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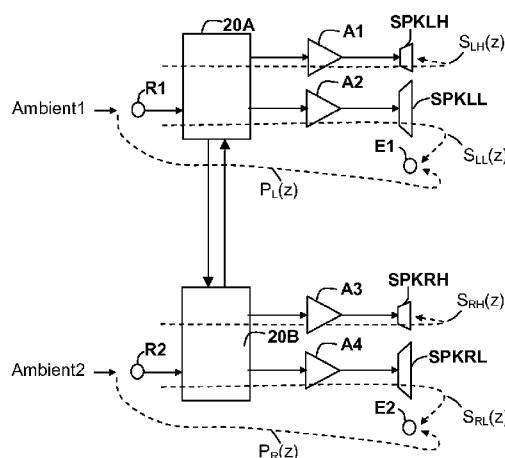
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(54) Title: LOW-LATENCY MULTI-DRIVER ADAPTIVE NOISE CANCELING (ANC) SYSTEM FOR A PERSONAL AUDIO DEVICE



(57) **Abstract:** A personal audio device including multiple output transducers for reproducing different frequency bands of a source audio signal, includes an adaptive noise canceling (ANC) circuit that adaptively generates an anti-noise signal for each of the transducers from at least one microphone signal that measures the ambient audio to generate anti-noise signals. The anti-noise signals are generated by separate adaptive filters such that the anti-noise signals cause substantial cancellation of the ambient audio at their corresponding transducers. The use of separate adaptive filters provides low-latency operation, since a crossover is not needed to split the anti-noise into the appropriate frequency bands. The adaptive filters can be implemented or biased to generate anti-noise only in the frequency band corresponding to the particular adaptive filter. The anti-noise signals are combined with source audio of the appropriate frequency band to provide outputs for the corresponding transducers.

Fig. 1B



Declarations under Rule 4.17:

- *as to applicant's entitlement to apply for and be granted a patent (Rule 4.17(ii))* — *with international search report (Art. 21(3))*
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LOW-LATENCY MULTI-DRIVER ADAPTIVE NOISE CANCELING (ANC) SYSTEM FOR A PERSONAL AUDIO DEVICE

FIELD OF THE INVENTION

[0001] The present invention relates generally to personal audio devices that include adaptive noise cancellation (ANC) and multiple drivers for differing frequency bands.

BACKGROUND OF THE INVENTION

[0002] Wireless telephones, such as mobile/cellular telephones, cordless telephones, and other consumer audio devices, such as MP3 players, are in widespread use. Performance of such devices with respect to intelligibility can be improved by providing ANC using a reference microphone to measure ambient acoustic events and then using signal processing to insert an anti-noise signal into the output of the device to cancel the ambient acoustic events.

[0003] While most audio systems implemented for personal audio devices rely on a single output transducer, in the case of transducers mounted on the housing of a wireless telephone, or a pair of transducers when earspeakers are used or when a wireless telephone or other device employs stereo speakers, for high quality audio reproduction, it may be desirable to provide separate transducers for high and low frequencies, as in high quality earspeakers. However, when implementing ANC in such systems, the latency introduced by the crossover that splits the signals between the low frequency transducer and the high frequency transducer introduces delay, which reduces the effectiveness of the ANC system, due to the increased latency of operation.

[0004] Therefore, it would be desirable to provide a personal audio device, including a wireless telephone and/or earspeakers that provide low-latency ANC operation while using multiple output transducers that handle different frequency bands.

DISCLOSURE OF THE INVENTION

[0005] The above-stated objectives of providing a personal audio device having ANC and employing multiple output transducers for handling different frequency bands, is accomplished in a personal audio system, a method of operation, and an integrated circuit.

[0006] The personal audio device includes both a low-frequency output transducer and a high-frequency transducer for reproducing a source audio signal for playback to a listener, and anti-noise signals for countering the effects of ambient audio sounds in the acoustic outputs of transducers. The personal audio device also includes the integrated circuit to provide adaptive noise-canceling (ANC) functionality. The method is a method of operation of the personal audio system and integrated circuit. A reference microphone is mounted on the device housing to provide a reference microphone signal indicative of the ambient audio sounds. The personal audio system further includes an ANC processing circuit for adaptively generating the anti-noise signals from the reference microphone signal, such that the anti-noise signals cause substantial cancellation of the ambient audio sounds at their corresponding transducers. Adaptive filters are used to generate the anti-noise signals by filtering the reference microphone signal.

[0007] The foregoing and other objectives, features, and advantages of the invention will be apparent from the following, more particular, description of the preferred embodiment of the invention, as illustrated in the accompanying drawings.

DESCRIPTION OF THE DRAWINGS

[0008] **Figure 1A** is an illustration of an exemplary wireless telephone **10** and a pair of earbuds **EB1** and **EB2**.

[0009] **Figure 1B** is a schematic diagram of circuits within wireless telephone **10**.

[0010] **Figure 2** is a block diagram of circuits within wireless telephone **10**.

[0011] **Figure 3** is a block diagram depicting signal processing circuits and functional blocks of various exemplary ANC circuits that can be used to implement ANC circuit **30** of CODEC integrated circuit **20A** of Figure 2.

[0012] **Figure 4** is a block diagram depicting signal processing circuits and functional blocks within CODEC integrated circuit **20**.

BEST MODE FOR CARRYING OUT THE INVENTION

[0013] The present invention encompasses noise canceling techniques and circuits that can be implemented in a personal audio system, such as a wireless telephone and connected earbuds. The personal audio system includes an adaptive noise canceling (ANC) circuit that measures and attempts to cancel the ambient acoustic environment at the earbuds or other output transducer location such as on the housing of a personal audio device that receives or generates the source audio signal. Multiple transducers are used, including a low-frequency and a high-frequency transducer that reproduce corresponding frequency bands of the source audio to provide a high quality audio output. The ANC circuit generates separate anti-noise signals which are provided to respective ones of the multiple transducers, to cancel ambient acoustic events at the

transducers. A reference microphone is provided to measure the ambient acoustic environment, which provides an input to separate adaptive filters that generate the anti-noise signals, so that low-latency is maintained by eliminating a need for crossover filtering of the generated anti-noise. The source audio crossover can then be placed ahead of the summation of source audio frequency band-specific components with their corresponding anti-noise signals, and the adaptive filters can be controlled to generate anti-noise only in the frequency ranges appropriate for their corresponding transducers.

