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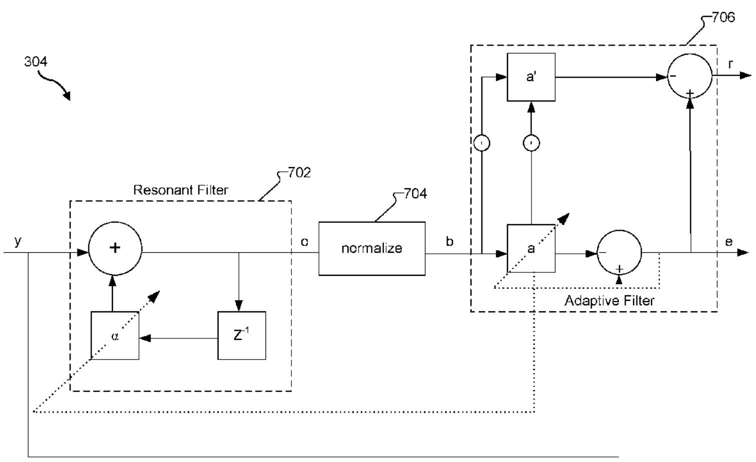

Fig. 7

(57) **Abstract**: An exemplary sound processor included in a cochlear implant system associated with a patient may include a filter system that 1) receives an input signal that includes audio content representative of an audio signal presented to the patient and tonal noise introduced by a noise source within the cochlear implant system, 2) tracks a fundamental frequency and a phase of the tonal noise within the input signal, 3) generates a reference noise signal based on the tracked frequency and the tracked phase, and 4) subtracts the reference noise signal from the input signal to generate a filtered version of the input signal. Corresponding systems and methods are also disclosed.

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SYSTEMS AND METHODS FOR CANCELLING TONAL NOISE IN A COCHLEAR IMPLANT SYSTEM

BACKGROUND INFORMATION

[0001] Many cochlear implant systems are equipped with a telecoil, which is configured to detect electromagnetic fields representative of audio content. Such electromagnetic fields may be generated by telephones, audio induction loops, and/or other types of assistive listening systems. A telecoil may be used in place of (or in addition to) a microphone as the audio input source to a cochlear implant system, and, in some cases, may provide a cochlear implant user with a clearer, more accurate representation of sound than the microphone.

[0002] Unfortunately, a telecoil can also detect electromagnetic fields produced by various sources associated with (e.g., internal to) a cochlear implant system. For example, a telecoil may detect tonal noise (i.e., discrete frequency noise) produced by a cochlear implant system's power supply. Clearly, it is desirable to remove this tonal noise from the signal output by the telecoil before the signal is processed and ultimately presented to the patient in the form of electrical stimulation. However, doing so has heretofore been difficult and costly to achieve. This is due, in part, to the fact that the fundamental frequency of such tonal noise may change over time. For example, as the voltage level of a battery included in a cochlear implant system's power supply depletes over time, the frequency at which the power supply operates may change to accommodate the drop in voltage.

BRIEF DESCRIPTION OF THE DRAWINGS

[0003] The accompanying drawings illustrate various embodiments and are a part of the specification. The illustrated embodiments are merely examples and do not limit the scope of the disclosure. Throughout the drawings, identical or similar reference numbers designate identical or similar elements.

[0004] FIG. 1 illustrates an exemplary cochlear implant system according to principles described herein.

[0005] FIG. 2 illustrates a schematic structure of the human cochlea according to principles described herein.

[0006] FIG. 3 illustrates exemplary components of a sound processor according to principles described herein.

[0007] FIGS. 4-6 illustrate exemplary configurations of the sound processor of FIG. 3 according to principles described herein.

[0008] FIG. 7 illustrates exemplary components of a filter system according to principles described herein.

[0009] FIG. 8 illustrates another exemplary configuration of the sound processor of FIG. 3 according to principles described herein.

[0010] FIG. 9 illustrates an exemplary method of cancelling tonal noise in a cochlear implant system according to principles described herein.

DETAILED DESCRIPTION

[0011] Systems and methods for cancelling tonal noise in a cochlear implant system are described herein. As will be described below, a sound processor included in a cochlear implant system associated with a patient may include a filter system that 1) receives an input signal that includes audio content representative of an audio signal presented to the patient and tonal noise introduced by a noise source within or outside the cochlear implant system, 2) tracks a fundamental frequency and a phase of the tonal noise within the input signal, 3) generates a reference noise signal based on the tracked frequency and the tracked phase, and 4) subtracts the reference noise signal from the input signal to generate a filtered version of the input signal.

[0012] As used herein, “tonal noise” refers to discrete frequency noise characterized by spectral tones that are pure tones in nature. In other words, tonal noise generated by a particular noise source (e.g., a power supply, a digital switching circuit, an electromagnetic source, etc.) is associated with a single fundamental frequency.

[0013] By cancelling (i.e., filtering out) tonal noise (either completely or partially) in a cochlear implant system, the systems and methods described herein may provide an improved hearing experience to the patient compared to conventional cochlear implant systems that do not account for and remove tonal noise. Moreover, the systems and methods described herein may track variations in the fundamental frequency and/or phase of tonal noise generated by a particular noise source, thereby ensuring that the tonal noise is appropriately removed from the cochlear implant system.

[0014] FIG. 1 illustrates an exemplary cochlear implant system 100. As shown, cochlear implant system 100 may include various components configured to be located external to a patient including, but not limited to, a microphone 102, a sound processor 104, and a headpiece 106. Cochlear implant system 100 may further include various components configured to be implanted within the patient including, but not limited to, a cochlear implant 108 and a lead 110 (also referred to as an electrode array) with a plurality of electrodes 112 disposed thereon. As will be described in more detail below, additional or alternative components may be included within cochlear implant system 100 as may serve a particular implementation. The components shown in FIG. 1 will now be described in more detail.

[0015] Microphone 102 may be configured to detect audio signals presented to the patient. Microphone 102 may be implemented in any suitable manner. For example, microphone 102 may include a microphone that is configured to be placed within the concha of the ear near the entrance to the ear canal, such as a T-MIC™ microphone from Advanced Bionics. Such a microphone may be held within the concha of the ear near the entrance of the ear canal by a boom or stalk that is attached to an ear hook configured to be selectively attached to sound processor 104. Additionally or alternatively, microphone 102 may be implemented by one or more microphones disposed within headpiece 106, one or more microphones disposed within sound processor 104, one or more beam-forming microphones, and/or any other suitable microphone as may serve a particular implementation.

[0016] Sound processor 104 (i.e., one or more components included within sound processor 104) may be configured to direct cochlear implant 108 to generate and apply electrical stimulation (also referred to herein as “stimulation current”) representative of one or more audio signals (e.g., one or more audio signals detected by microphone 102, input by way of an auxiliary audio input port, etc.) to one or more stimulation sites associated with an auditory pathway (e.g., the auditory nerve) of the patient. Exemplary stimulation sites include, but are not limited to, one or more locations within the cochlea, the cochlear nucleus, the inferior colliculus, and/or any other nuclei in the auditory pathway. To this end, sound processor 104 may process the one or more audio signals in accordance with a selected sound processing strategy or program to generate appropriate stimulation parameters for controlling cochlear implant 108. Sound processor 104 may include or be implemented by a behind-the-ear (“BTE”) unit, a body worn device, and/or any other sound processing unit as may serve a particular implementation. For example, sound processor 104 may be implemented by an electro-acoustic stimulation ("EAS") sound processor included in an EAS system configured to provide electrical and acoustic stimulation to a patient.

[0017] In some examples, sound processor 104 may wirelessly transmit stimulation parameters (e.g., in the form of data words included in a forward telemetry sequence) and/or power signals to cochlear implant 108 by way of a wireless communication link 114 between headpiece 106 and cochlear implant 108. It will be understood that communication link 114 may include a bi-directional communication link and/or one or more dedicated uni-directional communication links.

[0018] Headpiece 106 may be communicatively coupled to sound processor 104 and may include an external antenna (e.g., a coil and/or one or more wireless communication components) configured to facilitate selective wireless coupling of sound processor 104 to cochlear implant 108. Headpiece 106 may additionally or alternatively be used to selectively and wirelessly couple any other external device to cochlear implant 108. To this end, headpiece 106 may be configured to be affixed to the patient's head and positioned such that the external antenna housed within headpiece 106 is communicatively coupled to a corresponding implantable antenna (which may also be implemented by a coil and/or one or more wireless communication components) included within or otherwise associated with cochlear implant 108. In this manner, stimulation parameters and/or power signals may be wirelessly transmitted between sound processor 104 and cochlear implant 108 via a communication link 114 (which may include a bi-directional communication link and/or one or more dedicated uni-directional communication links as may serve a particular implementation).

