

(19) World Intellectual Property Organization
International Bureau



(43) International Publication Date
6 August 2009 (06.08.2009)

PCT

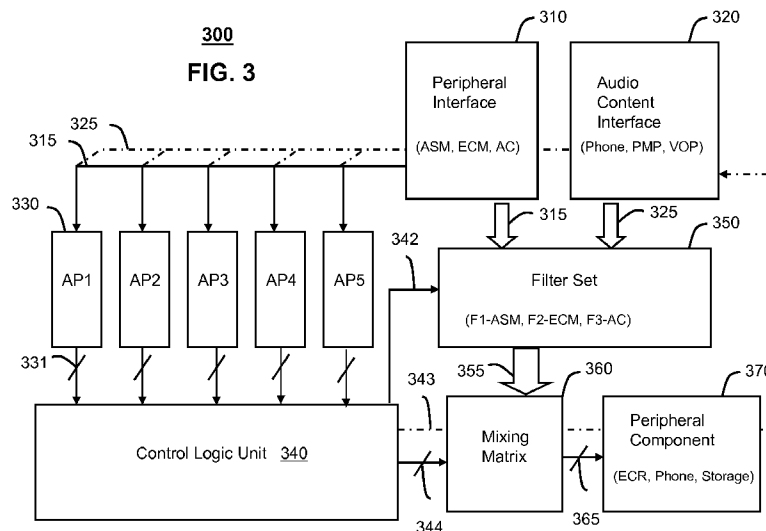
(10) International Publication Number
WO 2009/097009 A1

- (51) International Patent Classification:
H04R 5/02 (2006.01)
- (21) International Application Number:
PCT/US2008/073177
- (22) International Filing Date: 14 August 2008 (14.08.2008)
- (25) Filing Language: English
- (26) Publication Language: English
- (30) Priority Data:
60/955,848 14 August 2007 (14.08.2007) US
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- (81) Designated States (unless otherwise indicated, for every kind of national protection available): AE, AG, AL, AM, AO, AT, AU, AZ, BA, BB, BG, BH, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DO, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, GT, HN, HR, HU, ID, IL, IN, IS, JP, KE, KG, KM, KN, KP, KR, KZ, LA, LC, LK, LR, LS, LT, LU, LY, MA, MD, ME, MG, MK, MN, MW, MX, MY, MZ, NA, NG, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RS, RU, SC, SD, SE, SG, SK, SL, SM, ST, SV, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, ZA, ZM, ZW.
- (84) Designated States (unless otherwise indicated, for every kind of regional protection available): ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HR, HU, IE, IS, IT, LT, LU, LV, MC, MT, NL, NO, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

- Published:**
- with international search report
 - before the expiration of the time limit for amending the claims and to be republished in the event of receipt of amendments

(54) Title: METHOD AND DEVICE FOR LINKING MATRIX CONTROL OF AN EARPIECE



(57) Abstract: A device (100) and method (500) for configuring audio delivery on an earpiece is provided. The earpiece can include a peripheral interface (310) configured to receive a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules (330) that produce audio control information (331) responsive to an analysis of the sound signals, a logic control unit (340) to receive the audio control information from the plurality of audio processing modules and generate configuration data (342), and a mixing matrix (360) to mix the plurality of sound signals in accordance with the configuration data to produce output sound signals (365). The logic control unit can assign a priority to the plurality of audio processing modules, and modify the configuration data to mix the plurality of sound signals according to the priority. Other embodiments are disclosed.

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METHOD AND DEVICE FOR LINKING MATRIX CONTROL OF AN EARPIECE

FIELD

[0001] The present invention relates to a method of controlling an earpiece device, and more particularly, though not exclusively, a method of configuring and managing audio input and output on an earpiece.

BACKGROUND

[0002] Headphones or earpieces can be used for music enjoyment or voice communication. Use of these devices has steadily been increasing, and more products are expanding functionality to support audio delivery to headphones and earpieces.

[0003] The earpieces and associated products are becoming more intelligent as more communication features become available. However, the earpieces themselves have limited resources and must make efficient use of the many audio input and output configurations required to manage and support audio delivery.

[0004] A need therefore exists for improving audio configuration paths of headphones or earpieces.

SUMMARY

[0005] In a first embodiment, an earpiece, can include a peripheral interface configured to manage a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules that produce audio control information responsive to an analysis of the sound signals, a logic control unit operatively coupled to the plurality of audio processing modules to generate configuration data responsive to assessing the audio control information from the plurality of audio processing modules; and a mixing matrix operatively coupled to the control logic unit and peripheral interface to mix the plurality of sound signals in accordance with configuration data to produce output sound signals and route the output sound signals to at least one peripheral component. The earpiece can further include a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals that are presented to the mixing matrix.

[0006] The logic control unit can produce a linkage matrix of mixing gains that the mixing matrix applies to the plurality of filtered sound signals for producing the plurality of output sound signals. The logic control unit can assign a priority to the plurality of audio processing modules, and modifies the configuration data to mix the plurality of sound signals according to the priority. The priority can be event driven responsive to an audio processing module detecting a sound signature, responsive to an audio processing module detecting a sound exposure level that exceeds a predetermined safe hearing level, or responsive to an audio content interface receiving an audio stream from a phone, a media player, or a portable communication device.

[0007] The peripheral interface can include an Ambient Sound Microphone (ASM) configured to convert an ambient sound to an ambient sound signal, and an Ear Canal Microphone (ECM) configured to convert an internal sound from an ear canal of a user to an internal sound signal. The peripheral component can be an Ear Canal Receiver (ECR), a phone, a portable communication device, or a storage. The mixing matrix can associate the ECR with a first row or first column of mixing gains, associate the ECM with a second row or second column of mixing gains, and associate the ASM with a third row or third column of mixing gains. The mixing matrix can further associate an audio stream with a first row or first column of mixing gains, associate an audio storage with a second row or second column of mixing gains, and associate an audio controller with a third row or third column of mixing gains.

[0008] In a second embodiment, an earpiece can include a peripheral interface configured to manage a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules, an audio content interface configured to manage a plurality of audio streams and direct the plurality of audio streams to the plurality of audio processing modules, at least one signal analysis module operatively coupled to the peripheral interface and audio content interface to provide a shared analysis of the sound signals and the audio streams for the plurality of audio processing modules, a logic control unit operatively coupled to the plurality of audio processing modules and the at least one signal analysis module to receive the shared analysis and audio control information to generate configuration data, and a mixing matrix operatively coupled to the control logic unit, peripheral interface, and audio content interface to mix the plurality of filtered sound signals in accordance with the configuration data to produce output sound signals and route the output sound signals to at least one

peripheral component. The shared analysis can include spectral analysis, spectral band energy level analysis, spectral envelope analysis, voice activity detection analysis, and cross-correlation analysis.

[0009] The earpiece can further include a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals that are presented to the mixing matrix and mixed with mixing gains that are determined from the configuration data. The logic control unit can assign a priority to the plurality of audio processing modules, and modify the configuration data presented to the filter set to filter the plurality of sound signals according to the priority. The priority can be event driven responsive to an audio processing module detecting a sound signature, or an audio processing module detecting a sound exposure level that exceeds a predetermined safe hearing level.

[0010] In a third embodiment, a method for configuring audio delivery on an earpiece can include the steps of receiving a plurality of sound signals and a plurality of audio streams, analyzing the plurality of sound signals and the plurality of audio streams to produce audio control information, generating configuration data that prioritizes an audio delivery of the plurality of sound signals and the plurality of audio streams responsive to assessing the audio control information, and mixing the plurality of sound signals and the plurality of audio streams according to the configuration data to produce output sound signals that are routed to at least one peripheral component.

[0011] The mixing can include adjusting a volume of a first sound signal or first audio stream relative to one another or a second sound signal or second audio stream.

[0012] The prioritizing can be event driven responsive to detecting a warning sound, or event driven responsive to detecting a sound exposure level that exceeds a predetermined safe hearing level. The method can include associating an Ear Canal Receiver (ECR) signal with a first row or first column of mixing gains, associating an Ear Canal Microphone (ECM) signal with a second row or second column of mixing gains, and associating an Ambient Sound Microphone (ASM) signal with a third row or third column of mixing gains. The method can further include associating an audio stream with a first row or first column of mixing gains, associating an audio storage with a second row or second column of mixing gains, and associating an audio controller with a third row or third column of mixing gains.

BRIEF DESCRIPTION OF THE DRAWINGS

- [0013]** FIG. 1 is a pictorial diagram of an earpiece in accordance with an exemplary embodiment;
- [0014]** FIG. 2 is a block diagram of the earpiece in accordance with an exemplary embodiment;
- [0015]** FIG. 3 is an exemplary schematic of a software system for the earpiece in accordance with an exemplary embodiment;
- [0016]** FIG. 4 is a more detailed exemplary schematic of the software system of FIG. 3 for the earpiece in accordance with an exemplary embodiment;
- [0017]** FIG. 5 is a flowchart of a method for generating filter configuration data from a plurality of processing modules in accordance with an exemplary embodiment in accordance with an exemplary embodiment;
- [0018]** FIG. 6 is a flowchart of a method for generating and applying filter configuration data from a plurality of processing modules in accordance with an exemplary embodiment;
- [0019]** FIG. 7 is an exemplary schematic for configuring audio input and output via a mixing matrix in accordance with an exemplary embodiment;
- [0020]** FIG. 8 is another exemplary schematic of a software system for the earpiece providing separate analysis and re-synthesis modules for peripheral inputs in accordance with an exemplary embodiment; and
- [0021]** FIG. 9 is a more detailed schematic for analysis and re-synthesis for a particular peripheral input in accordance with an exemplary embodiment.

DETAILED DESCRIPTION

[0022] 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.

[0023] 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.

[0024] 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.

[0025] 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.

[0026] 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.