[0014] **Figure 1A** shows a wireless telephone **10** and a pair of earbuds **EB1** and **EB2**, each attached to a corresponding ear **5A**, **5B** of a listener. Illustrated wireless telephone **10** is an example of a device in which the techniques disclosed herein may be employed, but it is understood that not all of the elements or configurations illustrated in wireless telephone **10**, or in the circuits depicted in subsequent illustrations, are required. Wireless telephone **10** is connected to earbuds **EB1**, **EB2** by a wired or wireless connection, e.g., a BLUETOOTH™ connection (BLUETOOTH is a trademark of Bluetooth SIG, Inc.). Earbuds **EB1**, **EB2** each have a corresponding pair of transducers **SPKLH/SPKLL** and **SPKRH/SPKRL**, respectively, which reproduce source audio including distant speech received from wireless telephone **10**, ringtones, stored audio program material, and injection of near-end speech (i.e., the speech of the user of wireless telephone **10**). Transducers **SPKLH** and **SPKRH** are high-frequency transducers or “tweeters” that reproduce the higher range of audible frequencies and transducers **SPKLL** and **SPKRL** are low-frequency transducers or “woofers” that reproduce a lower range of audio frequencies. The source audio also includes any other audio that wireless telephone **10** is required to reproduce, such as source audio from web-pages or other network communications received by wireless telephone **10** and audio indications such as battery low and other system event notifications. Reference microphones **R1**, **R2** are provided on a surface of a housing of respective earbuds **EB1**, **EB2** for measuring the ambient acoustic environment. Another pair of

microphones, error microphones **E1**, **E2**, are provided in order to further improve the ANC operation by providing a measure of the ambient audio combined with the audio reproduced by respective transducer pairs **SPKLH/SPKLL** and **SPKRH/SPKRL** close to corresponding ears **5A**, **5B**, when earbuds **EB1**, **EB2** are inserted in the outer portion of ears **5A**, **5B**.

[0015] Wireless telephone **10** includes adaptive noise canceling (ANC) circuits and features that inject anti-noise signals into transducers **SPKLH**, **SPKLL**, **SPKRH** and **SPKRL** to improve intelligibility of the distant speech and other audio reproduced by transducers **SPKLH**, **SPKLL**, **SPKRH** and **SPKRL** An exemplary circuit **14** within wireless telephone **10** includes an audio integrated circuit **20** that receives the signals from reference microphones **R1**, **R2**, a near speech microphone **NS**, and error microphones **E1**, **E2** and interfaces with other integrated circuits such as an RF integrated circuit **12** containing the wireless telephone transceiver. In other implementations, the circuits and techniques disclosed herein may be incorporated in a single integrated circuit that contains control circuits and other functionality for implementing the entirety of the personal audio device, such as an MP3 player-on-a-chip integrated circuit.

Alternatively, the ANC circuits may be included within the housing of earbuds **EB1**, **EB2** or in a module located along wired connections between wireless telephone **10** and earbuds **EB1**, **EB2**.

For the purposes of illustration, the ANC circuits will be described as provided within wireless telephone **10**, but the above variations are understandable by a person of ordinary skill in the art and the consequent signals that are required between earbuds **EB1**, **EB2**, wireless telephone **10**, and a third module, if required, can be easily determined for those variations. Near speech microphone **NS** is provided at a housing of wireless telephone **10** to capture near-end speech, which is transmitted from wireless telephone **10** to the other conversation participant(s).

Alternatively, near speech microphone **NS** may be provided on the outer surface of the housing of one of earbuds **EB1**, **EB2**, on a boom affixed to one of earbuds **EB1**, **EB2**, or on a pendant located between wireless telephone **10** and either or both of earbuds **EB1**, **EB2**.

[0016] Figure 1B shows a simplified schematic diagram of audio integrated circuits **20A, 20B** that include ANC processing, as coupled to reference microphones **R1, R2**, which provide a measurement of ambient audio sounds **Ambient1, Ambient 2** that is filtered by the ANC processing circuits within audio integrated circuits **20A, 20B**, located within corresponding earbuds **EB1, EB2**. Audio integrated circuits **20A, 20B** may be alternatively combined in a single integrated circuit such as integrated circuit **20** within wireless telephone **10**. Audio integrated circuits **20A, 20B** generate outputs for their corresponding channels that are amplified by an associated one of amplifiers **A1-A4** and which are provided to the corresponding transducer pairs **SPKLH/SPKLL** and **SPKRH/SPKRL**. Audio integrated circuits **20A, 20B** receive the signals (wired or wireless depending on the particular configuration) from reference microphones **R1, R2**, near speech microphone **NS** and error microphones **E1, E2**. Audio integrated circuits **20A, 20B** also interface with other integrated circuits such as RF integrated circuit **12** containing the wireless telephone transceiver shown in Figure 1A. In other configurations, the circuits and techniques disclosed herein may be incorporated in a single integrated circuit that contains control circuits and other functionality for implementing the entirety of the personal audio device, such as a MP3 player-on-a-chip integrated circuit. Alternatively, multiple integrated circuits may be used, for example, when a wireless connection is provided from each of earbuds **EB1, EB2** to wireless telephone **10** and/or when some or all of the ANC processing is performed within earbuds **EB1, EB2** or a module disposed along a cable connecting wireless telephone **10** to earbuds **EB1, EB2**.

[0017] In general, the ANC techniques illustrated herein measure ambient acoustic events (as opposed to the output of transducers **SPKLH, SPKLL, SPKRH** and **SPKRL** and/or the near-end speech) impinging on reference microphones **R1, R2** and also measure the same ambient acoustic events impinging on error microphones **E1, E2**. The ANC processing circuits of integrated circuits **20A, 20B** individually adapt an anti-noise signal generated from the output of

the corresponding reference microphone **R1**, **R2** to have a characteristic that minimizes the amplitude of the ambient acoustic events at the corresponding error microphone **E1**, **E2**. Since acoustic path $P_L(z)$ extends from reference microphone **R1** to error microphone **E1**, the ANC circuit in audio integrated circuit **20A** is essentially estimating acoustic path $P_L(z)$ combined with removing effects of electro-acoustic paths $S_{LH}(z)$ and $S_{LL}(z)$ that represent, respectively, the response of the audio output circuits of audio integrated circuit **20A** and the acoustic/electric transfer function of transducers **SPKLH** and **SPKLL**. The estimated response includes the coupling between transducers **SPKLH**, **SPKLL** and error microphone **E1** in the particular acoustic environment which is affected by the proximity and structure of ear **5A** and other physical objects and human head structures that may be in proximity to earbud **EB1**. Similarly, audio integrated circuit **20B** estimates acoustic path $P_R(z)$ combined with removing effects of electro-acoustic paths $S_{RH}(z)$ and $S_{RL}(z)$ that represent, respectively, the response of the audio output circuits of audio integrated circuit **20B** and the acoustic/electric transfer function of transducers **SPKRH** and **SPKRL**.

[0018] Referring now to **Figure 2**, circuits within earbuds **EB1**, **EB2** and wireless telephone **10** are shown in a block diagram. The circuit shown in **Figure 2** further applies to the other configurations mentioned above, except that signaling between CODEC integrated circuit **20** and other units within wireless telephone **10** are provided by cables or wireless connections when audio integrated circuits **20A**, **20B** are located outside of wireless telephone **10**, e.g., within corresponding earbuds **EB1**, **EB2**. In such a configuration, signaling between a single integrated circuit **20** that implements integrated circuits **20A-20B** and error microphones **E1**, **E2**, reference microphones **R1**, **R2** and transducers **SPKLH**, **SPKLL**, **SPKRH** and **SPKRL** are provided by wired or wireless connections when audio integrated circuit **20** is located within wireless telephone **10**. In the illustrated example, audio integrated circuits **20A**, **20B** are shown as separate and substantially identical circuits, so only audio integrated circuit **20A** will be

described in detail below.