[0019] Cochlear implant 108 may include any type of implantable stimulator that may be used in association with the systems and methods described herein. For example, cochlear implant 108 may be implemented by an implantable cochlear stimulator. In some alternative implementations, cochlear implant 108 may include a brainstem implant and/or any other type of active implant or auditory prosthesis that may be implanted within a patient and configured to apply stimulation to one or more stimulation sites located along an auditory pathway of a patient.

[0020] In some examples, cochlear implant 108 may be configured to generate electrical stimulation representative of an audio signal processed by sound processor 104 (e.g., an audio signal detected by microphone 102) in accordance with one or more stimulation parameters transmitted thereto by sound processor 104. Cochlear implant

108 may be further configured to apply the electrical stimulation to one or more stimulation sites within the patient via one or more electrodes 112 disposed along lead 110 (e.g., by way of one or more stimulation channels formed by electrodes 112). In some examples, cochlear implant 108 may include a plurality of independent current sources each associated with a channel defined by one or more of electrodes 112. In this manner, different stimulation current levels may be applied to multiple stimulation sites simultaneously (also referred to as “concurrently”) by way of multiple electrodes 112.

[0021] FIG. 2 illustrates a schematic structure of the human cochlea 200 into which lead 110 may be inserted. As shown in FIG. 2, the cochlea 200 is in the shape of a spiral beginning at a base 202 and ending at an apex 204. Within the cochlea 200 resides auditory nerve tissue 206, which is denoted by Xs in FIG. 2. The auditory nerve tissue 206 is organized within the cochlea 200 in a tonotopic manner. Relatively low frequencies are encoded at or near the apex 204 of the cochlea 200 (referred to as an “apical region”) while relatively high frequencies are encoded at or near the base 202 (referred to as a “basal region”). Hence, each location along the length of the cochlea 200 corresponds to a different perceived frequency. Cochlear implant system 100 may therefore be configured to apply electrical stimulation to different locations within the cochlea 200 (e.g., different locations along the auditory nerve tissue 206) to provide a sensation of hearing.

[0022] FIG. 3 illustrates exemplary components of sound processor 104. It will be recognized that the components shown in FIG. 3 are merely representative of the many different components that may be included in sound processor 104 and that sound processor 104 may include additional or alternative components as may serve a particular implementation.

[0023] As shown, sound processor 104 may include an analog-to-digital converter (“ADC”) 302, a filter system 304, and processing circuitry 306. Each of these components will now be described in more detail.

[0024] As shown, ADC 302 may receive an analog input signal and convert the analog input signal into a digital input signal, represented by “y” in FIG. 3. In some cases, the analog input signal may include both audio content representative of an audio signal presented to the patient and tonal noise introduced by a noise source associated with (e.g., included within or external to) the cochlear implant system 100. Hence, both the analog input signal and its digital representation (i.e., the signal represented by “y”) may be referred to herein as a “corrupted input signal”. As used herein, the term “input signal” refers interchangeably to either the analog input signal supplied to ADC 302 or the digital representation of the analog input signal that is output by ADC 302.

[0025] The corrupted input signal may be provided by any suitable input source included within and/or otherwise associated with cochlear implant system 100. For example, FIG. 4 shows an exemplary configuration 400 in which the input signal may be provided by a telecoil 402 included in sound processor 104. As shown, telecoil 402 may detect an audio signal (which, in this case, is encoded in the form of an electromagnetic signal) presented to the patient. However, telecoil 402 may concurrently detect tonal noise, as also illustrated in FIG. 4. The audio signal and the tonal noise combine to form the analog input signal generated by telecoil 402 and input into ADC 302.

[0026] FIG. 5 illustrates an alternative configuration 500 in which the input signal is provided by a microphone 502. Microphone 502 may include any of the microphones described herein. For example, microphone 502 may include a microphone that is configured to be placed within the concha of the ear near the entrance to the ear canal, a microphone disposed within headpiece 106, and/or any other microphone as may serve a particular implementation. As shown, the signal detected by microphone 502 may include a combination of an audio signal presented to the patient and tonal noise generated by a noise source associated with cochlear implant system 100.

[0027] In some embodiments, the input signal may include a mixture of a signal provided by a microphone included in cochlear implant system 100 and a telecoil included in sound processor 104. To illustrate, FIG. 6 illustrates an exemplary configuration 600 in which the analog input signal input into ADC 302 is provided by a mixer 602. As shown, mixer 602 receives a first signal provided by microphone 502 and a second signal provided by telecoil 402. Mixer 602 may combine the first and second signals and provide the combined signal to ADC 302 in the form of the analog input signal. Alternatively, two ADCs may encode the two input analog signals separately, and the ensuing digital signals may be mixed together.

[0028] Filter system 304 may be configured to cancel the tonal noise from the input signal provided by ADC 302. As used herein, tonal noise may be “cancelled” by at least partially removing or filtering out the tonal noise from the input signal. Filter system 304 may output a filtered version of the input signal, which is represented by "r" in FIGS. 3-7 and has the tonal noise at least partially cancelled therefrom.

[0029] To generate the filtered version of the input signal, filter system 304 may track (e.g., in real time as the input signal is received) a fundamental frequency, a phase, and/or a magnitude of the tonal noise within the input signal, generate a reference noise signal based on the tracked attributes of the tonal noise, and subtract the reference noise signal from the input signal to generate a filtered version of the input signal. In some cases, the subtraction of the reference noise signal from the input signal at least partially cancels the tonal noise from the input signal.

[0030] FIG. 7 illustrates exemplary components of filter system 304. It will be recognized that filter system 304 may include additional or alternative components as may serve a particular implementation. As shown, filter system 304 may include a resonant filter 702 that receives the input signal "y" and outputs an output signal "o", which is a signal that has the tracked frequency. Filter system 304 may further include a normalizer 704, which normalizes the output signal "o" output by the resonant filter 702 to give a basis signal "b" (which is the reference noise signal). Filter system 304 may further include an adaptive filter 706, which processes the basis signal "b" and the input signal "y" to output the filtered version of the input signal, which is represented by the signal "e". Each of these components will now be described in more detail.

[0031] Resonant filter 702 is a single pole filter with a z-transform represented by $H(z) = \lambda / (1-\alpha z^{-1})$. Here, λ is a scale factor and α is a complex number, chosen to be is equal to $|\alpha|e^{j\omega}$. $|\alpha|$ is a real number constant on which the sharpness of the filter depends. The closer this constant is to 1, the narrower the filter is. An exemplary value of $|\alpha|$, which should always be less than one, is 0.9999. An exemplary value of λ is 1. Another exemplary value of λ is $(1-\alpha)$. The variable ω is an angular resonant frequency given by $2\pi f/f_s$, where f is the center frequency of the resonant filter 702 and f_s is the sampling frequency. This ω is set to the expected angular frequency of the fundamental in question. For example, ω may be set to be equal to the expected fundamental angular frequency of the tonal noise output by a particular noise source included in cochlear implant system 100.

[0032] The output "o" of the resonant filter 702 (i.e., the recurrence relation form of the resonant filter 702) is given by the equation $o(i) = \lambda y(i) + \alpha o(i-1)$, for every time step "i".

[0033] The output "o" is a complex signal whose phase is the same phase as the input complex sinusoidal of the tracked fundamental frequency, only if the frequency is being tracked accurately, i.e. only if ω is the accurate angular frequency of the tonal noise being tracked. Otherwise, the phase difference between "o" and the complex sinusoidal being tracked quickly rushes to +/- π , even for a small frequency tracking error.

[0034] Normalizer 704 normalizes the output signal "o" of the resonant filter 702. The normalized output of normalizer 704 is a basis signal "b". In some examples, the basis signal "b" is numerically equal to resonant filter output "o", i.e., b(i) = o(i). In some examples, "b" is obtained by dividing resonant filter output "o" with the magnitude (absolute value) of "o", i.e. b(i) = o(i) / |o(i)|.

[0035] This basis signal may also be referred to as a reference noise signal. Advantageously, the reference noise signal is generated from the signal corrupted by noise itself (i.e., from the input signal "y") and is used by adaptive filter 706 to cancel the noise signal from the corrupted input signal. This is because the resonant filter 702, when normalized by normalizer 704, creates an output waveform of nearly a single complex frequency, i.e., "b" will be something very close to $e^{j\omega 0}$ where ω0 is the true frequency of the tonal noise present in the input signal "y". This basis is used to predict the noise corresponding to frequency ω in input signal "y" and the best prediction is removed from the input to give a difference signal "e".

