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(12) **United States Patent**
Goldstein et al.

(10) **Patent No.: US 11,710,473 B2**
(45) **Date of Patent: Jul. 25, 2023**

(54) **METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION**

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(73) Assignee: **Staton Techiya LLC**, Delray Beach, FL (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **17/592,143**

(22) Filed: **Feb. 3, 2022**

(65) **Prior Publication Data**

US 2022/0230616 A1 Jul. 21, 2022

Related U.S. Application Data

(63) Continuation of application No. 17/321,892, filed on May 17, 2021, which is a continuation of application
(Continued)

(51) **Int. Cl.**
G10K 11/178 (2006.01)
H04R 3/00 (2006.01)
(Continued)

(52) **U.S. Cl.**
CPC ***G10K 11/17827*** (2018.01); ***G10K 11/002*** (2013.01); ***H04R 1/1016*** (2013.01);
(Continued)

(58) **Field of Classification Search**
CPC G10K 11/17827; G10K 11/002; H04R 1/1016; H04R 1/1083; H04R 3/002;
(Continued)

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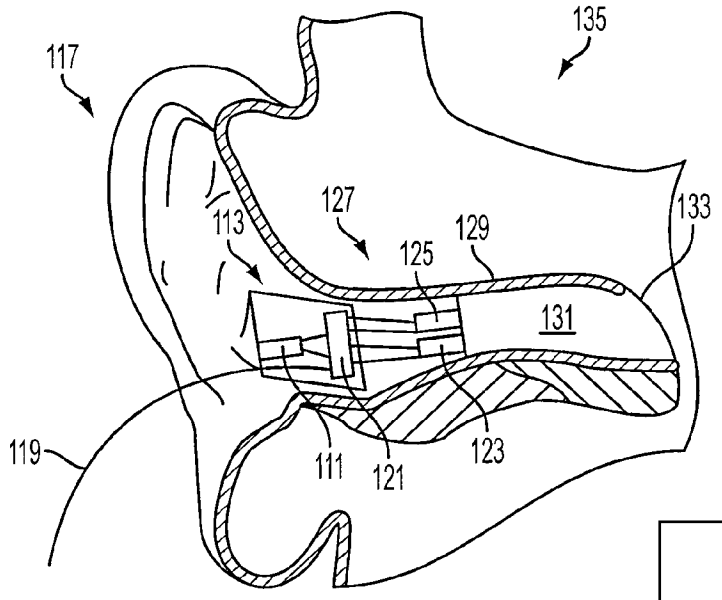
Primary Examiner — Andrew L Sniezek
(74) *Attorney, Agent, or Firm* — Akerman LLP; Peter A. Chiabotti

(57) **ABSTRACT**

Earpieces and methods for acute sound detection and reproduction are provided. A method can include measuring an external ambient sound level (xASL), monitoring a change in the xASL for detecting an acute sound, estimating a proximity of the acute sound, and upon detecting the acute sound and its proximity, reproducing the acute sound within an ear canal, where the ear canal is at least partially occluded by an earpiece. Other embodiments are disclosed.

20 Claims, 9 Drawing Sheets



Samsung v. Techiya
IPR2024-01004
Exhibit 1001

Related U.S. Application Data

No. 16/987,396, filed on Aug. 7, 2020, now Pat. No. 11,244,666, which is a continuation of application No. 16/669,490, filed on Oct. 30, 2019, now Pat. No. 10,810,989, which is a continuation of application No. 16/193,568, filed on Nov. 16, 2018, now Pat. No. 10,535,334, which is a continuation of application No. 14/574,589, filed on Dec. 18, 2014, now Pat. No. 10,134,377, which is a continuation of application No. 12/017,878, filed on Jan. 22, 2008, now Pat. No. 8,917,894.

(60) Provisional application No. 60/885,917, filed on Jan. 22, 2007.

(51) **Int. Cl.**
H04R 1/10 (2006.01)
G10K 11/00 (2006.01)
H04R 29/00 (2006.01)

(52) **U.S. Cl.**
CPC ***H04R 1/1083*** (2013.01); ***H04R 3/002*** (2013.01); ***H04R 3/005*** (2013.01); ***H04R 29/001*** (2013.01); *H04R 2225/41* (2013.01); *H04R 2460/05* (2013.01); *H04R 2499/11* (2013.01)

(58) **Field of Classification Search**
CPC .. H04R 3/005; H04R 29/001; H04R 2225/41; H04R 2460/05; H04R 2499/11
See application file for complete search history.

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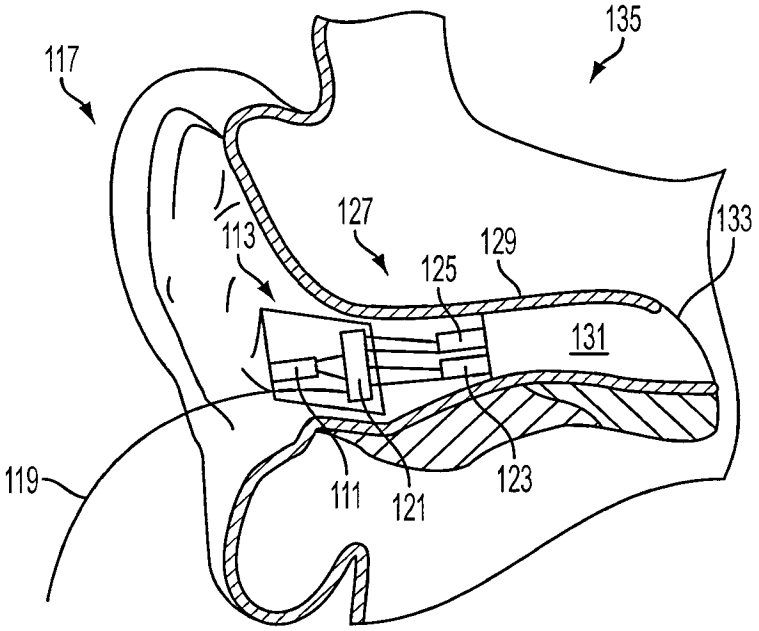

135
117
127
113
125
129
133
131
119
111
121
123

FIG. 1

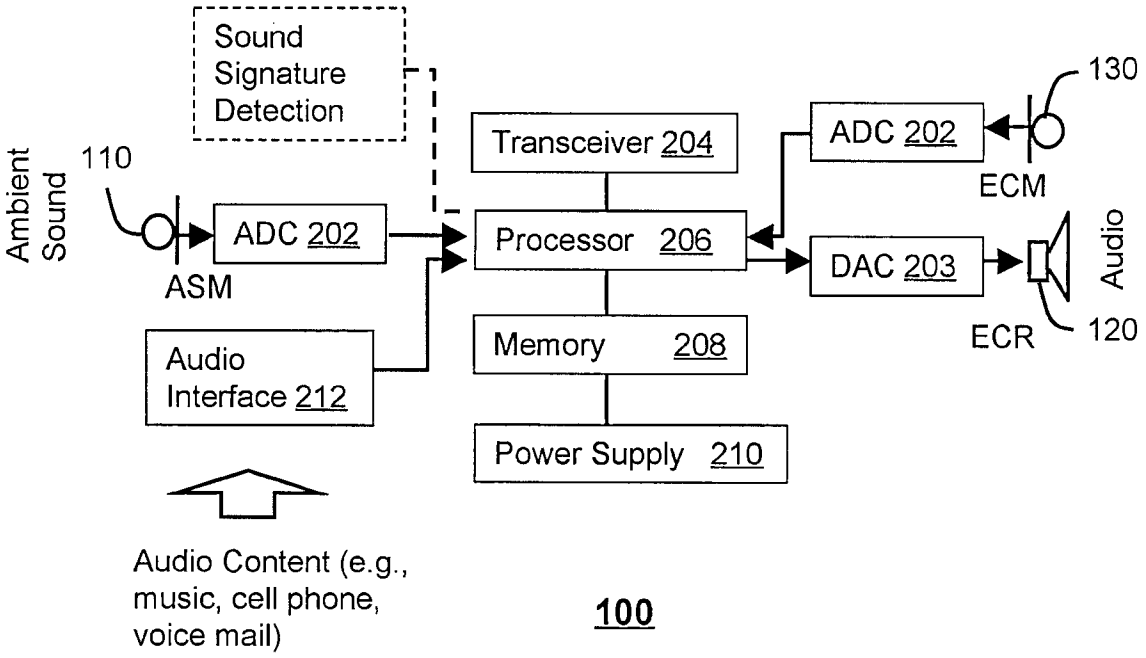
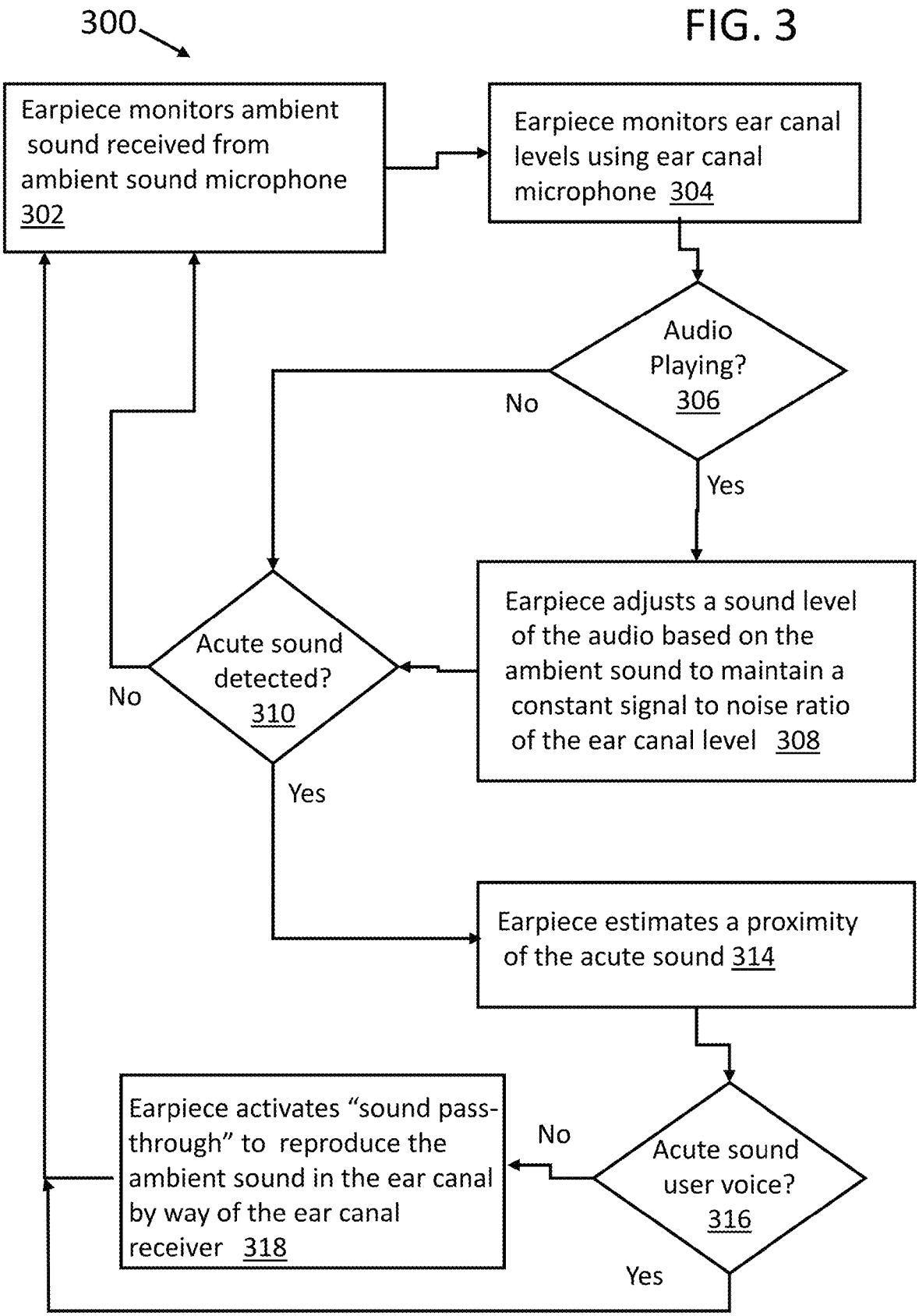