[0019] Audio integrated circuit **20A** includes an analog-to-digital converter (ADC) **21A** for receiving the reference microphone signal from reference microphone **R1** and generating a digital representation **ref** of the reference microphone signal. Audio integrated circuit **20A** also includes an ADC **21B** for receiving the error microphone signal from error microphone **E1** and generating a digital representation **err** of the error microphone signal, and an ADC **21C** for receiving the near speech microphone signal from near speech microphone **NS** and generating a digital representation of near speech microphone signal **ns**. (Audio integrated circuit **20B** receives the digital representation of near speech microphone signal **ns** from audio integrated circuit **20A** via the wireless or wired connections as described above.) Audio integrated circuit **20A** generates an output for driving transducer **SPKLH** from an amplifier **A1**, which amplifies the output of a digital-to-analog converter (DAC) **23A** that receives the output of a combiner **26A**. A combiner **26C** combines left-channel internal audio signal **ial** and source audio **ds**, which is received from a radio frequency (RF) integrated circuit **22**. Combiner **26A** combines source audio **ds_h+ia_{lh}**, which is the high-frequency band component of the output of combiner **26C** with high-frequency band anti-noise signal **anti-noise_{lh}** generated by a left-channel ANC circuit **30**, which by convention has the same polarity as the noise in reference microphone signal **ref** and is therefore subtracted by combiner **26A**. Combiner **26A** also combines an attenuated high-frequency portion of near speech signal **ns**, i.e., sidetone information **st_h**, so that the user of wireless telephone **10** hears their own voice in proper relation to downlink speech **ds**. Near speech signal **ns** is also provided to RF integrated circuit **22** and is transmitted as uplink speech to the service provider via an antenna **ANT**. Similarly, left-channel audio integrated circuit **20A** generates an output for driving transducer **SPKLL** from an amplifier **A2**, which amplifies the output of a digital-to-analog converter (DAC) **23B** that receives the output of a combiner **26B**. Combiner **26B** combines source audio **ds_l+ia_{ll}**, which is the low-frequency band

component of the output of combiner **26C** with low-frequency band anti-noise signal **anti-noise_{ll}** generated by ANC circuit **30**, which by convention has the same polarity as the noise in reference microphone signal **ref** and is therefore subtracted by combiner **26B**. Combiner **26B** also combines an attenuated portion of near speech signal **ns**, i.e., sidetone low-frequency information **st_l**.

[0020] Referring now to **Figure 3**, an example of details within ANC circuit **30** are shown, and as may be used to implement audio integrated circuit **20B** of Figure 2. An identical circuit is used to implement audio integrated circuit **20A**, with changes to the channel labels within the diagram as noted below. A high-frequency channel **50A** and a low-frequency channel **50B** are provided, for generating anti-noise signals **anti-noise_{rh}** and **anti-noise_{rl}**, respectively. In the description below, where signal and response labels contained the letter “r” indicating the right channel, the letter would be replaced with “l” to indicate the left channel in another circuit according to **Figure 3** as implemented within audio integrated circuit **20A** of Figure 2. Where signals and responses are labeled with the letter “h” for low-frequency in high-frequency channel **50A**, the corresponding elements in low-frequency channel **50B** would be replaced with signals and responses labeled with the letter “l”. An adaptive filter **32A** receives reference microphone signal **ref** and under ideal circumstances, adapts its transfer function $W_{rh}(z)$ to be $P_r(z)/S_{rh}(z)$ to generate anti-noise signal **anti-noise_{rh}**. The coefficients of adaptive filter **32A** are controlled by a W coefficient control block **31A** that uses a correlation of two signals to determine the response of adaptive filter **32A**, which generally minimizes, in a least-mean squares sense, those components of reference microphone signal **ref** that are present in error microphone signal **err**. While the example disclosed herein uses an adaptive filter **32A**, connected in a feed-forward configuration, the techniques disclosed herein can be implemented in a noise-canceling system having fixed or programmable filters, where the coefficients of adaptive filter **32A** are pre-set, selected or otherwise not continuously adapted, and also

alternatively or in combination with the fixed-filter topology, the techniques disclosed herein can be applied in feedback ANC systems or hybrid feedback/feed-forward ANC systems. The signals provided as inputs to W coefficient control block **31A** are the reference microphone signal **ref** as shaped by a copy of an estimate of the response of path $S_{rh}(z)$ provided by a filter **34B** and another signal provided from the output of a combiner **36C** that includes error microphone signal **err**. By transforming reference microphone signal **ref** with a copy of the estimate of the response of path $S_{rh}(z)$, $SE_{rhCOPY}(z)$, and minimizing the portion of the error signal that correlates with components of reference microphone signal **ref**, adaptive filter **32A** adapts to the desired response of $P_r(z)/S_{rh}(z)$.

[0021] In addition to error microphone signal **err**, the other signal processed along with the output of filter **34B** by W coefficient control block **31A** includes an inverted amount of the source audio ($ds+ia_r$) including downlink audio signal **ds** and internal audio **ian** processed by a secondary path filter **34A** having response $SE_{rh}(z)$, of which response $SE_{rhCOPY}(z)$ is a copy. Source audio ($ds+ia_r$) is first filtered before being provided to high-frequency channel **50A** by a high-pass filter **35A**, which passes only the frequencies to be rendered by the high-frequency transducer **SPKLH** or **SPKRH**. Similarly, the source audio ($ds+ia_r$) provided to low-frequency channel **50B** is first filtered by a low-pass filter **35B**, which passes only frequencies to be rendered by the low-frequency transducer **SPKLL** or **SPKRL**. Thus, high-pass filter **35A** and low-pass filter **35B** form a cross-over with respect to source audio ($ds+ia_r$), so that only the appropriate frequencies are passed to high-frequency channel **50A** and low-frequency channel **50B**, respectively, and having bandwidths appropriate to respective transducers **SPKLH**, **SPKLL** or **SPKRH**, **SPKRL**. By injecting an inverted amount of source audio ($ds+ia_r$) that has been filtered by response $SE_{rh}(z)$, adaptive filter **32A** is prevented from adapting to the relatively large amount of source audio present in error microphone signal **err**. By transforming the inverted copy of source audio ($ds+ia_r$) with the estimate of the response of path $S_{rh}(z)$, the

source audio that is removed from error microphone signal **err** before processing should match the expected version of source audio ($ds+ia_r$) reproduced at error microphone signal **err**. The source audio amounts match because the electrical and acoustical path of $S_{rh}(z)$ is the path taken by source audio ($ds+ia_r$) to arrive at error microphone **E**. Filter **34B** is not an adaptive filter, per se, but has an adjustable response that is tuned to match the response of secondary path adaptive filter **34A**, so that the response of filter **34B** tracks the adapting of secondary path adaptive filter **34A**. To implement the above, secondary path adaptive filter **34A** has coefficients controlled by an SE coefficient control block **33A**. Secondary path adaptive filter **34A** processes the low or high-frequency source audio ($ds+ia_r$) to provide a signal representing the expected source audio delivered to error microphone **E**. Secondary path adaptive filter **34A** is thereby adapted to generate a signal from source audio ($ds+ia_r$), that when subtracted from error microphone signal **err**, forms an error signal **e** containing the content of error microphone signal **err** that is not due to source audio ($ds+ia_r$). Combiner **36C** removes the filtered source audio ($ds+ia_r$) from error microphone signal **err** to generate the above-described error signal **e**.

