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(54) **RECURSIVE TRAINING OF ADAPTIVE FILTERS FOR ACTIVE NOISE CONTROL (ANC) SYSTEMS**

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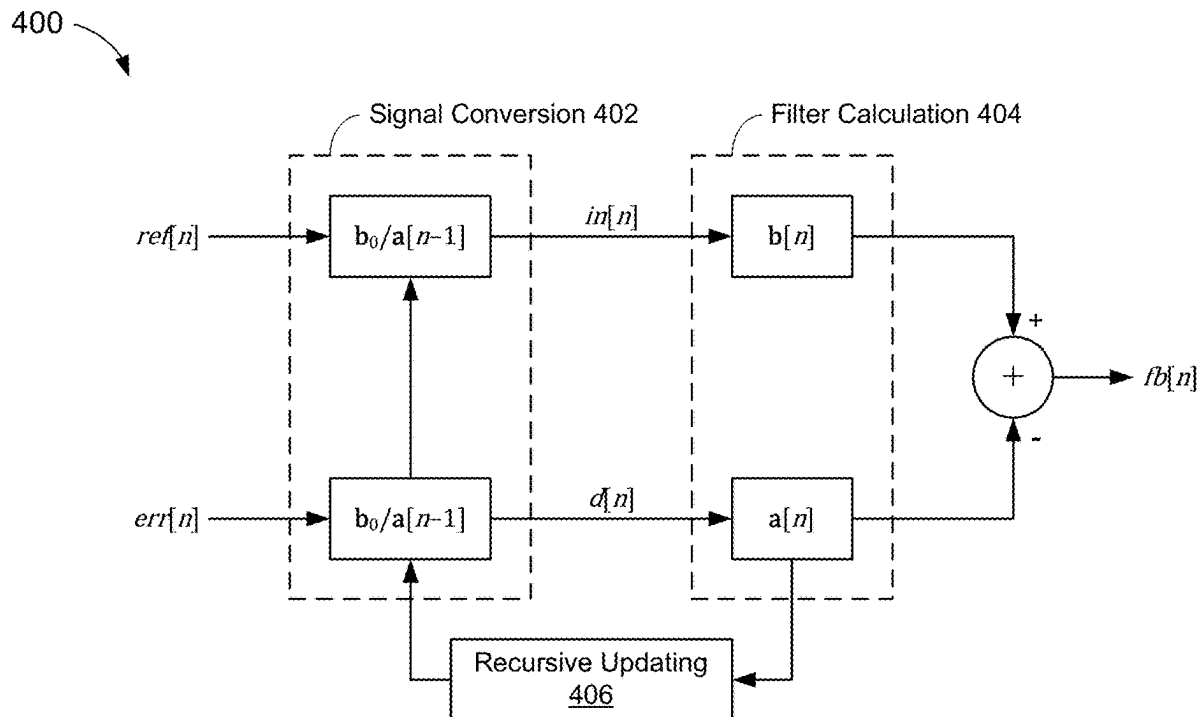
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(57) **ABSTRACT**

This disclosure provides methods, devices, and systems for active noise control (ANC). The present implementations more specifically relate to filter adaptation techniques for recursively training an infinite impulse response (IIR) filter to convert a reference audio signal to an anti-noise signal. In some aspects, an ANC system may record, via a feedforward microphone, a reference audio signal representing external noise, and may further record, via a feedback microphone, an error signal representing residual noise resulting from passive attenuation of the external noise via a primary path between the microphones. The ANC system may further estimate an IIR filter that converts the reference audio signal to the error signal based on a secondary path. In some implementations, the ANC system may recursively update the IIR filter coefficients based, at least in part, on the coefficients associated with previous frames of the reference audio signal and the error signal.