Sound Signature Detection
Ambient Sound
110
ASM
ADC 202
Audio Interface 212
Transceiver 204
Processor 206
Memory 208
Power Supply 210
ADC 202
130
ECM
DAC 203
Audio
ECR
120
Audio Content (e.g., music, cell phone, voice mail)
100

FIG. 2


300
FIG. 3
Earpiece monitors ambient sound received from ambient sound microphone 302
Earpiece monitors ear canal levels using ear canal microphone 304
Audio Playing? 306
No
Yes
Earpiece adjusts a sound level of the audio based on the ambient sound to maintain a constant signal to noise ratio of the ear canal level 308
Acute sound detected? 310
No
Yes
Earpiece estimates a proximity of the acute sound 314
Acute sound user voice? 316
No
Yes
Earpiece activates "sound pass-through" to reproduce the ambient sound in the ear canal by way of the ear canal receiver 318

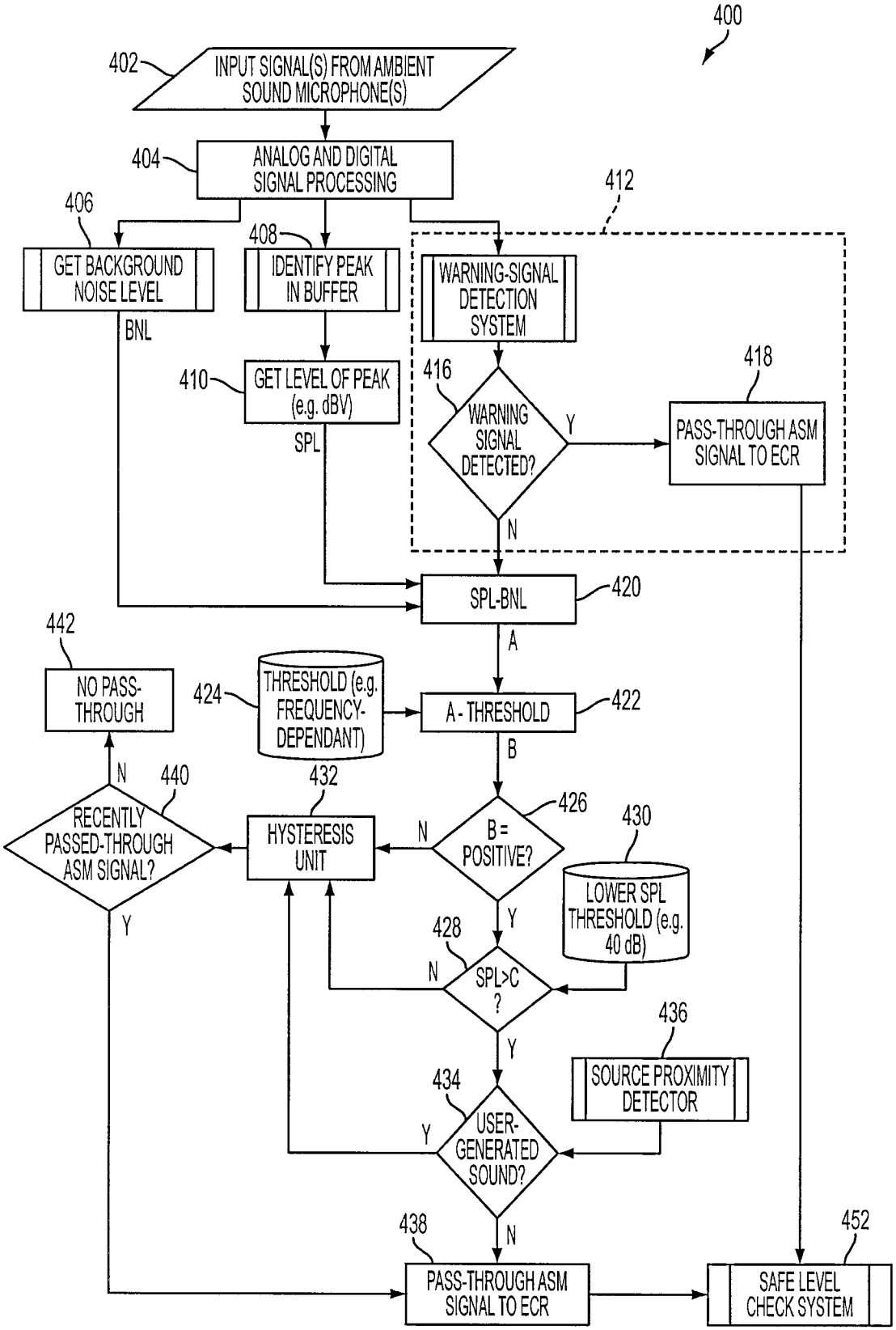

400
402
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONE(S)
404
ANALOG AND DIGITAL SIGNAL PROCESSING
406
GET BACKGROUND NOISE LEVEL
408
IDENTIFY PEAK IN BUFFER
412
WARNING-SIGNAL DETECTION SYSTEM
BNL
410
GET LEVEL OF PEAK (e.g. dBV)
416
WARNING SIGNAL DETECTED?
Y
418
PASS-THROUGH ASM SIGNAL TO ECR
SPL
N
SPL-BNL
420
A
442
NO PASS-THROUGH
424
THRESHOLD (e.g. FREQUENCY-DEPENDANT)
A - THRESHOLD
422
B
N
440
432
426
RECENTLY PASSED-THROUGH ASM SIGNAL?
HYSTERESIS UNIT
N
B = POSITIVE?
430
LOWER SPL THRESHOLD (e.g. 40 dB)
Y
Y
428
N
SPL>C ?
436
Y
SOURCE PROXIMITY DETECTOR
434
Y
USER-GENERATED SOUND?
438
N
452
PASS-THROUGH ASM SIGNAL TO ECR
SAFE LEVEL CHECK SYSTEM

FIG. 4

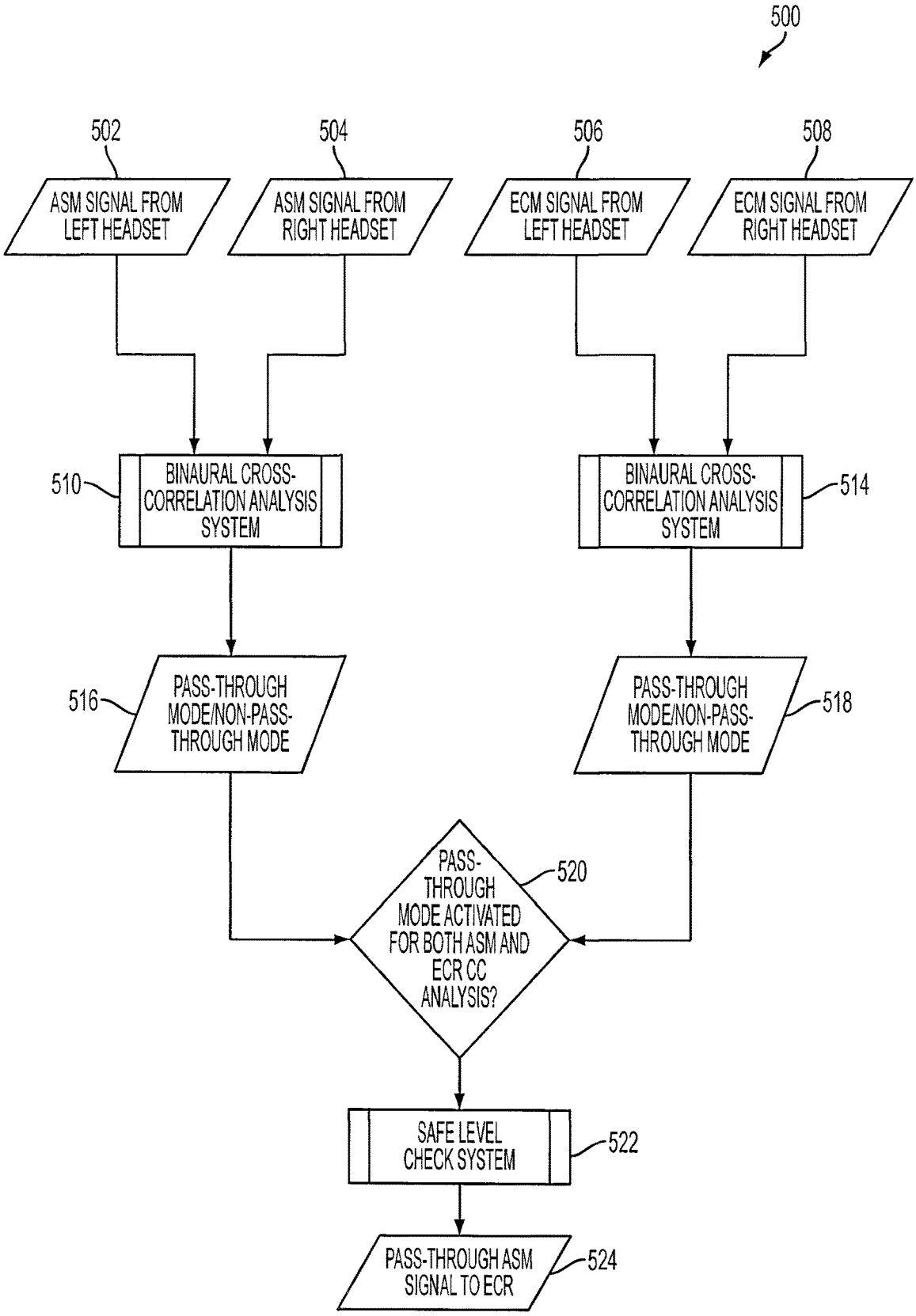

500
502
ASM SIGNAL FROM LEFT HEADSET
504
ASM SIGNAL FROM RIGHT HEADSET
506
ECM SIGNAL FROM LEFT HEADSET
508
ECM SIGNAL FROM RIGHT HEADSET
510
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
514
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
516
PASS-THROUGH MODE/NON-PASS-THROUGH MODE
518
PASS-THROUGH MODE/NON-PASS-THROUGH MODE
520
PASS-THROUGH MODE ACTIVATED FOR BOTH ASM AND ECR CC ANALYSIS?
522
SAFE LEVEL CHECK SYSTEM
524
PASS-THROUGH ASM SIGNAL TO ECR