[0022] Each of the high-frequency channel **50A** and low-frequency channel **50B** can operate independently to generate respective anti-noise signals **anti-noise_h** and **anti-noise_l**. However, since error signal **e** and reference microphone signal **ref** may contain frequencies of any frequency in the audio band, without band-limiting anti-noise signals **anti-noise_h** and **anti-noise_l**, they may contain components that should not be sent to their respective high- and low-frequency transducers **SPKRH/SPKLH** and **SPKRL/SPKLL**. Therefore, a noise injection technique is used to control the response $W_{rh}(z)$ of adaptive filter **32A**. A noise source **37** generates an output noise signal $n_h(z)$ that is supplied to a copy $W_{rhCOPY}(z)$ of the response $W_{rh}(z)$ of adaptive filter **32A** provided by an adaptive filter **32B**. A combiner **36A** adds noise signal $n_h(z)$ to the output of adaptive filter **34B** that is provided to W coefficient control **31A**. Noise signal $n_h(z)$, as shaped by filter **32B**, is subtracted from the output of combiner **36C** by a

combiner **36B** so that noise signal $n_h(z)$ is asymmetrically added to the correlation inputs to W coefficient control **31A**, with the result that the response $W_{rh}(z)$ of adaptive filter **32A** is biased by the completely correlated injection of noise signal $n_h(z)$ to each correlation input to W coefficient control **31A**. Since the injected noise appears directly at the reference input to W coefficient control **31A**, does not appear in error microphone signal **err**, and only appears at the other input to W coefficient control **31A** via the combining of the filtered noise at the output of filter **32B** by combiner **36B**, W coefficient control **31A** will adapt $W_{rh}(z)$ to attenuate the frequencies present in $n_h(z)$. The content of noise signal $n_h(z)$ does not appear in the anti-noise signal, only in the response $W_{rh}(z)$ of adaptive filter **32A** which will have amplitude decreases at the frequencies/bands in which noise signal $n_h(z)$ has energy.

[0023] In order to prevent low-frequencies from being generated in anti-noise signal **anti-noise_h**, noise source **37** generates noise having a spectrum that has energy in the low-frequency bands, which will cause W coefficient control **31A** to decrease the gain of adaptive filter **32A** in those low frequency bands in an attempt to cancel the apparent source of ambient acoustic sound due to injected noise signal $n_h(z)$. For example, a white noise source could be filtered by a response similar to the response of low-pass filter **35B** for use as noise source **37** in high-frequency channel **50A**, which will cause adaptive filter **32A** to have low gain in the regions of the pass-band of low-pass filter **35B**. By doing the same for low-frequency channel **50B**, i.e. filtering a white noise source with a response matching the response of high-pass filter **35A**, a cross-over is effectively formed by the adaptation of adaptive filters **32A** in high-frequency channel **50A** and low-frequency channel **50B** that prevents undesirable frequencies in respective anti-noise signals **anti-noise_h** and **anti-noise_l**. A similar construct could be formed around secondary path adaptive filter **34A**, but since the input to secondary path adaptive filter **34A** is already filtered by a respective one of filters **35A**, **35B** to remove out-of-band energy, such noise injection should not be needed to remove undesirable frequencies from the output of secondary path adaptive filter

34A. One advantage of using noise-injection, rather than additional filtering, to remove undesirable cross-over energy from anti-noise signals **anti-noise_h** and **anti-noise_l** is that additional latency is not introduced other than any latency due to the change in response due to noise source **37**.

[0024] Referring now to **Figure 4**, a block diagram of an ANC system is shown for implementing ANC techniques as depicted in Figure 3 and having a processing circuit **40** as may be implemented within audio integrated circuits **20A**, **20B** of Figure 2, which is illustrated as combined within one circuit, but could be implemented as two or more processing circuits that inter-communicate. Processing circuit **40** includes a processor core **42** coupled to a memory **44** in which are stored program instructions comprising a computer program product that may implement some or all of the above-described ANC techniques, as well as other signal processing. Optionally, a dedicated digital signal processing (DSP) logic **46** may be provided to implement a portion of, or alternatively all of, the ANC signal processing provided by processing circuit **40**. Processing circuit **40** also includes ADCs **21A-21E**, for receiving inputs from reference microphone **R1**, error microphone **E1**, near speech microphone **NS**, reference microphone **R2**, and error microphone **E2**, respectively. In alternative embodiments in which one or more of reference microphone **R1**, error microphone **E1**, near speech microphone **NS**, reference microphone **R2**, and error microphone **E2** have digital outputs or are communicated as digital signals from remote ADCs, the corresponding ones of ADCs **21A-21E** are omitted and the digital microphone signal(s) are interfaced directly to processing circuit **40**. DAC **23A** and amplifier **A1** are also provided by processing circuit **40** for providing the transducer output signal to transducer **SPKLH**, including anti-noise as described above. Similarly, DACs **23B-23D** and amplifiers **A2-A4** provide other transducer output signals to transducer pairs **SPKLH**, **SPKLL**, **SPKRH** and **SPKRL**. The transducer output signals may be digital output signals for provision to modules that reproduce the digital output signals acoustically.

[0025] While the invention has been particularly shown and described with reference to the preferred embodiments thereof, it will be understood by those skilled in the art that the foregoing and other changes in form, and details may be made therein without departing from the spirit and scope of the invention.

CLAIMS**WHAT IS CLAIMED IS:**

1. A personal audio system, comprising:

 a source of audio for reproduction, wherein the source of audio provides a source audio signal;

 a first transducer for reproducing high-frequency content of the source audio signal for playback to a listener and a first anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the first transducer;

 a second transducer for reproducing low-frequency content of the source audio signal for playback to the listener and a second anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the second transducer;

 at least one microphone for providing at least one microphone signal indicative of the ambient audio sounds; and

 a processing circuit that generates the first anti-noise signal and the second anti-noise signal from the at least one microphone signal using a first filter to reduce the presence of the ambient audio sounds at the first transducer and the second transducer in conformity with the at least one microphone signal, wherein the processing circuit generates the second anti-noise signal from the at least one microphone signal using a second filter to reduce the presence of the ambient audio sounds at the first transducer and the second transducer in conformity with the at least one microphone signal.

2. The personal audio system of Claim 1, wherein the first filter is a first adaptive filter having a first response that adapts to reduce the presence of the ambient audio sounds, and wherein the second filter is a second adaptive filter that adapts to reduce the presence of the ambient audio sounds.

3. The personal audio device of Claim 1, wherein the processing circuit restricts content of the first anti-noise signal to a first predetermined frequency range by limiting the first frequency response of the first adaptive filter to the first predetermined frequency range, and wherein the processing circuit restricts content of the second anti-noise signal to a second predetermined frequency range by limiting the second response of the second adaptive filter to a second predetermined frequency range, wherein the first predetermined frequency range and the second predetermined frequency range are substantially different.

4. The personal audio device of Claim 3, further comprising an error microphone for providing an error microphone signal indicative of the ambient audio sounds and acoustic outputs of the first transducer and the second transducer, wherein the first adaptive filter has a first coefficient generator that adapts to minimize components of the reference microphone signal present in the error microphone signal, and wherein the processing circuit restricts adaptation of the first frequency response by altering the frequency content of a first signal input to the first coefficient generator, and wherein the second adaptive filter has a second coefficient generator that adapts to minimize components of the reference microphone signal present in the error microphone signal, and wherein the processing circuit restricts adaptation of the first frequency response by altering the frequency content of a second signal input to the second coefficient generator.