[0027] At least one exemplary embodiment of the invention is directed to an earpiece for background noise mitigation. 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. 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 ear 117 of 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.

[0028] Earpiece 100 includes an Ambient Sound Microphone (ASM) 111 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 capture internal sounds within the ear canal and also 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 users ear canal 131, and to form an acoustic seal with the ear canal walls 129 at a location 127 between the entrance 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 5cc 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 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.

[0029] 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 ECM 123 can also be used for capturing voice that is resonant within the ear canal when the user is speaking to permit voice communication.

[0030] The ASM 111 is housed in an ear seal 113 and monitors sound pressure at the entrance to the occluded or partially occluded ear canal. The ASM 111 can also be used to capture the user's voice externally for permitting voice communication. All transducers shown can receive or transmit audio signals to a processor 121 that undertakes audio signal processing and provides a transceiver for audio or voice via the wired or wireless communication path 119.

[0031] The earpiece 100 can actively monitor a sound pressure level both inside and outside an ear canal and enhance spatial and timbral sound quality while maintaining supervision to ensure safe sound reproduction levels. The earpiece 100 in various embodiments can conduct listening tests, filter sounds in the environment, monitor warning sounds in the environment, present notification based on identified warning sounds, maintain constant audio content to ambient sound levels, and filter sound in accordance with a Personalized Hearing Level (PHL).

[0032] 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 measure ambient sounds in the environment received at the ASM 111 and internal sounds captured at the ECM 130. Ambient sounds correspond to sounds within the environment such as the sound of traffic noise, street noise, conversation babble, or any other acoustic sound.

[0033] Ambient sounds measured by the ASM 111 can also correspond to industrial sounds present in an industrial setting, such as, factory noise, lifting vehicles, automobiles, and robots. The processor 206 can monitor the ambient sound captured by the ASM 110 for sounds in the environment, such as an abrupt high energy sound corresponding to an 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.).

[0034] Internal sounds measured by the ECM 123 can correspond to sounds contained within the ear canal 131 such as spoken voice or audio content delivered by way of the ECR 125. The internal sounds can include residual background noise related to ambient sounds in the environment; for example, high level sounds that leak around the ear seal 127 and enter the ear canal 131. The processor 206 can monitor internal sounds captured by the ECM 123 and analyze the internal sounds. The processor 206 can also adjust a mixing between the ambient sound signals measured at the ASM 111 and the internal sound signals measured at the ECM 123, for example, responsive to assessing ambient background noise conditions.

[0035] 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 a 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.

[0036] The memory 208 can also store program instructions for execution on the processor 206. The memory 208 can be off-chip and external to the processor 208, and include a data buffer to temporarily capture the ambient sound and the internal sound, and a storage memory to save from the data buffer the recent portion of the history in a compressed format responsive to a directive by the processor. The data buffer can be a circular buffer that temporarily stores audio sound at a current time point to a previous time point. It should also be noted that the data buffer can in one configuration reside on the processor 206 to provide high speed data access. The storage memory can be non-volatile memory such as SRAM to store captured or compressed audio data.

[0037] The memory 208 can be a machine-readable medium. The term "machine-readable medium" should be taken to include a single medium or multiple media (e.g., a

centralized or distributed, and/or associated caches and external memory) that store the one or more sets of instructions. The term "machine-readable medium" shall also be taken to include any medium that is capable of storing, encoding or carrying a set of instructions for execution by the machine and that cause the machine to perform any one or more of the methodologies of the present disclosure.

[0038] The term "machine-readable medium" shall accordingly be taken to include, but not be limited to: solid-state memories such as a memory card or other package that houses one or more read-only (non-volatile) memories, random access memories, or other re-writable (volatile) memories; and/or magneto-optical or optical medium; and carrier wave signals such as a signal embodying computer instructions in a transmission medium. Accordingly, the disclosure is considered to include any one or more of a machine-readable medium or a distribution medium, as listed herein and including art-recognized equivalents and successor media, in which the software implementations herein are stored.

[0039] 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 ambient sounds can adjust the audio content and pass the ambient sounds directly to the ear canal. For instance, the processor 206 can lower a volume of the audio content played out the ECR 125 responsive to detecting an acute sound for transmitting the ambient sound to the ear canal. The processor 206 can also actively monitor the sound exposure level inside the ear canal via the ECM 123 and adjust the audio content to within a safe and subjectively optimized listening level range.

[0040] 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.

[0041] 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.

[0042] 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.

[0043] FIG. 3 is an exemplary schematic of a software system 300 for operating the earpiece 100. The software system 300 can reside at least in part or whole on the processor 206, the memory 208, and/or any associated machine readable storage medium operated on by the processor 206 (see FIG. 2). Generally stated, the software system 300 by way of the processor 206 can manage a configuration of audio input and output (paths) to the earpiece 100 to support audio delivery.

[0044] As illustrated, the software system 300 includes a peripheral interface 310 configured to manage a plurality of sound signals 315 and direct the plurality of sound signals 315 to a plurality of audio processing modules 330. The sound signals can be an ambient sound signal measured by the ASM 111 or an internal sound signal from measured by the ECM 123. The software system 300 can include an audio content interface 320 configured to manage a plurality of audio streams 325 and direct the plurality of audio streams 325 to the plurality of audio processing modules 330. The audio stream can be a voice signal from a Phone, a music signal from a personal media player (PMP), or an audio signal provided by the earpiece (e.g., loopback signal). The plurality of sound signals 315 and the plurality of audio streams 325 are also passed to a filter set 350 as shown by the wide arrows.

[0045] The plurality of audio processing modules 330 can produce audio control information 331 responsive to an analysis of the sound signals. The audio control information 331 is provided to the logic control unit 340 for configuring filtering and mixing operations on the plurality of sound signals 315 and the plurality of audio streams 325 at a filter set 350. The logic control 340 unit is operatively coupled to the plurality of audio processing modules 330 to receive the audio control information 331 from the plurality of audio processing modules 330 and generate configuration data. The configuration data can include filter data 342 for processing the plurality of sound

signals, audio control data 343 for assigning a priority to the plurality of sound signals, and router data 344 for mixing a plurality of filtered signals 355 according to the priority

[0046] The software system 300 includes the filter set 350 operatively coupled to the peripheral interface to process the plurality of sound signals 315 in accordance with the configuration data 342 for generating the filtered sound signals 355. The filter set 350 also processes the plurality of audio streams 325 in accordance with the configuration data 344 to produce filtered sound signals 355. The filtered sound signals 355 are passed to a mixing matrix 360.

[0047] The mixing matrix 360 is operatively coupled to the control logic unit 340 and filter set 350 to mix the plurality of filtered sound signals 355 in accordance with the configuration data 344 and generate output sound signals 365 that are routed to at least one output peripheral component 370. The peripheral component 370 can be the ECR 125, a phone, or a data storage. The mixing matrix 360 can include individual gains that are applied to the plurality of filtered sound signals 355 for producing the output sound signals.

[0048] FIG. 4 is a more detailed schematic of the software system 300 of FIG. 3. System components of FIG. 3 will be referred to when describing components of the detailed schematic.

[0049] The peripheral interface 310 (see FIG. 3) can include the Ambient Sound Microphone (ASM) 111 configured to convert an ambient sound to an ambient sound signal, and the Ear Canal Microphone (ECM) 123 configured to convert an internal sound from an ear canal of a user to an internal sound signal. The peripheral interface 310 receives the ASM sound signals from the external environment including background noise, warning sounds, and the ECM sound signal including the voice of a wearer of the earpiece 100 and any audio content playing out the ECR 125. The components (e.g., ASM-R, ECM-R) of the peripheral interface 310 can contain more than the components shown, for instance, a secondary ASM for sound localization or noise suppression. The peripheral interface 310 can also include components (ASM-L, ECM-L) from a second earpiece such as a L-left or R-right earpiece. In this case, the software system 300 can reside on one of the earpieces, although resources can be shared if the software system 300 is enabled on both earpieces.

[0050] The audio content interface 320 (see FIG. 3) can receive audio streams, for example, from a phone 421, a personal media player (PMP) 422, a portable

communication device (e.g., VOP) 423, or a component 411 of the earpiece. The local component 411 (e.g., ECR 125) can permit audio feedback for allowing the user to hear audio, for example, loop back when the user is speaking on the phone, or playing comfort noise during non-speech intervals. The audio content interface (AC) 420 can mix the plurality of audio streams to produce a single audio stream delivered to the processing modules (AP1-AP5). In one arrangement, the AC 420 can perform audio content mixing based on a user context.

[0051] The user context identifies an operation or mode of the earpiece, such as, an incoming call, a music session, or a voice mail. For example, if the user is listening to music and an incoming call is detected, the earpiece 100 by way of the software system 300 (see FIG. 3) further discussed herein can lower the volume of the music relative to the incoming call ring tone to permit a hearing of the incoming call (e.g., ring tone). As another example, if the user is using the earpiece in a transparent “safe” mode while listening to music, then any harmful ambient sounds (e.g., loud jackhammer, bus noises) can be attenuated with the music, for instance, by reducing the ASM to ECM pass through levels. Similarly, important warning sounds captured at the ASM 111 can be elevated in the mix with respect to other sounds by increasing the ratio of ASM 111 to ECM 123 sound levels. The AC 420 can also mix the sounds in accordance with manual intervention, for example, if the user adjusts a volume level manually by way of a user interface or directly mixes the audio streams.

[0052] Referring to both FIGS. 3 and 4, the sound signals from peripheral interface 310 and the mixed sound signals from the audio content interface 320 can be passed through a signal analysis module 410 operatively coupled to the peripheral interface 310 and the audio content interface 320 to provide a shared analysis of the sound signals and the audio streams. The sound signals and the shared analysis can then be sent to the plurality of audio processing modules (AP1 – AP5). The shared analysis can include spectral analysis, spectral band energy level analysis, spectral envelope analysis, voice activity detection analysis, and cross-correlation analysis.