[0036] Adaptive filter 706 receives the reference noise signal "b" and subtracts the reference noise signal "b" multiplied by a complex number gain "a" from the input signal "y" to generate the filtered version of the input signal "e" (also referred to as the error signal "e"). Thus e(i) = y(i) – a(i)b(i).

[0037] The output "e" will have the frequency ω removed as much as possible. The complex number "a" is a best predictor of the noise at frequency ω from the basis signal "b". As the input signal "y" is a real signal, it would be constituted of complex sinusoidals of both positive and negative frequencies. If the basis signal "b" models the positive noise frequency present in the input signal "y", then b' would model the corresponding negative frequency. As the filtered output "r" is just the noise cancelled version of "y", it has to be a real signal. Hence the best predictor of frequency (–ω) from b' in the noise is a'. Therefore, the filtered version of the input signal may be represented by r(i) = y(i) – a(i)b(i) – a'(i)b'(i). In this equation, (') indicates complex conjugation. Thus, by only predicting the component of one of the complex frequencies, the systems and methods described herein automatically determine the component for its conjugate frequency, which in turn cancels both the positive and negative frequency components of the noise. The above relation for calculating r(i) can also be computed using the equation $r(i) = y(i) - 2\,(\,Re(a(i))\,Re(b(i)) - Im(a(i))\,Im(b(i))\,)$, where Re and Im respectively stand for the real and imaginary parts of a complex number.

[0038] In the preceding equation, a(i), a complex number for each time step i, is the estimate of correlation between complex frequency ω present in y(i) and b(i). This estimate is adapted in such a way that output “e” will have the frequency ω removed as much as possible. This adaptation may be performed by the following update relation: $a(i + 1) = a(i) + \mu e(i)b'(i)$. A real number μ times the difference signal “e” times the conjugate of the present value of the basis signal is added to the present value of the estimate of correlation “a”, to give the estimate of correlation “a” for the next time step. In some examples, the adaptation of “a” may be performed by the following update relation: $a(i + 1) = a(i) + \mu e(i)b'(i) / |y(i)|^2$.

[0039] The real number μ can be selected in many ways, depending on the normalization equation used to generate basis signal “b” and the equation used to adapt “a”.

[0040] The angular frequency of tonal noise ω_0 can vary with time. Therefore, for more accurate and error free tracking, it is desirable to adapt ω, the resonant angular frequency of the resonant filter 702, to ω_0, the angular frequency of the noise to be tracked and removed i.e., it is desirable for ω to be equal to ω_0. To be able to achieve this, the systems and methods use a property of the resonant filter 702 itself, which is that the resonant filter output leads or lags the input depending upon whether the input noise frequency is lower or higher than the center frequency of the resonant filter. If “o” and hence “b” leads or lags the component of “y” having angular frequency ω, then the effect will be felt on the phase angle of “a”. This is because phase angle of “a” will adapt to minimize the effect of the phase lag or lead between “b” and the component of y having angular frequency ω. To give ω for the next time step, the following equation may be used: $\omega(i+1) = \omega(i) + \beta \arg(a(i))$. In this equation, β times the phase angle of “a” is added to the present value of the estimate of resonant angular frequency of the resonant filter 702.

[0041] In some examples, to avoid using trigonometric operations involved in calculating the phase angle of “a”, feedback is given from the imaginary part of “a” to update the center frequency ω. For example, feedback proportional to the imaginary part of "a" may be given by $\omega(i + 1) = \omega(i) + \beta\ Im\,(a(i))$, where *Im* stands for imaginary part.

[0042] In some examples, feedback proportional to the cube of imaginary part of "a" may be given by $\omega(i + 1) = \omega(i) + \beta(Im\,(a(i)))^3$, where *Im* stands for imaginary part.

[0043] The above algorithm uses two rate of adaptation parameters, μ and β. Frequency (ω) as well as phase and gain ("a") adaptation works better under no-signal condition, i.e., when only noise is audible. Hence, the rate of adaptation is reduced as the input signal (sound) level "l" increases. In some examples, the level "l" is detected by averaging the half wave rectified version of the input signal over the past few samples. In another example, "l" is detected by using. the equation $l(i) = \max(\ y(i), \eta l(i - 1)\ )$. The maximum value between a constant η times the input signal level "l" and the instantaneous value of input signal "y" is selected as the estimate of input signal level "l" for the next time step. In yet another example, $l(i) = \max(\ |y(i)|, \eta l(i - 1)\ )$. The maximum value between a constant η times the input signal level "l" and the absolute value of the instantaneous value of input signal "y" is selected as the estimate of input signal level "l" for the next time step. Here decay parameter η dictates how quickly the the level signal follows the input. The larger the value of η, slower is the rate of change of the level signal. Thus μ and β are chosen such that, the more the value of level signal "l" , the slower the adaption of phase and gain will be. Thus, both μ and β, are chosen to be inversely related to "l". In some examples, adaption parameters μ and β are calculated using equations: $\mu = C1 \times (C2)^{-l(i)}$ and $\beta = C3 \times (C4)^{-l(i)}$. Exemplary values of the these constants are $C1 = 2\times10^{-4}$, $C2 = 2^{1000}$, $C3 = 3\times10^{-9}$, and $C4 = 2^{1000}$.

[0044] Filter system 304 may also filter harmonic and inter-modulation harmonics of the tonal noise without having to track them individually. For example, if a particular harmonic of angular frequency ω is related to its fundamentals (of angular frequency ω1, ω2, ...) by $\omega = n1\omega1 + n2\omega2 \ldots$, then the basis signal "b" corresponding to the harmonic of angular frequency ω is generated by taking complex product $b1^{n1} * b2^{n2} * \ldots$, where b1, b2, ... are the basis signals generated for the fundamentals of angular frequencies ω1, ω2... . Thus, $b = b1^{n1}b2^{n2}\ldots$ Once the harmonic basis is generated, the removal follows the same adaptive filter technique used to remove the fundamentals, namely $e(i) = y(i) - a(i)b(i)$, $a(i+1) = a(i) + \delta e(i)b'(i)$ and $r(i) = y(i) - a(i)b(i) - a'(i)b'(i)$. In these equations, (') indicates complex conjugation. δ is the adaptation parameter for "a". An exemplary value of adaption parameter δ for the preceding equations is

$\delta = C5 \times C6^{-l(i)}$. Exemplary values of the these constants are $C5 = 1\times10^{-4}$ and $C6 = 2^{1000}$. Like the other adaptation parameters μ and β, δ also has an inverse relation to input signal level "l".

[0045] If the adaptation parameters μ, β or δ are chosen to be large, the amount with which the values of adapted parameters (namely, "a" and ω) change depends heavily on instantaneous signal values and would thus keep oscillating with a large variance about the optimal value. On the other hand, if the adaptation parameters are chosen to be too small, time to converge to the optimal values will be too large.

[0046] In some examples, the algorithm directly uses filtered output signal "r" instead of adaptive filter output "e", in order to adapt correlation "a" and tracked angular frequency ω. Both "e" and "r" are the filtered versions of input "y". "e" is generated by filtering out only the positive complex sinusoid from the input, while "r" is generated by filtering out both positive and negative complex sinusoids from the input. The equations used in this algorithm are as follows: $o(i) = \lambda y(i) + \alpha o(i-1)$, $r(i) = y(i) - a(i)b(i) - a'(i)b'(i)$, and $a(i+1) = a(i) + \mu r(i)b'(i)$. The basis signal "b" is obtained by normalizing resonant filter output "o". In some examples, this normalization is carried out using one of the following two equations: $b(i) = o(i)$.

$b(i) = o(i) / |o(i)|$.

[0047] An exemplary use case in which filter system 304 may be used to cancel tonal noise will now be described. In some examples, the tonal noise may be generated by a power source included within sound processor 104. For example, a switched mode power supply may induce a noise at the switching frequency, and possibly also harmonic frequencies. An input source (e.g., a telecoil) may detect the tonal noise and thereby include the tonal noise in an input signal provided to filter system 304. To track and cancel out this tonal noise from the input signal, the filter system 304 may be provided (e.g., by a fitting system, by a clinician, and/or during a manufacturing process) with an estimated or expected value for the fundamental frequency of the tonal noise. Filter system 304 may then track variations in the fundamental frequency and/or phase of the tonal noise, and cancel the tonal noise from the input signal as described above.

[0048] Continuing with this example, the fundamental frequency of the tonal noise generated by the power supply may change over time. For example, as the voltage level of a battery included in the power supply depletes over time, the frequency at which the power supply operates may change to accommodate the drop in voltage.