5. The personal audio device of Claim 4, wherein the processing circuit alters the frequency content of the first signal input to the first coefficient generator by injecting a first additional signal having first predetermined frequency content in the first predetermined frequency range into the first signal input to the first coefficient generator, and wherein the processing circuit alters the frequency content of the second signal input to the second coefficient generator by injecting a second additional signal having second predetermined frequency content in the second predetermined frequency range into the second signal input to the second coefficient generator.

6. The personal audio device of Claim 5, wherein the first additional signal and the second additional signal are noise signals.

7. The personal audio device of Claim 1, wherein the processing circuit receives the source audio signal and filters the source audio signal to provide a crossover that generates a higher-frequency content source audio signal and a lower-frequency content source audio signal, and wherein the processing circuit further combines the higher-frequency content source audio signal with the first anti-noise signal and combines the lower-frequency content source audio signal with the second anti-noise signal.

8. The personal audio device of Claim 1, wherein the first transducer is a high-frequency transducer of an earspeaker and wherein the second transducer is a low-frequency transducer of the earspeaker.

9. The personal audio device of Claim 8, further comprising:

a third transducer for reproducing high-frequency content of a second source audio signal and a third anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the third transducer; and

a fourth transducer for reproducing low-frequency content of the second source audio signal and a fourth anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the fourth transducer, and wherein the processing circuit further generates the third anti-noise signal and the fourth anti-noise signal from the at least one microphone signal using a third filter to reduce the presence of the ambient audio sounds at the third transducer in conformity with the at least one microphone signal, wherein the processing circuit generates the fourth anti-noise signal from the at least one microphone signal using a fourth filter to reduce the presence of the ambient audio sounds at the fourth transducer in conformity with the at least one microphone signal.

10. A method of counteracting effects of ambient audio sounds by a personal audio system, the method comprising:

measuring ambient audio sounds with at least one microphone to produce at least one microphone signal;

first generating a first anti-noise signal from the at least one microphone signal using a first filter to reduce the presence of the ambient audio sounds at the first transducer in conformity with the at least one microphone signal;

second generating a second anti-noise signal from the at least one microphone signal using a second filter to reduce the presence of the ambient audio sounds at the second transducer in conformity with the at least one microphone signal;

providing a source of audio for reproduction, wherein the source of audio provides a source audio signal;

reproducing high-frequency content of the source audio signal and the first anti-noise signal with the first transducer; and

reproducing low-frequency content of the source audio signal and the second anti-noise signal with the second transducer.

11. The method of Claim 10, wherein the first filter is a first adaptive filter having a first response that adapts to reduce the presence of the ambient audio sounds, and wherein the second filter is a second adaptive filter that adapts to reduce the presence of the ambient audio sounds.

12. The method of Claim 10, wherein the first generating comprises restricting content of the first anti-noise signal to a first predetermined frequency range by limiting the first frequency response of the first adaptive filter to the first predetermined frequency range, and wherein the second generating further comprises restricting content of the second anti-noise signal to a second predetermined frequency range by limiting the second response of the second adaptive filter to a second predetermined frequency range, and wherein the first predetermined frequency range and the second predetermined frequency range are substantially different.

13. The method of Claim 12, further comprising measuring the ambient audio sounds and acoustic outputs of the first transducer and the second transducer with an error microphone to generate an error microphone signal, wherein the first generating comprises adapting coefficients of a first coefficient generator that controls the first frequency response to minimize components of the reference microphone signal present in the error microphone signal, and wherein the second generating comprises adapting coefficients of a second coefficient generator that controls the second frequency response to minimize components of the reference microphone signal present in the error microphone signal, wherein the first generating restricts adaptation of the first frequency response by altering frequency content of a first signal input to the first coefficient generator, and wherein the second generating restricts adaptation of the second frequency response by altering frequency content of a second signal input to the second coefficient generator.

14. The method of Claim 13, wherein the first generating restricts adaptation of the first frequency response by injecting a first additional signal having a first predetermined frequency content in the first predetermined frequency range into at least one first signal input to the first coefficient generator, and wherein the second generating restricts adaptation of the second frequency response by injecting a second additional signal having a second predetermined frequency content in the second predetermined frequency range into at least one second signal input to the second coefficient generator.

15. The method of Claim 14, wherein the first additional signal and the second additional signal are noise signals.

16. The method of Claim 10, further comprising:

receiving the source audio signal and filtering the source audio signal to implement a crossover that generates a higher-frequency content source audio signal and a lower-frequency content source audio signal; and

combining the higher-frequency content source audio signal with the first anti-noise signal; and

combining the lower-frequency content source audio signal with the second anti-noise signal.

17. The method of Claim 10, wherein the first transducer is a high-frequency transducer of an earspeaker and wherein the second transducer is a low-frequency transducer of the earspeaker.

18. The method of Claim 17, further comprising:

reproducing high-frequency content of a second source audio signal and a third anti-noise signal with a third transducer for countering the effects of ambient audio sounds in an acoustic output of the third transducer; and

reproducing low-frequency content of the second source audio signal and a fourth anti-noise signal with a fourth transducer for countering the effects of ambient audio sounds in an acoustic output of the fourth transducer;

generating the third anti-noise signal and the fourth anti-noise signal from the at least one microphone signal using a third filter to reduce the presence of the ambient audio sounds at the third transducer and the fourth transducer in conformity with the at least one microphone signal; and

generating the fourth anti-noise signal from the at least one microphone signal using a fourth filter to reduce the presence of the ambient audio sounds at the third transducer and the fourth transducer in conformity with the at least one microphone signal.

19. An integrated circuit for implementing at least a portion of a personal audio system, comprising:

a source of audio for reproduction, wherein the source of audio provides a source audio signal;

a first output for providing a first output signal to a first transducer for reproducing high-frequency content of the source audio signal and a first anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the first transducer;

a second output for providing a second output signal to a second transducer for reproducing a second audio signal including both second source audio for playback to a listener and a second anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the second earspeaker;

at least one microphone input for providing at least one microphone signal indicative of the ambient audio sounds; and

a processing circuit that generates the first anti-noise signal and the second anti-noise signal from the at least one microphone signal using a first filter to reduce the presence of the ambient audio sounds at the first transducer and the second transducer in conformity with the at least one microphone signal, wherein the processing circuit generates the second anti-noise signal from the at least one microphone signal using a second filter to reduce the presence of the ambient audio sounds at the first transducer and the second transducer in conformity with the at least one microphone signal.

20. The integrated circuit of Claim 19, wherein the first filter is a first adaptive filter having a first response that adapts to reduce the presence of the ambient audio sounds, and wherein the second filter is a second adaptive filter that adapts to reduce the presence of the ambient audio sounds.

21. The integrated circuit of Claim 19, wherein the processing circuit restricts content of the first anti-noise signal to a first predetermined frequency range by limiting the first frequency response of the first adaptive filter to the first predetermined frequency range, and wherein the processing circuit restricts content of the second anti-noise signal to a second predetermined frequency range by limiting the second response of the second adaptive filter to a second predetermined frequency range, wherein the first predetermined frequency range and the second predetermined frequency range are substantially different.