[0053] The shared analysis module 410 permits a sharing of processing resources among the processing modules (AP1 – AP5) to minimize individual resource use of the processing modules from performing a common analysis. For instance, instead of each processing module performing an FFT analysis, the shared module 410 can perform an FFT and share output results with the modules. In another arrangement, output results

from the shared analysis can be conveyed to the processing modules based on frequency band requirements or trigger events and thresholds. For instance, AP1 may register as an event listener for ASM signals that exceed a certain level in a frequency band; AP2 may register as an event listener to receive ECM signals that match a particular masking profile, and so on.

[0054] Alternatively, each processing module can interpret the output results individually and make their own respective determination as how to process the sound signals. For instance, each processing module (AP1 – AP5) can selectively process the plurality of sound signals in accordance with its own requirements. For example, AP1 can be a sound detection module to identify one or more sounds in the user's environment, for example, in accordance with the teachings presented in U.S. Patent Application No. 11/966,457 filed on December 28, 2007 entitled "Method and Device for Sound Signature Detection" herein incorporated by reference in its entirety. AP 2 can be a sound exposure monitoring module to assess safe listening levels, for instance, in accordance with the teachings presented in U.S. provisional patent application No. 60/887,165 filed on January 30, 2007 entitled "Sound Pressure Level Monitoring and Notification System" herein incorporated by reference in its entirety. AP 3 can be a sound enhancement module or voice control module, for instance, in accordance with the teachings presented in U.S. Provisional Application No. 60/911,691 filed on April 13, 2007 entitled "Method and Device for Voice Operated Control", and U.S. Provisional Application No. 60/885,917 filed on January 22, 2007 entitled "Method and Device for Acute Sound Detection and Reproduction" herein incorporated by reference in its entirety. AP 4 can be a sound correction module to modify a sound signal based on a sound level exposure within safe listening levels, for instance, in accordance with the teachings presented in U.S. Provisional Application No. 60/866,420 filed on November 18, 2006, entitled "Method and Device for Personalized Hearing" the entire disclosure of which is incorporated herein by reference.

[0055] Each of the processing modules (AP1 – AP5) generates separate audio control information 331 in view of the audio signals that can be used by the logic control unit 340 to make an informed decision related to filtering and mixing the sound signals and audio streams. The audio control information 331 includes filter control data 342, audio control data 343, and router control data 344. The filter control data 342 can include filter coefficients that each processing module (AP) identifies as being significant

to their intended function. The audio control data 343 can control a mixing of the plurality of audio streams at the AC 420 based on an established priority in view of the processing module decisions (e.g., warning signal detected, sound exposure level exceeded, echo feedback condition). The router control data 344 can include mixing gains (g1-g2) to amplify or attenuate the filtered sound streams.

[0056] The logic control unit 340 responsive to receiving the audio control information 331 generates a priority that is used for generating the configuration data (341,342 and 343) used to filter the sound signals and mix the filtered sound signals. The priority controls how the filtered signals are filtered at the filter set 350 (e.g. F1, F2, F3, see also FIG. 3). As illustrated, F1 is a first filter for the ambient sound signal, and F2 is a second filter for the internal sound signal in the ear canal. The logic control unit 340 varies the level of filtering at both F1 and F2 in accordance with the filter configuration data 342 as established by the determined priority. The filter coefficients for the F1 filter and F2 filter are provided on a frame-by-frame basis in real-time with the filter configuration data 342 from the logic control unit 340. The priority also controls how the audio streams are mixed in the audio control unit 320 (e.g., see VOP, Phone, PMP, see also FIG. 3) based on the audio control data, and how the filtered signals are mixed in the mixing matrix 360 (e.g., g1-g9, see also FIG. 3).

[0057] For example, an AP1 module configured for warning sound detection upon identifying a warning sound in an ASM sound signal, can generate filter coefficients to amplify one or a group of frequency bands of the warning sound (e.g., horn or siren). The AP1 module can consider the other frequency bands as don't cares since they are not part of the warning sound. AP1 can also raise or flag a priority level that a warning sound has been detected. Similarly, AP3 module for speech enhancement may determine from a simultaneous ECM audio stream that the wearer is speaking and generate filter coefficients that accentuate certain portions of the sound spectrum. AP3 can raise a flag or priority level based on an energy level of the spoken voice. AP4 may determine an echo condition feedback that could potentially damage the users hearing and generate filter coefficients to null out the feedback. AP4 can raise a flag or priority level indicating an immediate danger condition. AP2 may determine that a sound level exposure is being exceeded and generate audio control data to turn volume down on a media player. The logic control unit 340 upon receiving the audio control information and

corresponding priorities from the respective processing modules (AP1-AP5) can then determine the appropriate filter configuration given the filter coefficients and the priority.

[0058] Notably, the logic control unit 340 evaluates the audio control information from all the processing modules (AP1-APN) individually, and then collectively as a whole, for prioritizing the configuration data. For instance, warning sounds that are given a higher priority over voice from the ECM or music from the AC, will be mixed according to the priority. Thus, the output sound signal generated by the mixing matrix 360 emphasizes the warning sound, followed by the voice, followed by the audio content. If voice commands are given a higher priority over music, than the mixing matrix 360 can reduce the music levels from the AC module when an AP module detects voice. Again, the priority can be established manually (e.g., via user interface) or automatically (e.g., user context, or presence). As another example, the mixing matrix 360 can increase ASM to ECM pass through upon detection of a warning sound, while simultaneously lowering a phone volume if the user is checking voice mail messages. As yet another example, the mixing matrix 360 can reduce ASM to ECM pass and simultaneously lower a music level if an incoming phone call is detected. The priority can also be event driven responsive to detecting a sound signature, a background noise condition, a battery life indication, a manual interaction, or a voice recognition command.

[0059] In the foregoing, a detailed description for one exemplary operation of audio input/output configuration of the software system 400 shown in FIG. 4 is provided. It should be noted that other configurations apply, for instance, using more or less than the number of mixing coefficients (g1-g9) or filters (F1-F9) shown.

[0060] As illustrated, The AC control unit 420 mixes audio content (e.g., phone, VOP, PMP) and outputs an AC signal (or pair of stereo channels if two earphone devices are used). The mixing is controlled in accordance with configuration data 343 received from the control logic unit 340 and also automatically by the AC control unit 420. The AC signal generated from the AC 420 is then filtered by audio content filter F3. The internal ear-canal audio signal from the ECM 123 is filtered by ECM filter F1 205, and the ambient audio signal from the ASM is filtered by ASM filter F2. All three filters (F1, F2 and F3) are updated and controlled by the logic control unit 340 via filter configuration data 342.

[0061] The filtered audio signals 355 are then passed to the mixing matrix 360. The 9 mixing coefficients (g1-g9) of the mixing matrix 360 are controlled and updated by the

control logic unit 340 via configuration data 344. Each mixing coefficient (e.g., g1-g9) can be a time-varying positive number, negative number, or zero. As part of the mixing, the output sound signals are routed to the three peripheral output components (Phone 421, ECR 125, and Storage 430). The mixing gains can be normalized so as to permit balanced audio delivery. The peripheral output components comprise the ear canal receiver ECR 125 for delivering audio to the ear canal, the output signal to the phone 421 (i.e. the signal that is transmitted to another remote individual), and the audio storage device 430 (e.g. hard drive on a PMP).

[0062] The filter configuration data 342 for the 2 filters, the AC configuration data 343 for the AC control unit, and the routing configuration data 344 for the mixing matrix 360 is generated responsive to receiving the audio control information from the plurality of processing modules (AP1-AP5). Recall, each module (e.g., modules 1-5, or more) receives as its inputs at least one of the unfiltered audio input signals (e.g. the ECM and ASM signals) or audio sound streams (e.g., music signal from PMP). As illustrated, each module analyzes the input audio signal and generates 3 output types: coefficients for the router (e.g., g1-g9), filter coefficients for the 3 signal filters (e.g., F1, F2 and F3), and audio control signals for the AC control unit. Each of these output types can have "don't care" states- which can be represented by a reserved number value (e.g., -1).

[0063] The control logic unit 340 prioritizes the sound signals and audio streams for each signal type, for instance, it can prioritize the router coefficient values (e.g., mixing gains g1-g9) higher for one AP module than the other. In some cases, the control logic can maximally prioritize the signal type from one module so that the output router coefficients, for example, from one module is copied directly to the output of the control logic unit. In other cases, the output signals of the same type from different modules can be combined by a weighted addition. Similarly, the filter coefficients for a particular filter (F1 – F3) can be generated by a weighted addition (frequency domain) or convolution (time domain).

[0064] **FIG. 5** is a flowchart of a method 500 for generating filter configuration data in accordance with an exemplary embodiment. The method 500 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.

[0065] In particular, method 500 describes one exemplary arrangement in which the logic circuit 340 generates filter configuration data from the audio content received by individual AP modules to produce a combination filter output. In the exemplary embodiment, only one filter of the three filters shown in FIG. 4 is explained (e.g. generation of F1 coefficients for the ECM audio path). In other exemplary embodiments, the filter output for method 300 includes all 3 filters (F1, F2 and F3), for instance, the ECM sound signal, the ASM sound signal and the AC sound signal.

[0066] The method 500 can start at step **502**. At step **504**, the logic control unit 340 can receive N filter vectors from the N processing modules **506** (AP1-APN). Each processing module generates a filter vector that is a set of filter coefficients (time or frequency domain). Each of the N processing modules (AP1-APN) also outputs an associated module priority.

