operational parameter is an operational parameter of the low-latency filter part, wherein the second operational parameter is an operational parameter of the high-latency filter part, and wherein the third operational parameter is an operational parameter of the gain unit.

13. The method of claim 10, wherein determining the second operational parameter based on the first operational parameter comprises minimizing a cost function of the device with respect to the second operational parameter.

14. The method of claim 11, wherein determining the second operational parameter and the third operational parameter based on the first operational parameter comprises minimizing a cost function of the device with respect to the second operational parameter and the third operational parameter.

15. A program element, which, when being executed by a processor, is adapted to control or carry out a method of any of claims 10 to 14.

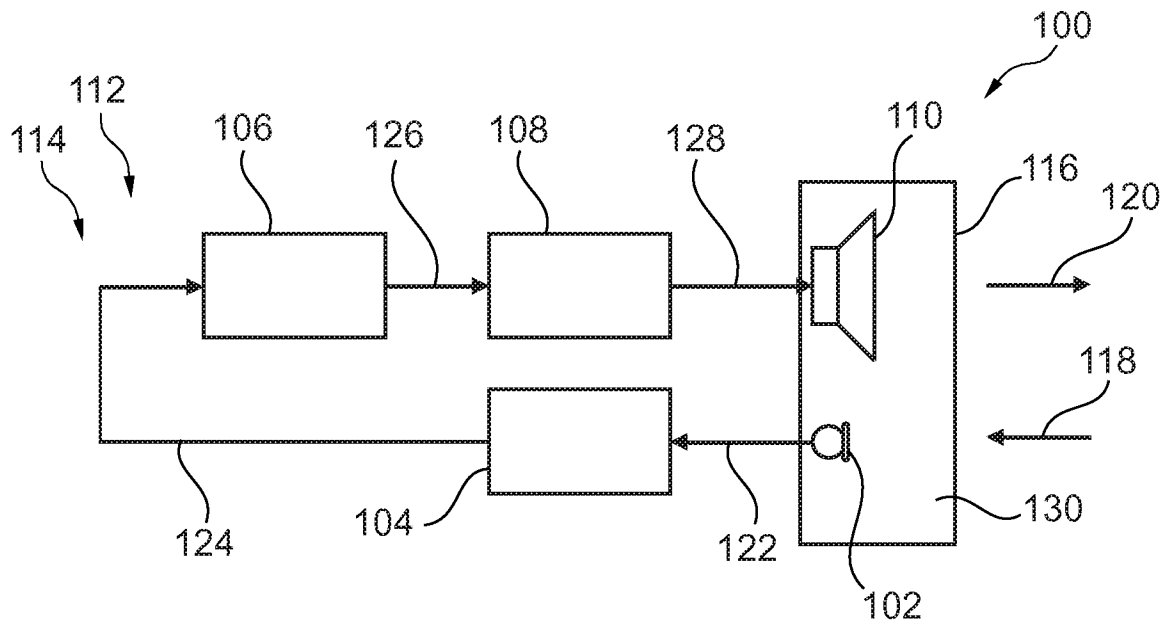


Fig. 1