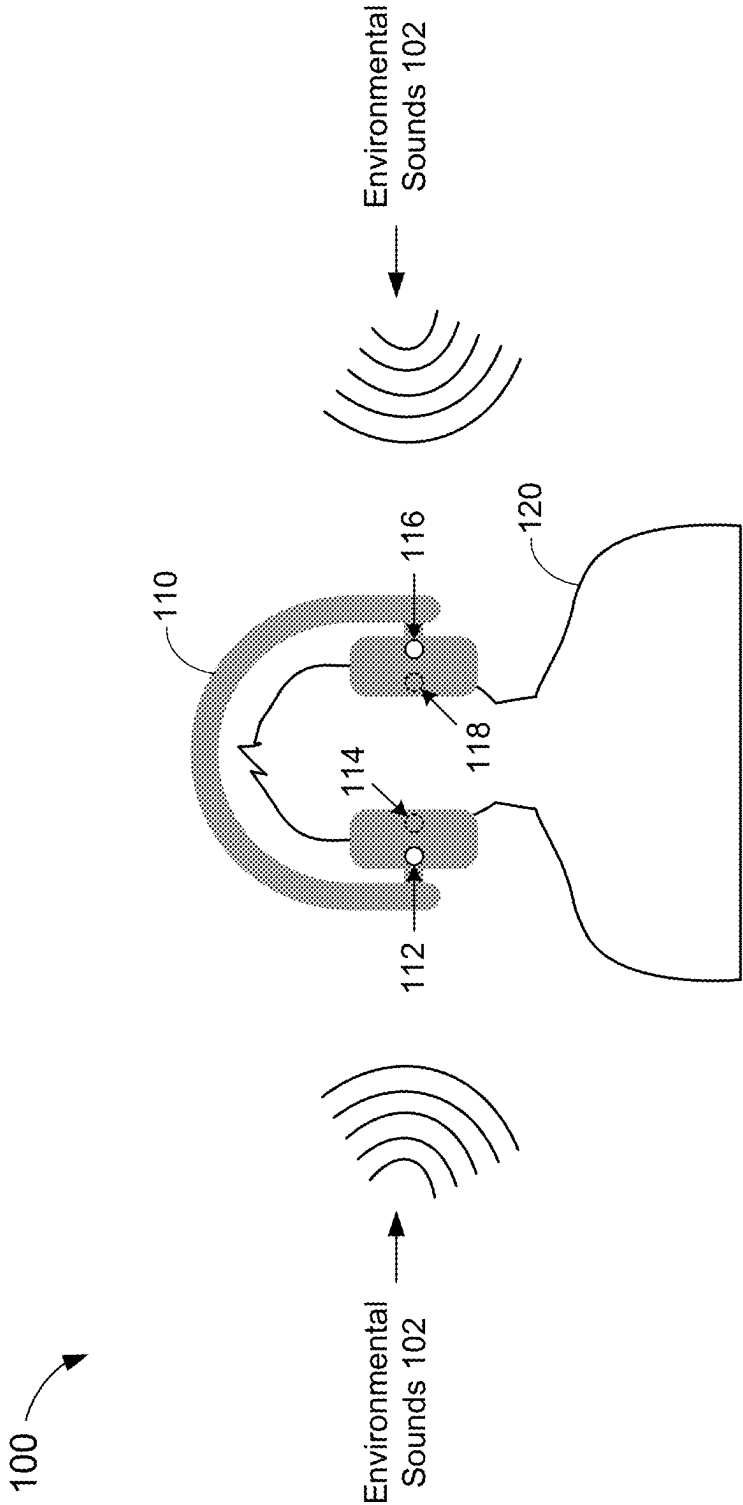


FIG. 1

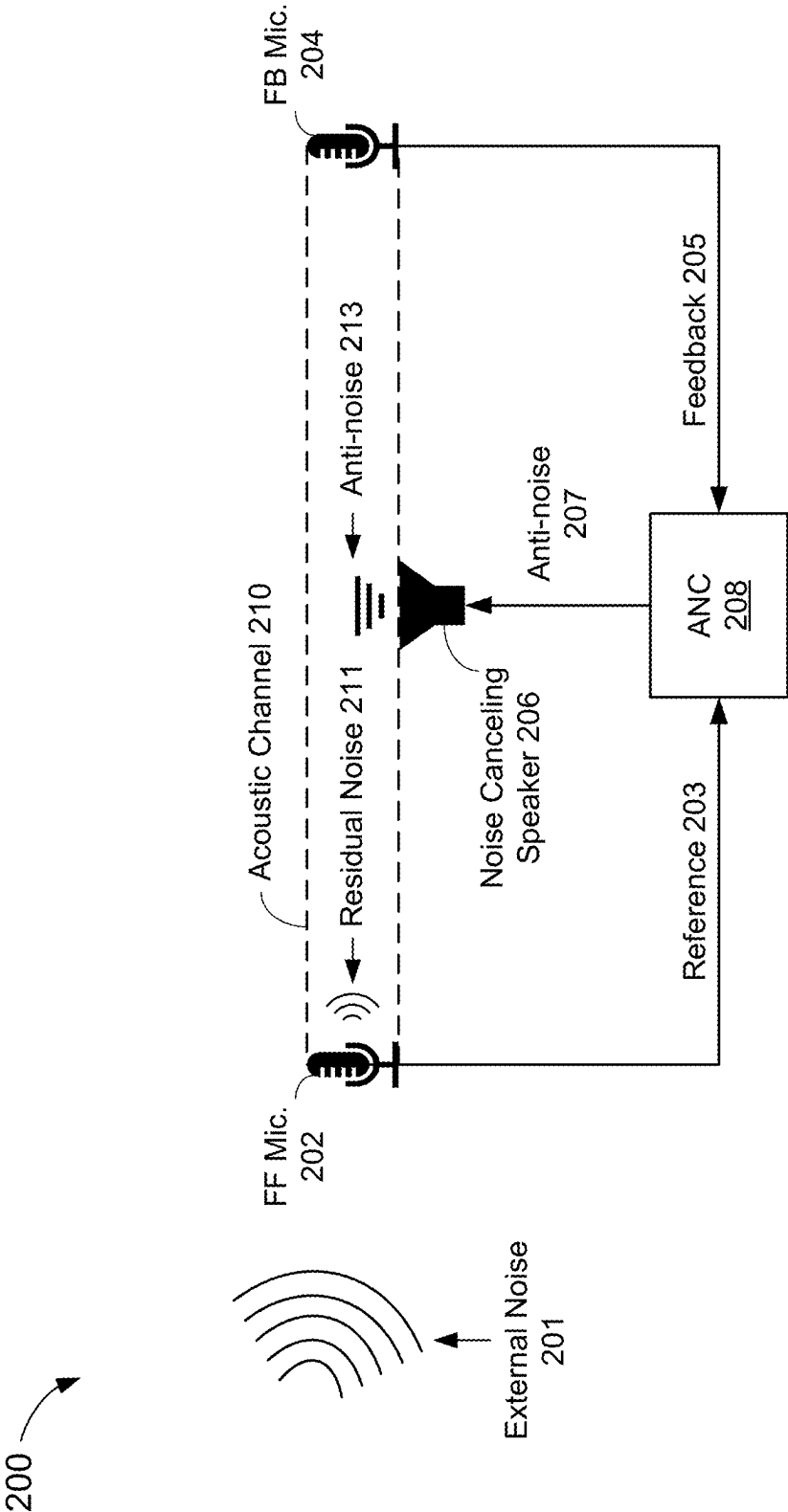


FIG. 2

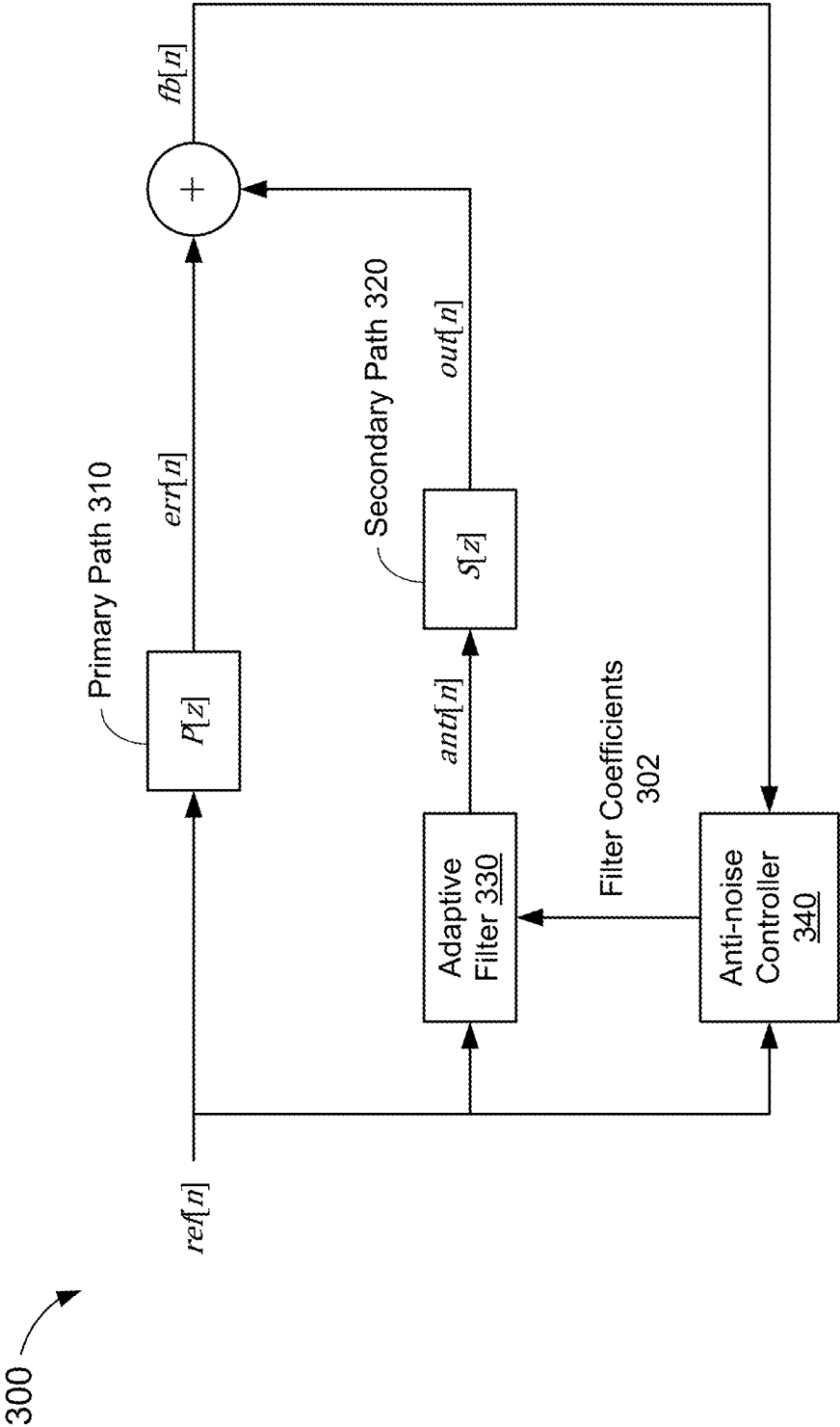


FIG. 3

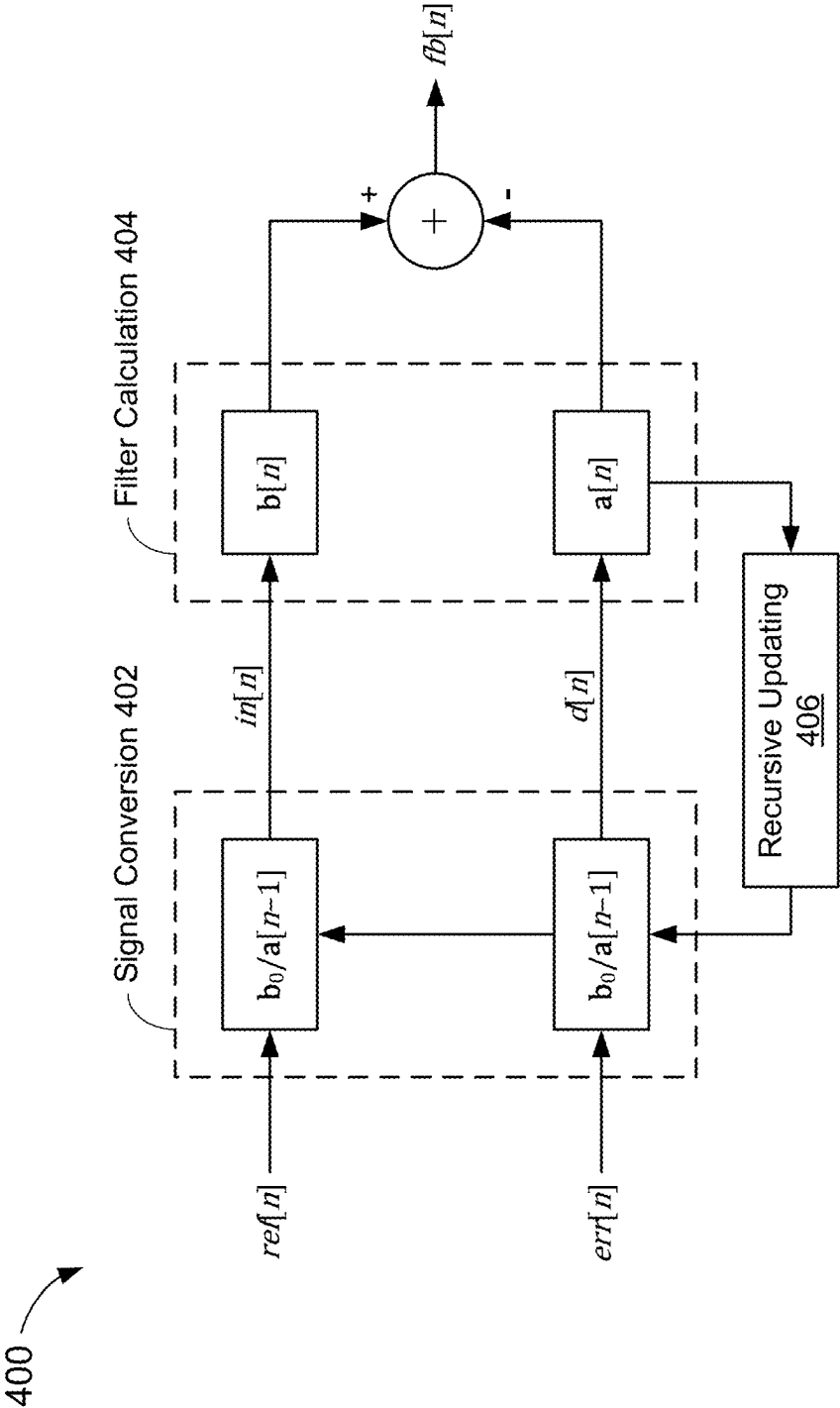


FIG. 4

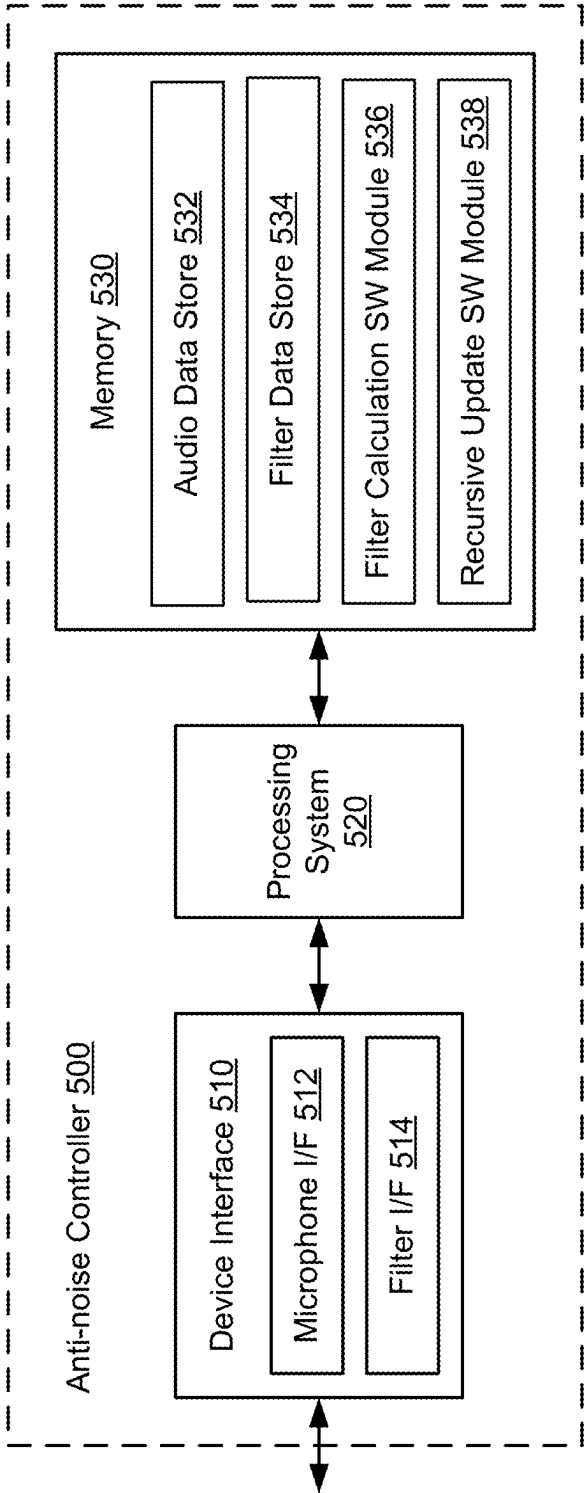


FIG. 5

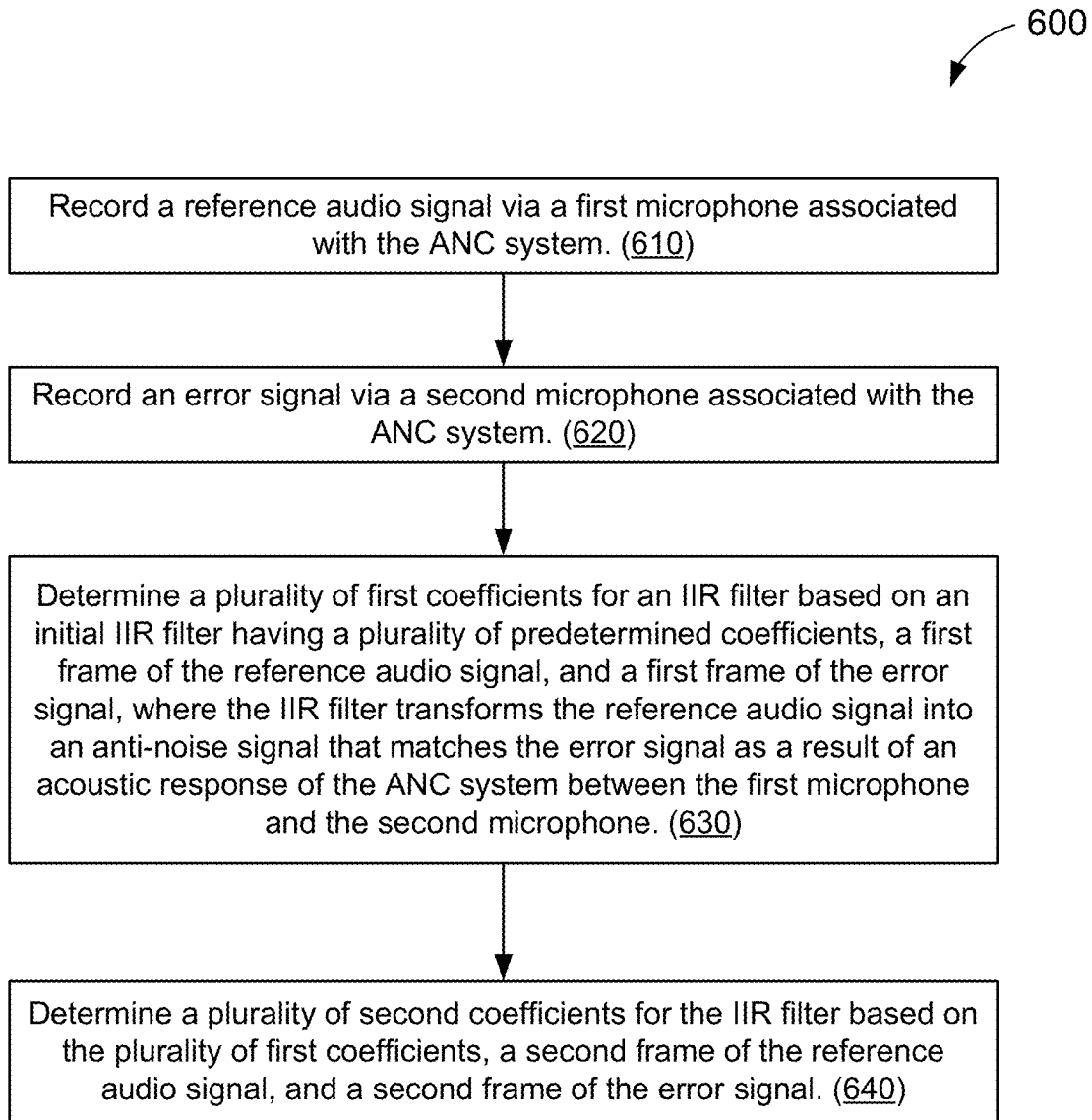


FIG. 6

RECURSIVE TRAINING OF ADAPTIVE FILTERS FOR ACTIVE NOISE CONTROL (ANC) SYSTEMS

TECHNICAL FIELD

[0001] The present implementations relate generally to acoustic noise suppression, and specifically to recursive training of adaptive filters for active noise control (ANC) systems.