[0067] At step **508**, the logic control unit 340 assesses the priorities raised by each processing module and evaluates the priorities against a corresponding reference module priority at step **510**. If a processing module does not assign a particular priority value, then a default priority value is retrieved from the reference module priority list. At step **512**, the logic control unit 340 identifies the processing module with the highest priority. If a first processing module returns the same module priority value as a second processing module, then the logic control unit 340 assigns a unique module priority value to both the second and first modules, for example, based on a user context.

[0068] Upon selecting the processing module with the highest priority, the logic control unit 340 proceeds to determine if any configuration data is a “don’t care” as shown in step **514**. A “don’t care” value is a value that does not affect the filter state of the respective processing module. The filter configuration data 342 generated by each processing module can include filter values of a time domain filter vector (e.g. for an FIR type filter) of frequency domain filter (e.g., bi-quads) with “don’t care” value; the logic control unit 340 can assign a reserved numerical value of symbol for “don’t care” filter states. A “don’t care” flag can be associated with each filter output from each module.

[0069] At step **516**, the logic control unit 340 can copy the configuration data (e.g., filter data, router data, audio content data) for “care” states to internal memory for output to the filter set 350, mixing matrix 360, and audio control 420 (see FIG. 4). The logic

control unit 340 can continue to update the configuration data for “care” states as it is retrieved from each of the respective processing modules according to the priorities. Starting with the next highest priority module, as shown in step **520**, the filter values for this high priority module are then checked at step **514** to see if they correspond to a “don’t care” state in the method loop. The method loop continues until the end of the module priority list as shown in step **518**.

[0070] If the logic control unit 340 identifies a “don’t care” state is present, the filter values for that state is not updated; else, the filter values for this module are copied to the output and no further update of the filters is conducted until new filter vectors are received from all modules as shown in step **522**. In such regard, the logic control unit 340 output filter states using filter values directly from the processing modules based strictly on priority. If all module filter values are “don’t care” states, then the filter output is the same as a previous filter output. The method 500 continues back at step **504** to evaluate the priorities of each module in real-time as results are received. The logic control unit 340 thus continually updates the filter values in view of priority for the respective filters (F1, F2 and F3) to process audio signals and audio streams in real time.

[0071] **FIG. 6** is a flowchart of a method 600 for generating filter configuration data where filter data is collectively combined based on priority. The method 600 is an extension of method 500 of FIG. 5 and can practiced with more or less than the number of steps shown and is not limited to the order shown. Briefly, FIG. 6 presents an embodiment wherein processing modules can have a same priority value. In such a case when two modules have the same priority value, and neither contain (don’t care” filter states, then the filter values from each of these two module with the same priority value are combined. The combination can be a simple summation of filter weights (e.g. for frequency domain filters) or a convolution of the two sequences (e.g. for time domain weights).

[0072] The method 600 can start at step **602**. At step **604**, the logic control unit 340 can receive N filter vectors from the N processing modules **606** (AP1-APN). Each processing module generates a filter vector that is a set of filter coefficients (time or frequency domain). Each of the N processing modules (AP1-APN) also outputs an associated module priority. At step **608**, the logic control unit 340 assesses the priorities raised by each processing module. If a processing module does not assign a particular

priority value, then a default priority value can be retrieved from a reference module priority list.

[0073] At step **610**, the logic control unit 340 identifies the processing module with the highest priority. The logic control unit 340 at step **618** then examines the audio control information from each of the N processing modules to identify “don’t care” states. Filter values with “don’t care” are not considered in generating output filter configuration data 342 or router configuration data 344 (see FIG. 4). Accordingly, the method 600 continues to get the next most important priority modules at step **620** until the end of module priority list is reached at step **622**.

[0074] Filter values other than “don’t care” are however used in generating a combined filter output; that is, a filter output representative of the collective priority of all the processing modules (AP1-APN) based on their processing results. The combination also depends on whether the module is uniquely defined; for instance, it has precedence over combination with other filter values. Thus at step **614**, the logic control unit 340 determines whether a processing module with a “care” filter value is uniquely defined. If yes, then the filter values for that unique model are copied to the output (e.g., router configuration, filter configuration data, audio configuration data) at step **616**. If however, the module is not unique, then the logic control unit 340 at step **612** combines the filter value with filter data from modules of the same priority. These filter values are then copied to the output at step **616** if no other modules eclipse their collective priority, or until a module with higher priority is identified. Upon evaluating all new filter from all the modules at step **624**, the method 600 can return back to step 604 to continue to retrieve the filter vectors in real-time.

[0075] FIG. 7 is an exemplary schematic 700 for configuring audio input and output via a mixing matrix in accordance with an exemplary embodiment. Reference will be made to FIG. 4 when describing the schematic 700. For instance, the filtered ECM signal 702 is the output of the ECM filter (F1) of FIG. 4. The mixing gains g1 (704), g4 (704) and g7 (707) correspond to the first column of the mixing matrix 360 as shown in FIG. 4. The phone out 710, ECR 712, and audio storage 714 correspond to the output peripheral components (421, 125, and 430) of the mixing matrix 360.

[0076] The mixing gains (g1-g9) of the mixing matrix 360 can be applied to the filtered audio signals 355 (see FIG. 3) before they are routed to the corresponding peripheral output components (phone out 710, ECR 712, and audio storage 714). FIG. 7

shows a signal path diagram for the filtered ECM input signal 702 to the 3 peripheral output components of the matrix 260. The mixing gains can be normalized coefficients whose total contribution across a column of the mixing matrix is unity. For example, mixing gains of $g_1=0.2$, $g_4=0.3$, and $g_7=0.5$ are applied to the filtered ECM signal 702 to produce phone out 710, ECR out 125, and audio storage out 714. The coefficient values can be time variant for each new input audio sample into the mixing matrix 360, or can remain constant for a block of input samples.

[0077] In one exemplary embodiment, the 9 coefficients (g_1 - g_9) for each of the mixing gains in mixing matrix 360 can be generated using a similar logic process as described in FIG. 4 and FIG. 4, except the inputs to the logic system are not filter vectors but rather 9 coefficient values from each module. The priority value assigned to each module, or the priority value automatically generated by that module, can be a different value for the filter vectors than for the 9 coefficient values. Moreover, 9 coefficients are shown since there are 3 audio paths (ASM, ECM, and AC) and 3 output peripheral components (Phone, ECR, data storage). The number of mixing gains and structure of the mixing matrix 360 can thus be a function of the number of audio input and audio output paths.

[0078] FIG. 8 is another exemplary schematic 800 of the software system 300 providing separate analysis and re-synthesis modules for peripheral inputs in accordance with an exemplary embodiment. As illustrated, the schematic 800 combines the three audio filters F1, F2 and F3 with the shared analysis module 410 of the software system shown in FIG 4. Analysis/Re-synthesis unit 810 performs shared analysis of the ECM sound signal for processing modules (AP1-AP3). Analysis/Re-synthesis unit 820 performs shared analysis of the ASM sound signal for processing modules (AP1-AP3). Analysis/Re-synthesis unit 830 performs shared analysis of the Audio Content (AC) sound signal for processing modules (AP1-AP3).

[0079] The combined analysis/re-synthesis units 810, 820 and 830 filter the 3 input audio signals; the ECM signal; the ASM signal, and Audio Content (AC) signal. The AC signal comprises at least one of; an earcon signal (e.g. a "low battery" auditory warning cue); a signal from a mobile telephone; and a stereo signal from a PMP (e.g. portable DVD player, music audio player etc). The output signal of the analysis/resynthesis units 810, 820 and 830 are passed to the mixing columns (g_1 , g_4 , g_7 ; g_2 , g_5 , g_8 ; and g_3 , g_6 , g_9) of the mixing matrix 360. The mixing gains g_1 - g_9 are controlled via the router

configuration data from the logic control unit 340, which in turn were determined from the audio control information generated by the processing modules (AP1-AP3).

[0080] As illustrated the mixing matrix 360 can associate a phone with a first row of a linkage matrix, associates an ECR with a second row of a linkage matrix, associate an audio storage with a third row of a linkage matrix, associate an ECM with a first column of a linkage matrix, associate an ASM with a second column of a linkage matrix, and associate an AC controller with a third column of a linkage matrix. The mixing matrix can modify the output sound signal of the phone using values in the first row of the linkage matrix, modify the output sound signal of the ECR using values in the second row of the linkage matrix, and modify the output sound signal sent to the audio storage using values in the third row of the linkage matrix.

[0081] Recall each processing module has two or more output signals: filter gain coefficients which control the resynthesis of the signals in the analysis/resynthesis units; 9 coefficients for the router, and audio configuration data for the audio control. The processing modules (AP1-AP3) create AC Control signals to control the routing of the input AC signals using AC control unit (420, see FIG. 4). The processing modules are configured and their internal state can be queried by the user interface 850 (e.g. a computer connected to the DSP unit via USB).

[0082] FIG. 9 is a more detailed schematic 900 of an analysis and re-synthesis unit for a particular peripheral input in accordance with an exemplary embodiment. For instance, the analysis and re-synthesis unit 910 corresponds to the ASM sound signal path of the software system 800 shown in FIG. 8. That is, the analysis and re-synthesis unit 910 can receive audio input 901 from the ASM 111. Notably, the schematic 900 can be applied for producing filtered output for also the ECM 123 sound signals and AC 420 sound signals (see FIG. 4) or other signals.

[0083] As illustrated in FIG. 9, the analysis and re-synthesis unit 910 incorporates a filter-bank 912 (e.g., cascade of band-pass filters (BPF)) structure for decomposing the ASM sound signal into a plurality of filter-banks. The filter-bank can be based on a linear frequency scale or a non-linear frequency scale, such as, an octave, 1/3 octave, critical band, equivalent rectangular band (ERB), or melody (mel)-frequency scale. Each output of each filter-bank can then be presented to each of the N processing modules (AP1 920-AP2 930). As previously indicated, the processing modules 920 and 930 generate audio control information 331 (see FIG. 3) that the logic control unit 340 uses in

generating filter configuration data 342 (see FIG. 4). In the embodiment shown, the filter configuration data 342 is used by the analysis and re-synthesis unit 910 to scale the filter-bank output signals. For instance, the filter-bank outputs are gain scaled to amplify or attenuate the individual filter-banks with the filter coefficients in the filter configuration data 342. The filter-bank output signals can then be summed to generate a composite filtered sound signal. The filtered sound signal is then presented to the mixing matrix (see FIG. 8) for routing and mixing to the one or more output peripheral components (e.g., Phone, ECR, data storage).