This, in turn, changes the fundamental frequency of the tonal noise. Filter system 304 may track this change and continuously remain locked to it, thereby allowing filter system 304 to accurately and effectively cancel the tonal noise from the input signal despite the variations in frequency.

[0049] In some examples, multiple filter systems 304 may be included in sound processor 104 in order to track and remove tonal noise originating from multiple different noise sources. For example, multiple noise sources may include multiple power supplies, or circuits or algorithms working at different rates of operation. The noise from each noise source can include a fundamental frequency and harmonics of that fundamental frequency (frequencies at an integer multiple of the fundamental frequency). The noise from various noise sources can also intermodulate (usually because of the presence of sum non-linearity), and produce cross harmonics. A cross harmonic of various fundamental frequencies is at a frequency which is the numerical sum of some integer multiple of each of the fundamental frequencies.

[0050] As shown in FIGS. 3-6, the filtered version of the input signal “e” may be input into processing circuitry 306. Processing circuitry 306 may include any suitable processing components configured to generate control data used to direct cochlear implant 108 to generate and apply electrical stimulation representative of the filtered version of the filtered input signal “e” to the patient (i.e., to one or more stimulation sites within the cochlea of the patient).

[0051] FIG. 8 shows an exemplary configuration 800 in which sound processor 104 includes a mixer circuit 802 configured to mix the filtered version of the input signal “e” with a signal output by a microphone 804 included within cochlear implant system 100. Microphone 804 may include any of the microphones described herein. Configuration 800 may be used in scenarios in which it is desirable to combine audio content detected by a microphone and by a different input source (e.g., a telecoil).

[0052] The systems and methods described above cancel tonal noise in the time domain. In some alternative embodiments, tonal noise may be cancelled in the frequency domain. For example, sound processor 104 may take a “snapshot” of the system noise (i.e., tonal noise generated by each noise source within the cochlear implant system 100). This may be performed in any suitable manner. For example, while no audio is being presented to the user, sound processor 104 may process an input signal provided by an input source (e.g., a microphone, a telecoil, and/or any other input source) for a predetermined amount of time and then compute the average of the FFT output. This snapshot may be representative of the system noise, and may be stored and then subtracted from frequency domain representations of input signals provided by the input source during normal operation of the cochlear implant system 100 (i.e,. while audio is being presented to the patient).

[0053] FIG. 9 illustrates an exemplary method 900 of cancelling tonal noise in a cochlear implant system. While FIG. 9 illustrates exemplary steps according to one embodiment, other embodiments may omit, add to, reorder, and/or modify any of the steps shown in FIG. 9. One or more of the steps shown in FIG. 9 may be performed by filter system 304 and/or any implementation thereof.

[0054] In step 902, a filter system included in a sound processor that is a part of a cochlear implant system associated with a patient receives an input signal that includes audio content representative of an audio signal presented to the patient and tonal noise introduced by a noise source associated with the cochlear implant system. Step 902 may be performed in any of the ways described herein.

[0055] In step 904, the filter system tracks a fundamental frequency and a phase of the tonal noise within the input signal. Step 904 may be performed in any of the ways described herein.

[0056] In step 906, the filter system generates a reference noise signal based on the tracked frequency and the tracked phase. The reference noise signal could have a frequency nearly equal to the fundamental frequency or its harmonics. Step 906 may be performed in any of the ways described herein.

[0057] In step 908, the filter system subtracts the reference noise signal from the input signal to generate a filtered version of the input signal. Step 908 may be performed in any of the ways described herein.

[0058] In the preceding description, various exemplary embodiments have been described with reference to the accompanying drawings. It will, however, be evident that various modifications and changes may be made thereto, and additional embodiments may be implemented, without departing from the scope of the invention as set forth in the claims that follow. For example, certain features of one embodiment described herein may be combined with or substituted for features of another embodiment described herein. The description and drawings are accordingly to be regarded in an illustrative rather than a restrictive sense.

CLAIMS

What is claimed is:

1. A sound processor included in a cochlear implant system associated with a patient and comprising:

a filter system that

receives an input signal that includes audio content representative of an audio signal presented to the patient and tonal noise introduced by a noise source associated with the cochlear implant system,

tracks a fundamental frequency and a phase of the tonal noise within the input signal,

generates a reference noise signal based on the tracked frequency and the tracked phase, and

subtracts the reference noise signal from the input signal to generate a filtered version of the input signal.

2. The sound processor of claim 1, wherein the subtracting of the reference noise signal from the input signal at least partially cancels the tonal noise from the input signal.

3. The sound processor of claim 1, wherein the generated reference noise has the same frequency as the tracked frequency.

4. The sound processor of claim 1, wherein the generated reference noise has a frequency which is a harmonic of the tracked frequency.

5. The sound processor of claim 1, wherein the filter system tracks a magnitude of the tonal noise, and wherein the generation of the reference noise signal is further based on the tracked magnitude.

6. The sound processor of claim 1, wherein the filter system comprises:

a resonant filter that receives the input signal and outputs an output signal that has the tracked frequency; and an adaptive filter that receives the output signal from the resonant filter and that outputs the filtered version of the input signal.

7. The sound processor of claim 6, wherein the resonant filter is a single pole filter with a z-transform represented by $H(z) = \lambda \ / \ (1-\alpha z^{-1})$, where α is equal to $|\alpha|e^{j\omega}$, $|\alpha|$ is a constant, and ω is an expected angular frequency of the fundamental frequency.

8. The sound processor of claim 7, wherein the output signal output by the resonant filter is represented by $o(i) = \lambda\, y(i) + \alpha o(i-1)$.

9. The sound processor of claim 8, wherein the filter system further comprises a normalizer that normalizes the output signal output by the resonant filter to give a basis signal represented by $b(i) = o(i) \ / \ |o(i)|$, wherein the basis signal is the reference noise signal.

10. The sound processor of claim 9, wherein the adaptive filter receives the reference noise signal and subtracts the reference noise signal from the input signal to generate the filtered version of the input signal, wherein the filtered version of the input signal is represented by $r(i) = y(i) - a(i)b(i) - a'(i)b'(i)$, where ' indicates a complex conjugation and where a(i) is a complex number that is an estimate of a correlation between a complex frequency present in y(i) and b(i).

11. The sound processor of claim 10, wherein the adaptive filter uses a complex least means square algorithm to generate a(i).

12. The sound processor of claim 1, wherein the sound processor further comprises a telecoil that provides the input signal.

13. The sound processor of claim 1, further comprising a mixer circuit that mixes a signal output by a microphone included the cochlear implant system with the filtered version of the input signal.

14. The sound processor of claim 1, wherein the input signal is provided by a microphone included in the cochlear implant system.

15. The sound processor of claim 1, wherein the input signal includes a mixture of a signal provided by a microphone included in the cochlear implant system and a telecoil included in the sound processor.

16. The sound processor of claim 1, wherein the sound processor further comprises processing circuitry that directs a cochlear implant included in the cochlear implant system to generate and apply electrical stimulation representative of the filtered version of the input signal to the patient.

17. The sound processor of claim 1, wherein the noise source is a power supply included in the sound processor.

18. The sound processor of claim 1, further comprising an analog-to-digital converter that outputs input signal.

19. A system comprising:

a telecoil that outputs an analog input signal that includes audio content representative of an audio signal presented to a patient and tonal noise introduced by a noise source;

an analog-to-digital converter that converts the analog input signal into a digital input signal;

a filter system that

receives the digital input signal,

generates a reference noise signal associated with the digital input signal, and

subtracts the reference noise signal from the digital input signal to generate a filtered version of the digital input signal; and

processing circuitry that processes the filtered version of the digital input signal and directs a cochlear implant to generate and apply electrical stimulation representative of the filtered version of the digital input signal to the patient.

20. A method comprising:

receiving, by a filter system included in a sound processor that is a part of a cochlear implant system associated with a patient, an input signal that includes audio content representative of an audio signal presented to the patient and tonal noise introduced by a noise source associated with the cochlear implant system;

tracking, by the filter system, a fundamental frequency and a phase of the tonal noise within the input signal;

generating, by the filter system, a reference noise signal based on the tracked frequency and the tracked phase; and

subtracting, by the filter system, the reference noise signal from the input signal to generate a filtered version of the input signal.