BACKGROUND OF RELATED ART

[0002] Active noise control (ANC) systems suppress noise or unwanted sounds in an acoustic channel (such as a human ear) by producing “anti-noise” waveforms (also referred to as “anti-sound” waveforms) that cancel or attenuate the noise in the channel. More specifically, the anti-noise waveform is equal in amplitude and opposite in phase to the noise waveform so that the waveforms cancel each other based on the principles of superposition. ANC systems are often implemented in wearable devices that are designed to be worn in or over a user’s ears (such as headsets, headphones, and earbuds, among other examples). Such wearable devices generally include a microphone to record or detect the unwanted noise, an adaptive filter to convert the recorded noise to an anti-noise signal, and a noise cancelling speaker to output the anti-noise signal as an acoustic waveform that cancels the noise in a user’s ear.

[0003] In feedforward (FF) ANC systems, a reference microphone (also referred to as a “feedforward microphone”) is disposed on an outer surface of an earpiece (such as an earcup or an earbud) to record external noise that may filter into a user’s ear. Such external noise may be attenuated or altered by the physical properties of the earpiece (also referred to as an “acoustic response”) before reaching the user’s ear. The acoustic response of the earpiece can be modeled as a transfer function $P[z]$ that converts the external noise to residual noise incident upon the user’s ear. Existing FF ANC systems rely on estimates of the transfer function $P[z]$ to calculate a finite impulse response (FIR) filter that can convert the recorded noise to an anti-noise signal.

[0004] FIR filters tend to require a significant amount of power, which may not be practical for many ANC applications (such as wireless headsets). By comparison, infinite impulse response (IIR) filters consume significantly less power and are thus much more efficient than FIR filters. As such, FIR filters are often converted (or mapped) to IIR filters for implementation in ANC systems. However, existing ANC systems can accumulate significant errors from the estimation of the transfer function $P[z]$ and the mapping of the FIR filter to an IIR filter. Thus, new filter adaptation techniques are needed to improve the quality of noise suppression in ANC systems.

SUMMARY

[0005] This Summary is provided to introduce in a simplified form a selection of concepts that are further described below in the Detailed Description. This Summary is not intended to identify key features or essential features of the claimed subject matter, nor is it intended to limit the scope of the claimed subject matter.

[0006] One innovative aspect of the subject matter of this disclosure can be implemented in a method of acoustic noise

suppression performed by an active noise control (ANC) system. The method includes recording a reference audio signal via a first microphone associated with the ANC system; recording an error signal via a second microphone associated with the ANC system; determining a plurality of first coefficients for an infinite impulse response (IIR) filter based on, an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, where the IIR filter transforms the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone; and determining a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal.

[0007] Another innovative aspect of the subject matter of this disclosure can be implemented in a controller for an ANC system, including a processing system and a memory. The memory stores instructions that, when executed by the processing system, cause the controller to record a reference audio signal via a first microphone associated with the ANC system; record an error signal via a second microphone associated with the ANC system; determine a plurality of first coefficients for an IIR filter based on an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, where the IIR filter transforms the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone; and determine a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal.

BRIEF DESCRIPTION OF THE DRAWINGS

[0008] The present implementations are illustrated by way of example and are not intended to be limited by the figures of the accompanying drawings.

[0009] FIG. 1 shows an example environment for which acoustic noise suppression may be implemented.

[0010] FIG. 2 shows an example active noise control (ANC) system.

[0011] FIG. 3 shows a simplified block diagram of an example ANC system, according to some implementations.

[0012] FIG. 4 shows a block diagram of an example anti-noise controller, according to some implementations.

[0013] FIG. 5 shows another block diagram of an example anti-noise controller, according to some implementations.

[0014] FIG. 6 shows an illustrative flowchart depicting an example operation for acoustic noise suppression, according to some implementations.

DETAILED DESCRIPTION

[0015] In the following description, numerous specific details are set forth such as examples of specific components, circuits, and processes to provide a thorough understanding of the present disclosure. The term “coupled” as used herein means connected directly to or connected through one or more intervening components or circuits. The terms “electronic system” and “electronic device” may be used interchangeably to refer to any system capable of

electronically processing information. Also, in the following description and for purposes of explanation, specific nomenclature is set forth to provide a thorough understanding of the aspects of the disclosure. However, it will be apparent to one skilled in the art that these specific details may not be required to practice the example embodiments. In other instances, well-known circuits and devices are shown in block diagram form to avoid obscuring the present disclosure. Some portions of the detailed descriptions which follow are presented in terms of procedures, logic blocks, processing and other symbolic representations of operations on data bits within a computer memory.

[0016] These descriptions and representations are the means used by those skilled in the data processing arts to most effectively convey the substance of their work to others skilled in the art. In the present disclosure, a procedure, logic block, process, or the like, is conceived to be a self-consistent sequence of steps or instructions leading to a desired result. The steps are those requiring physical manipulations of physical quantities. Usually, although not necessarily, these quantities take the form of electrical or magnetic signals capable of being stored, transferred, combined, compared, and otherwise manipulated in a computer system. It should be borne in mind, however, that all of these and similar terms are to be associated with the appropriate physical quantities and are merely convenient labels applied to these quantities.

[0017] Unless specifically stated otherwise as apparent from the following discussions, it is appreciated that throughout the present application, discussions utilizing the terms such as “accessing,” “receiving,” “sending,” “using,” “selecting,” “determining,” “normalizing,” “multiplying,” “averaging,” “monitoring,” “comparing,” “applying,” “updating,” “measuring,” “deriving” or the like, refer to the actions and processes of a computer system, or similar electronic computing device, that manipulates and transforms data represented as physical (electronic) quantities within the computer system’s registers and memories into other data similarly represented as physical quantities within the computer system memories or registers or other such information storage, transmission or display devices.

[0018] In the figures, a single block may be described as performing a function or functions; however, in actual practice, the function or functions performed by that block may be performed in a single component or across multiple components, and/or may be performed using hardware, using software, or using a combination of hardware and software. To clearly illustrate this interchangeability of hardware and software, various illustrative components, blocks, modules, circuits, and steps have been described below generally in terms of their functionality. Whether such functionality is implemented as hardware or software depends upon the particular application and design constraints imposed on the overall system. Skilled artisans may implement the described functionality in varying ways for each particular application, but such implementation decisions should not be interpreted as causing a departure from the scope of the present disclosure. Also, the example input devices may include components other than those shown, including well-known components such as a processor, memory and the like.

[0019] The techniques described herein may be implemented in hardware, software, firmware, or any combination thereof, unless specifically described as being implemented

in a specific manner. Any features described as modules or components may also be implemented together in an integrated logic device or separately as discrete but interoperable logic devices. If implemented in software, the techniques may be realized at least in part by a non-transitory processor-readable storage medium including instructions that, when executed, performs one or more of the methods described above. The non-transitory processor-readable data storage medium may form part of a computer program product, which may include packaging materials.

[0020] The non-transitory processor-readable storage medium may comprise random access memory (RAM) such as synchronous dynamic random-access memory (SDRAM), read only memory (ROM), nonvolatile random access memory (NVRAM), electrically erasable programmable read-only memory (EEPROM), FLASH memory, other known storage media, and the like. The techniques additionally, or alternatively, may be realized at least in part by a processor-readable communication medium that carries or communicates code in the form of instructions or data structures and that can be accessed, read, and/or executed by a computer or other processor.

[0021] The various illustrative logical blocks, modules, circuits and instructions described in connection with the embodiments disclosed herein may be executed by one or more processors (or a processing system). The term “processor,” as used herein may refer to any general-purpose processor, special-purpose processor, conventional processor, controller, microcontroller, and/or state machine capable of executing scripts or instructions of one or more software programs stored in memory.

[0022] As described above, in feedforward (FF) active noise control (ANC) systems, a reference microphone (also referred to as a “feedforward microphone”) is disposed on an outer surface of an earpiece (such as an earcup or an earbud) to record external noise that may filter into a user’s ear. Such external noise may be attenuated or altered by the physical properties of the earpiece (also referred to as an “acoustic response”) before reaching the user’s ear. The acoustic response of the earpiece can be modeled as a transfer function $P[z]$ that converts the external noise to residual noise incident upon the user’s ear. Existing FF ANC systems rely on estimates of the transfer function $P[z]$ to calculate a finite impulse response (FIR) filter that can convert the recorded noise to an anti-noise signal.

[0023] FIR filters tend to require a significant amount of power, which may not be practical for many ANC applications (such as wireless headsets). By comparison, infinite impulse response (IIR) filters consume significantly less power and are thus much more efficient than FIR filters. As such, FIR filters are often converted (or mapped) to IIR filters for implementation in ANC systems. However, existing ANC systems can accumulate significant errors from the estimation of the transfer function $P[z]$ and the mapping of the FIR filter to an IIR filter. Aspects of the present disclosure recognize that such errors can be avoided by directly calculating the IIR filter, without estimating the transfer function $P[z]$ or calculating an FIR filter.

