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APPLICATION NUMBER	FILING or 371(c) DATE	GRP ART UNIT	FIL FEE REC'D	ATTY. DOCKET NO	DRAWINGS	TOT CLAIMS	IND CLAIMS
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CONFIRMATION NO. 6535

32146
STEVE W. GOLDSTEIN
16390 MADOLENA PLACE
DEL RAY BEACH, FL33446

FILING RECEIPT

Date Mailed: 01/22/2007

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Applicant(s)

Stevem Wayne Goldstein, Delray Beach, FL;

Power of Attorney: None

If Required, Foreign Filing License Granted: 01/22/2007

The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is
US60/883,013

Projected Publication Date: None, application is not eligible for pre-grant publication

Non-Publication Request: No

Early Publication Request: No

** SMALL ENTITY **

Title

Personalized Hearing Measurement and Listening System (PHMLS.2):

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Application Data Sheet 37 CFR 1.76		Attorney Docket Number	
		Application Number	
Title of Invention	A Personalized Hearing Measurement and Listening System (PHMLS.2):		
The application data sheet is part of the provisional or nonprovisional application for which it is being submitted. The following form contains the bibliographic data arranged in a format specified by the United States Patent and Trademark Office as outlined in 37 CFR 1.76. This document may be completed electronically and submitted to the Office in electronic format using the Electronic Filing System (EFS) or the document may be printed and included in a paper filed application.			

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☐	Portions or all of the application associated with this Application Data Sheet may fall under a Secrecy Order pursuant to 37 CFR 5.2 (Paper filers only. Applications that fall under Secrecy Order may not be filed electronically.)
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Prefix	Given Name		Middle Name		Family Name		Suffix
	Stevem		Wayne		Goldstein		
Residence Information (Select One) ☒ US Residency ☐ Non US Residency ☐ Active US Military Service							
City	Delray Beach		State/Province	FI	Country of Residence i	US	
Citizenship under 37 CFR 1.41(b) i			US				
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City	Delray Beach			State/Province	FI		
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Application Data Sheet 37 CFR 1.76		Attorney Docket Number	
		Application Number	
Title of Invention	A Personalized Hearing Measurement and Listening System (PHMLS.2):		

Application Information:

Title of the Invention	A Personalized Hearing Measurement and Listening System (PHMLS.2):		
Attorney Docket Number		Small Entity Status Claimed	☒
Application Type	Provisional		
Subject Matter	Utility		
Suggested Class (if any)		Sub Class (if any)	
Suggested Technology Center (if any)			
Total Number of Drawing Sheets (if any)	21	Suggested Figure for Publication (if any)	
Publication Information:			
☐ Request Early Publication (Fee required at time of Request 37 CFR 1.219)			
☐ Request Not to Publish. I hereby request that the attached application not be published under 35 U.S.C. 122(b) and certify that the invention disclosed in the attached application has not been and will not be the subject of an application filed in another country, or under a multilateral agreement, that requires publication at eighteen months after filing.			

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Prior Application Status		Remove	
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
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Application Data Sheet 37 CFR 1.76		Attorney Docket Number	
		Application Number	
Title of Invention	A Personalized Hearing Measurement and Listening System (PHMLS.2):		

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Application Number	Country ⁱ	Parent Filing Date (YYYY-MM-DD)	Priority Claimed
			☒ Yes ☐ No
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Signature	/stevegoldstein/			Date (YYYY-MM-DD)	2006-12-31
First Name	Steven	Last Name	Goldstein	Registration Number	

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Abstract:

A Personalized Hearing Measurement and Listening System (PHMLS) is described. The PHMLS is a self contained Personalized Sound Field Monitoring system for recording, diagnostic and Safe Level Audio Playback comprising an active Canalphone (canaltrasciever) arrangement utilizing in-ear and exterior microphones, a minimum of one miniature loudspeaker and an active audio signal processor platform. Primarily intended for users with healthy auditory functioning, auditioned content and noise, environmental (ambient) sound levels are actively monitored to ensure that personalized reproduction levels inside the ear canal are within a safe and subjectively optimized listening level range. In addition, active compensation for environmental sound stimuli and auditory warning signal detection is undertaken using microphones both within and outside the ear canal, and system integrity is monitored to ensure against problems due to ill fitting of the Canalphone by the user. Furthermore, PHMLS allows for in-situ measurement of the user's outer ear anatomy and additional personalization is undertaken using an interactive setup. Communication of audio content and personalized auditory system measurements (e.g. HRTF's) are possible with a wireless transceiver in the Canalphone.