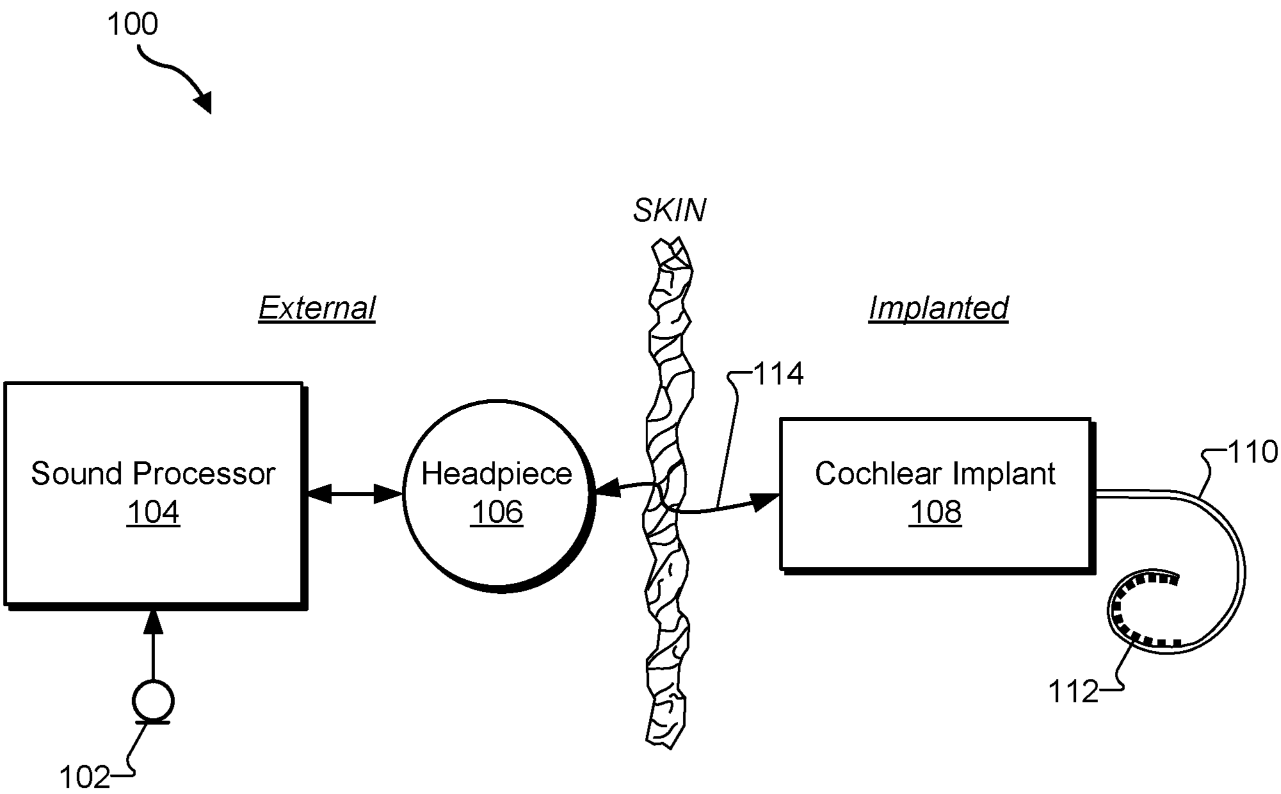

100
SKIN
External
Implanted
114
Sound Processor
104
Headpiece
106
Cochlear Implant
108
110
112
102

Fig. 1

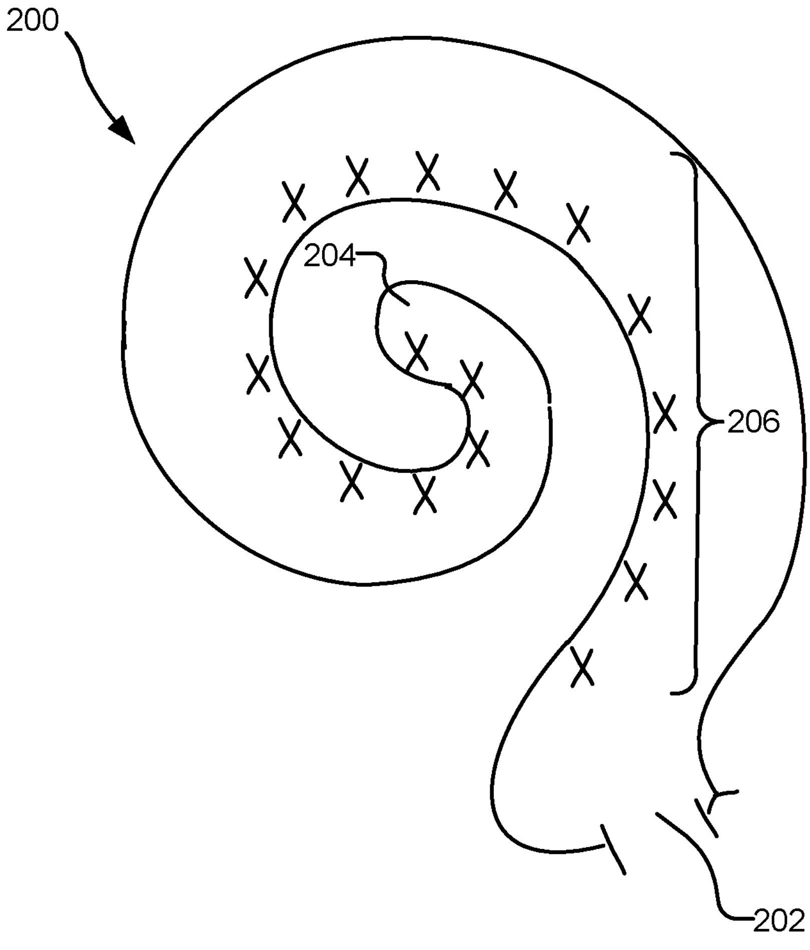


Fig. 2

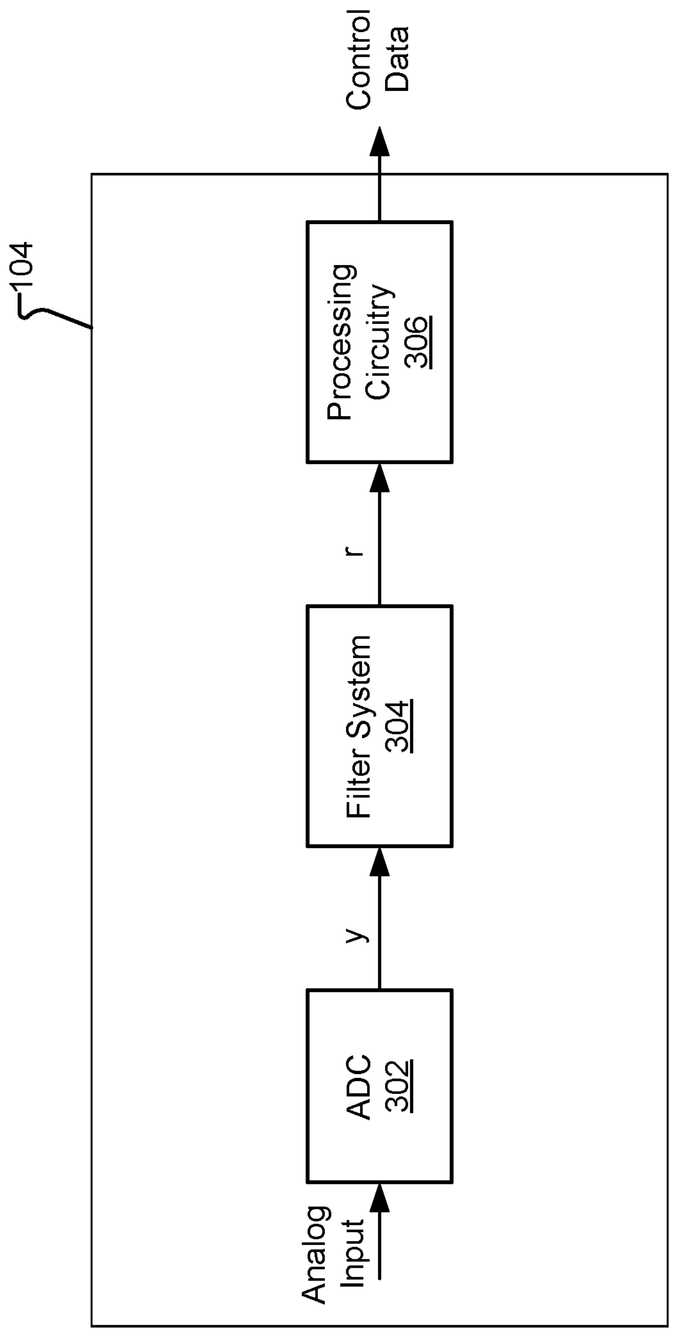

104
Analog Input
ADC 302
y
Filter System 304
r
Processing Circuitry 306
Control Data

Fig. 3