[0024] Various aspects relate generally to ANC systems, and more particularly, to filter adaptation techniques for recursively training an IIR filter to convert a reference audio signal to an anti-noise signal. In some aspects, an ANC system may record, via an FF microphone, a reference audio signal ($ref[n]$) representing external noise, and may further

record, via a feedback (FB) microphone, an error signal ($err[n]$) representing residual noise resulting from passive attenuation of the external noise, where n is a frame (or sampling) index in a discrete time domain. The passive attenuation represents an acoustic response ($P[z]$) of a primary path of the ANC system between the FF and FB microphones. The ANC system may further estimate an IIR filter ($b[n]/a[n]$) that converts the reference audio signal $ref[n]$ to the error signal $err[n]$, where

$$ref[n] * s[n] * \frac{b[n]}{a[n]} = err[n],$$

and where $[*]$ denotes a convolution and $s[n]$ represents an impulse response of a secondary path of the ANC system. In some implementations, the ANC system may recursively update the IIR filter coefficients $a[n]$ and $b[n]$ based, at least in part, on the coefficients $a[n-1]$ associated with the previous frame of the reference audio signal $ref[n-1]$ and error signal $err[n-1]$ until $a[n] \approx a[n-1]$.

[0025] Particular implementations of the subject matter described in this disclosure can be implemented to realize one or more of the following potential advantages. As described above, existing ANC systems rely on estimates for the acoustic response $P[z]$ of the primary path, and $S[z]$ of the secondary path, of the ANC system to calculate an FIR filter that can be converted to an IIR filter, where $S[z]$ is a transfer function of the secondary path in the z -domain. As a result, estimation errors associated with the acoustic response $P[z]$, and transfer errors associated with the mapping from the FIR domain to the IIR domain, may accumulate in the IIR filter. By contrast, the IIR filter of the present implementations is trained only on the secondary path of the ANC system. By directly calculating an IIR filter based on the secondary path, aspects of the present disclosure can avoid estimation errors associated with the acoustic response $P[z]$ as well as transfer errors associated with the mapping from the FIR domain to the IIR domain. As a result, the IIR filters of the present implementations may significantly improve the quality and performance of acoustic noise suppression compared to the IIR filters implemented by existing ANC systems.

[0026] FIG. 1 shows an example environment 100 for which acoustic noise suppression may be implemented. The environment 100 includes a wearable audio device 110 and a user 120. In the example of FIG. 1, the wearable audio device 110 is depicted as a headset having earcups designed to be worn over a user's ears. However, in some other implementations, the wearable audio device 110 may include a pair of earbuds that are designed to be worn inside a user's ear canals. As used herein, the term "earpiece" may generally refer to any portions of the wearable audio device 110 that at least partially covers a user's ears (such as an earcup or an earbud).

[0027] In some aspects, the wearable audio device 110 may include a number of microphones 112-118 disposed on the earpieces. As shown in FIG. 1, the left earpiece includes a first pair of microphones 112 and 114 and the right earpiece includes a second pair of microphones 116 and 118. More specifically, the microphones 114 and 118 are disposed on an inner surface of each earpiece in close proximity to the user's ears, whereas the microphones 112 and 116 are disposed on an outer surface of each earpiece further away

from the user's ears. In the example of FIG. 1, the wearable audio device 110 is shown to include only four microphones 112-118. However, in actual implementations, the wearable audio device 110 may include more microphones than those depicted in FIG. 1.

[0028] The microphones 112 and 116 are positioned or otherwise configured to detect external sounds 102 in the surrounding environment 100 (depicted as a series of acoustic waves). Thus, the microphones 112 and 116 may be referred to herein as feedforward (FF) microphones (or "reference" microphones"). Each of the microphones 112 and 116 is configured to convert the detected external sounds 102 to an electrical signal representative of the acoustic waveform. Such electrical signals may be referred to herein as "reference audio signals."

[0029] The microphones 114 and 118 are positioned or otherwise configured to detect residual sounds from the surrounding environment 100 (not shown for simplicity) that filter through the earpieces and enter the user's ears. Thus, the microphones 114 and 118 may be referred to herein as feedback (FB) microphones (or "error microphones"). Each of the microphones 114 and 118 may convert the detected residual sounds to an electrical signal representative of the acoustic waveform. Such electrical signals may be referred to herein as "error signals."

[0030] For some applications, the environmental sounds 102 may be distracting to the user 120 or otherwise perceived as unwanted noise (such as when listening to audio playback via the wearable audio device 110). In some aspects, the wearable audio device 110 may suppress or attenuate the residual sounds that filter or pass through the earpieces. For example, the wearable audio device 110 may use the FF microphones 112 and 116 to detect the characteristics or waveform of the environmental sounds 102 and produce an anti-noise waveform that can cancel or attenuate the residual sounds at the user's ears. More specifically, the anti-noise waveform may be equal in amplitude and opposite in phase to the waveform of the residual sounds so that the waveforms cancel each other based on the principles of superposition (also referred to as "destructive interference").

[0031] FIG. 2 shows an example active noise control (ANC) system 200. The ANC system 200 includes an FF microphone 202, an FB microphone 204, a noise canceling speaker 206, and an ANC component 208. In some implementations, the ANC system 200 may be one example of the wearable audio device 110 of FIG. 1. With reference to FIG. 1, the FF microphone 202 and FB microphone 204 may be examples of the microphones 112 and 114, respectively, of the left earpiece or the microphones 116 and 118, respectively, of the right earpiece. More specifically, the FB microphone 204 may be disposed on an inner surface of an earpiece or housing that is in close proximity to a user's ear when the earpiece is worn by the user, and the FF microphone 202 may be disposed on an outer surface of the earpiece or housing that is further away from the user's ear when the earpiece is worn by the user.

[0032] The FF microphone 202 is configured to convert acoustic waves representing external noise 201 from the surrounding environment (such as the environmental sound 102 of FIG. 1) to a reference audio signal 203. As described with reference to FIG. 1, at least some of the external noise 201 filters through the earpiece as residual noise 211. In other words, the residual noise 211 is a result of the external noise 201 being attenuated or altered by the physical prop-

erties of the earpiece (also referred to as “passive attenuation”). In some implementations, the ANC component **208** may convert the reference audio signal **203** to an anti-noise signal **207** for output, by the noise cancelling speaker **206**, as an acoustic “anti-noise” waveform **213**. More specifically, the ANC component **208** models responses associated with the passive attenuation, loudspeaker response, air inside a channel (or tub), and various other factors that may affect the anti-noise waveform **213** or the residual noise **211**, so that the anti-noise waveform **213** has substantially the same amplitude but opposite phase as the waveform of the residual noise **211**.

[0033] The residual noise **211** combines with the anti-noise **213** in an acoustic channel **210** that guides or directs sounds towards the user’s ear (such as an acoustic duct or cavity of the earpiece). As a result, the anti-noise **213** cancels or attenuates the amount of residual noise **211** that reaches the user’s ear. The amount of noise suppression or attenuation depends on how closely the anti-noise waveform **213** matches the waveform of the residual noise **211**, and thus, the accuracy with which the ANC component **208** models the acoustic response of the earpiece or the ANC system **200**. The FB microphone **204** is configured to detect any sounds that enter the user’s ear and convert such acoustic waves to a feedback signal **205**. Thus, the feedback signal **205** may indicate how much of the residual noise **211** is attenuated by the anti-noise **213**. In some implementations, the ANC component **208** may use the feedback signal **205** to dynamically improve the accuracy of the anti-noise signal **207**.

[0034] FIG. 3 shows a simplified block diagram of an example ANC system **300**, according to some implementations. In some implementations, the ANC system **300** may be one example of the ANC system **200** of FIG. 2. More specifically, the ANC system **300** is configured to produce an anti-noise signal (anti[n]) based on a reference audio signal (ref[n]), where n is a frame (or sampling) index in a discrete time domain. With reference to FIG. 2, the reference audio signal ref[n] may be one example of the reference audio signal **203** and the anti-noise signal anti[n] may be one example of the anti-noise signal **207**.

[0035] The ANC system **300** includes a primary acoustic path **310**, an adaptive filter **330**, a secondary acoustic path **320**, and an anti-noise controller **340**. The primary path **310** represents an acoustic response of the ANC system **300** that transforms the reference audio signal ref[n] into an error signal (err[n]). With reference for example to FIG. 2, the error signal err[n] may represent the residual noise **211** that enters the acoustic channel **210**. Thus, the primary path **310** describes the passive attenuation of the earpiece between the microphones **202** and **204**. The primary path **310** can be modeled as a transfer function $P[z]$, in the z-domain, using ref[n] as the input and err[n] as the output, where $\text{ref}[n] + p[n] = \text{err}[n]$ and $p[n]$ is the impulse response of the primary path.