FIG. 5

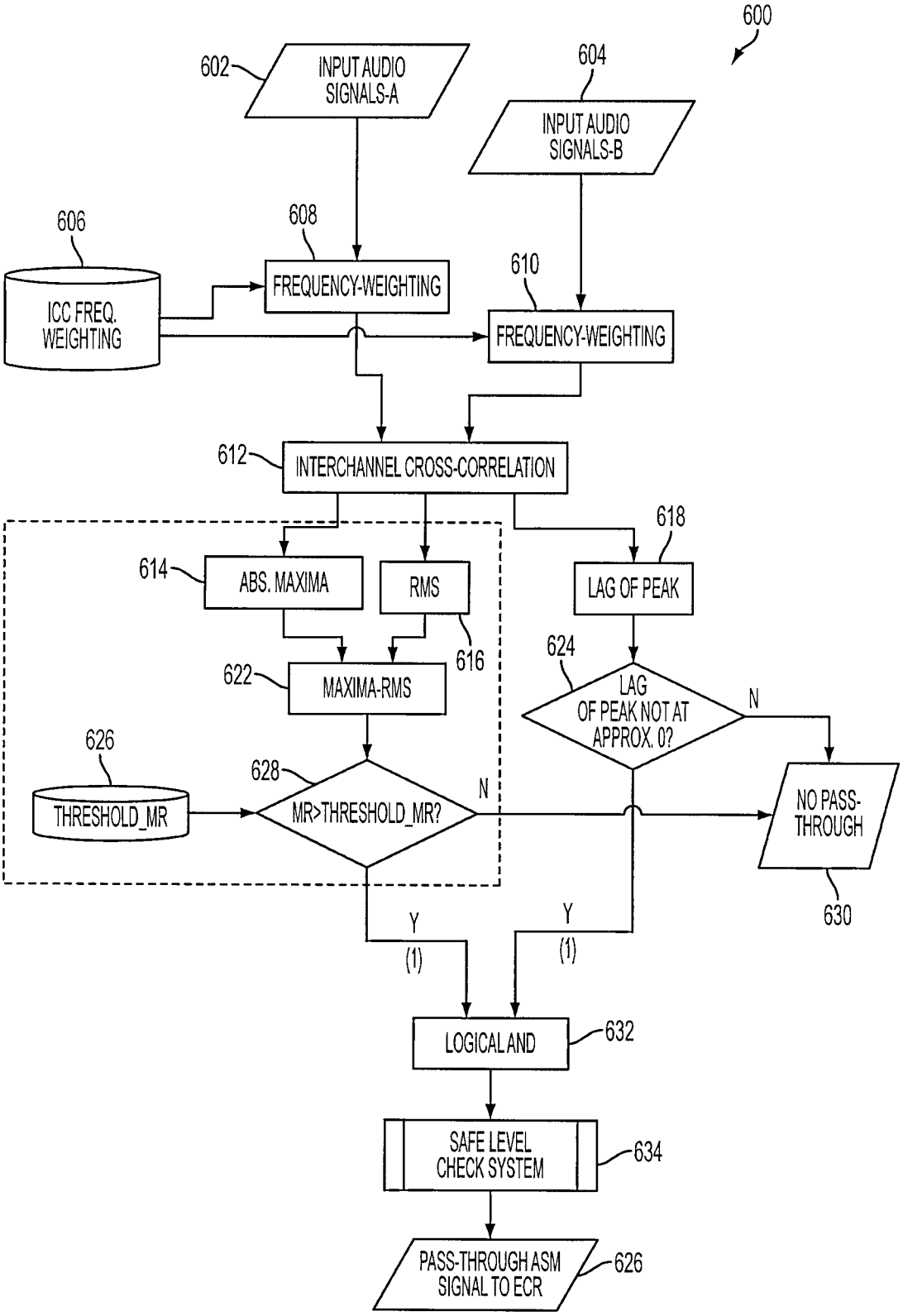

600
602
INPUT AUDIO SIGNALS-A
604
INPUT AUDIO SIGNALS-B
606
ICC FREQ. WEIGHTING
608
FREQUENCY-WEIGHTING
610
FREQUENCY-WEIGHTING
612
INTERCHANNEL CROSS-CORRELATION
614
ABS. MAXIMA
RMS
616
618
LAG OF PEAK
622
MAXIMA-RMS
624
LAG OF PEAK NOT AT APPROX. 0?
N
626
THRESHOLD_MR
628
MR>THRESHOLD_MR?
N
NO PASS-THROUGH
630
Y (1)
Y (1)
LOGICAL AND
632
SAFE LEVEL CHECK SYSTEM
634
PASS-THROUGH ASM SIGNAL TO ECR
626

FIG. 6

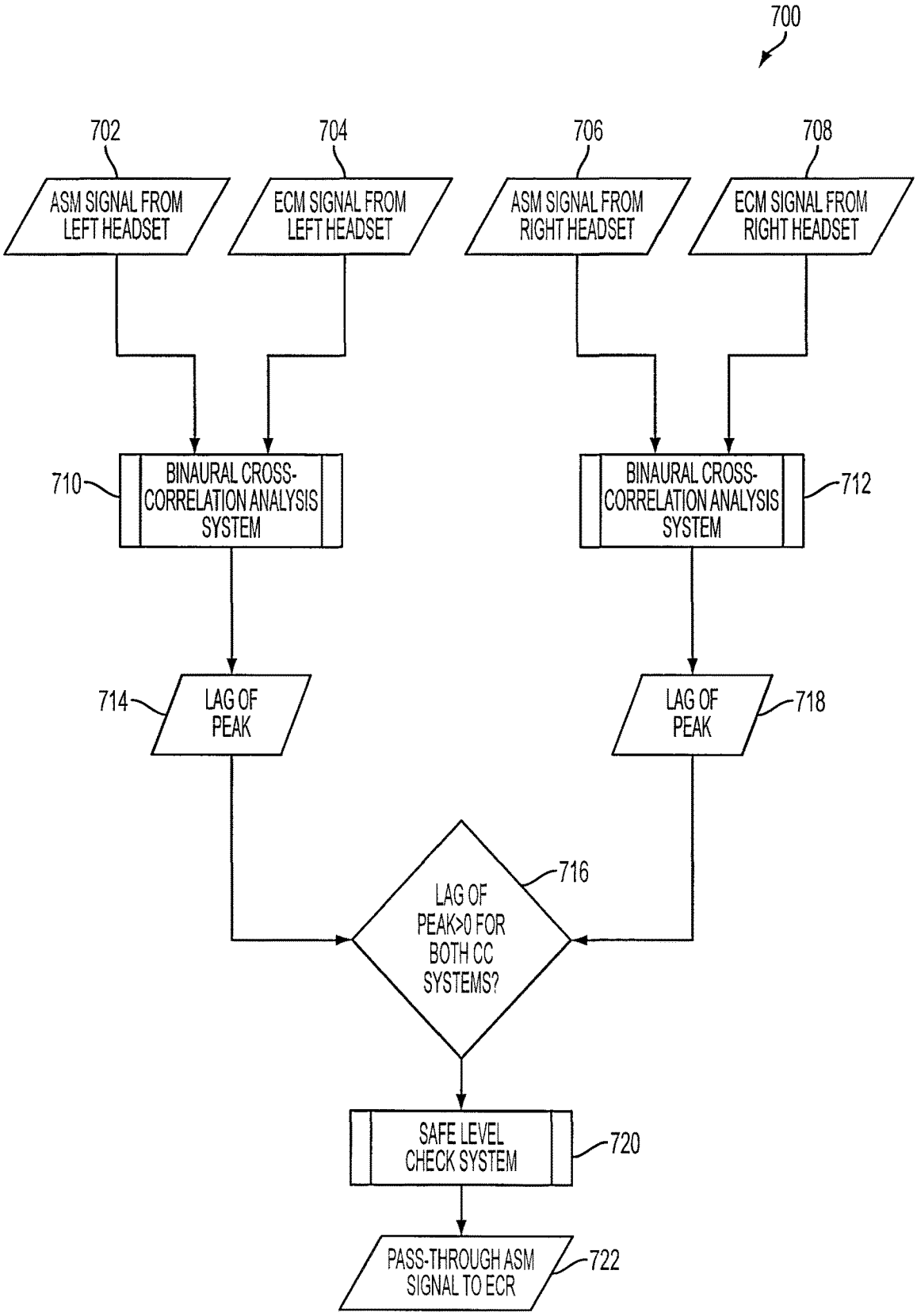

700
702
ASM SIGNAL FROM LEFT HEADSET
704
ECM SIGNAL FROM LEFT HEADSET
706
ASM SIGNAL FROM RIGHT HEADSET
708
ECM SIGNAL FROM RIGHT HEADSET
710
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
712
BINAURAL CROSS-CORRELATION ANALYSIS SYSTEM
714
LAG OF PEAK
718
LAG OF PEAK
716
LAG OF PEAK>0 FOR BOTH CC SYSTEMS?
SAFE LEVEL CHECK SYSTEM
720
PASS-THROUGH ASM SIGNAL TO ECR
722