Background

1. Field of the invention

This invention pertains to sound reproduction, sound recording, audio communications and hearing protection using Canalphones designed to provide variable acoustical isolation from ambient sounds while being able to audition both environmental and desired audio stimuli. Particularly, the new audio system describes a method for active monitoring of the sound pressure level both inside and outside the ear canal to allow for enhanced spatial and timbral sound quality whilst maintaining supervision to ensure safe reproduction levels.

2. Description of related art

Since the advent of industrialization over two centuries ago, the human auditory system has been increasingly stressed to tolerate high noise levels it had hitherto been unexposed. Recently, human knowledge of the causes of hearing damage have been researched intensively, and models for predicting hearing loss developed and verified with empirical data from decades of scientific research. And 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. In fact, music reproduction levels and urban noise are antagonistic; we play our personal audio devices louder to hear over the traffic noise. And use of personal audio devices is rapidly increasing, especially in the younger generation who are suffering permanent hearing damage at increasingly younger ages.

Noise is a constant in industrialized societies, given the ubiquity of external sound intrusions, such as people talking on their cell phones, blaring music in health clubs, or the constant hum of HVAC systems in schools and office buildings. To combat the undesired cacophony, consumers are arming themselves with portable audio playback devices to drown out intrusive noise. The majority of devices providing the consumer with audio content do so using insert (or in-ear) earbuds, which delivers sound directly to the ear canal, generating levels sufficient to perceptually mask background noise even though the Earbuds provide little to no ambient sound isolation. With earbuds, personal audio reproduction levels can reach in excess of 100 dB (Fligor and Cox, 2004); enough to exceed recommended daily sound exposure levels in less than a minute and to cause permanent acoustic trauma (Price, 1981). Furthermore, rising population densities have continually increased sound levels in society. According to Berger (2003), 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 level to compensate for ambient noise.

To overcome the increase in ambient noise level, it is increasingly common to incorporate passive noise cancelling systems, by embedding the earphone loudspeaker in an acoustical isolation assembly, such as an ear plug (“insert-headphones”); as described in US patent nos. 4,677,679, 4,763,753, and 5,887,070. Such passive sound attenuation devices are often used in live-sound for monitoring a music-audio mix. However, for everyday use of listening to reproduced Audio Content (such as from a Personal Media Player or cell phone) these devices are potentially dangerous, as the user is isolated from ambient auditory warning devices, such as car horns and fire alarms. To overcome this limitation, US patent application no. 2006/0182287 describes an audio monitoring system using an ambient sound microphone mounted at the entrance to an occlude ear canal in an ear plug, and to simultaneously monitor the sound mix created by an ear-canal receiver and detected in the ambient sound field. In US patent application no. 2006/0182287, the relative ambient sound signal level and reproduced audio content level are manually adjusted by the user. The present invention improves such extant Ambient Sound Monitoring systems by adjusting the level of auditioned ambient sound automatically, on a frequency-selective basis; by monitoring the ambient sound field for warning sounds such as alarms and horns. If a warning sound is detected in the ambient sound field, audio content being reproduced by a personal media player (for example) may be attenuated on a frequency-selective basis so as to allow the user to hear the local warning sound. Alternatively, a warning message may be reproduced to alert the user to the presence of a local auditory warning device.

Headphones by their very nature are “one size fits all”; in terms of appearance but more importantly in terms of sonic prorogation. Conversely, human hearing is unique and changes dramatically with age and environmental factors. Furthermore, an individual’s left and right ear can have a significantly different hearing sensitivity.

So a sound reproduction system that can tailor the audio reproduction level to the hearing sensitivity of each of the users' ear, and yet also be tailored for the specific transduction device (i.e. the frequency response of the loudspeaker coupled with the ear of the user) allows for normalization of loudness within a given level range.

The Personalized Hearing Measurement and Listening System (PHMLS) described in the present invention brings a new degree of personalized full-band width general audio reproduction to the user and ensures that timbral equalization is possible using a multiband level normalization to account for inter-individual differences in hearing sensitivity. A technique which maintains the sound level within specific loudness boundaries for hearing aid use is described in US patent application 2002/0076072, and the PHMLS improves upon this technique of active gain control by personalizing the loudness boundaries to a specific individual dependant on preference and hearing thresholds.