22. The integrated circuit of Claim 21, further comprising an error microphone for providing an error microphone signal indicative of the ambient audio sounds and acoustic outputs of the first transducer and the second transducer, wherein the first adaptive filter has a first coefficient generator that adapts to minimize components of the reference microphone signal present in the error microphone signal, and wherein the processing circuit restricts adaptation of the first frequency response by altering the frequency content of a first signal input to the first coefficient generator, and wherein the second adaptive filter has a second coefficient generator that adapts to minimize components of the reference microphone signal present in the error microphone signal, and wherein the processing circuit restricts adaptation of the first frequency response by altering the frequency content of a second signal input to the second coefficient generator.

23. The integrated circuit of Claim 22, wherein the processing circuit alters the frequency content of the first signal input to the first coefficient generator by injecting a first additional signal having a first predetermined frequency content in the first predetermined frequency range into the first signal input to the first coefficient generator, and wherein the processing circuit alters the frequency content of the second signal input to the second coefficient generator by injecting a second additional signal having a second predetermined frequency content in the second predetermined frequency range into the second signal input to the second coefficient generator.

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24. The integrated circuit of Claim 23, wherein the first additional signal and the second additional signal are noise signals.

25. The integrated circuit of Claim 19, wherein the processing circuit receives the source audio signal and filters the source audio signal to provide a crossover that generates a higher-frequency content source audio signal and a lower-frequency content source audio signal, and wherein the processing circuit further combines the higher-frequency content source audio signal with the first anti-noise signal and combines the lower-frequency content source audio signal with the second anti-noise signal.

26. The integrated circuit of Claim 19, wherein the first transducer is a high-frequency transducer of an earspeaker and wherein the second transducer is a low-frequency transducer of the earspeaker.

27. The integrated circuit of Claim 26, further comprising:

a third output for providing a third output signal to a third transducer for reproducing high-frequency content of a second source audio signal and a third anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the third transducer; and

a fourth output for providing a fourth output signal to a fourth transducer for reproducing low-frequency content of the second source audio signal and a fourth anti-noise signal for countering the effects of ambient audio sounds in an acoustic output of the fourth transducer, and wherein the processing circuit further generates the third anti-noise signal and the fourth anti-noise signal from the at least one microphone signal using a third filter to reduce the presence of the ambient audio sounds at the third transducer and the fourth transducer in conformity with the at least one microphone signal, wherein the processing circuit generates the fourth anti-noise signal from the at least one microphone signal using a fourth filter to reduce the presence of the ambient audio sounds at the third transducer and the fourth transducer in conformity with the at least one microphone signal.

28. A personal audio system, comprising:

multiple output transducers;

at least one microphone for providing at least one microphone signal indicative of the ambient audio sounds; and

a processing circuit that implements adaptive noise-canceling in which multiple adaptive filters generate multiple anti-noise signals for corresponding ones of the multiple output transducers and operate as cross-overs for separating the at least one microphone signal into multiple frequency bands corresponding to the multiple output transducers by generating the multiple anti-noise signals in the corresponding ones of the multiple frequency bands.

29. A method of countering effects of ambient audio sounds by a personal audio system, the method comprising:

measuring ambient audio sounds with at least one microphone to generate at least one microphone signal;

generating multiple anti-noise signals for providing to corresponding ones of multiple output transducers using corresponding ones of multiple adaptive filters that operate as crossovers for separating the at least one microphone signal into multiple frequency bands corresponding to the multiple output transducers by generating the multiple anti-noise signals in the corresponding ones of the multiple frequency bands.

30. An integrated circuit for implementing at least a portion of a personal audio system, comprising:

multiple outputs for providing multiple output signals to corresponding ones of multiple output transducers;

at least one microphone input for receiving at least one microphone signal indicative of the ambient audio sounds; and

a processing circuit that implements adaptive noise-canceling in which multiple adaptive filters generate multiple anti-noise signals at corresponding ones of the multiple outputs and operate as cross-overs for separating the at least one microphone signal into multiple frequency bands corresponding to the multiple output transducers by generating the multiple anti-noise signals in the corresponding ones of the multiple frequency bands.

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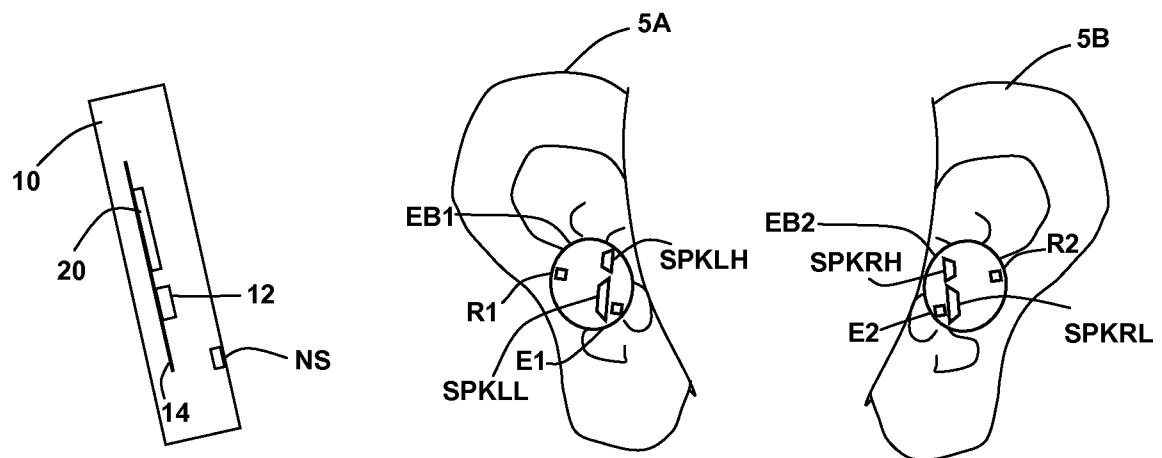