FIG. 7

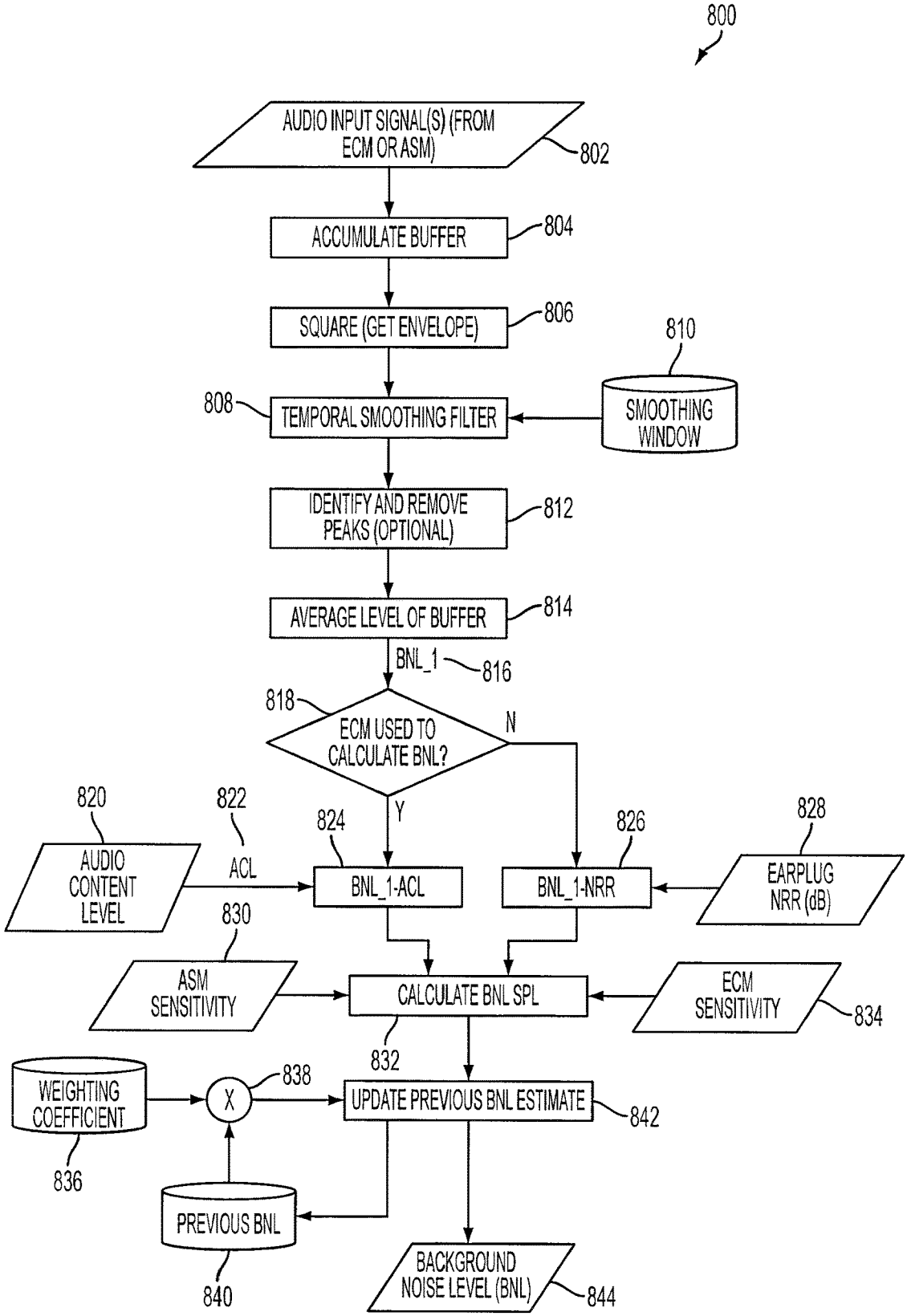

800
AUDIO INPUT SIGNAL(S) (FROM ECM OR ASM)
802
ACCUMULATE BUFFER
804
SQUARE (GET ENVELOPE)
806
810
SMOOTHING WINDOW
808
TEMPORAL SMOOTHING FILTER
IDENTIFY AND REMOVE PEAKS (OPTIONAL)
812
AVERAGE LEVEL OF BUFFER
814
BNL_1
816
818
ECM USED TO CALCULATE BNL?
N
Y
820
AUDIO CONTENT LEVEL
822
ACL
824
BNL_1-ACL
826
BNL_1-NRR
828
EARPLUG NRR (dB)
830
ASM SENSITIVITY
CALCULATE BNL SPL
ECM SENSITIVITY
834
832
WEIGHTING COEFFICIENT
X
838
UPDATE PREVIOUS BNL ESTIMATE
842
836
PREVIOUS BNL
840
BACKGROUND NOISE LEVEL (BNL)
844

FIG. 8

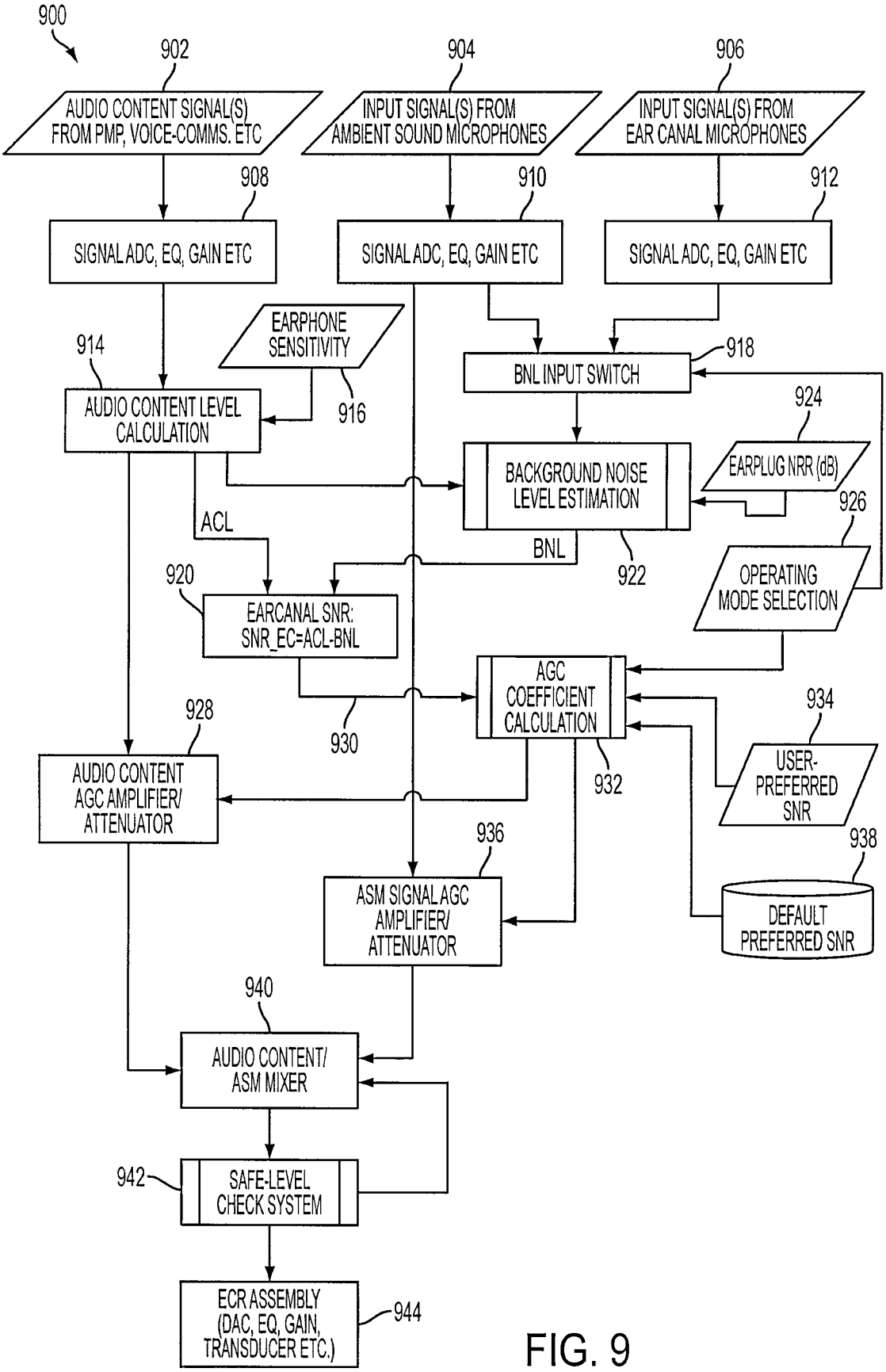

900
902
AUDIO CONTENT SIGNAL(S) FROM PMP, VOICE-COMMS. ETC
904
INPUT SIGNAL(S) FROM AMBIENT SOUND MICROPHONES
906
INPUT SIGNAL(S) FROM EAR CANAL MICROPHONES
908
SIGNAL ADC, EQ, GAIN ETC
910
SIGNAL ADC, EQ, GAIN ETC
912
SIGNAL ADC, EQ, GAIN ETC
914
AUDIO CONTENT LEVEL CALCULATION
EARPHONE SENSITIVITY
916
BNL INPUT SWITCH
918
BACKGROUND NOISE LEVEL ESTIMATION
924
EARPLUG NRR (dB)
926
OPERATING MODE SELECTION
ACL
BNL
922
920
EARCANAL SNR: SNR_EC=ACL-BNL
AGC COEFFICIENT CALCULATION
930
932
934
USER-PREFERRED SNR
928
AUDIO CONTENT AGC AMPLIFIER/ ATTENUATOR
936
ASM SIGNAL AGC AMPLIFIER/ ATTENUATOR
938
DEFAULT PREFERRED SNR
940
AUDIO CONTENT/ ASM MIXER
942
SAFE-LEVEL CHECK SYSTEM
ECR ASSEMBLY (DAC, EQ, GAIN, TRANSDUCER ETC.)
944