To provide the compensatory process for differences in hearing sensitivity among consumers (or between the two ears of the individual), the PHMLS can determine a minimum threshold of audibility and maximum loudness comfort metric using a stand-alone, self-administered sound threshold measurement. This equalization can be determined by using swept or static pure tones or band-limited noise or band-limited music to account for the frequency dependency of hearing status and Canalphone frequency response. This self-test could be undertaken in conjunction with a computer interface, or as a self-contained system on the PHMLS device, with instructions provided from a stored voice messaging system.

This self-test system can be used to monitor the hearing status of the consumer, and prompt the user to seek consultation from a hearing healthcare professional should abnormalities or changes in hearing status be suspected. Extant approaches to measuring auditory sensitivity, such as disclosed in US patent no. 6,496,585 are administered with a computer, and lack the ability to monitor ambient sound level to instruct the subject to relocate if the background noise level.

An ambient noise level detector works in conjunction with a hearing sensitivity self-test system to ensure that a suitably high signal-to-noise ratio is attained before the audiogram is invoked, and would inform the listener to relocate to a quieter environment if the ambient noise level were deemed too high. The system disclosed in US patent 7,037,274 also monitors ear canal SPL whilst simultaneously conducting a hearing threshold audiogram administered using a loudspeaker on an ear-canal probe. However, the later system does not measure the audiogram in-situ in a self-contained portable system, whereas the present invention allows the user to relocate to a quieter environment to conduct the audiogram.

To maintain the reproduced Audio Content Sound Pressure Level within a range bounded the Uncomfortable Listening Level (UCL), yet also to keep the total sound pressure level exposure within safe limits, a Sound Pressure Level Dosimeter (SPLD) constitutes a major feature of the PHMLS.

The Dosimeter continuously monitors ear canal SPL and can upload an “Envirogram” history of SPL exposure to a central database server for analysis by audiologists and the user.

Standard practice for conducting noise surveys of occupational settings includes the use of dosimetry, which integrates the sound level over time and compares data to established damage risk criteria, such as those promulgated by the Occupational Safety and Health Administration (e.g. OSHA, 1983). However, unlike current sound exposure monitoring strategies, which simply monitor ambient noise over a given time period (typically a work-day of eight hours) and reset to zero at the beginning of the next time period, or are averaged across several time periods (as in an average weekly noise exposure), the Dosimeter in the present invention describes a process for estimating the time required for recovery from the sound exposure based on exposure history and scientific evidence of the physiologic basis for this process. Furthermore, the “always on”

The Ear Recovery Function is the algorithm calculating the decay in the SPL Dose describing the theoretical recovery from potentially hazardous sound exposure when sound levels are below Effective Quiet. “Effective Quiet” is defined as sound levels considered to not contribute to noise-induced permanent threshold shift (NIPTS); 76-78 dBA (Melnik, 1991) for broadband noise. In the current invention, a value of 78 dBA may represent the level that will not contribute to increasing or decreasing SPL Dose, but rather hold the SPL Dose steady.

Alternatively, the value defined as Effective Quiet may be frequency-dependent; Effective Quiet may be defined as 74 dB SPL for the octave band centered at 4000 Hz, 78 dB SPL for the octave band centered at 2000 Hz, and 82 dB SPL for the octave bands centered at 500 Hz and 1000 Hz. It is based on audiological research of growth and decay of temporary threshold shift (TTS), which is the temporary decrease in hearing sensitivity that arises from metabolic exhaustion of the sensory cells of the inner ear from exposure to high levels of sound. Sound exposure that results in a TTS is known to be sufficient to eventually result in a permanent hearing loss. The recovery from TTS is thought to reflect the improvement in cellular function in the inner ear with time, and proceeds in an exponential and predictable fashion. The Recovery Function estimates the time required for recovery from sound exposure based on exposure history and scientific evidence of the physiological basis for this process. Using either a visual or auditory message, the user can be informed of the time remaining until sound exposure levels may cause hearing damage.

The PHMLS detects when the Canalphone is inserted in the ear of a user. Unlike existing proximity detectors, such as described in US patent no. 6,542,436 for devices such as wireless telephone, the present invention does not use complex adaptive filtering systems to detect the presence of a reflecting surface, but instead functions by looking at SPL differences between the two microphones in the same Canalphone (i.e. the difference in SPL between the Ambient Sound Microphone and the Ear Canal Microphone): the differences will be smaller if the Canalphones are removed.

Also, the PHMLS can determine the distance between the two Canalphones using either an acoustic SONAR mechanism or by analyzing the signal strength of transmitted radio waves between each Canalphone. Therefore, the operating state of the system can be logged; such as if the user has the Canalphones inserted in their ear or not.