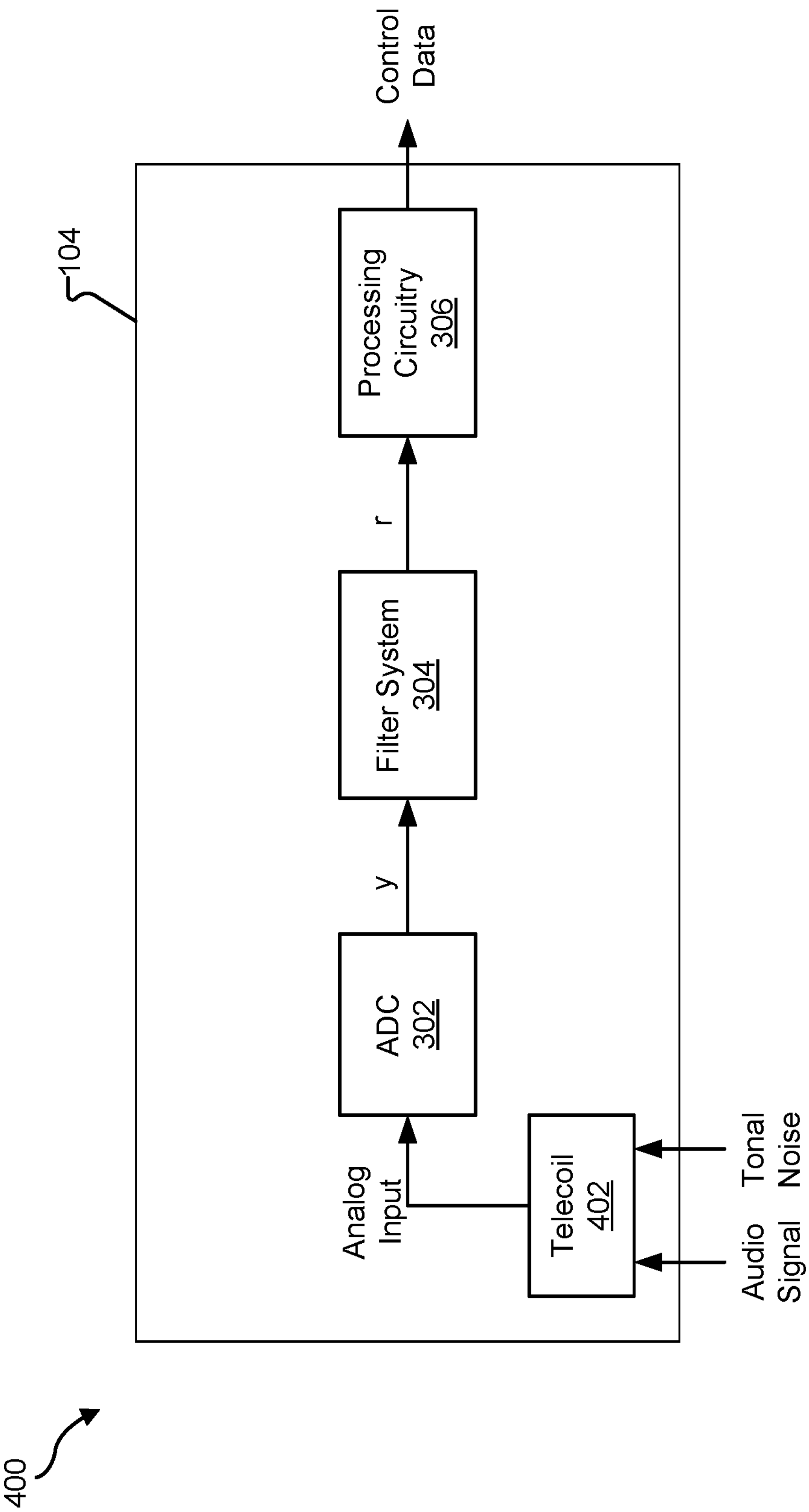

Control Data
104
Processing Circuitry 306
r
Filter System 304
y
ADC 302
Analog Input
Telecoil 402
Tonal Noise
Audio Signal
400

Fig. 4

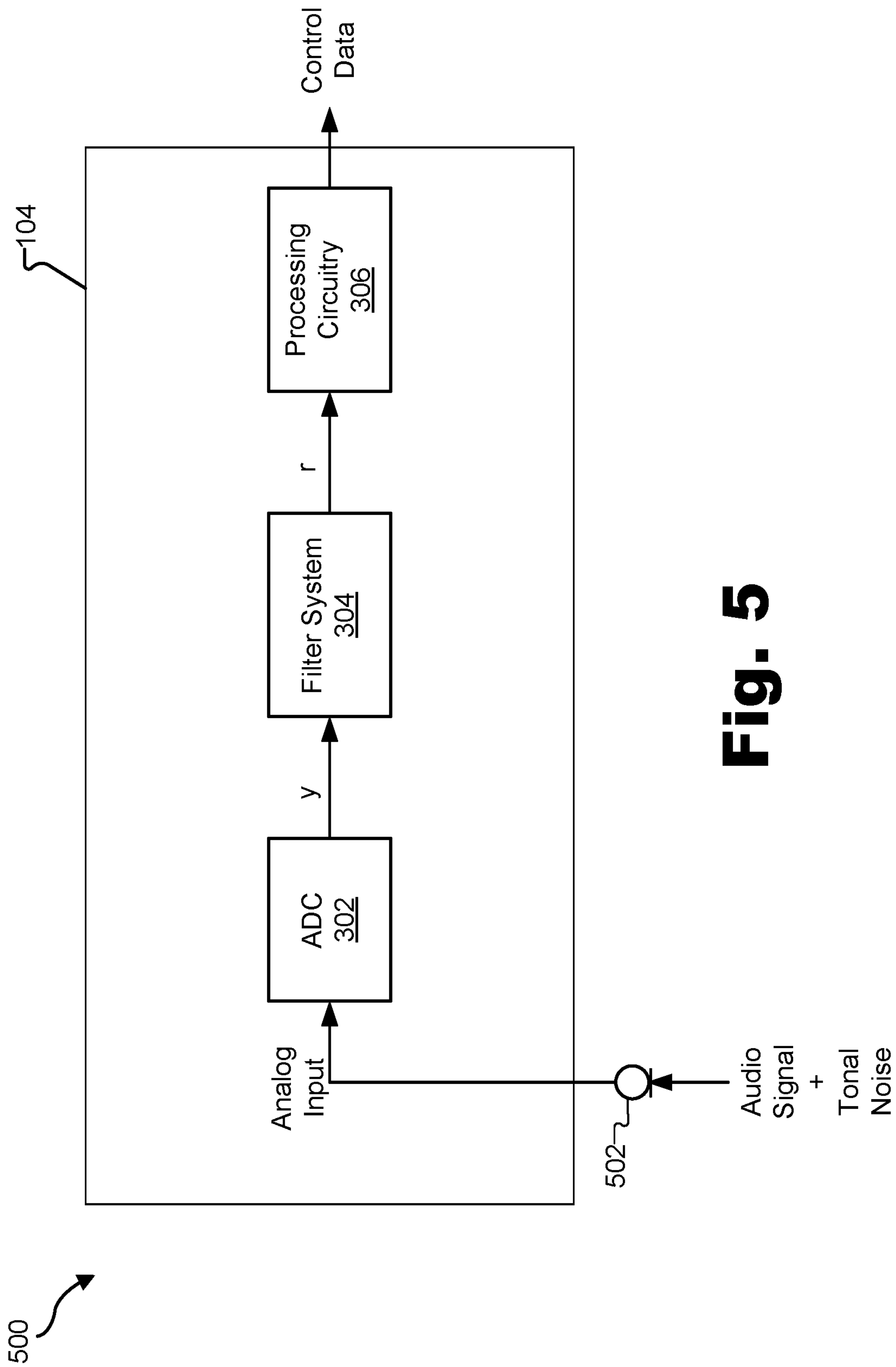

500
104
Control Data
Processing Circuitry 306
r
Filter System 304
y
ADC 302
Analog Input
502
Audio Signal + Tonal Noise