Fig. 1A

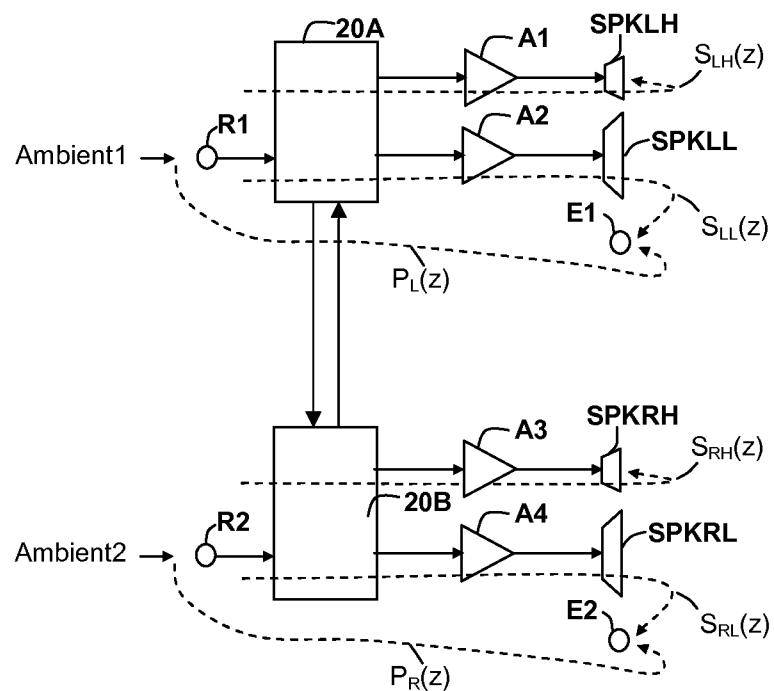


Fig. 1B

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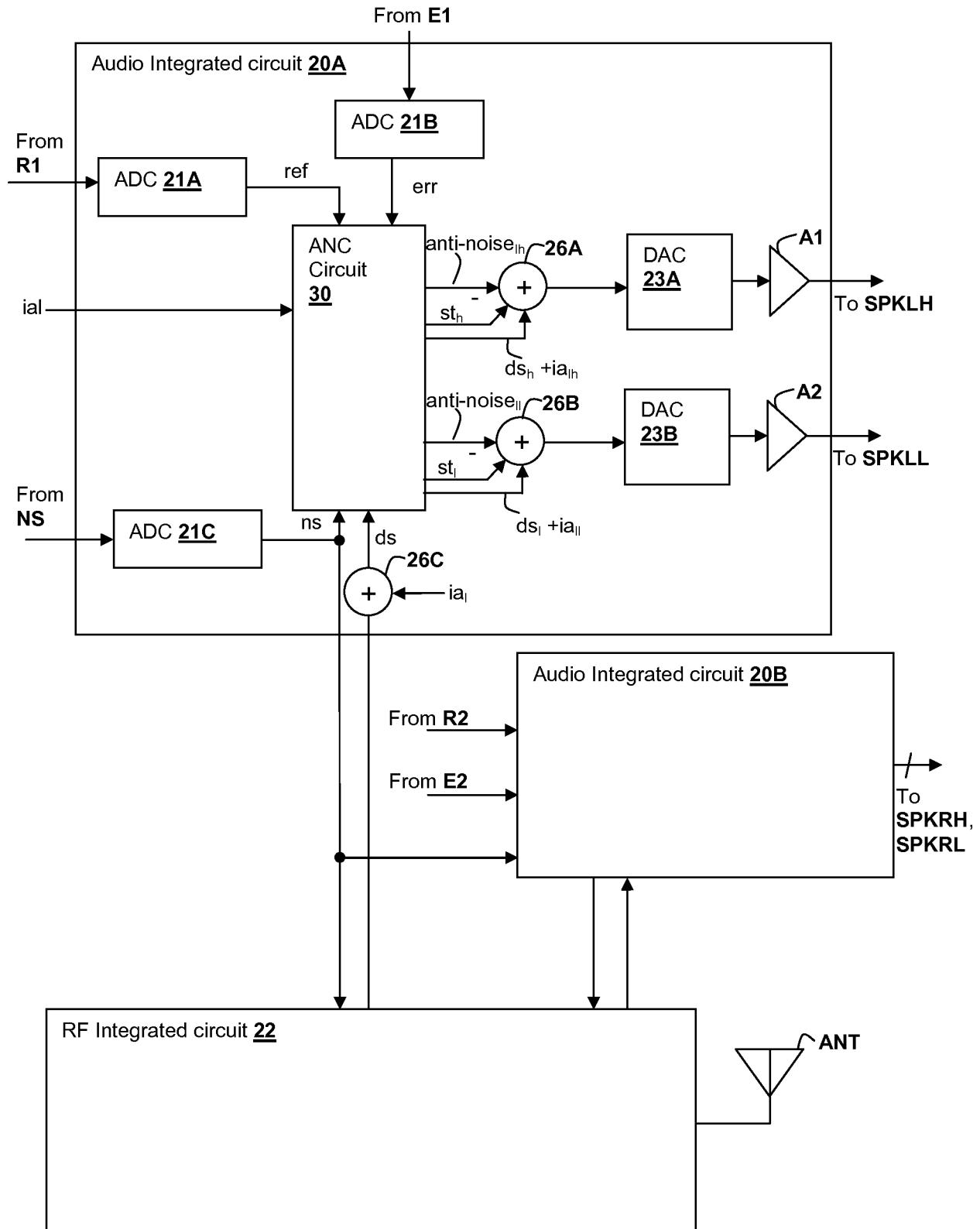


Fig. 2

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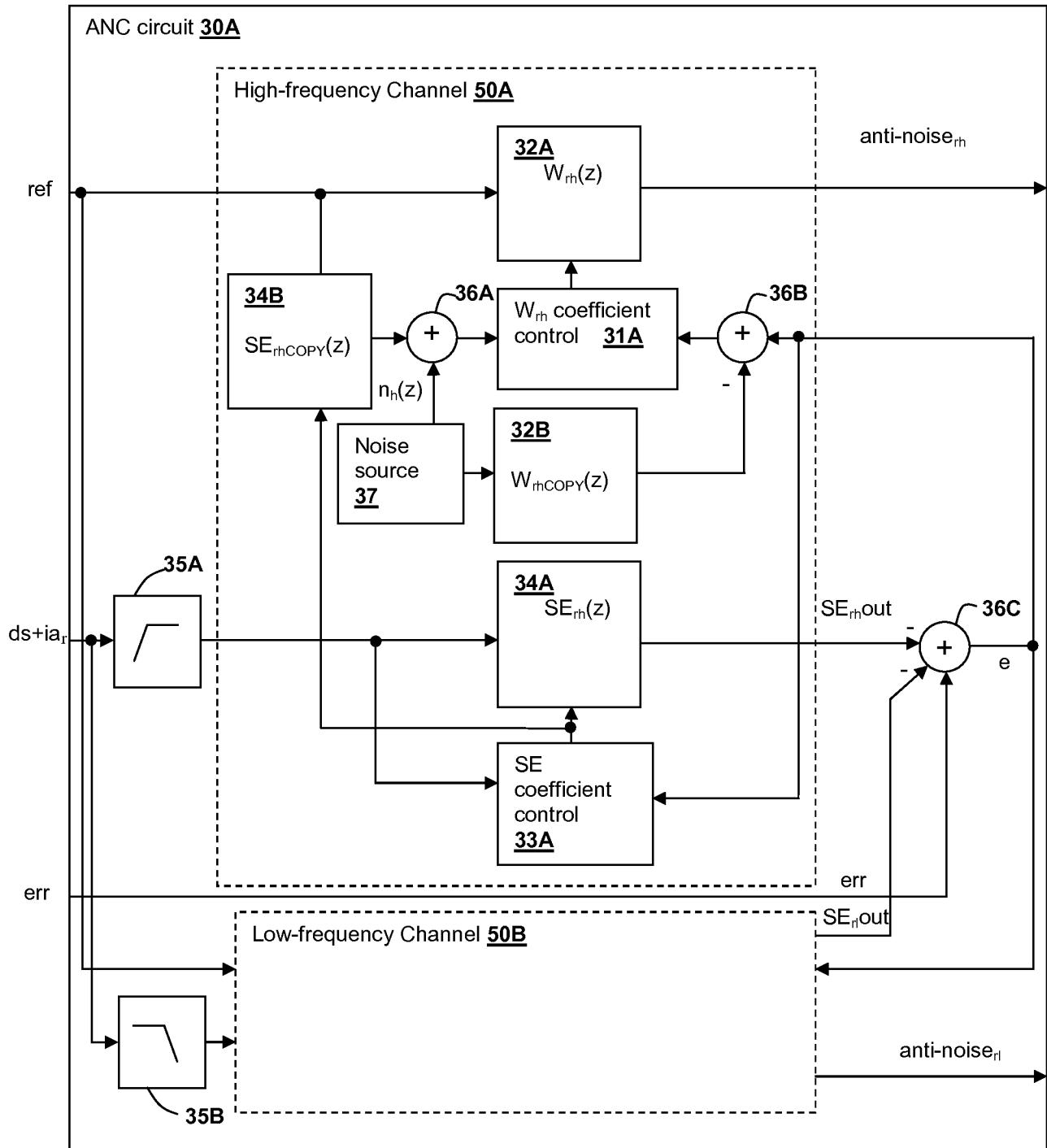