With the SPL Dosimeter, the SPL in the ear canal affected by Audio content reproduced with the Ear Canal Receiver can therefore be kept within a user-personalized comfortable listening level range and can give a uniform sound quality for different listeners, whilst mitigating the risk of hearing. Furthermore, with a multiband analysis system the PHMLS actively monitors ambient (environmental) sound levels to ensure that the signal-to-noise ratio of the sound within the occluded ear canal(s) is maintained at a level conducive for good intelligibility of the audio content, yet also at a personalized safe listening level. Active control of reproduced level falls under the category of Automatic Gain Control (AGC).

AGC systems are often used in speech audio signal processing systems, such as with telephones- as described in US patent Nos. 5,146,504 and 6,363,343. The SPLD informs a subsystem of the PHMLS to affect the gain of the reproduced Audio Content via an AGC to keep the sound level in the users' occluded ear canal(s) at a safe value. The PHMLS differs significantly from extant AGC-for-speech systems by assuming that full-bandwidth Audio Content is reproduced, and therefore a full audio bandwidth SPLD measurement is given, whereas the suggested embodiment for systems such as given in Patent 6,363,343 do not consider frequencies above 4 kHz in the level determination system.

Furthermore, the present system may utilize a "look ahead" analysis system for reproduction of pre-recorded audio content, using a data buffer to offset the reproduction of the audio signal. This look-ahead system allows the AGC circuit to analyze audio content either received from an external media device, or detected with the ambient microphones, in-situ before it is reproduced and can therefore reduce the problems of timbral distortion caused by, for instance, attenuating high level transients.

Car audio systems often use AGC to increase the reproduction level of Audio Content in response to increased ambient sound levels (such as from engine noise); as disclosed in US patent no. 5,081,682. These systems can therefore maintain a signal-to-noise ratio, and therefore maintain a good intelligibility, in spite of increases in background noise levels. The problem with these loudspeaker-based systems is that the noise level estimation also includes the contribution from the reproduced Audio Content and the noise level is overestimated. The affect of this is that reproduction level increases due to this overestimated noise level; a phenomenon called "gain chase". To overcome this problem of gain chase, solutions using complex adaptive filters are used, as suggested in US patent no. 6,529,605.

The present invention does not suffer from the problem of “gain chase” because the reproduced Audio Content transducer (i.e. the loudspeaker within the ear canal) is acoustically decoupled from the ambient noise sensor microphone on the outside of the Canalphone. Furthermore, the PHMLS differs from existing “signal-to-noise level maintenance” systems by considering psychoacoustic functions such as level dependant equal loudness curves (i.e. “Fletcher – Munson” curves; Blauert, 1998) to maintain a perceptually equal signal-to-noise loudness ratio, and yet ensuring that dangerous SPL exposure limits are not exceeded.

It is increasingly common practice for circum-aural headphone audio presentation to use an Active Noise Cancellation (ANC) system to reduce the transmission of local ambient noise to the ear canal, and thereby allow reduce the overall sound exposure to the listener’s auditory system. The basic premise for such ANC systems is to generate an inverse phase soundwave at the eardrum for countering low frequency noise. An ANC system is described in US patent nos. 5,305,387 and 5,181,252; wherein a microphone placed in front of the left and right supra-aural loudspeakers. Besides the cumbersome circum-aural headphone design, the major limitation of phase-inversion ANC systems is that their efficacy is greatly reduced above about 1 kHz, and below this level the degree of noise cancellation is, at best, only a modest 15 dB at 200-450 Hz (Pawelczyk, 2002).

Due to the limitations of extant ANC systems, the Phase Inversion ANC system is not present in the preferred embodiment of the PHMLS as a means to reduce transmission of ambient noise to the inner ear canal, as the sound attenuation (i.e. Noise Reduction Rating) provided by the Earphone assembly is approximately 25 dB from 125 Hz- 8000 kHz. The Canalphone forms an acoustic seal in the ear canal, and by electronically transmitting certain sounds through the ear seal using the Ambient Microphone and Ear Canal Receiver, this NRR can effectively changed in-situ by the listener.

This may be particularly beneficial for live sound performances with amplified sound, where the SPL in the concert hall may be approximately 100 dB, yet the listeners preferred listening level is 85 dB, so the 25 dB attenuation by the acoustic seal alone would be too large. Alternatively, the user may select a net gain by the PHMLS system, for instance, to enhance speech intelligibility whilst seated at the back of a theatre.