[0084] In the following, a detailed description for operation of the analysis and re-synthesis unit 910 for use with an audio input signal is provided. The input signal 901 is called "audio input 1", and can be any one of the exemplary three signals described previously (i.e. ECM, ASM or AC signal). The audio input signal is split and processed by the different band-pass-filters 912 (i.e. with different centre frequencies). In one arrangement, the response of each respective band pass filter 912 is such that if the filter outputs are summed, then the power spectral density is the same as the original audio input signal. The band pass filters 912 can be time-domain FIR type filters or IIR type filters, or the analysis could be with a frequency domain FFT. The output signal of each band-pass filter is fed to at least one module (for the sake of clarity, 2 modules are shown in the exemplary embodiment of figure 8). Each module then creates an output signal to control the re-synthesis of the band-pass filtered input signals. This output signal is a vector of weights to control the gains of the filtered signals (i.e. the number of gains is equal to the number of filter used in the analysis). The logic control unit 340 determines a single set of gain coefficients for the gains (i.e. G1, G2, G3 and G4 in the exemplary embodiment of figure 8). After the first set of band pass filtered signals have been multiplied with the gains (G1-G4) to create a second set of amplified (or attenuated) band-pass filtered signal, the second set of signals are summed with the summing unit to create a single filtered ASM output sound signal, which form one of the three exemplary inputs to the previously described mixing matrix 360.

[0085] 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.

CLAIMS

What is claimed is,

1. An earpiece, comprising:

a peripheral interface configured to manage a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules that produce audio control information responsive to an analysis of the plurality of sound signals;

a logic control unit operatively coupled to the plurality of audio processing modules to generate configuration data responsive to assessing the audio control information from the plurality of audio processing modules; and

a mixing matrix operatively coupled to the logic control unit and peripheral interface to mix the plurality of sound signals in accordance with configuration data to produce output sound signals and route the output sound signals to at least one peripheral component.

2. The earpiece of claim 1, further comprising a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals that are presented to the mixing matrix.

3. The earpiece of claim 1, wherein the logic control unit produces a linkage matrix of mixing gains that the mixing matrix applies to the plurality of filtered sound signals for producing the plurality of output sound signals.

4. The earpiece of claim 1, wherein the logic control unit assigns a priority to the plurality of audio processing modules, and modifies the configuration data to mix the plurality of sound signals according to the priority.

5. The earpiece of claim 4, wherein the priority is event driven responsive to an audio processing module detecting a sound signature.

6. The earpiece of claim 4, wherein the priority is event driven responsive to an audio processing module detecting a sound exposure level that exceeds a predetermined safe hearing level.

7. The earpiece of claim 4, wherein the priority is event driven responsive to an audio content interface receiving an audio stream from a phone, a media player, or a portable communication device.

8. The earpiece of claim 1, wherein the peripheral interface comprises:

an Ambient Sound Microphone (ASM) configured to convert an ambient sound to an ambient sound signal; and

an Ear Canal Microphone (ECM) configured to convert an internal sound from an ear canal of a user to an internal sound signal,

and the peripheral component is an Ear Canal Receiver (ECR), a phone, a portable communication device, or a storage.

9. The earpiece of claim 8, wherein the mixing matrix:

associates the ECR with a first row or first column of mixing gains;

associates the ECM with a second row or second column of mixing gains; and

associates the ASM with a third row or third column of mixing gains.

10. The earpiece of claim 1, wherein the mixing matrix:

associates an audio stream with a first row or first column of mixing gains;

associates an audio storage with a second row or second column of mixing gains;

associates an audio controller with a third row or third column of mixing gains.

11. An earpiece, comprising:

a peripheral interface configured to manage a plurality of sound signals and direct the plurality of sound signals to a plurality of audio processing modules;

an audio content interface configured to manage a plurality of audio streams and direct the plurality of audio streams to the plurality of audio processing modules;

at least one signal analysis module operatively coupled to the peripheral interface and audio content interface to provide a shared analysis of the sound signals and the audio streams for the plurality of audio processing modules;

a logic control unit operatively coupled to the plurality of audio processing modules and the at least one signal analysis module to receive the shared analysis and audio control information to generate configuration data; and

a mixing matrix operatively coupled to the control logic unit, peripheral interface, and audio content interface to mix the plurality of filtered sound signals in accordance with the configuration data to produce output sound signals and route the output sound signals to at least one peripheral component.

12. The earpiece of claim 11, wherein the shared analysis comprises spectral analysis, spectral band energy level analysis, spectral envelope analysis, voice activity detection analysis, and cross-correlation analysis.

13. The earpiece of claim 11, comprising a filter set operatively coupled to the peripheral interface to process the plurality of sound signals in accordance with the configuration data to produce filtered sound signals that are presented to the mixing matrix and mixed with mixing gains that are determined from the configuration data.

14. The earpiece of claim 13, wherein the logic control unit assigns a priority to the plurality of audio processing modules, and modifies the configuration data presented to the filter set to filter the plurality of sound signals according to the priority.

15. The earpiece of claim 14, wherein the priority is event driven responsive to an audio processing module detecting a sound signature.

16. The earpiece of claim 14, wherein the priority is event driven responsive to an audio processing module detecting a sound exposure level that exceeds a predetermined safe hearing level.

17. A method for configuring audio delivery on an earpiece, the method comprising the steps of:

receiving a plurality of sound signals and a plurality of audio streams;
analyzing the plurality of sound signals and the plurality of audio streams to produce audio control information;
generating configuration data that prioritizes an audio delivery of the plurality of sound signals and the plurality of audio streams responsive to assessing the audio control information; and
mixing the plurality of sound signals and the plurality of audio streams according to the configuration data to produce output sound signals that are routed to at least one peripheral component.

18. The method of claim 17, wherein the mixing includes adjusting a volume of a first sound signal or first audio stream relative to one another or a second sound signal or second audio stream.

19. The method of claim 17, wherein the mixing includes adjusting a volume of a first sound signal or first audio stream relative to one another or a second sound signal or second audio stream.

20. The method of claim 17, wherein the prioritizing is event driven responsive to detecting a sound exposure level that exceeds a predetermined safe hearing level.

21. The method of claim 17, comprising:

associating an Ear Canal Receiver (ECR) signal with a first row or first column of mixing gains;

associating an Ear Canal Microphone (ECM) signal with a second row or second column of mixing gains; and

associating an Ambient Sound Microphone (ASM) signal with a third row or third column of mixing gains.

22. The method of claim 17, comprising

associating an audio stream with a first row or first column of mixing gains;

associating an audio storage with a second row or second column of mixing gains; and

associating an audio controller with a third row or third column of mixing gains.

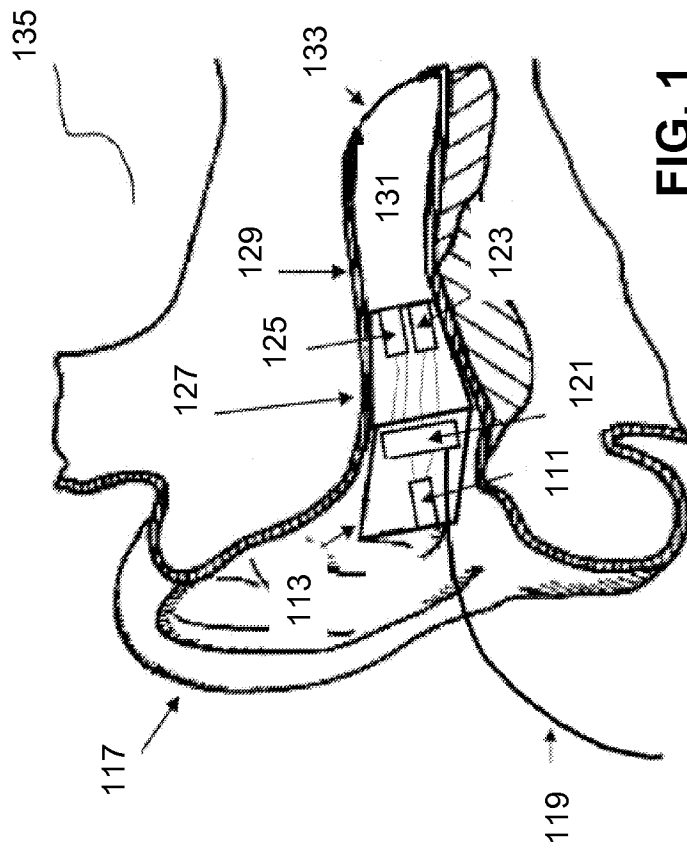
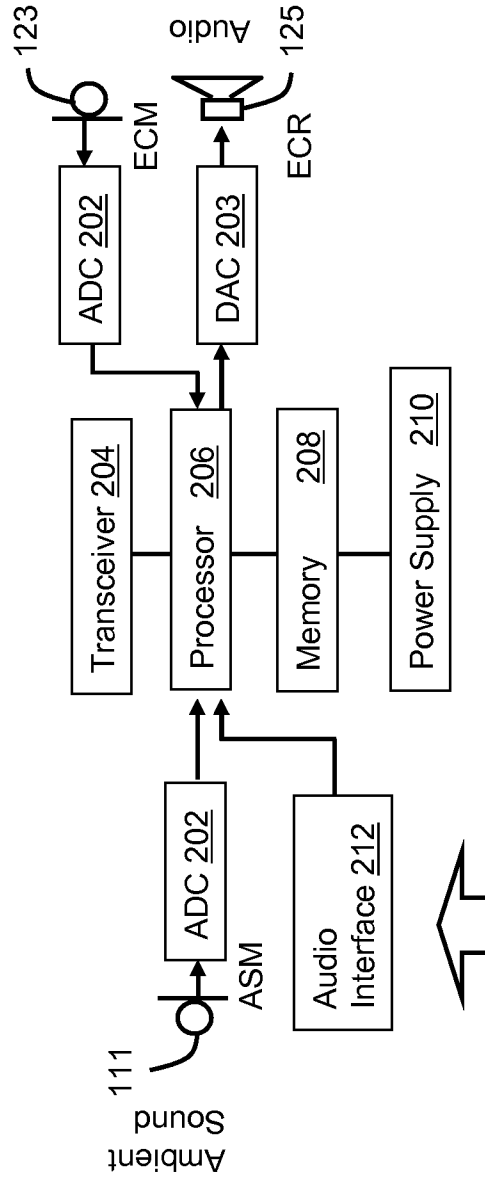