Fig. 3

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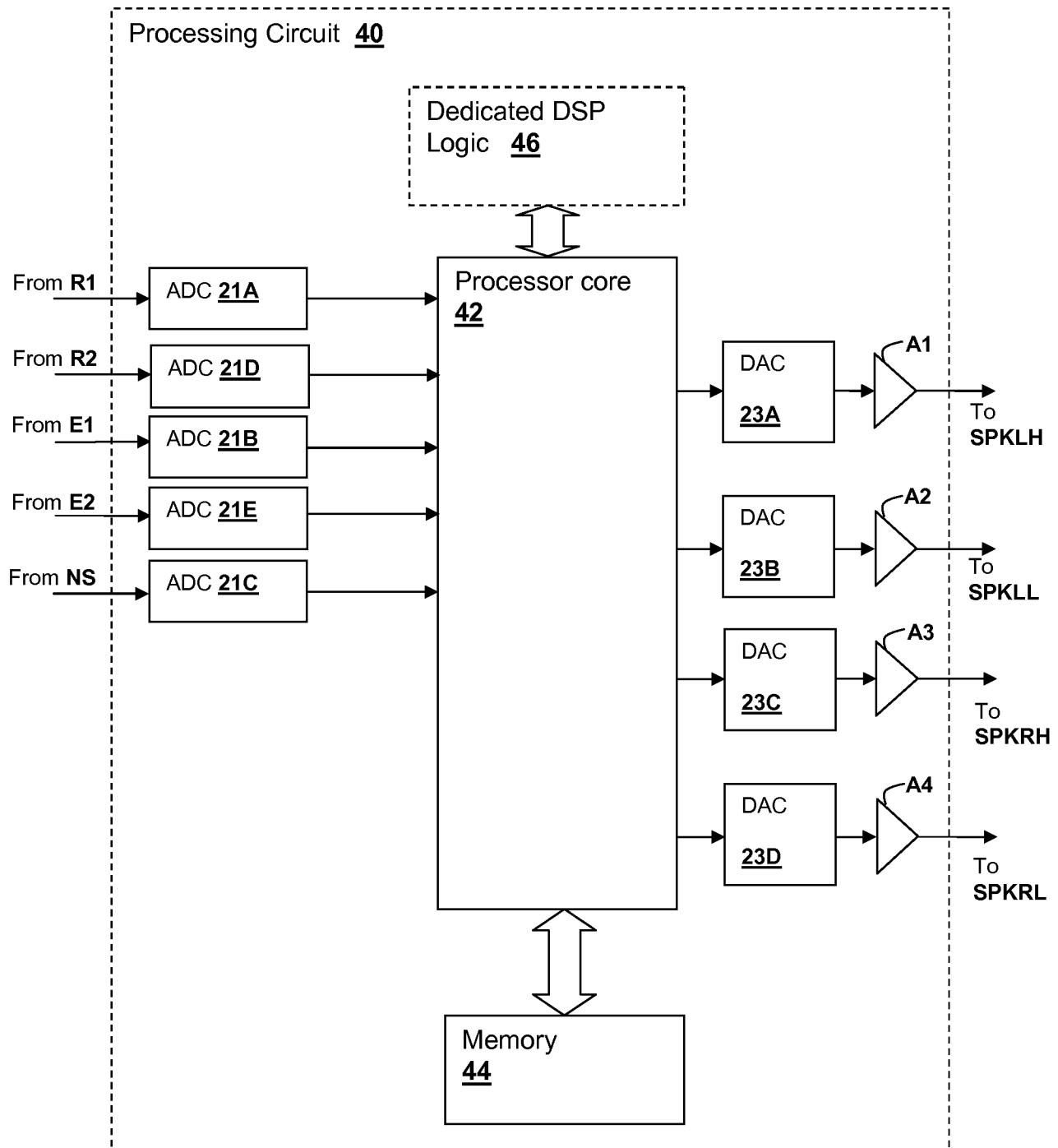


Fig. 4

INTERNATIONAL SEARCH REPORT

International application No
PCT/US2014/016833

A. CLASSIFICATION OF SUBJECT MATTER

INV. G10K11/178

ADD. H04R3/12 H04R1/10

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)

G10K H04R

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	US 2012/259626 A1 (LI REN [US] ET AL) 11 October 2012 (2012-10-11)	1,7-10, 16-19, 25-27
Y	paragraphs [0028] - [0034], [0046] - [0050], [0112] - [0113]; figure 2	2-4, 11-13, 20-22
A	-----	5,6,14, 15,23, 24,28-30
Y	US 5 410 605 A (SAWADA HIDESHI [JP] ET AL) 25 April 1995 (1995-04-25) column 4, line 27 - column 8, line 63; figure 5 ----- -/-	2-4, 11-13, 20-22



Further documents are listed in the continuation of Box C.



See patent family annex.

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

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"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

20 May 2014

02/06/2014

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INTERNATIONAL SEARCH REPORT

International application No
PCT/US2014/016833

C(Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP 1 880 699 A2 (PHONAK AG [CH]) 23 January 2008 (2008-01-23) paragraph [0027] - paragraph [0037]; figure 2 -----	1-3, 10-12, 19-21
A	US 2012/250873 A1 (BAKALOS PERICLES [US] ET AL) 4 October 2012 (2012-10-04) paragraphs [0027] - [0042], [0096] - [0107]; figure 3 -----	4,13,22

INTERNATIONAL SEARCH REPORT

Information on patent family members

International application No PCT/US2014/016833

Patent document cited in search report	Publication date	Patent family member(s)			Publication date
US 2012259626	A1	11-10-2012	CN	103460716 A	18-12-2013
			EP	2695394 A1	12-02-2014
			KR	20130137046 A	13-12-2013
			US	2012259626 A1	11-10-2012
			WO	2012138435 A1	11-10-2012
<hr style="border-top: 1px dashed black;"/>					
US 5410605	A	25-04-1995	DE	4221292 A1	14-01-1993
			GB	2257327 A	06-01-1993
			JP	3471370 B2	02-12-2003
			JP	H0511783 A	22-01-1993
			US	5410605 A	25-04-1995
<hr style="border-top: 1px dashed black;"/>					
EP 1880699	A2	23-01-2008	EP	1629808 A1	01-03-2006
			EP	1880699 A2	23-01-2008
<hr style="border-top: 1px dashed black;"/>					
US 2012250873	A1	04-10-2012	CN	103460715 A	18-12-2013
			EP	2692145 A1	05-02-2014
			US	2012250873 A1	04-10-2012
			WO	2012134874 A1	04-10-2012
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