[0036] The adaptive filter **330** converts the reference audio signal ref[n] to the anti-noise signal anti[n]. The secondary path **320** transforms the anti-noise signal anti[n] into an output signal (out[n]) that can be used to cancel or suppress the error signal err[n]. With reference for example to FIG. 2, the output signal out[n] may represent the anti-noise waveform **213** that is output to the acoustic channel **210** via the noise canceling speaker **206**. Thus, the secondary path **320** may include a digital-to-analog converter (DAC), recon-

struction filter, power amplifier (PA), loudspeaker, FB microphone, acoustic medium (such as air) between the loudspeaker and the FB microphone, preamplifier, antialiasing filter, and analog-to-digital converter (ADC). The secondary path **320** can be modeled as a transfer function $S[z]$, where the output signal out[n] is produced as a result of the anti-noise signal anti[n] passing through $S[z]$.

[0037] The anti-noise controller **340** determines a set of coefficients (or weights) **302** for the adaptive filter **330** that causes the output signal out[n] to match the error signal err[n]. More specifically, the filter coefficients **302** may cause the adaptive filter **330** to convert the reference audio signal ref[n] to an anti-noise signal anti[n] that transforms into an output signal out[n], via the secondary path **320**, having substantially the same amplitude and opposite phase as the error signal err[n]. With proper tuning of the filter coefficients **302**, the output signal out[n] may cancel or attenuate the error signal err[n] at the user’s ear. The result of the attenuation is detected by an FB microphone (such as the FB microphone **204** of FIG. 2) as a feedback signal (fb[n]) that can be used by the anti-noise controller **340** to fine-tune or update the filter coefficients **302**.

[0038] When the output signal out[n] is perfectly matched to the error signal err[n], the feedback signal fb[n] is nullified or otherwise approaches zero. Thus, the performance of the ANC system **300** depends on the accuracy of the anti-noise signal anti[n] produced by the adaptive filter **330**. In existing ANC systems, the adaptive filter **330** is initially modeled as a finite impulse response (FIR) filter $W[z]$ based on the transfer functions $P[z]$ and $S[z]$, where:

$$W[z] = \frac{P[z]}{S[z]} \quad (1)$$

[0039] With reference to Equation 1, the transfer function $S[z]$ can be determined relatively accurately based on known properties or characteristics of various circuit components and media associated with the secondary path **320** (such as a DAC, reconstruction filter, PA, loudspeaker, FB microphone, air, preamplifier, antialiasing filter, and ADC). Many existing ANC systems estimate the primary path transfer function $P[z]$ associated with the primary path **310** based on the reference audio signal ref[n] and the error signal err[n], where:

$$\text{ref}[n] * p[n] = \text{err}[n] \quad (2)$$

[0040] FIR filters tend to require a significant amount of power, which may not be practical for many ANC applications (such as wireless headsets). Thus, the FIR filter $W[z]$ is often mapped to an IIR filter for use as the adaptive filter **330**. However, the IIR filters used in existing ANC systems can accumulate significant errors from the estimation of the transfer function $P[z]$ and the mapping from the FIR domain to the IIR domain.

[0041] In some aspects, the ANC controller **340** may avoid any errors associated with estimating the transfer function $P[z]$, or mapping between the FIR and IIR domains, by directly calculating an IIR filter ($b[n]/a[n]$) to be used as the adaptive filter **330** based on the reference audio signal ref[n] and the error signal err[n], where:

$$ref[n] * \frac{b[n]}{a[n]} * s[n] = err[n] \quad (3)$$

[0042] With reference to Equation 3, the reference audio signal $ref[n]$ can be recorded via an FF microphone (such as the FF microphone **202** of FIG. 2) and the error signal $err[n]$ can be recorded via an FB microphone (such as the FB microphone **204** of FIG. 2) when no output signal $out[n]$ is provided to the same acoustic channel. The secondary path transfer function $S[z]$ can be calculated with a relatively high degree of accuracy based on known properties or characteristics of various circuit components and media associated with the secondary path **320**. Accordingly, the ANC controller **340** can be trained to determine the IIR filter $b[n]/a[n]$ based on known parameters of Equation 3.

[0043] FIG. 4 shows a block diagram of an example anti-noise controller **400**, according to some implementations. In some implementations, the anti-noise controller **400** may be one example of the anti-noise controller **340** of FIG. 3. With reference to Equation 3, the anti-noise controller **400** may be trained to determine the coefficients $a[n]$ and $b[n]$ for the IIR filter based on a reference audio signal $ref[n]$ and an error signal $err[n]$. In some implementations, the reference audio signal $ref[n]$ may be recorded via an FF microphone associated with an ANC system (such as the FF microphone **202** of FIG. 2) and the error signal $err[n]$ may be recorded via an FB microphone associated with the ANC system (such as the FB microphone **204** of FIG. 2). Thus, the IIR filter $b[n]/a[n]$ also may be referred to as “ANC filter.”

[0044] The anti-noise controller **400** includes a signal conversion component **402**, a filter calculation component **404**, and a recursive updating component **406**. The signal conversion component **402** is configured to convert the reference audio signal $ref[n]$ and the error signal $err[n]$ to an input signal $in[n]$ and a desired signal $d[n]$, respectively, based on a recursive IIR filter

$$\left(\frac{b_0}{a[n-1]} \right),$$

where:

$$ref[n] * \frac{b_0}{a[n-1]} = in[n] \quad (4)$$

and refrain

$$err[n] * \frac{b_0}{a[n-1]} = d[n] \quad (5)$$

[0045] The numerator of the recursive IIR filter includes a set of predetermined filter coefficients b_0 associated with a range of frequencies to be attenuated by the ANC system. For example, the filter coefficients b_0 may be configured to filter out unwanted frequencies of the reference audio signal $ref[n]$ and the error signal $err[n]$. In some implementations, the filter coefficients b_0 may be configured as a low-pass filter. In some other implementations, the filter coefficients b_0 may be configured as a bandpass filter. Still further, in

some implementations, the filter coefficients b_0 may be configured as a high-pass filter.

[0046] The denominator of the recursive IIR filter includes a set of filter coefficients $a[n-1]$ derived from a previous frame of the reference audio signal $ref[n-1]$ and a previous frame of the error signal $err[n-1]$, for $n \geq 0$, where:

$$\begin{aligned} ref[n-1] &= \left[ref\left[overlap - \frac{N}{2}\right], \right. \\ &\quad \left. ref\left[overlap - \frac{N}{2} + 1\right], \dots, ref\left[overlap + \frac{N}{2} - 1\right] \right] \\ err[n-1] &= \left[err\left[overlap - \frac{N}{2}\right], \right. \\ &\quad \left. err\left[overlap - \frac{N}{2} + 1\right], \dots, err\left[overlap + \frac{N}{2} - 1\right] \right] \end{aligned}$$

where

$$overlap = n * \frac{N}{2}$$

(representing a 50% signal overlap) and N is an even numbered frame length.

[0047] In some implementations, the denominator of the recursive IIR filter may be initialized to a set of predetermined filter coefficients a , when calculating the ANC filter based on the first recorded frame of the reference audio signal $ref[0]$ and the first recorded frame of the error signal $err[0]$ (such that $a[n-1]=a_0$ for $n=0$).

[0048] The filter calculation component **404** determines a set of filter coefficients $a[n]$ and $b[n]$ for the ANC filter based on the following equation:

$$in[n] * b[n] = d[n] * a[n] \quad (6)$$

[0049] By combining Equations 4-6, the input signal $in[n]$ and the desired signal $d[n]$ can be calculated as follows:

$$\begin{cases} in[n] = \sum_{k=0}^N b_k ref[n-k] - \sum_{k=1}^N a_k in[n-k] \\ d[n] = \sum_{k=0}^N b_k err[n-k] - \sum_{k=1}^N a_k d[n-k] \end{cases} \quad (7)$$

where $b_0=[b_0[0], b_1[0], \dots, b_N[0]]$ and $a_0[n-1]=[1, a_1[n-1], \dots, a_N[n-1]]$.