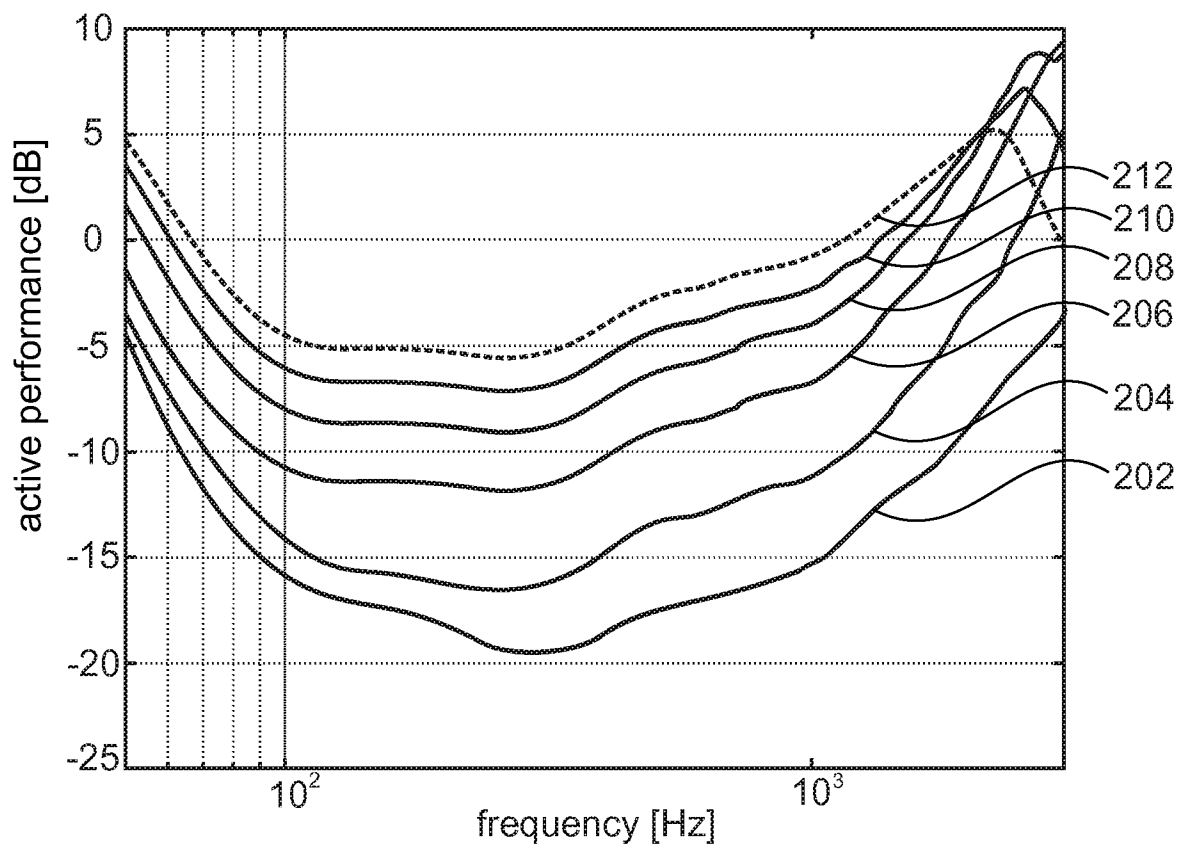


Fig. 2

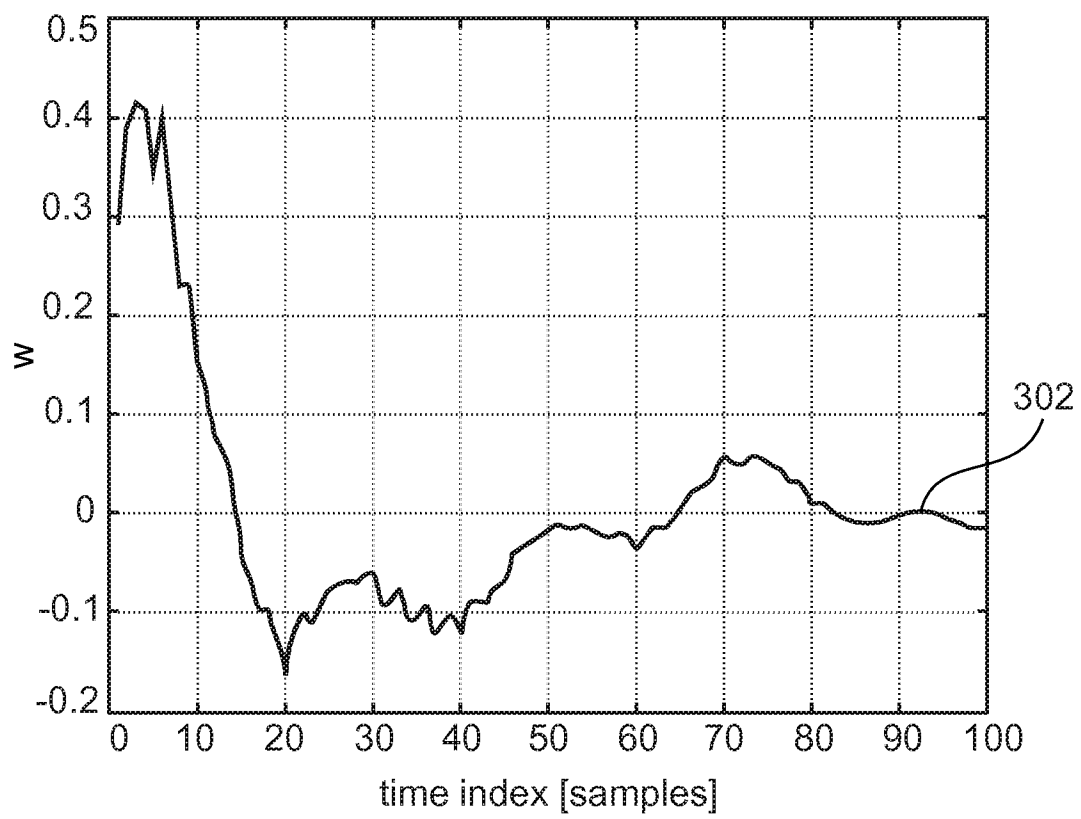


Fig. 3

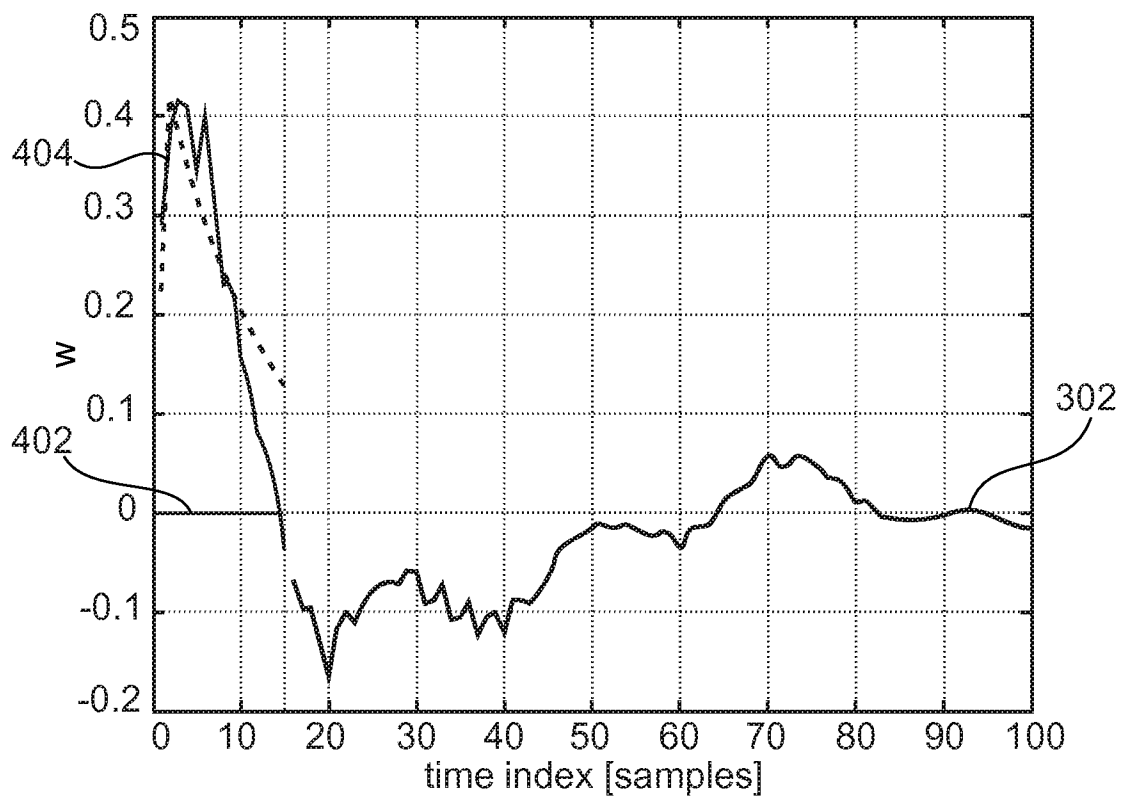


Fig. 4

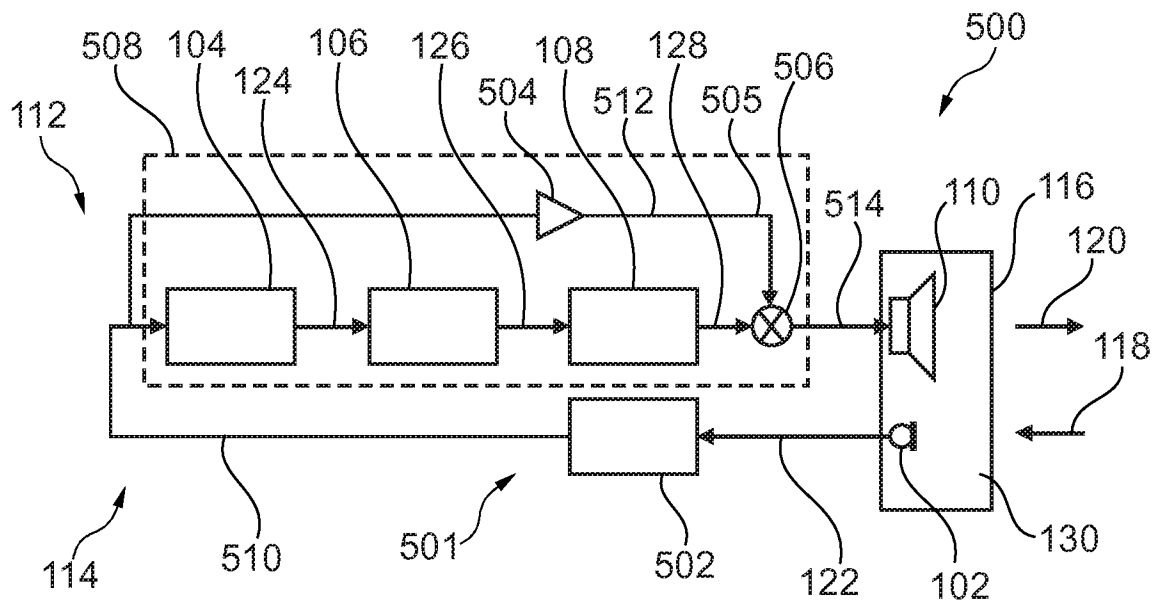


Fig. 5