Fig. 5

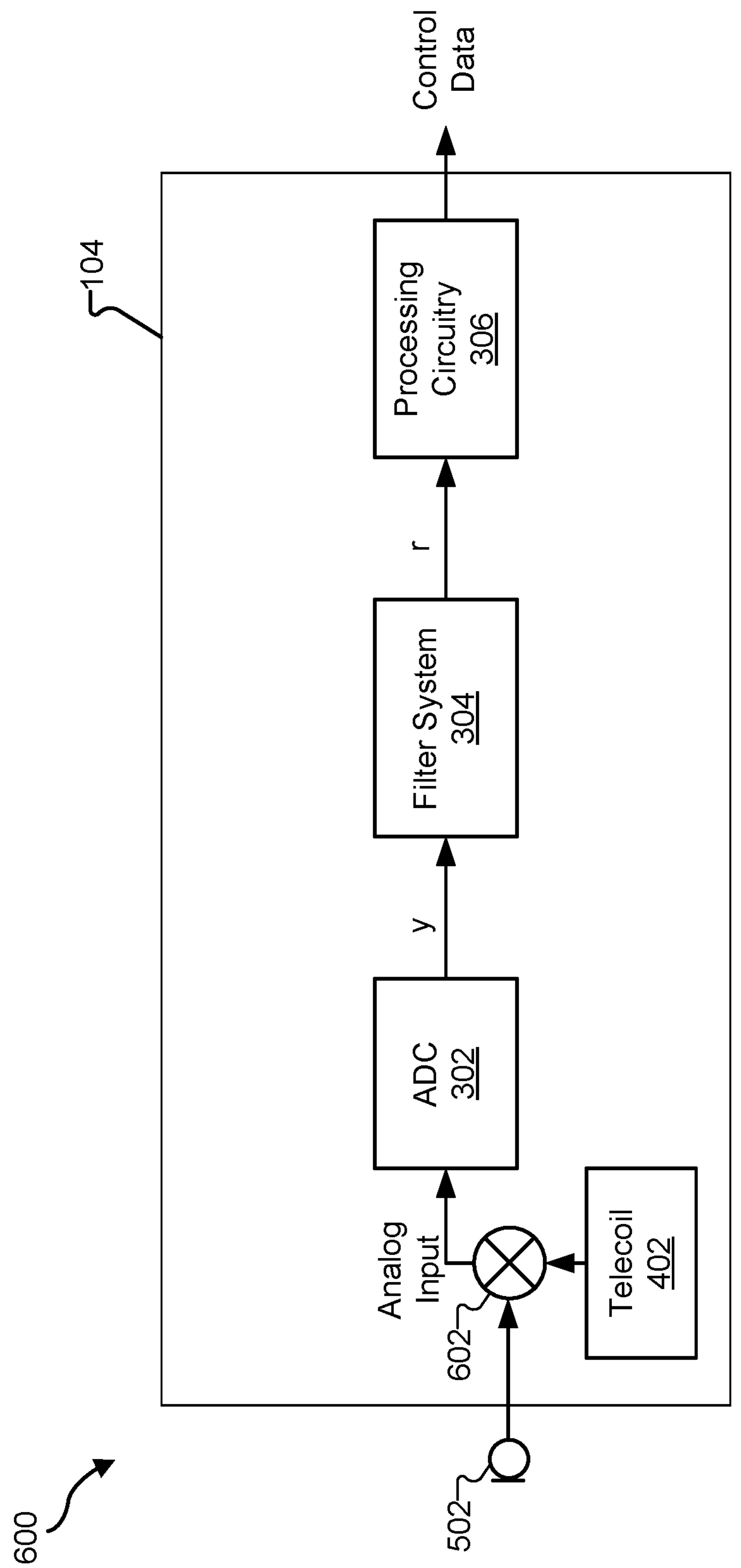
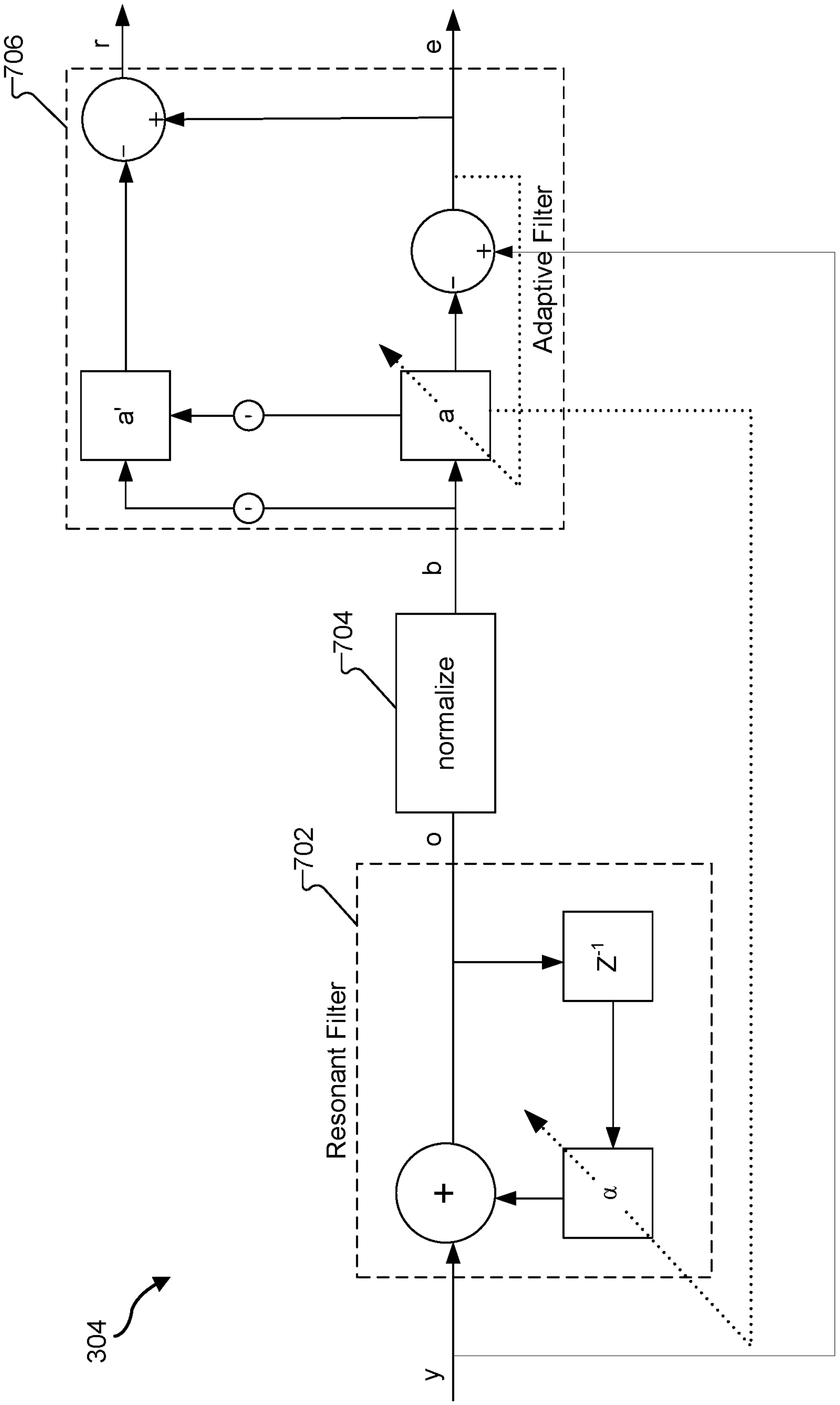

600
104
Control Data
Processing Circuitry 306
r
Filter System 304
y
ADC 302
Analog Input
602
Telecoil 402
502