FIG. 9

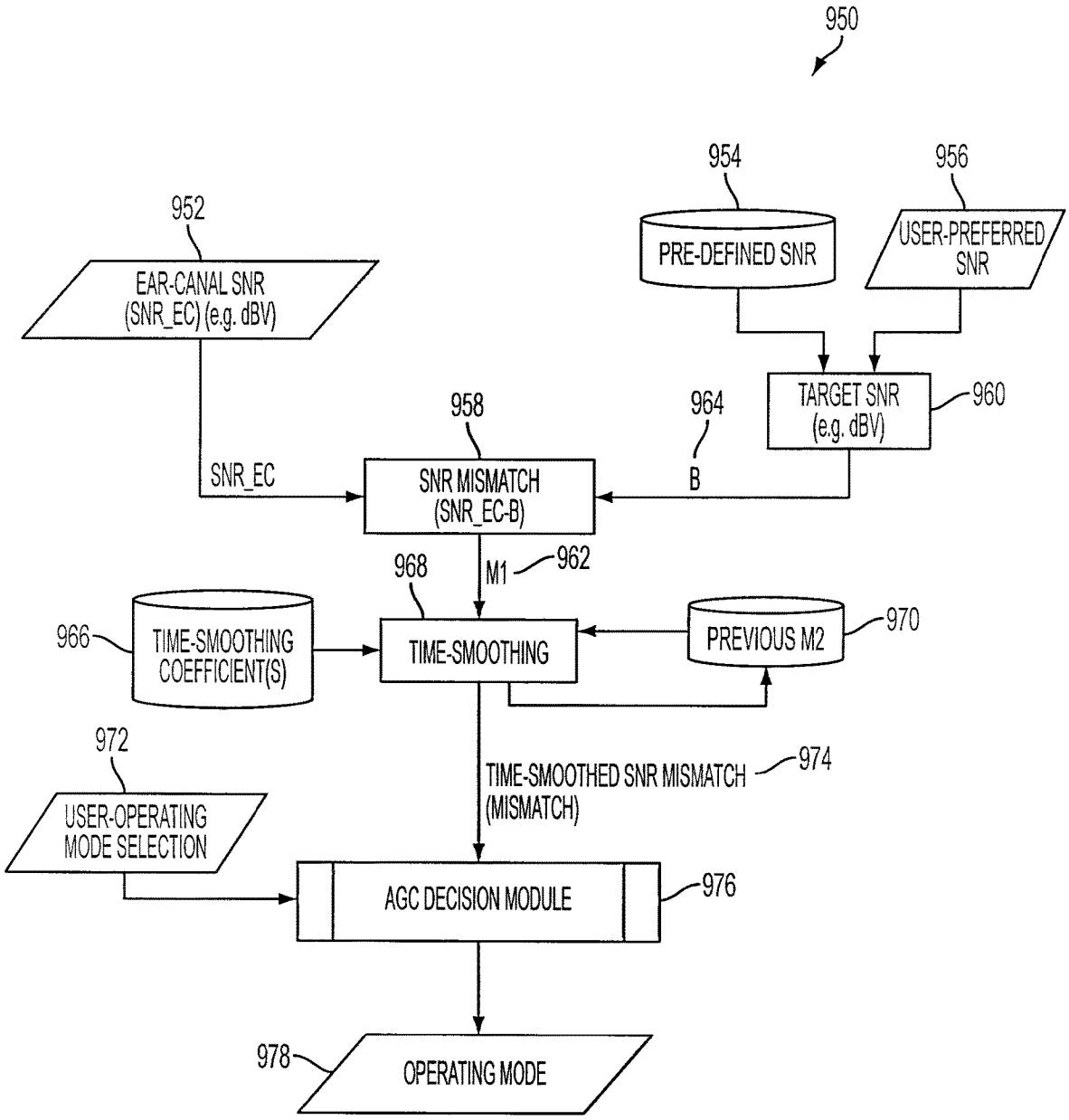

950
954
956
952
PRE-DEFINED SNR
USER-PREFERRED SNR
EAR-CANAL SNR (SNR_EC) (e.g. dBV)
TARGET SNR (e.g. dBV)
960
958
964
SNR_EC
SNR MISMATCH (SNR_EC-B)
B
968
M1
962
966
TIME-SMOOTHING COEFFICIENT(S)
TIME-SMOOTHING
PREVIOUS M2
970
972
TIME-SMOOTHED SNR MISMATCH (MISMATCH)
974
USER-OPERATING MODE SELECTION
AGC DECISION MODULE
976
978
OPERATING MODE

FIG. 10

METHOD AND DEVICE FOR ACUTE SOUND DETECTION AND REPRODUCTION

CROSS-REFERENCE TO RELATED APPLICATION(S)

This is a continuation of and claims priority to U.S. patent application Ser. No. 17/321,892, filed 17 May 2021, which is a continuation of and claims priority to U.S. patent application Ser. No. 16/987,396, filed 7 Aug. 2020, which is a continuation of and claims priority to U.S. patent application Ser. No. 16/669,490, filed 30 Oct. 2019, which is a continuation of and claims priority to U.S. patent application Ser. No. 16/193,568, filed 16 Nov. 2018, now U.S. Pat. No. 10,535,334, which is a continuation of and claims priority to U.S. patent application Ser. No. 14/574,589, filed on Dec. 18, 2014, now U.S. Pat. No. 10,134,377, which claims priority to and is a continuation of U.S. patent application Ser. No. 12/017,878, filed on Jan. 22, 2008, now U.S. Pat. No. 8,917,894, which claims the benefit of U.S. Provisional Patent Application Ser. No. 60/885,917, filed on Jan. 22, 2007, all of which are herein incorporated by reference in their entireties.

FIELD OF THE INVENTION

The present invention relates to a device that monitors sound directed to an occluded ear, and more particularly, though not exclusively, to an earpiece and method of operating an earpiece that detects acute sounds and allows the acute sounds to be reproduced in an ear canal of the occluded ear.

BACKGROUND

Since the advent of industrialization over two centuries ago, the human auditory system has been increasingly stressed to tolerate high noise levels to which it had hitherto been unexposed. Recently, human knowledge of the causes of hearing damage have been researched intensively and models for predicting hearing loss have been developed and verified with empirical data from decades of scientific research. Yet it can be strongly argued that the danger of permanent hearing damage is more present in our daily lives than ever, and that sound levels from personal audio systems in particular (i.e. from portable audio devices), live sound events, and the urban environment are a ubiquitous threat to healthy auditory functioning across the global population.

Environmental noise is constantly presented in industrialized societies given the ubiquity of external sound intrusions. Examples include people talking on their cell phones, blaring music in health clubs, or the constant hum of air conditioning systems in schools and office buildings.

Excess noise exposure can also induce auditory fatigue, possibly comprising a person's listening abilities. On a daily basis, people are exposed to various environmental sounds and noises within their environment, such as the sounds from traffic, construction, and industry.

To combat the undesired cacophony of annoying sounds, people are arming themselves with portable audio playback devices to drown out intrusive noise. The majority of devices providing the person with audio content do so using insert (or in-ear) earbuds. These earbuds deliver sound directly to the ear canal at high sound levels over the background noise even though the earbuds generally provide little to no ambient sound isolation. Moreover, when people wear earbuds (or headphones) to listen to music, or engage

in a call using a telephone, they can effectively impair their auditory judgment and their ability to discriminate between sounds. With such devices, the person is immersed in the audio experience and generally less likely to hear warning sounds within their environment. In some cases, the user may even turn up the volume to hear their personal audio over environmental noises. It also puts them at high sound exposure risk which can potentially cause long term hearing damage.

With earbuds, personal audio reproduction levels can reach in excess of 100 dB. This is enough to exceed recommended daily sound exposure levels in less than a minute and to cause permanent acoustic trauma. Furthermore, rising population densities have continually increased sound levels in society. According to researchers, 40% of the European community is continuously exposed to transportation noise of 55 dBA and 20% are exposed to greater than 65 dBA. This level of 65 dBA is considered by the World Health Organization to be intrusive or annoying, and as mentioned, can lead to users of personal audio devices increasing reproduction levels to compensate for ambient noise.

A need therefore exists for enhancing the user's ability to listen in the environment without harming his or her hearing faculties.

SUMMARY

Embodiments in accordance with the present invention provide a method and device for acute sound detection and reproduction.

In a first embodiment, an earpiece can include an Ambient Sound Microphone (ASM) to capture ambient sound, at least one Ear Canal Receiver (ECR) to deliver audio to an ear canal; and a processor operatively coupled to the ASM and the at least one ECR. The processor can monitor a change in the ambient sound level to detect an acute sound from the change. The acute sound can be reproduced within the ear canal via the ECR responsive to detecting the acute sound.

The processor can pass (transmit) sound from the ASM directly to the ECR to produce sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal. In one arrangement, the processor can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal. In one arrangement, the processor can measure an external ambient sound level (xASL) of the ambient sound with the ASM and subtract an attenuation level of the earpiece from the xASL to estimate the internal ambient sound level (iASL) within the ear canal.

The earpiece can further include an Ear Canal Microphone (ECM) to measure an ear canal sound level (ECL) within the ear canal. In this configuration, the processor can estimate the internal ambient sound level (iASL) within the ear canal by subtracting an estimated audio content sound level (ACL) from the ECL. For instance, the processor can measure a voltage level of the audio content sent to the ECR, and apply a transfer function of the ECR to convert the voltage level to the ACL. The processor can be located external to the earpiece on a portable computing device.

In a second embodiment, an earpiece can comprise an Ambient Sound Microphone (ASM) to capture ambient sound, at least one Ear Canal Receiver (ECR) to deliver audio to an ear canal, an audio interface operatively coupled to the processor to receive audio content, and a processor

operatively coupled to the ASM and the at least one ECR. The processor can monitor a change in the ambient sound level to detect an acute sound from the change, adjust an audio content level (ACL) of the audio content delivered to the ear canal, and reproduce the acute sound within the ear canal via the ECR responsive to detecting the acute sound and based on the ACL.

The audio interface can receive the audio content from at least one among a portable music player, a cell phone, and a portable communication device. During operation, the processor can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal. In one arrangement, the processor can mute the audio content and pass the acute sound to the ECR for reproducing the acute sound within the ear canal. In another arrangement, the processor can amplify the acute sound with respect to the audio content level (ACL).

In a third embodiment, a method for acute sound detection and reproduction can include the steps of measuring an ambient sound level (xASL) of ambient sound external to an ear canal at least partially occluded by the earpiece, monitoring a change in the xASL for detecting an acute sound, and reproducing the acute sound within the ear canal responsive to detecting the acute sound. The reproducing can include enhancing the acute sound over the ambient sound. The step of reproducing can produce sound within the ear canal at a same sound pressure level (SPL) as the acute sound measured at an entrance to the ear canal.

The method can further include receiving audio content from an audio interface that is directed to the ear canal, and maintaining an approximately constant ratio between a level of the audio content (ACL) and a level of an internal ambient sound level (iASL) measured within the ear canal. The ACL can be determined by measuring a voltage level of the audio content sent to the ECR, and applying a transfer function of the ECR to convert the voltage level to the ACL. The method can further include measuring an Ear Canal Level (ECL) within the ear canal, and subtracting the ACL from the ECL to estimate the iASL. The iASL can be estimated by subtracting an attenuation level of the earpiece from the xASL.

In a fourth embodiment, a method for acute sound detection and reproduction suitable for use with an earpiece can include the steps of measuring an external ambient sound level (xASL) in an ear canal at least partially occluded by the earpiece, monitoring a change in the xASL for detecting an acute sound, estimating a proximity of the acute sound, and reproducing the acute sound within the ear canal responsive to detecting the acute sound based on the proximity. The step of estimating a proximity can include performing a cross correlation analysis between at least two microphones, identifying a peak in the cross correlation and an associated time lag, and determining the direction from the associated time lag. The method can further include identifying whether the acute sound is a vocal signal produced by a user operating the earpiece or a sound source external from the user.