[0050] Based on Equation 7, the frame base of $in[n]$ and $d[n]$ can be composed as follows:

$$in[n] = [in[overlap], in[overlap+1], \dots, in[overlap+N-1]]$$

$$d[n] = [d[overlap], d[overlap+1], \dots, d[overlap+N-1]]$$

[0051] Based on the first frame of the reference audio signal $ref[n]$ and the error signal $err[n]$, Equation 7 can further be rewritten in matrix form:

$$A[n] = \begin{bmatrix} \text{in}[0] & 0 & 0 & 0 & \dots & 0 & 0 & \dots & 0 \\ \text{in}[1] & \text{in}[0] & 0 & 0 & \dots & d[0] & 0 & \dots & 0 \\ \text{in}[2] & \text{in}[1] & 0 & 0 & \dots & d[1] & d[0] & \dots & 0 \\ \vdots & \vdots & 0 & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\ \text{in}[N-1] & \text{in}[N-2] & \dots & \text{in}[N-L+1] & \text{in}[N-L] & d[N-2] & d[N-3] & \dots & d[N-L+1] \end{bmatrix}$$

$$x[n] = \begin{bmatrix} b[0] \\ b[1] \\ \vdots \\ b[L-1] \\ a[1] \\ a[2] \\ \vdots \\ a[L-1] \end{bmatrix}$$

$$y[n] = \begin{bmatrix} d[0] \\ d[1] \\ \vdots \\ d[N] \end{bmatrix}$$

[0052] In some implementations, the filter calculation component 404 may calculate the filter coefficients $a[n]$ and $b[n]$ associated with the current frame n using Wiener-Hopf equations:

$$R_{AA}x = R_{Ay} \quad (8)$$

where $R_{AA} = A[n]^H A[n]$ and $R_{Ay} = A[n]^H y[n]$. For example, after solving for the unknown vector $x = R_{AA}^{-1} R_{Ay}$, the coefficients a and b can be extracted from the vector x , where:

$$b = [b[0] \dots b[L]]$$

$$a = [a[1]]$$

and where $a = [a[1] \ a[2] \ \dots \ a[L-1]]$.

[0053] In some implementations, the filter calculation component 404 may use diagonal loading techniques to improve the robustness of the calculations associated with Equation 8, where:

$$AI = A + \beta I$$

where β is a constant having a value less than 1 and I is the identity matrix. Then the vector x can be rewritten as:

$$x = (AI^H AI)^{-1} AI^H y$$

[0054] In some implementations, the filter calculation component 404 may use QR decomposition techniques to further improve the robustness of the calculations associated with Equation 8, where:

$$QR = AI$$

[0055] Assuming the matrix A (or AI) can be subdivided into two parts according to the filter coefficients b and a , the contribution from the vector a in the matrix AI can be expressed as:

$$Q(:, L+1:2L-1)R(L+1:2L-1, L+1:2L-1)$$

where $Q(:, L+1:2L-1)$ extracts all column elements, and row elements from length $L+1$ to $2L-1$, of the Q matrix, and $R(L+1:2L-1, L+1:2L-1)$ extracts column and row elements from length $L+1$ to $2L-1$ of the R matrix. As such, the vector a can be calculated according to the following equation:

$$a = R(L+1:2L-1, L+1:2L-1)^{-1} Q(:, L+1:2L-1)^T y$$

[0056] To calculate the filter coefficients b , the matrix A (or AI) may be reshaped so that:

$$AIr = [AI(:, L+1:2L-1), AI(:, 1:L)]$$

[0057] where $AI(:, L+1:2L-1)$ extracts all column elements, and row elements from length $L+1$ to $2L-1$, of the AI matrix, and $AI(:, 1:L)$ extracts all column elements, and row elements from length 1 to L , of the AI matrix. As such, the QR decomposition can be expressed as:

$$Q_r R_r = AIr$$

and the filter coefficients b can be calculated according to the following equation:

$$b = R_r(L+1:2L-1, L+1:2L-1)^{-1} Q_r(:, L+1:2L-1)^T y$$

[0058] The recursive updating component 406 is configured to compare the filter coefficients $a[n]$ calculated for the current frame n with the filter coefficients $a[n-1]$ calculated for the previous frame $n-1$ to determine whether to update the coefficients a and b of the ANC filter. For example, the recursive updating component 406 may update the filter coefficients a and b by providing the filter coefficients $a[n]$ for the current frame n back to the signal conversion

component **402**, to be used in the recursive IIR filter (as $a[n-1]$) applied to the next frame of audio. In some implementations, the recursive updating component **406** may continuously update the filter coefficients a and b until the filter coefficients $a[n]$ converge or otherwise remain relatively unchanged across multiple updates (such as where $a[n] \sim a[n-1]$). For example, the recursive updating component **406** may calculate a distance D between the filter coefficients $a[n]$ and $a[n-1]$ and compare the distance D with a threshold (T). More specifically, the recursive updating component **406** may determine that the filter coefficients $a[n]$ have converged when $D \leq T$.

[0059] Upon determining that the filter coefficients $a[n]$ have converged, the recursive updating component **406** may refrain from further updating the filter coefficients a and b of the ANC filter. At this time, the anti-noise controller **400** may use the ANC filter to actively cancel or attenuate the error signal $err[n]$. With reference for example to FIG. 3, the anti-noise controller **400** may tune the adaptive filter **330** using the filter coefficients a and b . As a result, the adaptive filter **330** converts the reference signal $ref[n]$ to an anti-noise signal $anti[n]$ based on the filter coefficients a and b . The secondary path **320** transforms the anti-noise signal $anti[n]$ into an output signal $out[n]$ that can be combined with the error signal $err[n]$ to achieve the desired noise suppression or attenuation (such as described with reference to FIGS. 2 and 3).

[0060] FIG. 5 shows another block diagram of an example anti-noise controller **500**, according to some implementations. In some implementations, the anti-noise controller **500** may be one example of the ANC component **208** of FIG. 2 or any of the anti-noise controllers **340** or **400** of FIGS. 3 and 4, respectively.

[0061] The anti-noise controller **500** includes a device interface **510**, a processing system **520**, and a memory **530**. The device interface **510** is configured to communicate with one or more components of an ANC system (such as the wearable audio device **110** of FIG. 1 or any of the ANC systems **200** or **300** of FIGS. 2 and 3, respectively). More specifically, the device interface **510** may include a microphone interface (I/F) **512** and a filter interface (I/F) **514**. The microphone interface **512** is configured to communicate with one or more microphones of the ANC system (such as the FF microphone **202** and the FB microphone **204** of FIG. 2). In some implementations, the microphone interface **512** may record a reference audio signal via a first microphone associated with the ANC system and also may record an error signal via a second microphone associated with the ANC system. The filter interface **514** is configured to communicate with an adaptive filter of the ANC system (such as the adaptive filter **330** of FIG. 3).

[0062] The memory **530** may include an audio data store **532** and a filter data store **534**. The audio data store **532** is configured to store frames of the reference audio signal and the error signal as well as any intermediate signals that may be produced by the anti-noise controller **500** as a result of performing active noise control. The filter data store **534** is configured to store filter weights or coefficients for the adaptive filter of the ANC system. The memory **530** also may include a non-transitory computer-readable medium (including one or more nonvolatile memory elements, such as EPROM, EEPROM, Flash memory, or a hard drive, among other examples) that may store at least the following software (SW) modules:

[0063] a filter calculation SW module **536** to determine a plurality of first coefficients for an IIR filter based on an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, where the IIR filter transforms the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone; and

[0064] a recursive update SW module **538** to determine a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal.

Each software module includes instructions that, when executed by the processing system **520**, causes the anti-noise controller **500** to perform the corresponding functions.

[0065] The processing system **520** may include any suitable one or more processors capable of executing scripts or instructions of one or more software programs stored in the anti-noise controller **500** (such as in the memory **530**). For example, the processing system **520** may execute the filter calculation SW module **536** to determine a plurality of first coefficients for an IIR filter based on an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, where the IIR filter transforms the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone. The processing system **520** also may execute the recursive update SW module **538** to determine a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal.

[0066] FIG. 6 shows an illustrative flowchart depicting an example operation **600** for acoustic noise suppression, according to some implementations. In some implementations, the example operation **600** may be performed by an ANC system such as the wearable audio device **110** of FIG. 1, any of the ANC systems **200** or **300** of FIGS. 2 and 3, respectively, or any of the anti-noise controllers **400** or **500** of FIGS. 4 and 5, respectively.

[0067] The ANC system records a reference audio signal via a first microphone associated with the ANC system (**610**). The ANC system records an error signal via a second microphone associated with the ANC system (**620**). In some implementations, the first microphone may be disposed on an outer surface of an earpiece, and the second microphone may be disposed on an inner surface of the earpiece that is closer than the outer surface to an ear of a user when the earpiece is worn by the user.

[0068] The ANC system determines a plurality of first coefficients for an IIR filter based on an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, where the IIR filter transforms the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone (**630**). In some implementations, the determining of the plurality of first coefficients may include converting the first frame of the reference audio signal to an input signal based on the initial IIR filter having a plurality of predetermined

coefficients; converting the first frame of the error signal to a desired signal based on the initial IIR filter; and determining a first IIR filter that transforms the input signal into the desired signal, where the first IIR filter includes the plurality of first coefficients.