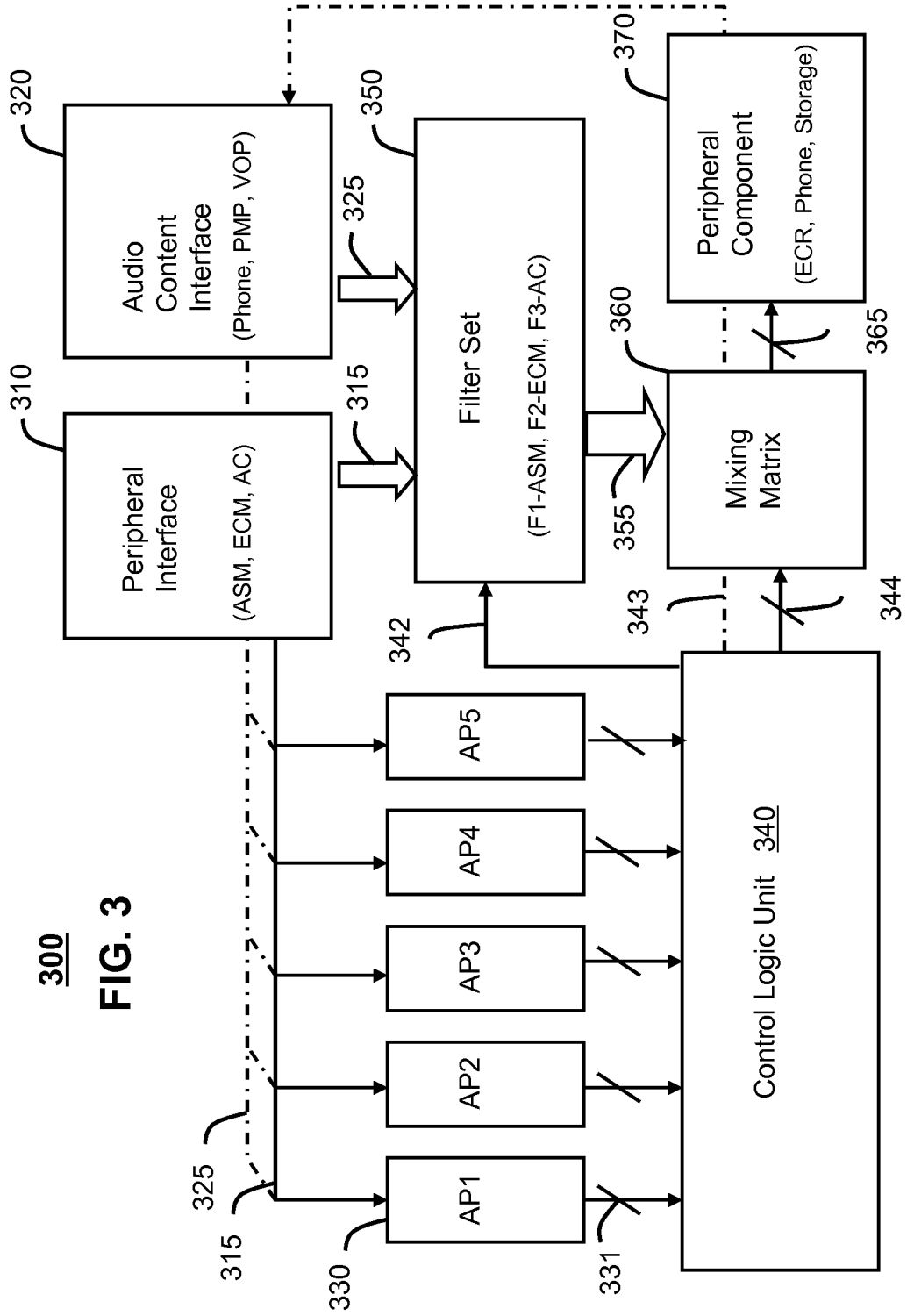
FIG. 1



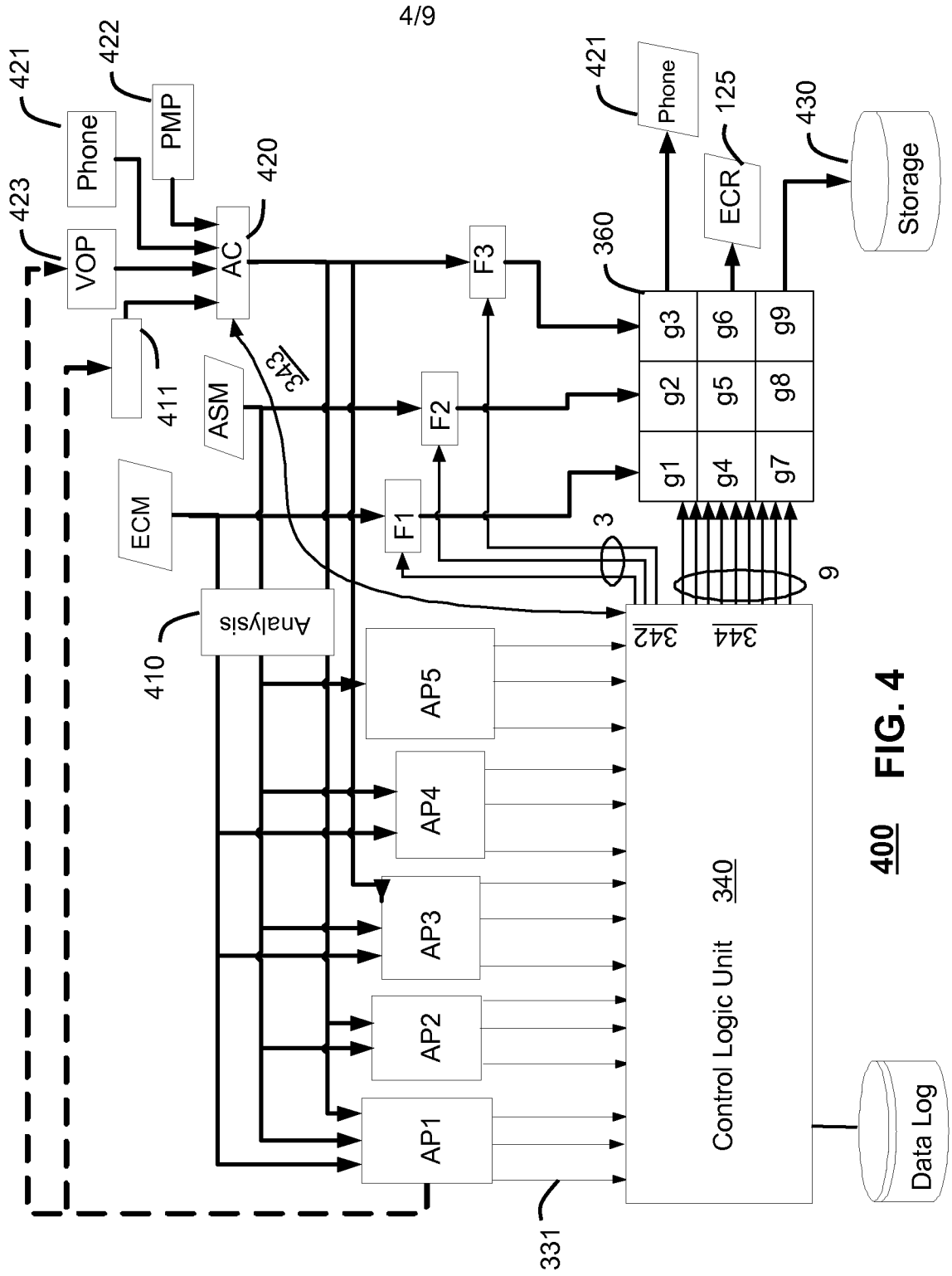
Audio Content (e.g.,
music, cell phone,
voice mail)