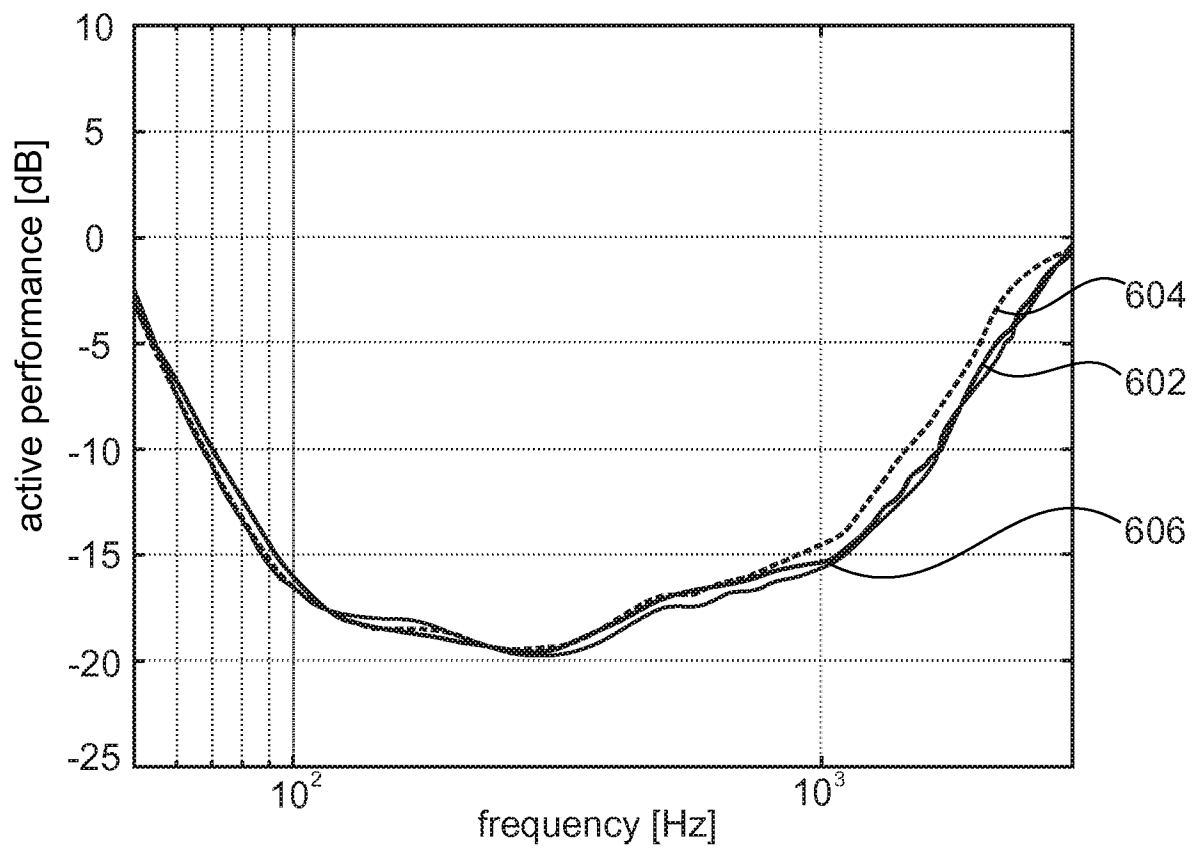


Fig. 6

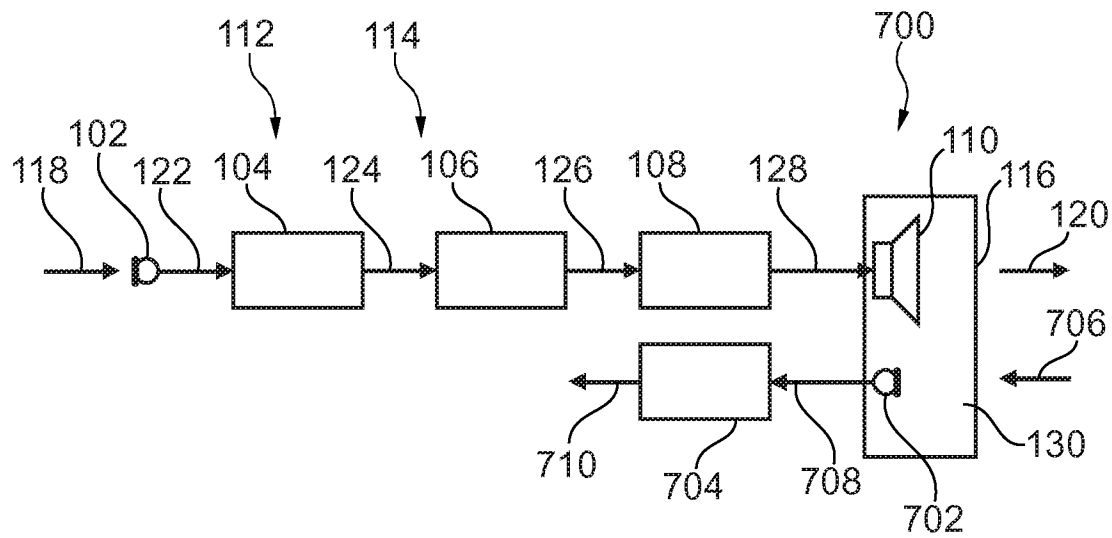


Fig. 7

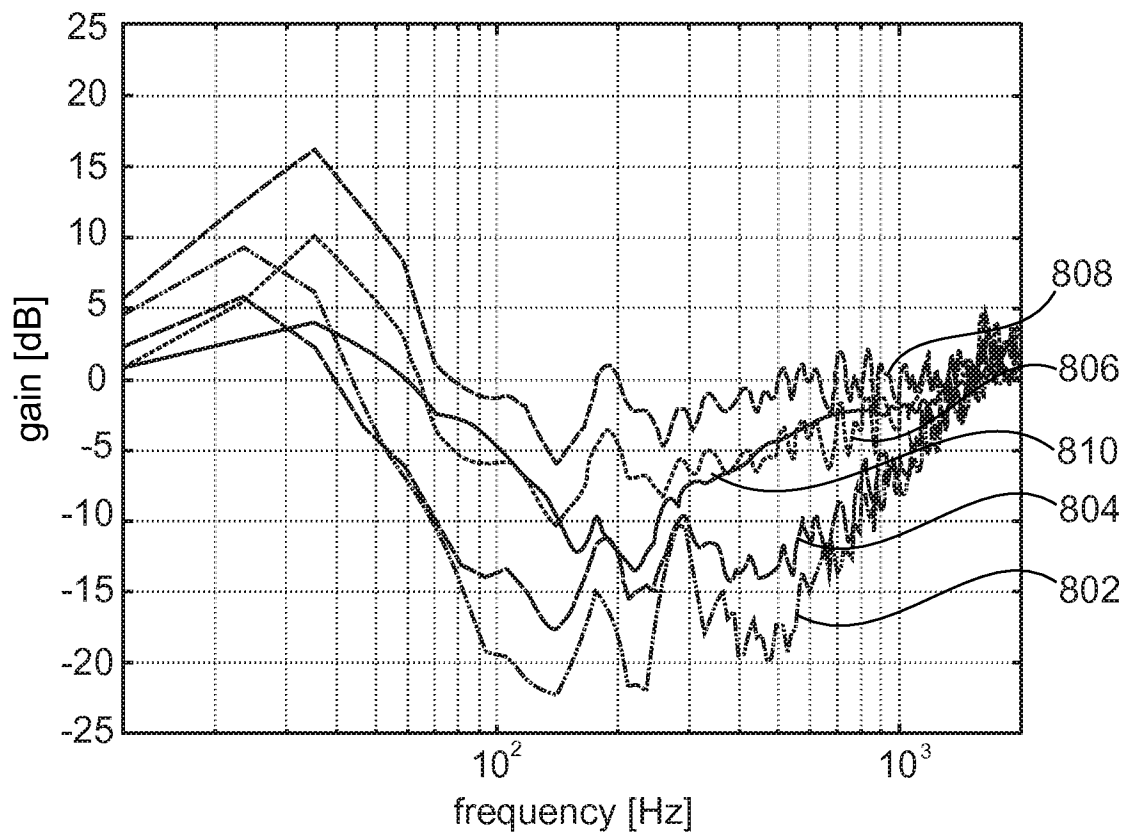


Fig. 8

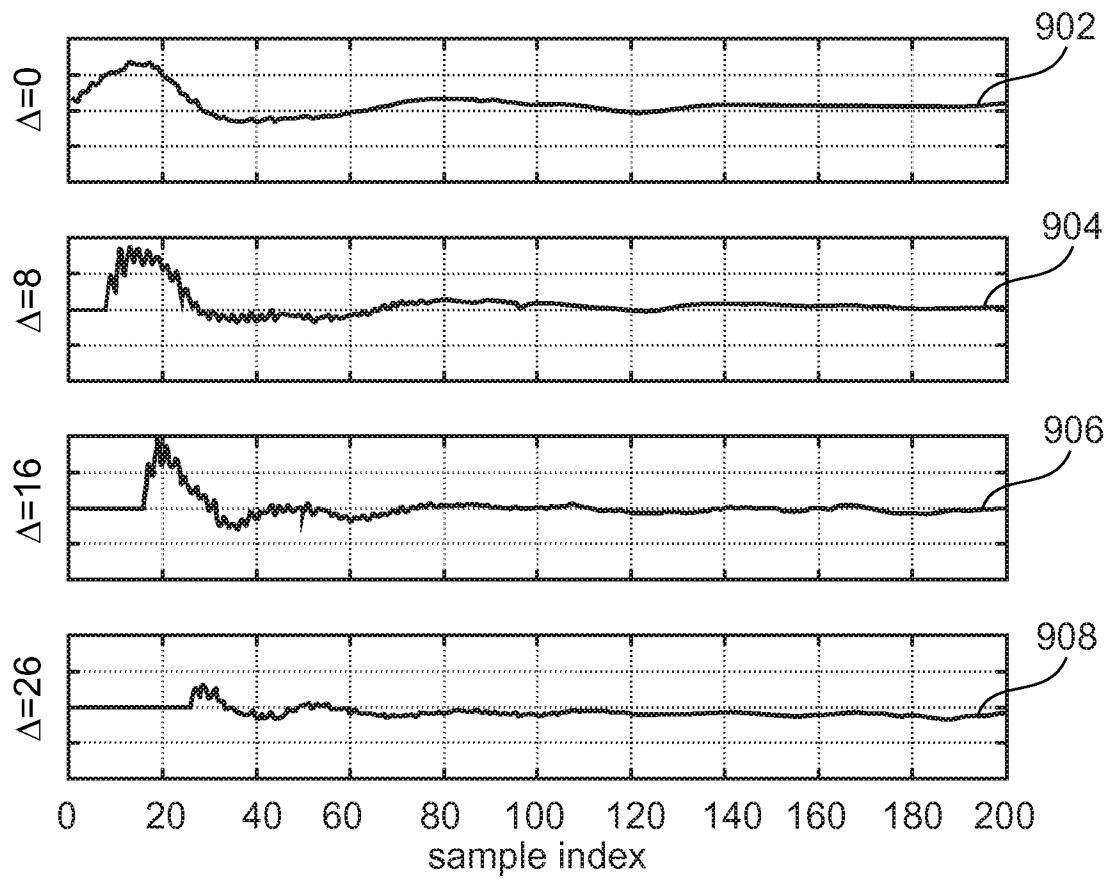


Fig. 9