In a fifth embodiment, a method for acute sound detection and reproduction suitable for use with an earpiece can include measuring an external ambient sound level (xASL) due to ambient sound outside of an ear canal at least partially occluded by the earpiece, measuring an internal ambient sound level (iASL) due to residual ambient sound within the ear canal at least partially occluded by the earpiece, monitoring a high frequency change between the xASL and the iASL with respect to a low frequency change between the xASL and the iASL for detecting an acute sound, and reproducing the xASL within the ear canal responsive to detecting the high frequency change. The method can further include determining a proximity of a sound source producing the acute sound.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;

FIG. **2** is a block diagram of the earpiece in accordance with an exemplary embodiment;

FIG. **3** is a flowchart of a method for acute sound detection in accordance with an exemplary embodiment;

FIG. **4** is a more detailed approach to the method of FIG. **3** in accordance with an exemplary embodiment;

FIG. **5** is a flowchart of a method for acute sound source proximity in accordance with an exemplary embodiment;

FIG. **6** is a flowchart of a method for binaural analysis in accordance with an exemplary embodiment;

FIG. **7** is a flowchart of a method for logic control in accordance with an exemplary embodiment;

FIG. **8** is a flowchart of a method for estimating background noise level in accordance with an exemplary embodiment;

FIG. **9** is a flowchart of a method for maintaining constant audio content level (ACL) to internal ambient sound level (iASL) in accordance with an exemplary embodiment; and

FIG. **10** is a flowchart of a method for adjusting audio content gain in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

The following description of at least one exemplary embodiment is merely illustrative in nature and is in no way intended to limit the invention, its application, or uses.

Processes, techniques, apparatus, and materials as known by one of ordinary skill in the relevant art may not be discussed in detail but are intended to be part of the enabling description where appropriate, for example the fabrication and use of transducers. Additionally in at least one exemplary embodiment the sampling rate of the transducers can be varied to pick up pulses of sound, for example less than 50 milliseconds.

In all of the examples illustrated and discussed herein, any specific values, for example the sound pressure level change, should be interpreted to be illustrative only and non-limiting. Thus, other examples of the exemplary embodiments could have different values.

Note that similar reference numerals and letters refer to similar items in the following figures, and thus once an item is defined in one figure, it may not be discussed for following figures.

Note that herein when referring to correcting or preventing an error or damage (e.g., hearing damage), a reduction of the damage or error and/or a correction of the damage or error are intended.

At least one exemplary embodiment of the invention is directed to an earpiece for ambient sound monitoring and warning detection. Reference is made to FIG. **1** in which an earpiece device, generally indicated as earpiece **100**, is constructed in accordance with at least one exemplary embodiment of the invention. As illustrated, earpiece **100** depicts an electro-acoustical assembly **113** for an in-the-ear acoustic assembly, as it would typically be placed in the ear canal **131** of a user **135**. The earpiece **100** can be an in the ear earpiece, behind the ear earpiece, receiver in the ear, open-fit device, or any other suitable earpiece type. The earpiece **100** can be partially or fully occluded in the ear

canal, and is suitable for use with users having healthy or abnormal auditory functioning.

Earpiece **100** includes an Ambient Sound Microphone (ASM) Ill to capture ambient sound, an Ear Canal Receiver (ECR) **125** to deliver audio to an ear canal **131**, and an Ear Canal Microphone (ECM) **123** to assess a sound exposure level within the ear canal. The earpiece **100** can partially or fully occlude the ear canal **131** to provide various degrees of acoustic isolation. The assembly is designed to be inserted into the user's ear canal **131**, and to form an acoustic seal with the walls **129** of the ear canal at a location **127** between the entrance **117** to the ear canal and the tympanic membrane (or ear drum) **133**. Such a seal is typically achieved by means of a soft and compliant housing of assembly **113**. Such a seal is pertinent to the performance of the system in that it creates a closed cavity **131** of approximately 5 cc between the in-ear assembly **113** and the tympanic membrane **133**. As a result of this seal, the ECR (speaker) **125** is able to generate a full range bass response when reproducing sounds for the user. This seal also serves to significantly reduce the sound pressure level at the user's eardrum **133** resulting from the sound field at the entrance to the ear canal. This seal is also the basis for the sound isolating performance of the electro-acoustic assembly **113**.

Located adjacent to the ECR **125**, is the ECM **123**, which is acoustically coupled to the (closed) ear canal cavity **131**. One of its functions is that of measuring the sound pressure level in the ear canal cavity **131** as a part of testing the hearing acuity of the user as well as confirming the integrity of the acoustic seal and the working condition of itself and the ECR. The ASM **111** is housed in an assembly **113** and monitors sound pressure at the entrance to the occluded or partially occluded ear canal. All transducers shown can receive or transmit audio signals to a processor **121** that undertakes audio signal processing and provides a transceiver for audio via the wired or wireless communication path **119**.

Referring to FIG. **2**, a block diagram of the earpiece **100** in accordance with an exemplary embodiment is shown. As illustrated, the earpiece **100** can include a processor **206** operatively coupled to the ASM **111**, ECR **125**, and ECM **123** via one or more Analog to Digital Converters (ADC) **202** and Digital to Analog Converters (DAC) **203**. The processor **206** can monitor the ambient sound captured by the ASM **111** for acute sounds in the environment, such as an abrupt high energy sound corresponding to the on-set of a warning sound (e.g., bell, emergency vehicle, security system, etc.), siren (e.g., police car, ambulance, etc.), voice (e.g., "help", "stop", "police", etc.), or specific noise type (e.g., breaking glass, gunshot, etc.). The processor **206** can utilize computing technologies such as a microprocessor, Application Specific Integrated Chip (ASIC), and/or digital signal processor (DSP) with associated storage memory **208** such as Flash, ROM, RAM, SRAM, DRAM or other like technologies for controlling operations of the earpiece device **100**. The memory **208** can store program instructions for execution on the processor **206** as well as captured audio processing data.

The earpiece **100** can include an audio interface **212** operatively coupled to the processor **206** to receive audio content, for example from a media player or cell phone, and deliver the audio content to the processor **206**. The processor **206** responsive to detecting acute sounds can adjust the audio content and pass the acute sounds directly to the ear canal. For instance, the processor can lower a volume of the audio content responsive to detecting an acute sound for transmitting the acute sound to the ear canal. The processor **206** can also actively monitor the sound exposure level inside the ear canal and adjust the audio to within a safe and subjectively optimized listening level range.

The earpiece **100** can further include a transceiver **204** that can support singly or in combination any number of wireless access technologies including without limitation Bluetooth™, Wireless Fidelity (WiFi), Worldwide Interoperability for Microwave Access (WiMAX), and/or other short or long range communication protocols. The transceiver **204** can also provide support for dynamic downloading over-the-air to the earpiece **100**. It should be noted also that next generation access technologies can also be applied to the present disclosure.

The power supply **210** can utilize common power management technologies such as replaceable batteries, supply regulation technologies, and charging system technologies for supplying energy to the components of the earpiece **100** and to facilitate portable applications. A motor (not shown) can be a single supply motor driver coupled to the power supply **210** to improve sensory input via haptic vibration. As an example, the processor **206** can direct the motor to vibrate responsive to an action, such as a detection of a warning sound or an incoming voice call.

The earpiece **100** can further represent a single operational device or a family of devices configured in a master-slave arrangement, for example, a mobile device and an earpiece. In the latter embodiment, the components of the earpiece **100** can be reused in different form factors for the master and slave devices.

FIG. **3** is a flowchart of a method **300** for acute sound detection and reproduction in accordance with an exemplary embodiment. The method **300** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **300**, reference will be made to components of FIG. **2**, although it is understood that the method **300** can be implemented in any other manner using other suitable components. The method **300** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

The method **300** can start in a state wherein the earpiece **100** has been inserted and powered on. As shown in step **302**, the earpiece **100** can monitor the environment for ambient sounds received at the ASM **111**. Ambient sounds correspond to sounds within the environment such as the sound of traffic noise, street noise, conversation babble, or any other acoustic sound. Ambient sounds can also correspond to industrial sounds present in an industrial setting, such as factory noise, lifting vehicles, automobiles, and robots to name a few.

Although the earpiece **100** when inserted in the ear can partially occlude the ear canal, the earpiece **100** may not completely attenuate the ambient sound. During the monitoring of ambient sounds in the environment, the earpiece **100** also monitors ear canal levels via the ECM **123** as shown in step **304**. The passive aspect of the physical earpiece **100**, due to the mechanical and sealing properties, can provide upwards of a 22-26 dB noise reduction. However, portions of ambient sounds higher than 26 dB can still pass through the earpiece **100** into the ear canal. For instance, high energy low frequency sounds are not completely attenuated. Accordingly, residual sound may be resident in the ear canal and heard by the user.

Sound within the ear canal **131** can also be provided via the audio interface **212**. The audio interface **212** can receive the audio content from at least one among a portable music player, a cell phone, and a portable communication device. The audio interface **212** responsive to user input can direct

sound to the ECR **125**. For instance, a user can elect to play music through the earpiece **100** which can be audibly presented to the ear canal **131** for listening. The user can also elect to receive voice communications (e.g., cell phone, voice mail, messaging) via the earpiece **100**. For instance, the user can receive audio content for voice mail or a phone call directed to the ear canal via the ECR **125**. As shown in step **304**, the earpiece **100** can monitor ear canal levels due to ambient sound and user selected sound via the ECM **123**.

If at step **306**, audio is playing (e.g., music, cell phone, etc.), the earpiece **100** adjusts a sound level of the audio based on the ambient sound to maintain a constant signal to noise ratio with respect to the ear canal level at step **308**. For instance, the processor **206** can selectively amplify or attenuate audio content received from the audio interface **212** before it is delivered to the ECR **125**. The processor **206** estimates a background noise level from the ambient sound received at the ASM **111**, and adjusts the audio level of delivered audio content (e.g., music, cell phone audio) to maintain a constant signal (e.g., audio content) to noise level (e.g., ambient sound). By way of example, if the background noise level increases due to traffic sounds, the earpiece **100** automatically increases the volume of the audio content. Similarly, if the background noise level decreases, the earpiece **100** automatically decreases the volume of the audio content. The processor **206** can track variations on the ambient sound level to adjust the audio content level.

If at step **310**, an acute sound is detected within the ambient sound, the earpiece **100** activates "sound pass-through" to reproduce the ambient sound in the ear canal by way of the ECR **125**. The processor **206** permits the ambient sound to pass through the ECR **125** to the ear canal **131** directly for example by replicating the ambient sound external to the ear canal within the ear canal. This is important if the acute sound corresponds to an on-set for a warning sound such as a bell, a car, or an object. In such regard, the ambient sound containing the acute sound is presented directly to the ear canal in an original form. Although, the earpiece **100** inherently provides attenuation due to the physical and mechanical aspects of the earpiece and its sealing properties, the processor **206** can reproduce the ambient sound within the ear canal **131** at an original amplitude level and frequency content to provide "transparency". For instance, the processor **206** measures and applies a transfer function of the ear canal to the passed ambient sound signal to provide an accurate reproduction of the ambient sound within the ear canal.