[0069] The ANC system further determines a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal (640). In some implementations, the second frame of the reference audio signal and the second frame of the error signal may follow the first frame of the reference audio signal and the first frame of the error signal, respectively, in time. In some implementations, the determining of the plurality of second coefficients may include converting the second frame of the reference audio signal to an input signal based on a first IIR filter having a numerator associated with a predetermined frequency bandwidth and a denominator associated with the plurality of first coefficients; converting the second frame of the error signal to a desired signal based on the first IIR filter; and determining a second IIR filter that transforms the input signal into the desired signal, the second IIR filter including the plurality of second coefficients.

[0070] In some aspects, the ANC system may further determine a distance between the plurality of first coefficients and the plurality of second coefficients and selectively update the IIR filter based on the distance between the plurality of first coefficients and the plurality of second coefficients. In some implementations, the selective updating of the IIR filter may include determining that the distance between the plurality of first coefficients and the plurality of second coefficients is greater than a threshold distance and updating the IIR filter responsive to determining that the distance between the plurality of first coefficients and the plurality of second coefficients is greater than the threshold distance. In some implementations, the updating of the IIR filter may include determining a plurality of third coefficients for the IIR filter based on the plurality of second coefficients, a third frame of the reference audio signal, and a third frame of the error signal.

[0071] In some other implementations, the selective updating of the IIR filter may include determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than a threshold distance and refraining from updating the IIR filter responsive to determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than the threshold distance. In some aspects, the ANC system may further output the anti-noise signal via one or more speakers associated with the ANC system based on determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than the threshold distance.

[0072] Those of skill in the art will appreciate that information and signals may be represented using any of a variety of different technologies and techniques. For example, data, instructions, commands, information, signals, bits, symbols, and chips that may be referenced throughout the above description may be represented by voltages, currents, electromagnetic waves, magnetic fields or particles, optical fields or particles, or any combination thereof.

[0073] Further, those of skill in the art will appreciate that the various illustrative logical blocks, modules, circuits, and algorithm steps described in connection with the aspects

disclosed herein may be implemented as electronic hardware, computer software, or combinations of both. To clearly illustrate this interchangeability of hardware and software, various illustrative components, blocks, modules, circuits, and steps have been described above generally in terms of their functionality. Whether such functionality is implemented as hardware or software depends upon the particular application and design constraints imposed on the overall system. Skilled artisans may implement the described functionality in varying ways for each particular application, but such implementation decisions should not be interpreted as causing a departure from the scope of the disclosure.

[0074] The methods, sequences or algorithms described in connection with the aspects disclosed herein may be embodied directly in hardware, in a software module executed by a processor, or in a combination of the two. A software module may reside in RAM memory, flash memory, ROM memory, EPROM memory, EEPROM memory, registers, hard disk, a removable disk, a CD-ROM, or any other form of storage medium known in the art. An exemplary storage medium is coupled to the processor such that the processor can read information from, and write information to, the storage medium. In the alternative, the storage medium may be integral to the processor.

[0075] In the foregoing specification, embodiments have been described with reference to specific examples thereof. It will, however, be evident that various modifications and changes may be made thereto without departing from the broader scope of the disclosure as set forth in the appended claims. The specification and drawings are, accordingly, to be regarded in an illustrative sense rather than a restrictive sense.

What is claimed is:

1. A method of acoustic noise suppression performed by an active noise control (ANC) system, comprising:
 - recording a reference audio signal via a first microphone associated with the ANC system;
 - recording an error signal via a second microphone associated with the ANC system;
 - determining a plurality of first coefficients for an infinite impulse response (IIR) filter based on an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, the IIR filter transforming the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone; and
 - determining a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal.
2. The method of claim 1, wherein the first microphone is disposed on an outer surface of an earpiece, and the second microphone is disposed on an inner surface of the earpiece that is closer than the outer surface to an ear of a user when the earpiece is worn by the user.
3. The method of claim 1, wherein the second frame of the reference audio signal and the second frame of the error signal follow the first frame of the reference audio signal and the first frame of the error signal, respectively, in time.
4. The method of claim 1, wherein the determining of the plurality of first coefficients comprises:

converting the first frame of the reference audio signal to an input signal based on the initial IIR filter;
 converting the first frame of the error signal to a desired signal based on the initial IIR filter; and
 determining a first IIR filter that transforms the input signal into the desired signal, the first IIR filter including the plurality of first coefficients.

5. The method of claim 1, wherein the determining of the plurality of second coefficients comprises:

converting the second frame of the reference audio signal to an input signal based on a first IIR filter having a numerator associated with a predetermined frequency bandwidth and a denominator associated with the plurality of first coefficients;

converting the second frame of the error signal to a desired signal based on the first IIR filter; and

determining a second IIR filter that transforms the input signal into the desired signal, the second IIR filter including the plurality of second coefficients.

6. The method of claim 1, further comprising:

determining a distance between the plurality of first coefficients and the plurality of second coefficients; and
 selectively updating the IIR filter based on the distance between the plurality of first coefficients and the plurality of second coefficients.

7. The method of claim 6, wherein the selective updating of the IIR filter comprises:

determining that the distance between the plurality of first coefficients and the plurality of second coefficients is greater than a threshold distance; and

updating the IIR filter responsive to determining that the distance between the plurality of first coefficients and the plurality of second coefficients is greater than the threshold distance.

8. The method of claim 7, wherein the updating of the IIR filter comprises:

determining a plurality of third coefficients for the IIR filter based on the plurality of second coefficients, a third frame of the reference audio signal, and a third frame of the error signal.

9. The method of claim 6, wherein the selective updating of the IIR filter comprises:

determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than a threshold distance; and

refraining from updating the IIR filter responsive to determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than the threshold distance.

10. The method of claim 9, further comprising:

outputting the anti-noise signal via one or more speakers associated with the ANC system based on determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than the threshold distance.

11. A controller for an active noise control (ANC) system comprising:

a processing system; and

a memory storing instructions that, when executed by the processing system, causes the controller to:

record a reference audio signal via a first microphone associated with the ANC system;

record an error signal via a second microphone associated with the ANC system;

determine a plurality of first coefficients for an infinite impulse response (IIR) filter based on an initial IIR filter having a plurality of predetermined coefficients, a first frame of the reference audio signal, and a first frame of the error signal, the IIR filter transforming the reference audio signal into an anti-noise signal that matches the error signal as a result of an acoustic response of the ANC system between the first microphone and the second microphone; and

determine a plurality of second coefficients for the IIR filter based on the plurality of first coefficients, a second frame of the reference audio signal, and a second frame of the error signal.

12. The controller of claim 11, wherein the first microphone is disposed on an outer surface of an earpiece, and the second microphone is disposed on an inner surface of the earpiece that is closer than the outer surface to an ear of a user when the earpiece is worn by the user.

13. The controller of claim 11, wherein the second frame of the reference audio signal and the second frame of the error signal follow the first frame of the reference audio signal and the first frame of the error signal, respectively, in time.

14. The controller of claim 11, wherein the determining of the plurality of first coefficients comprises:

converting the first frame of the reference audio signal to an input signal based on the initial IIR filter;

converting the first frame of the error signal to a desired signal based on the initial IIR filter; and

determining a first IIR filter that transforms the input signal into the desired signal, the first IIR filter including the plurality of first coefficients.

15. The controller of claim 11, wherein the determining of the plurality of second coefficients comprises:

converting the second frame of the reference audio signal to an input signal based on a first IIR filter having a numerator associated with a predetermined frequency bandwidth and a denominator associated with the plurality of first coefficients;

converting the second frame of the error signal to a desired signal based on the first IIR filter; and

determining a second IIR filter that transforms the input signal into the desired signal, the second IIR filter including the plurality of second coefficients.

16. The controller of claim 11, wherein execution of the instructions further causes the controller to:

determine a distance between the plurality of first coefficients and the plurality of second coefficients; and

selectively update the IIR filter based on the distance between the plurality of first coefficients and the plurality of second coefficients.

17. The controller of claim 16, wherein the selective updating of the IIR filter comprises:

determining that the distance between the plurality of first coefficients and the plurality of second coefficients is greater than a threshold distance; and

updating the IIR filter responsive to determining that the distance between the plurality of first coefficients and the plurality of second coefficients is greater than the threshold distance.

18. The controller of claim 17, wherein the updating of the IIR filter comprises:

determining a plurality of third coefficients for the IIR filter based on the plurality of second coefficients, a third frame of the reference audio signal, and a third frame of the error signal.

19. The controller of claim **16**, wherein the selective updating of the IIR filter comprises:

determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than a threshold distance; and

refraining from updating the IIR filter responsive to determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than the threshold distance.

20. The controller of claim **19**, wherein execution of the instructions further causes the controller to:

output the anti-noise signal via one or more speakers associated with the ANC system based on determining that the distance between the plurality of first coefficients and the plurality of second coefficients is less than the threshold distance.

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