ANC is, however, a component of a system to attenuate the level of transmitted or recorded user auscultation (i.e. sounds created by the user, such as coughs or sneezes). This auscultation attenuation system also reduces the level of unwanted user-generated sounds by attenuating those frequencies of the auscultation. A multiband ANC system is disclosed in US patent no. 6,801,629, which provides a means to automatically attenuate the transmission of loud transients through an earplug, which might otherwise cause hearing damage, using a microphone mounted in the open end, and a miniature loudspeaker mounted in the occluded ear-canal end. The problem with such a compression technique is that of the response time of the ANC system, and secondly that the release-time of the ANC system may reduce the level of ambient sounds which the user wishes to hear.

The present invention improves upon extant ambient sound ANC techniques with the inclusion of a look-ahead ambient SPL monitoring system; wherein the signal monitored at the entrance to the occluded ear canal is stored in a circular data buffer, with data memory either within the canalphone or on a remote device, and is dynamically processed to reduce the level of sudden transients in time, and can return the gain to its previous setting by monitoring for an end of the transient.

In a particular user-selected operating paradigm called “Learn Mode”, the user can expose sounds to the PHMLS to “teach” the device about sounds that are pertinent to the user, and to process the ambient sound field to either enhance or reduce the transmission of these sounds to the PHMLS user. These taught sounds are called sound signatures. In addition to the user-provided library of specific sound signatures, the PHMLS has access to an online library of sounds; such as car-horns and sirens from emergency vehicles are an obvious example of sounds which the PHMLS would allow to be enhanced (in terms of intelligibility, for example by attenuating other sound frequencies where the target sound is at low level) and “pass-through”, as well as the pre-amble chime on information systems such as at airports or train stations. There are an equally large number of examples of sounds that the PHMLS could attenuate; such as a dentist drill- which could be attenuated on a frequency-selective basis to prevent hearing damage from excessive exposure to both the dentist and patient.

Prior art disclosed in US patent application no. 2006/0140416 discusses a source proximity detecting assembly contained within an earplug for determining if a dominant source is active within approximately 3 meters of an individual. This prior art uses two microphones spaced apart mounted on the exterior of the earplug. The outputs of the two microphones are used to detect the angle of incidence of the source and to predict source distance based on the delay time of any detected echoes (it is assumed that the premise is that for close sources the echoes will arrive much later and weaker than the direct sound, compared with the far-source condition).

The Source Proximity Detector is a particular aspect of the invention that can allow sound from a nearby location to be reproduced by the inner loudspeakers of the PHMLS. Therefore, if someone nearby wishes to speak to the PHMLS user, the user can hear them. Unlike prior art, this system functions by looking at sound level differences between the four microphones within the pair of Canalphones; i.e. the two ASM’s and the two ECM’s. Due to sound reflections, the head shadowing affect is stronger for close sources than far sources, so the difference in SPL at the two ears will be larger for close sources than farther sources.

As well as addressing the problem of potentially hearing damage SPL, the PHMLS improves the overall sonic experience by optimizing the spatial auditory cues in the audio content. Enhancement of spatial sound fidelity is described by the process of auralization; the perceptual creation of a real or artificial acoustic space. For headphone audio presentation in general, this involves recreating the directional characteristics of both the sound source and those sounds that contribute to a listeners feeling of the recording environment (ambiance).

Extracted ambience sound components (i.e. recorded reverberation) can then be filtered by the users personalized or semi-personalized HRTF's to create the impression of enveloping reverberation.

The system described in US patent application 2004/0076301A1 describes a Canalphone auralization system employing several microphones and a head-tracking device to achieve a high sense of spatial auditory realism by simulating the effect of the listeners' head in an acoustic space. A similar approach uses a number of assumptions about the user's anatomical ear features to increase the sense of realism and add a feeling of the virtual sound source being located outside of the listeners' head. Such features are the directional-dependant acoustic filtering effect of human head and torso, which combined are called the Head Related Transfer Function (HRTF). Use of HRTF-based filtering in conjunction with acoustic space simulation can be used to process a sound and control its perceived spatial location, and to give a sense of an immersive sound field as might be perceived in a natural acoustic space such as a concert hall. Naturalness of imagery in sound reproduction is a highly esteemed sound quality issue, and has been shown to correlate highly with preference in formal listening tests (e.g. Berg and Rumsey, 2000).