100

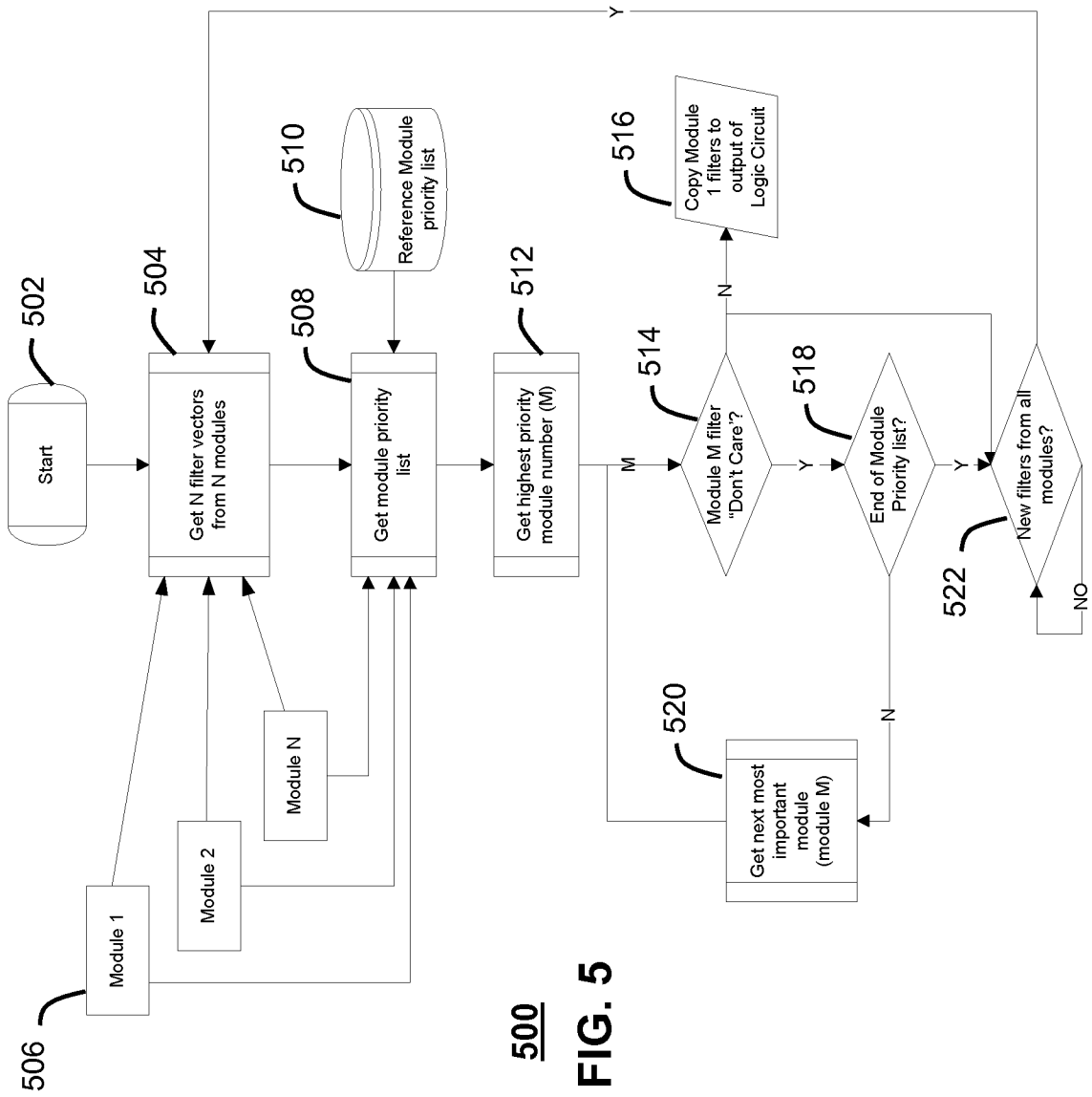
FIG. 2



300
FIG. 3

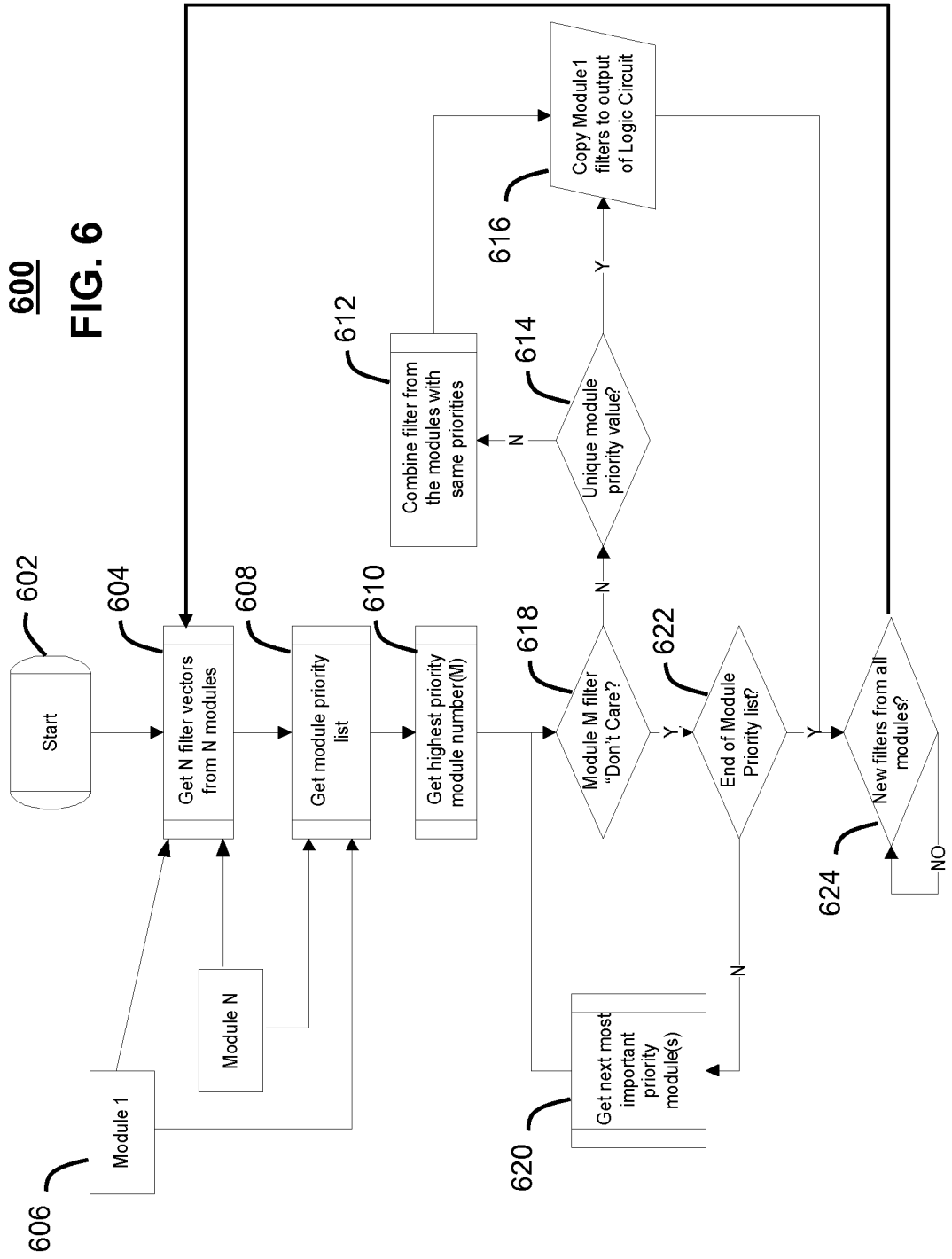


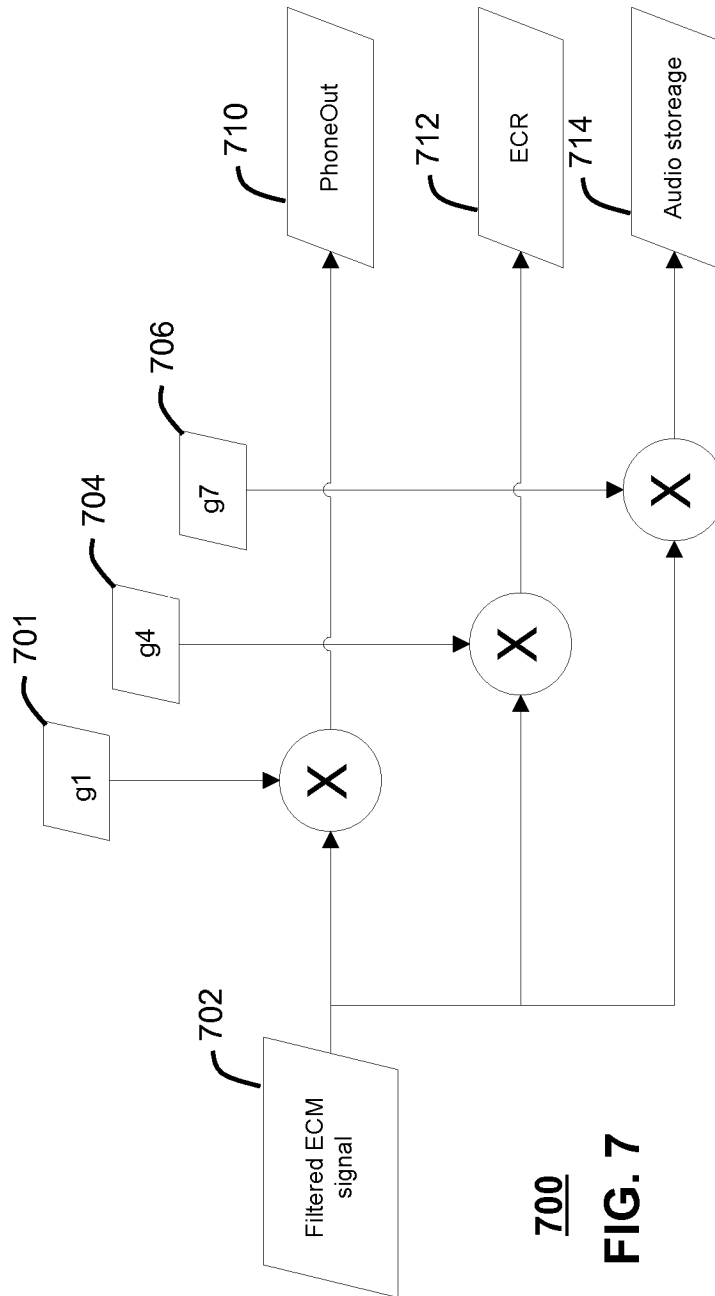
400 FIG. 4

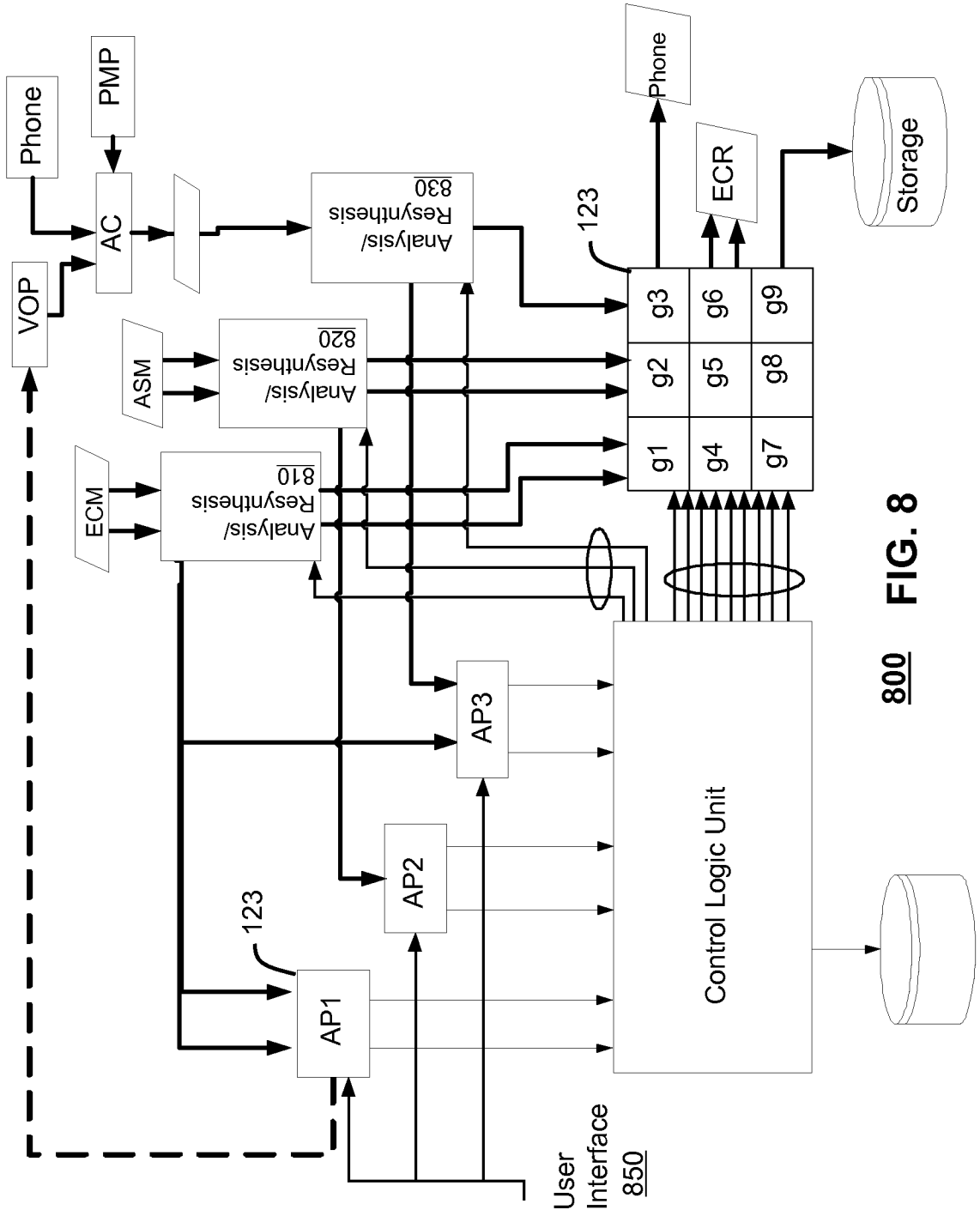


500
FIG. 5

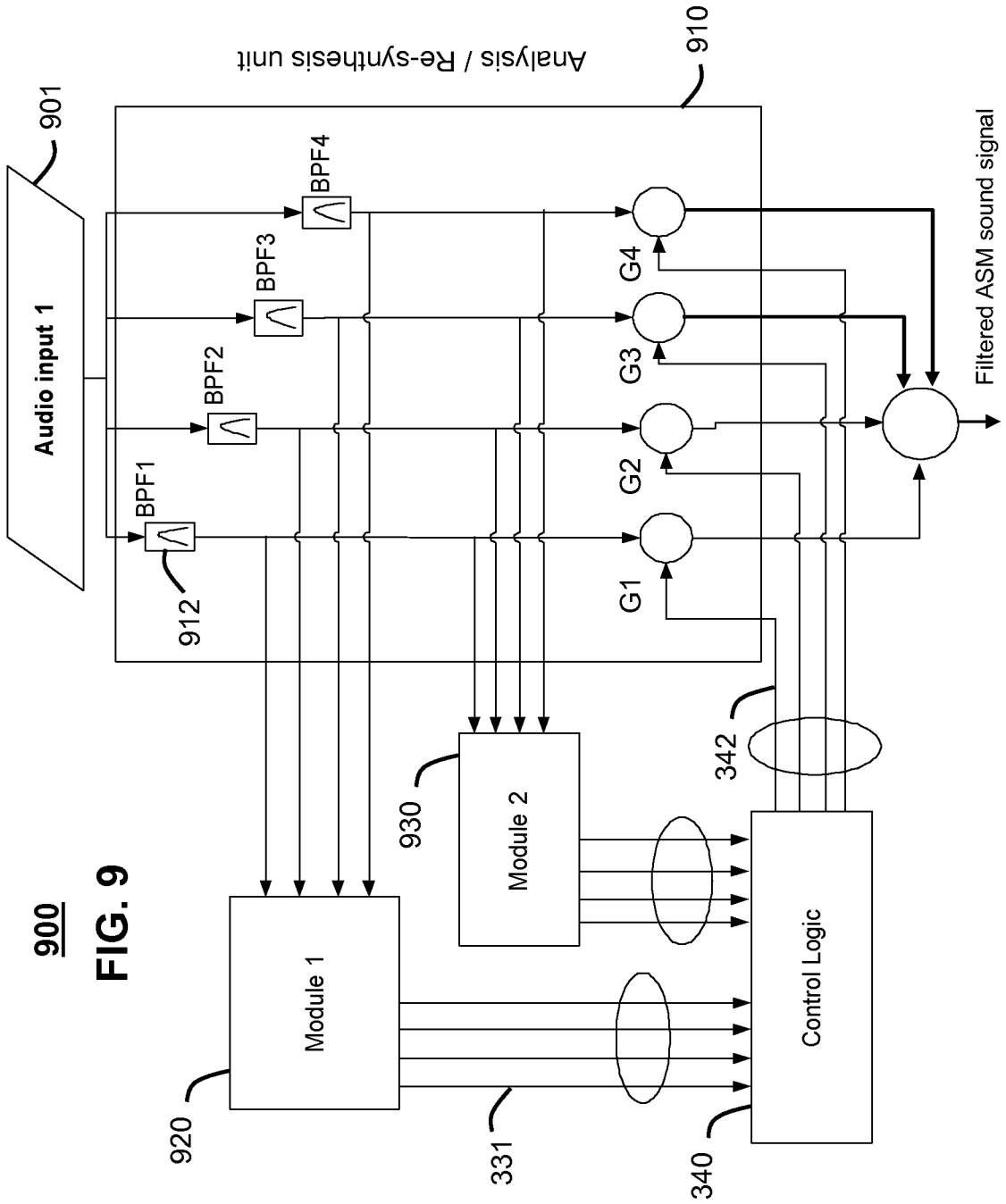
600
FIG. 6







800 FIG. 8



900
FIG. 9

INTERNATIONAL SEARCH REPORT

International application No.

PCT/US 08/73177

A. CLASSIFICATION OF SUBJECT MATTER

IPC(8) - H04R 5/02 (2009.01)

USPC - 381/309

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)
USPC - 381/309Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
USPC - 381/1, 150, 309 (keyword limited, see terms below)Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
PubWEST(PGPB,USPT,EPAB,JPAB); Google. Search Terms Used: hearing aid, earpiece, earphone, microphone, speaker, priority, row, column, matrix, filter, device, piece, object, assistance, signal, sound, process, condition, modify, mix, add, sum, logic, control, ear, audio, sound, ambient, event

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 6,023,517 A (Ishige) 08 February 2000 (08.02.2000) entire document, especially: Fig. 1, col. 1, ln 5-7, col. 3, ln 54-60, col. 4, ln 4-11, col. 5, ln 64-67	1-6, 11-20
Y	to col. 6, ln 1-44, col. 7, ln 29-33	7-10, 21, 22
Y	US 2002/0071583 A1 (Stetzler et al.) 13 June 2002 (13.06.2002) para. [0039]	7
Y	US 2007/0160243 A1 (Dijkstra et al) 12 July 2007 (12.07.2007) Fig.1 , [0013]-[0016], [0042]	8-10, 21, 22
A	US 2006/0078150 A1 (Knorr et al.) 13 April 2006 (13.04.2006) entire document	1-22

 Further documents are listed in the continuation of Box C.

* Special categories of cited documents:

"A" document defining the general state of the art which is not considered to be of particular relevance

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

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

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

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

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

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

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

"&" document member of the same patent family

Date of the actual completion of the international search

21 June 2009 (21.06.2009)

Date of mailing of the international search report

06 JUL 2009Name and mailing address of the ISA/US
Mail Stop PCT, Attn: ISA/US, Commissioner for Patents
P.O. Box 1450, Alexandria, Virginia 22313-1450
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Lee W. YoungPCT Helpdesk: 571-272-4300
PCT OSP: 571-272-7774