Fig. 6



304
y
Resonant Filter
702
α
Z-1
o
normalize
704
b
a
a'
Adaptive Filter
e
r
706

Fig. 7

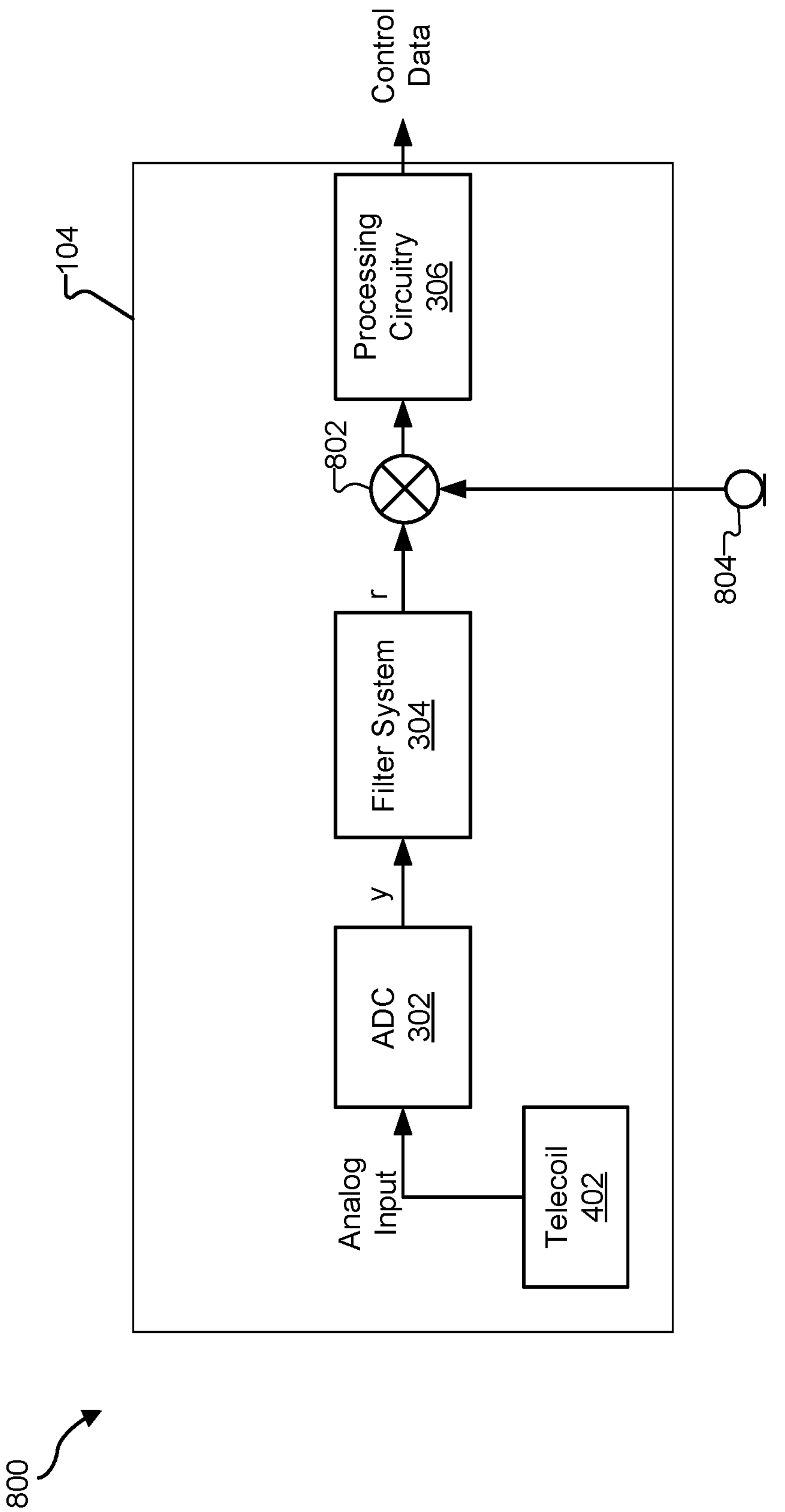

Control Data
104
Processing Circuitry 306
802
Filter System 304
r
804
ADC 302
y
Analog Input
Telecoil 402
800

Fig. 8

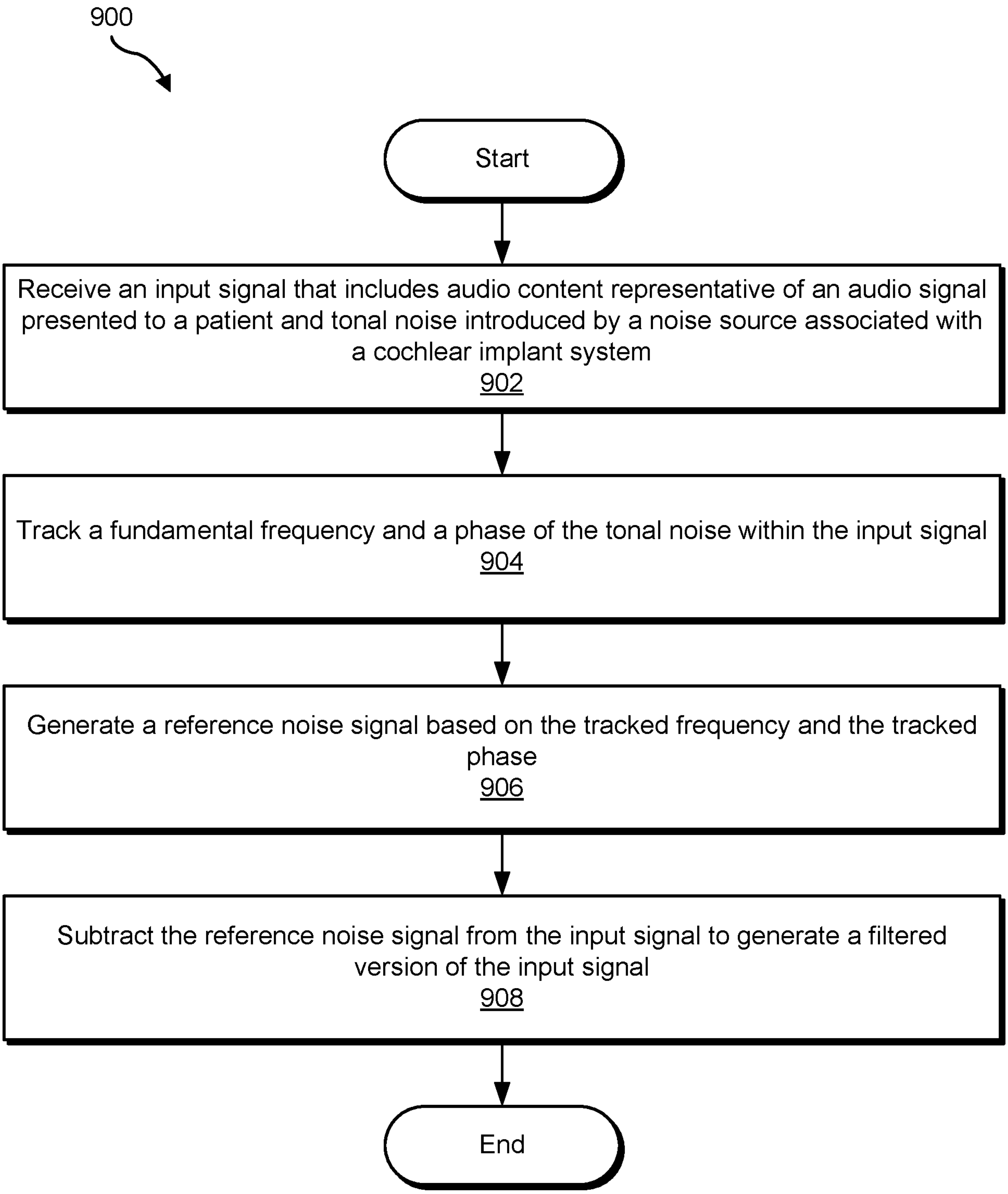

900
Start
Receive an input signal that includes audio content representative of an audio signal presented to a patient and tonal noise introduced by a noise source associated with a cochlear implant system
902
Track a fundamental frequency and a phase of the tonal noise within the input signal
904
Generate a reference noise signal based on the tracked frequency and the tracked phase
906
Subtract the reference noise signal from the input signal to generate a filtered version of the input signal
908
End

Fig. 9

INTERNATIONAL SEARCH REPORT

International application No
PCT/IB2014/061224

A. CLASSIFICATION OF SUBJECT MATTER
INV. A61N1/36 G10L21/02 G10L21/0208
ADD.

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

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
A61N

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

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

EPO-Internal, WPI Data

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X Y	US 2005/288923 A1 (KOK CHI-WAH [CN]) 29 December 2005 (2005-12-29) abstract; figures 1-6 paragraphs [0007] - [0054]; claims 1-3	1-11,15, 17,20 12-14, 16,18,19
Y	US 2010/310084 A1 (HERSBACH ADAM [AU]) 9 December 2010 (2010-12-09) abstract; figures 1-9b paragraphs [0027] - [0090]	12-14, 16,18
Y	WO 2013/065010 A1 (COCHLEAR LTD [AU]; MAUGER STEFAN J [AU]; HERSBACH ADAM A [AU]; DAWSON) 10 May 2013 (2013-05-10) abstract; figures 1-13 paragraphs [0030] - [0195]	18,19

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[X] Further documents are listed in the continuation of Box C.

[X] See patent family annex.

* Special categories of cited documents :

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"E" earlier application or patent but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
7 October 2014	16/10/2014

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

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C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US 2007/167671 A1 (MILLER SCOTT A III [US] MILLER III SCOTT ALLAN [US]) 19 July 2007 (2007-07-19) abstract; figures 1-13 paragraphs [0046] - [0097] -----	1-20
A	US 2011/150257 A1 (JENSEN JESPER [DK] ET AL) 23 June 2011 (2011-06-23) abstract; figures 1-6b paragraphs [0007] - [0210] -----	1-20
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2

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Information on patent family members

International application No
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