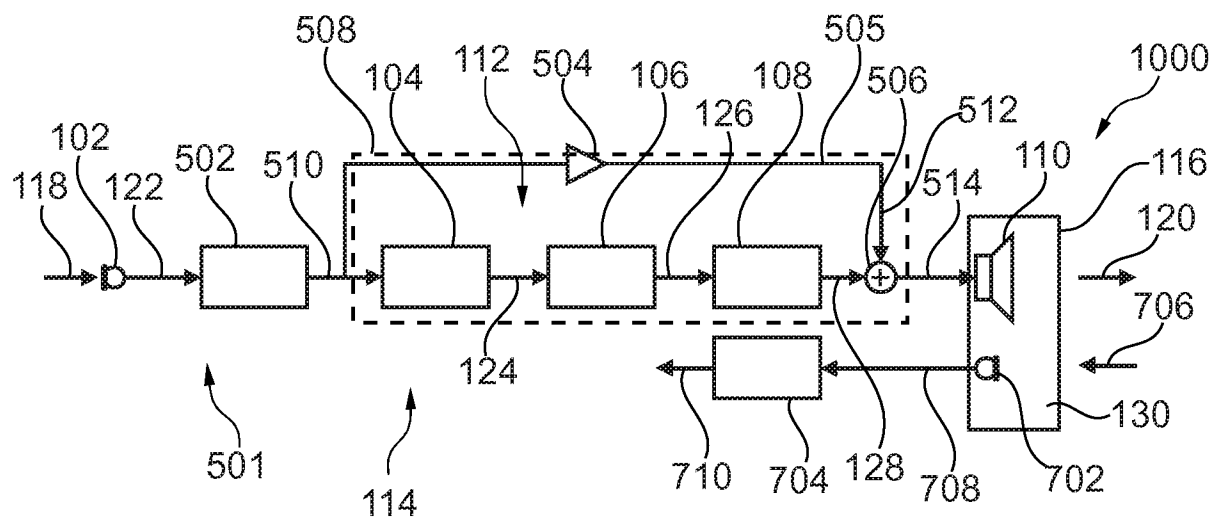


Fig. 10

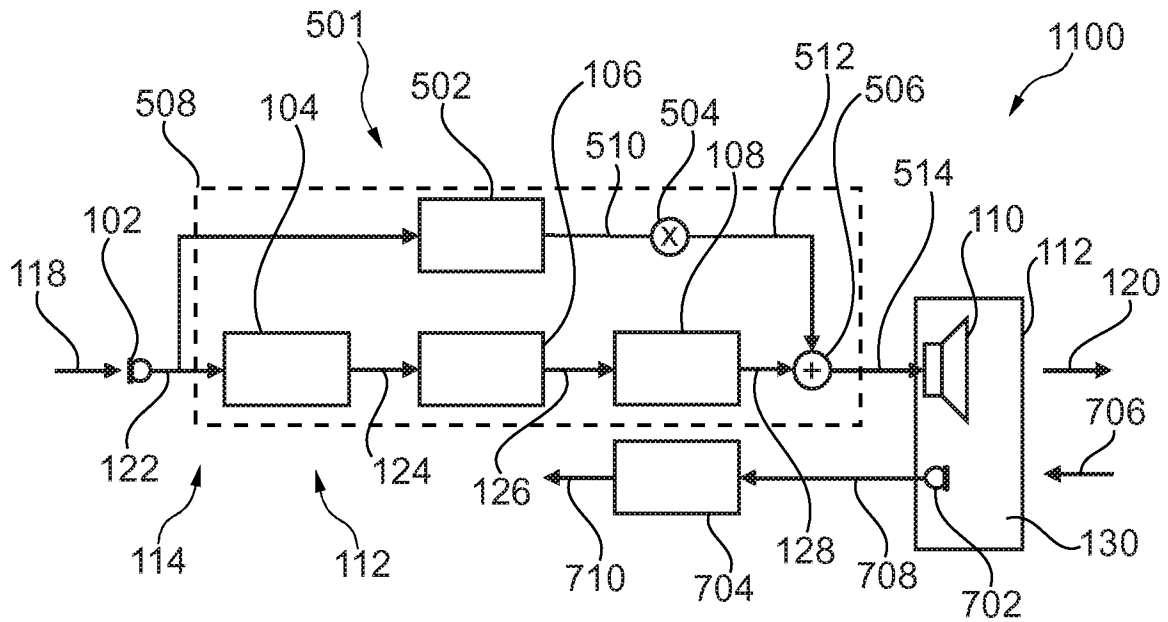


Fig. 11

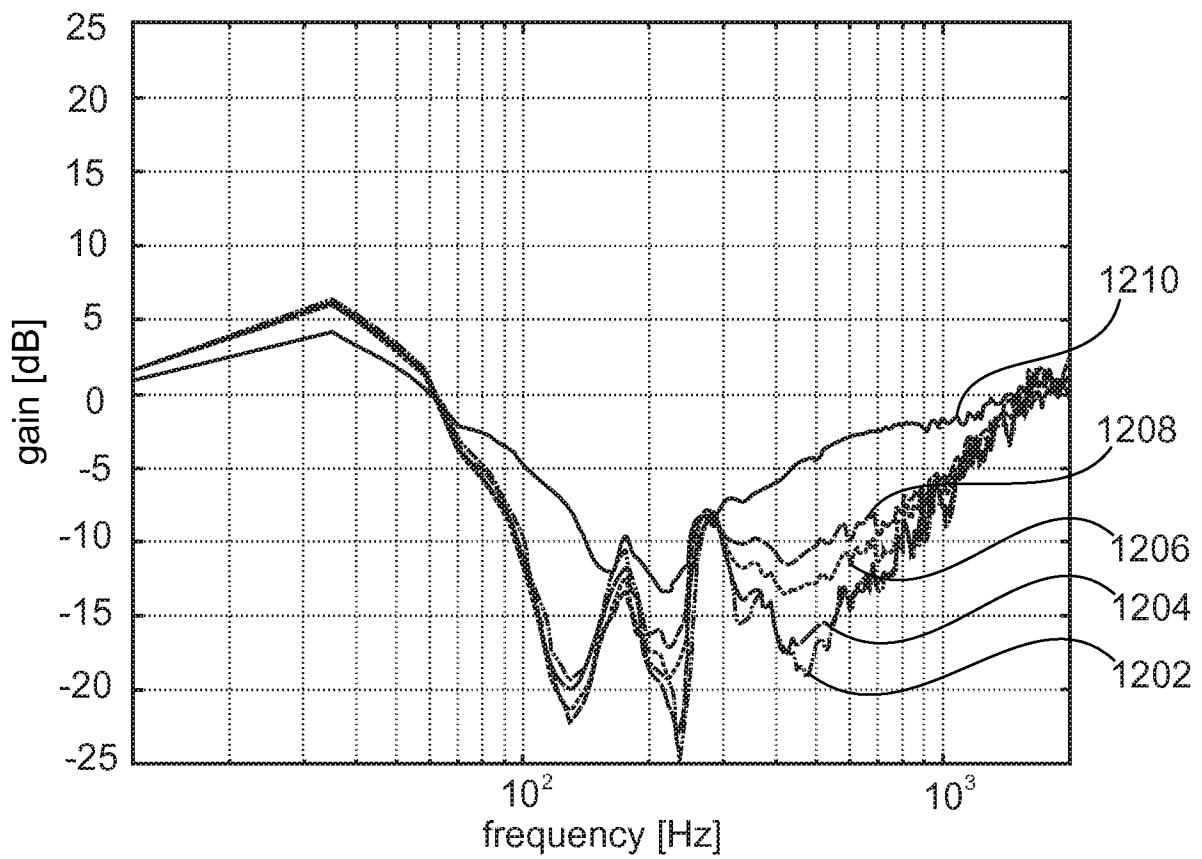


Fig. 12

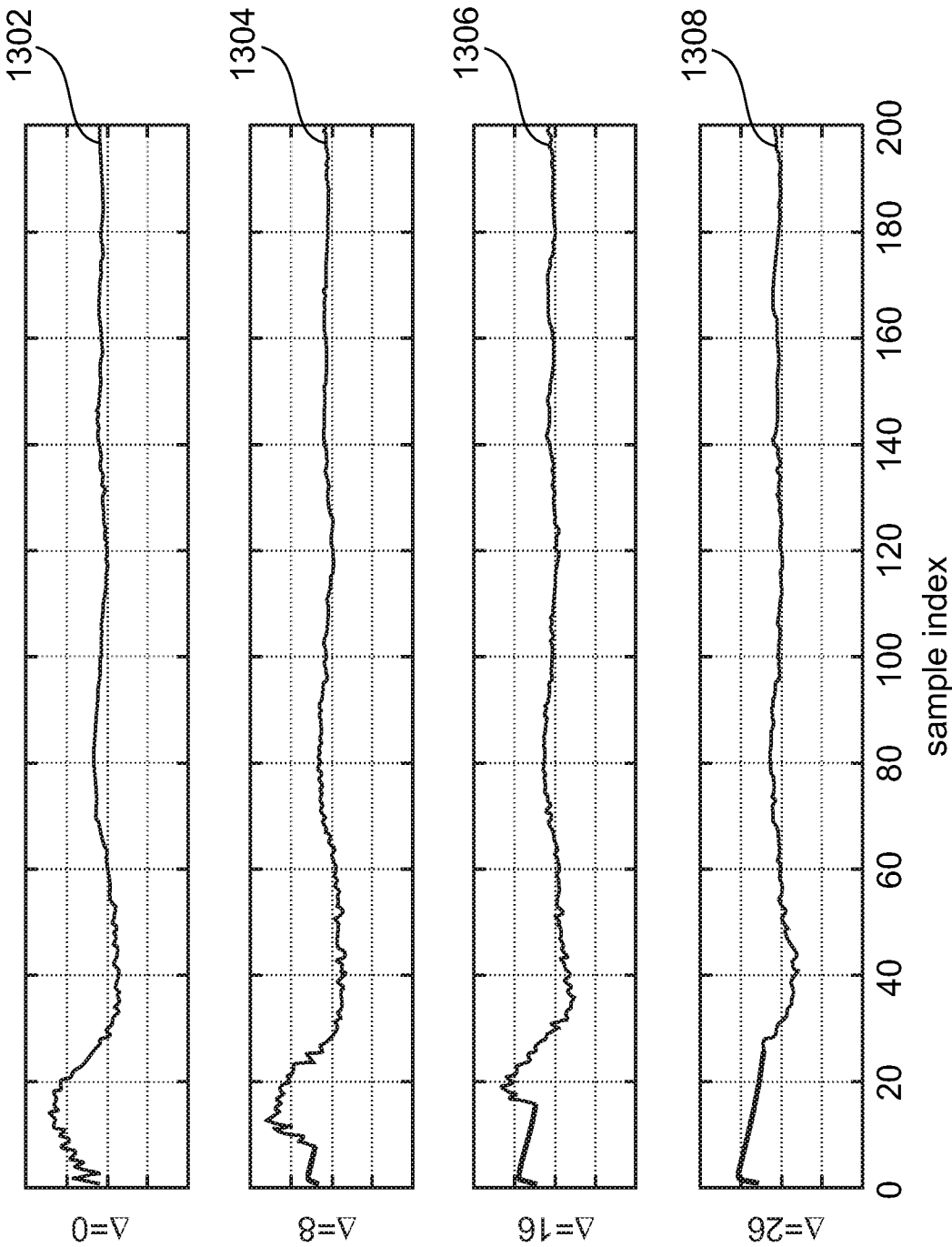


Fig. 13