In one embodiment, the earpiece **100** looks for temporal and spectral characteristics in the ambient sound for detecting acute sounds. For instance, as will be explained ahead, the processor **206** looks for an abrupt change in the Sound Pressure Level (SPL) of an ambient sound across a small time period. The processor **206** can also detect abrupt magnitude changes across frequency sub-bands (e.g. filter-bank, FFT, etc.). Notably, the processor **206** can search for on-sets (e.g., fast rising amplitude wave-front) of an acute sound or other abrupt feature characteristics without initially attempting to initially identify or recognize the sound source. That is, the processor **206** is actively listening for a presence of acute sounds before identifying the type of sound source.

Even though the earplug inherently provides a certain attenuation level (e.g., noise reduction rating), the processor **206** in view of the ear canal level (ECL) and ambient sound level (ASL) can reproduce the ambient sound within the ear canal to allow the user to make an informed decision with regard to the acute sound. The ECL corresponds to all sounds within the ear canal and includes the internal ambient sound level (iASL) resulting from residual ambient sounds through the earpiece and the audio content level (ACL) resulting from the audio delivered via the audio interface **212**. Briefly, xASL is the external ambient sound external to the ear canal and the earpiece (e.g., ambient sound outside the ear canal). iASL is the residual ambient sound that remains internal in the ear canal. The following equations describe the relationship among terms:

$$iASL = xASL - NRR \quad \text{(EQ 1)}$$

$$iASL = ECL - ACL \quad \text{(EQ 2)}$$

As EQ 1 shows, the iASL is the difference between the external ambient sound (xASL) and the attenuation of the earpiece (Noise Reduction Rating) due to the physical and sealing properties of the earpiece. The processor **206** can measure an external ambient sound level (xASL) of the ambient sound with the ASM **111** and subtracts an attenuation level of the earpiece (NRR) from the xASL to estimate the internal ambient sound level (iASL) within the ear canal.

EQ 2 is an alternate, or supplemental, method for calculating the iASL as the difference between the ECL and the Audio Content Level (ACL). By way of the ECM **123**, the processor **206** can estimate an internal ambient sound level (iASL) within the ear canal by subtracting the estimated audio content sound level (ACL) from the ECL. The processor **206** measures a voltage level of the audio content sent to the ECR **125**, and applies a transfer function of the ECR **125** to convert the voltage level to the ACL.

The processor **206** evaluates the equations above to pass sound from the ASM **111** directly to the ECR **125** to produce sound within the ear canal at a same sound pressure level (SPL) and frequency representation as the acute sound measured at an entrance to the ear canal. Further, the processor **206** can maintain an approximately constant ratio between an audio content level (ACL) and an internal ambient sound level (iASL) measured within the ear canal.

At step **314**, the earpiece **100** can estimate a proximity of the acute sound. For instance, as will be shown ahead, the processor **206** can perform a correlation analysis on at least two microphones to determine whether the sound source is internal (e.g., the user) or external (e.g., an object other than the user). At step **316**, the earpiece **100** determines whether it is the user's voice that generates the acute sound when the user speaks, or whether it is an external sound such as a vehicle approaching the user. If at step **316**, the processor **206** determines that the acute sound is a result of the user speaking, the processor **206** does not activate a pass-through mode, since this is not considered an external warning sound. The pass-through mode permits ambient sound detected at the ASM **111** to be transmitted directly to the ear canal. If however, the acute sound corresponds to an external sound source, such as an on-set of a warning sound, the earpiece at step **318** activates "sound pass-through" to reproduce the ambient sound in the ear canal by way of the ECR **125**. The earpiece **100** can also present an audible notification to the user indicating that an external sound source generating the acute sound has been detected. The method **300** can proceed back to step **302** to continually monitor for acute sounds in the environment.

FIG. **4** is a detailed approach to the method **400** of FIG. **3** for an Acute-Sound Pass-Through System (ACPTS) in accordance with an exemplary embodiment. The method **400** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **400**, reference will be made to com-

ponents of FIG. **2**, although it is understood that the method **400** can be implemented in any other manner using other suitable components. The method **400** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

At step **402**, the earpiece **100** captures ambient sound signals from the ASM **111**. At step **404**, the processor **206** applies analog and discrete time signal processing to condition and compensate the ambient sound signal for the ASM **111** transducer. At step **406**, the processor **206** estimates a background noise level (BNL) as will be discussed ahead. At step **408**, the processor **206** identifies at least one peak in a data buffer storing a portion of the ambient sound signal. The processor **206** at step **410** gets a level of the peak (e.g., dBV). Block **412** presents a method for warning signal detection (e.g. car horns, klaxons). When a warning signal is detected at step **416**, the processor **206** invokes at step **418** a pass-through mode whereby the ASM signal is reproduced with the ECR **125**. Upon activating pass-through mode, the processor **206** can perform a safe level check at step **452**. If a warning signal is not detected, the method **400** proceeds to step **420**.

At step **420**, the processor **206** subtracts the estimated BNL from an SPL of the ambient sound signal to produce signal "A". A high energy level transient signal is indicative of an acute sound. At step **422**, a frequency dependent threshold is retrieved at step **424**, and subtracted from signal "A", as shown in step **422** to produce signal "B". At step **426**, the processor **206** determines if signal "B" is positive. If not, the processor **206** performs a hysteresis to determine if the acute sound has already been detected. If not, the processor at step **428** determines if an SPL of the ambient sound is greater than a signal "C" (e.g. threshold). If the SPL is greater than signal "C", the earpiece generates a user generated sound at step **434**. The signal "C" is used to ensure that the SPL between the signal and background noise is positive and greater than a predetermined amount. For instance, a low SPL threshold (e.g., "C" 40 dB) can be used as shown in step **430**, although it can adapt to different environmental conditions. The low SPL threshold provides an absolute measure to the SPL difference. At step **436**, a proximity of a sound source generating the acute sound can be estimated as will be discussed ahead. The method **400** can continue to step **432**.

Briefly, if a transient, high-level sound (or acute sound) is detected in the ambient sound signal (ASM input signal), then it is converted to a level, and its magnitude compared with the BNL is calculated. The magnitude of this resulting difference (signal "A") is compared with the threshold (see step **422**). If the value is positive, and the level of the transient is greater than a predefined threshold (see step **428**), the processor **206** invokes the optional Source Proximity Detector at step **436**, which determines if the acute sound was created by the User's voice (i.e., a user generated sound). If a user-generated sound is NOT detected, then Pass-through operation at step **438** is invoked, whereby the ambient sound signal is reproduced with the ECR **125**. If the difference signal at step **428** is not positive, or the level of the identified transient is too low, then the hysteresis is invoked at step **432**. The processor **206** decides if the pass-through was recently used at step **440** (e.g. in the last 10 ms). If pass-through mode was recently activated, then processor **206** invokes the pass-through system at step **438**; otherwise there is no pass-through of the ASM signal to the ECR as shown at step **442**. Upon activating pass-through mode, the processor **206** can perform a safe level check at step **452**.

FIG. **5** is a flowchart of a method **500** for acute sound source proximity. The method **500** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **500**, reference will be made to components of FIG. **2**, although it is understood that the method **500** can be implemented in any other manner using other suitable components. The method **500** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery device.

Briefly, FIG. **5** describes a method **500** for Source Proximity Detection (SPD) to determine if the Acute sound detected was created by the User's voice operating the earpiece **100**. The SPD method **500** uses as its inputs the external ambient sound signals from left and right electroacoustic earpiece **100** assemblies (e.g., a headphone). In some embodiments the SPD method **500** employs Ear Canal Microphone (ECM) signals from left and right earpiece **100** assemblies placed on left and right ears respectively. The processor **206** performs an electronic cross-correlation between the external ambient sound signals to determine a Pass-through or Non Pass-through operating mode. In the described embodiment whereby the cross-correlation of both the ASM and ECM signals is involved, a pass-through mode is invoked when the cross-correlation analysis for both the left and right earpiece **100** assemblies return a "Pass-through" operating mode, as determined by a logical AND unit.

For instance, at step **502** a left ASM signal from a left headset incorporating the earpiece **100** assembles is received. Simultaneously, at step **504** a right ASM signal from a right headset is received. At step **510**, the processor **206** performs a binaural cross correlation on the left ASM signal and the right ASM signal to evaluate a pass through mode **516**. At step **506** a left ECM signal from the left headset is received. At step **508**, a right ECM signal from the right headset is received. At step **514**, the processor **206** performs a binaural cross correlation on the left ECM signal and the right ECM signal to evaluate a pass through mode **518**. A pass through mode **524** is invoked if both the ASM and ECM cross correlation analysis are the same as determined in step **520**. A safe level check can be performed by processor **206** at step **522**.

FIG. **6** is a flowchart of a method **600** for binaural analysis. The method **600** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **600**, reference will be made to components of FIG. **2**, although it is understood that the method **600** can be implemented in any other manner using other suitable components. The method **600** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **6** describes a component of the SPD method **500** wherein a cross-correlation of two input audio signals **602** and **604** (e.g., left and right ASM signals) is calculated. The input signals may first be weighted using a frequency-dependent filter (e.g. an FIR-type filter) using filter coefficients **606** and filtering networks **608** and **610**. Alternatively, an interchannel cross-correlation calculated with function **612** can return a frequency-dependent correlation such as a coherence function. The absolute maximum peak of a calculated cross-correlation **614** can be subtracted from a mean (or RMS) **616** correlation, with subtractor **622**, and compared **628** with a predefined threshold **626**, to determine if the peak is significantly greater than the average correlation (i.e. a test for peakedness). Alternatively, the maxima of the peak may simply be compared with the threshold **628**

without the subtraction process **622**. If the lag-time of the peak **618** is at approximately lag-sample 0, then the sound source is determined, at step **624**, as being on the interaural axis-indicative of User-generated speech, and a no-pass through mode is returned **630** (a further function described in FIG. **7** may be used to confirm that the sound source originates in the User-head, rather than external to the user- and further confirming that the acute sound is a User-generated voice sound). The logical AND unit **632** activates the pass-through mode **636** if both criteria in the decision units **628** and **624** confirm that the absolute maxima of the peak is above a predefined threshold **626**, AND the lag of the peak is NOT at approximately lag sample zero. A safe level check may be performed by processor **206** at step **634**.

FIG. **7** is a flowchart of a method **700** for logic control. The method **700** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **700**, reference will be made to components of FIG. **2**, although it is understood that the method **700** can be implemented in any other manner using other suitable components. The method **700** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **7** describes a further component of the SPD method **500**, which is optional to confirm that the acute sound source is from a location indicative of user-generated speech; i.e. inside the head. Method steps **702-712** are similar to Method steps **502-514** of FIG. **5**. The cross-correlations of step **710** and **712** provide a time-lag of the maximum absolute peak for a pair of input signals; the ASM and ECM signals for the same headset (e.g. the ASM and ECM for the left headset). At step **714** a left lag of a peak of the left cross correlation is determined, and simultaneously, a right lag of a peak of the right cross correlation is determined at step **718**. If a lag of a respective peak is greater than zero—this indicates that the sound arrived at the ECM signal before the ASM signal. Decision step **716** determines if the lag is greater than zero for both the left and right headsets- and activates the pass-through mode **722** if so. A safe level check may be performed by processor **206** at step **720**.

FIG. **8** is a flowchart of a method **800** for estimating background sound level. The method **800** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **800**, reference will be made to components of FIG. **2**, although it is understood that the method **800** can be implemented in any other manner using other suitable components. The method **800** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, method **800** receives as its input **802** either or both the ASM signal from ASM **111** and a signal from the ECM **123**. An audio buffer **804** of the input audio signal is accumulated (e.g. 10 ms of data), which is then processed by squaring step **806** to obtain the temporal envelope. The envelope is smoothed (e.g. an FIR-type low-pass digital filter) at step **808** using a smoothing window **810** stored in data memory (e.g. a Hanning or Hamming shaped window). At step **812**, transient peaks in the input buffer can be identified and removed to determine a "steady-state" Background Noise Level (BNL). At step **814** an average BNL **816** can be obtained (similar to, or the same as, the RMS) that is frequency dependent or a single value averaged over all frequencies. If the ECM **123** is used to determine the BNL, then decision step **818** adjusts the ambient BNL estimation to provide an equivalent ear-canal BNL SPL, by deducting an Earpiece Noise Reduction Rating **828** from the BNL estimate **826**. Alternatively, if the ECM **123** is used, then the Audio Content SPL level (ACL) **822** of any reproduced Audio Content **820** is deducted from the ECM level at step **824**. The updated BNL estimate is then converted to a Sound Pressure Level (SPL) equivalent **832** (i.e. substantially equal to the SPL at the ear-drum in which the earphone device is inserted) by taking into account the sensitivity (e.g. measured in V per dB) of either the ASM **111** or ECM **123** at steps **830** and **834** respectively. The resulting BNL SPL is then combined at step **842** with the previous BNL estimate **840**, by averaging **838** a weighted previous BNL (weighted with coefficient **836**), to give a new ear-canal BNL **844**.

FIG. **9** is a flowchart of a method **900** for maintaining constant audio content level (ACL) to internal ambient sound level (iASL). The method **900** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **900**, reference will be made to components of FIG. **2**, although it is understood that the method **900** can be implemented in any other manner using other suitable components. The method **900** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Briefly, FIG. **9** describes a method **900** for Constant Signal-to-Noise Ratio (CSNRS). At step **904** an input signal is captured from the ASM **111** and processed at step **910** (e.g. ADC, EQ, gain). Similarly, at step **906** an input signal from the ECM **123** is captured and processed at step **912**. The method **900** also receives as input an Audio Content signal **902**, e.g. a music audio signal from a portable Media Player or mobile-phone, which is processed with an analog and digital signal processing system as shown in step **908**. An Audio Content Level (ACL) is determined at step **914** based on an earpiece sensitivity from step **916**, and returns a dBV value.

In one exemplary embodiment, method **900** calculates a RMS value over a window (e.g. the last 100 ms). The RMS value can then be first weighted with a first weighting coefficient and then averaged with a weighted previous level estimate. The ACL is converted to an equivalent SPL value (ACL), which may use either a look-up-table or algorithm to calculate the ear-canal SPL of the signal if it was reproduced with the ECR **125**. To calculate the equivalent ear canal SPL, the sensitivity of the ear canal receiver can be factored in during processing.

At step **922** the BNL is estimated using inputs from either or both the ASM signal at step **902**, and/or the ECM signal at step **906**. The BNL may be adjusted by the earpiece noise reduction rating **924**. These signals are selected using the BNL input switch at step **918**, which may be controlled automatically or with a specific user-generated manual operation at step **926**. The Ear-Canal SNR is calculated at step **920** by differencing the ACL from step **914** and the BNL from step **922** and the resulting SNR **930** is passed to the method step **932** for AGC coefficient calculation. The AGC coefficient calculation **932** calculates gains for the Audio Content signal and ASM signal from the Automatic Gain Control steps **928** and **936** (for the Audio Content and ASM signals, respectively). AGC coefficient calculation **932** may use a default preferred SNR **938** or a user-preferred SNR **934** in its calculation. After the ASM signal and Audio content signal have been processed by the AGCs **928** and **936**, the two signals are mixed at step **940**.

At step **942**, a safe-level check determines if the resulting mixed signal is too high, if it were reproduced with the ECR **125** as shown in block **944**. The safe-level check can use

information regarding the user's listening history to determine if the user's sound exposure is such that it may cause a temporary or a permanent hearing threshold shift. If such high levels are measured, then the safe-level check reduces the signal level of the mixed signals via a feedback path to step **940**. The resulting audio signal generated after step **942** is then reproduced with the ECR **125**.

FIG. **10** is a flowchart of a method **950** for maintaining a constant signal to noise ratio based on automatic gain control (AGC). The method **950** can be practiced with more or less than the number of steps shown and is not limited to the order shown. To describe the method **950**, reference will be made to components of FIG. **2**, although it is understood that the method **950** can be implemented in any other manner using other suitable components. The method **950** can be implemented in a single earpiece, a pair of earpieces, headphones, or other suitable headset audio delivery devices.

Method **950** describes calculation of AGC coefficients. The method **950** receives as its inputs an Ear Canal SNR **952** and a target SNR **960** to provide a SNR mismatch **958**. The target SNR **964** is chosen from a pre-defined SNR **954**, sorted in computer memory or a manually defined SNR **956**. At step **958**, a difference is calculated between the actual ear-canal SNR and the target SNR to produce the mismatch **962**. The mismatch level **962** is smoothed over time at step **968**, which uses a previous mismatch **970** that is weighted using single or multiple weighting coefficients **966**, to give a new time-smoothed SNR mismatch **974**. Depending on the magnitude of this mismatch, various operating modes **972**, **978** can be invoked, for example, as described by the AGC decision module **976** (step **932** in FIG. **9**).

While the present invention has been described with reference to exemplary embodiments, it is to be understood that the invention is not limited to the disclosed exemplary embodiments. The scope of the following claims is to be accorded the broadest interpretation so as to encompass all modifications, equivalent structures and functions of the relevant exemplary embodiments. Thus, the description of the invention is merely exemplary in nature and, thus, variations that do not depart from the gist of the invention are intended to be within the scope of the exemplary embodiments of the present invention. Such variations are not to be regarded as a departure from the spirit and scope of the present invention.

What is claimed is:

1. A device that can control ambient passthrough in an earpiece comprising:
- a first memory that stores instructions; and
- a processor that executes the instructions to perform operations, the operations comprising:
  - communicatively connecting to an earpiece, wherein the earpiece includes a first microphone, a second microphone, a speaker, a second memory and an earpiece processor;
  - sending an audio content gain value and an ambient sound gain value to the earpiece; and
  - sending an audio content signal to the earpiece, wherein the earpiece processor
    - applies the audio content gain value to the audio content signal to generate a modified audio content signal, wherein the earpiece processor
    - applies the ambient sound gain value to an ambient sound signal to generate a modified ambient sound signal, and wherein the earpiece processor
    - mixes the modified ambient sound signal with the modified audio content signal to generate a mixed signal.

2. The device according to claim **1**, wherein the earpiece processor calculates a noise level based upon a first microphone signal from the first microphone, wherein the earpiece processor modifies the mixed signal based upon the noise level.

3. The device according to claim **1**, wherein the earpiece processor generates the ambient sound signal from a first microphone signal from the first microphone.

4. The device according to claim **1** further including a user interface.

5. The device according to claim **1**, wherein the earpiece processor applies an ambient gain to a first microphone signal from the first microphone to generate the ambient sound signal.

6. The device according to claim **1**, wherein the device is a portable communication device.

7. The device according to claim **6**, where the device portable communication device is at least one of a phone, a watch, a tablet, earpiece, a laptop computer or a combination thereof.

8. A device that can control ambient sound passthrough comprising:
- a first microphone that measures a first acoustic environment to generate a first microphone signal;
- a second microphone that is positioned closer to a user's ear canal than the first microphone when the device is worn, where the second microphone generates a second microphone signal;
- a speaker;
- a memory that stores instructions; and
- a processor that executes the instructions to perform operations, the operations comprising:
  - receiving an audio content signal;
  - modifying the audio content signal to generate a modified audio content signal;
  - modifying the first microphone signal to generate a modified first microphone signal;
  - modifying the second microphone signal to generate a modified second microphone signal;
  - generating an ambient sound signal from at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof;
  - detecting an acute sound by analyzing at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof;
  - modifying the modified audio content signal if an acute sound is detected to generate a new modified audio content signal;
  - generating a mixed signal by mixing the ambient sound signal with the modified audio content signal or the new modified audio content signal; and
  - sending the mixed signal to the speaker.

9. The device according to claim **8**, wherein the operations further comprise:
- determining if the acute sound is a user's voice by analyzing the temporal and spectral characteristics of at least one of the modified first microphone signal, the modified second microphone signal or a combination thereof.

10. The device according to claim **9**, wherein the operations further comprise:
- generating a noise signal from at least one of the modified first microphone signal, the modified second microphone signal, or both;
- generating a noise reduction signal.

11. The device according to claim **10**, wherein the operations further comprise:

mixing the noise reduction signal with the mixed signal to generate a new mixed signal, and sending the new mixed signal to the speaker instead of the mixed signal.

12. The device according to claim **11**, wherein the operations further comprise:

sending a modified ambient signal to the speaker instead of the new mixed signal if the user's voice is detected.

13. The device according to claim **9**, wherein the operations further comprise:

sending a modified ambient signal to the speaker instead of the mixed signal if the user's voice is detected.

14. The device according to claim **9**, wherein the operations further comprise:

detecting whether the acute sound is at least one of a warning sound, siren, voice, specific noise type, or a combination, by analyzing the spectrum of at least one of the modified first microphone signal, the modified second microphone signal, the modified ambient sound signal, or a combination.

15. The device according to claim **14**, wherein the operations further comprise:

decreasing a volume of the modified audio content signal when a warning or siren is detected.

16. The device according to claim **14**, where the operations further comprise:

sending a notification signal to the speaker if the acute sound is a warning sound or siren.

17. The device according to claim **16**, wherein the warning sound is at least one of a bell, an emergency vehicle, the sound of a security system or a combination thereof.

18. The device according to claim **14**, wherein if an alarm is detected it is at least one of a police siren, an ambulance siren, a car honking or a combination thereof.

19. The device according to claim **8**, wherein the operations further comprise:

decreasing a volume of the modified audio content signal when a user's voice is detected.

20. The device according to claim **8**, wherein the acute sound is a voice.

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