The inherent problem with such extant binaural auralization systems just described is that of relating the users specific anatomical features to those parameters in the system. In other words, the efficacy of these binaural auralization systems is dependant on the degree of similarity between the users (personalized or semi-personalized) HRTF and the direction dependant filtering coefficients used by the particular auralization system. Measuring an individual's HRTF is traditionally a very time consuming process involving a trained engineer and the use of high quality microphones located in the subjects ears, and reproducing an audio test signal from loudspeakers located around the subject in an anechoic chamber (Blauert, 1997).

In a particular embodiment of the PHMLS, HRTF's can be measured and determined using the exterior microphones of the Canalphone, and reproducing the test signal from an additional loudspeaker around the listener (which may be inaudible to the listener due to the passive attenuation of the Canalphone). This external measurement loudspeaker could be the external loudspeaker in the PHMLS (i.e. it could be detachable), or alternatively the user could use their own loudspeaker, and would enter information such as manufacturer and model number to allow for equalization by the HRTF measurement system using a large database of loudspeaker responses. Unless temporal truncation was applied, this measured HRTF would include the room impulse response, so the user could measure their HRTF in a preferred listening environment. The general problem with circum-aural and supra-aural headphones rather than in-ear Canalphones (as with the PHMLS) is that the HRTF-based spatial signal processing methods must take into account the additional filtering effect of the ear concha.

The PHMLS provides a novel means to undertake headtracking, and therefore use the personalized HRTF's to process the Audio Content in a way to auralize the recorded sound source. This differs from extant headtracking systems; which may mount a pair of loudspeakers on the user, and detect head movement using a fixed microphone (US patent no. 6,961,439), or using an ultrasonic audio transmission from a pair of transducers to a pair of microphones, or using a magnetic-based (US patent no. 5,812,257) or inertial-based (US patent no. 5,645,077) tracking system. In the present invention, the headtracking is accomplished by reproducing a test signal (which may be recorded music) from a stationary loudspeaker, such as in a cell phone or a laptop loudspeaker. The Ambient Sound Microphone in the left and right Canalphone of the PHMLS can then be used to detect head motion of the user, by an analysis of changes in the relative output level and phase of the two microphone signals. This approach is similar to detecting motion of a nearby sound source (as disclosed in US patent application no. 2006/0140416; which uses two microphone mounted in the same Canalphone); but the advantage of using the two Ambient Sound Microphones is that they are far apart, so will be more sensitive to head movement than if the microphones were closer together (as the head-shadowing effect is greater for the spaced ASM's, so inter-microphone signal changes are greater for a given head movement).

Besides physical HRTF measurement, a novel measurement system involving correlating photographs of the user's ears to a library of measured HRTFS with associated photographs is part of the PHMLS. The user would upload a photograph of their ear to a website, and an algorithm would select features of the ear (such as concha size) and match the users ear to a "closest-fit" ear in the HRTF online database, and would then provide the user with a set of binaural HRTF coefficients from the empirically measured "closest-fit" ear.

Besides stored audio content reproduction, the PHMLS is well suited for peer-to-peer (ear-to-ear) speech communication. A particular embodiment for the invention is for wireless two-way communication with either a group of people (such as for emergency service use, particularly in high-noise environments such as EMS, race car events or avionics) or with another person. A one-way operating mode may also be of use for audition by a supervisor, such as at a call-center, or by translators at international conferences etc. A Virtual Private Network (VPN) may be used to enable secure communication with another person (or group of people). Communication with multiple VPN's, for instance by a supervisor, may be used in a way whereby the different groups are spatially separated using personalized HRTF filtering to increase intelligibility for the supervisor. Wireless VPN communication using earphones-transceivers is discussed in patent WO 2006/104887, and the present invention improves upon the reproduced audio fidelity by monitoring ambient sound levels to maintain a suitably high signal-to-noise ratio in the ear canal, and also monitors ear-canal SPL to ensure against potential hearing damage. Furthermore, the present invention allows for personalization of the VPN using personalized or semi-personalized HRTF's to spatialize the perceived direction of the other individuals relative to each other, using GPS or transmitted signal strength cues to determine the relative location of other individuals.

Using either the Ear Canal Microphones or the Ambient Sound Microphones, the binaural advantage of the PHMLS increases speech intelligibility for speech communication. Spatial release from masking (Shinn-Cunningham et al, 2001) is the cognitive unmasking process of making it easier to listen to individual sound sources by spatially redistributing them. Spatial release from masking applies not just to speech sound scenes but also to musical sound scenes. For instance, by spatially redistributing the reverberation, the sound image corresponding to the recorded musical instrument becomes more spatially focused. The ability to use personalized or semi-personalized HRTF's, measured in-site with the PHMLS, allows for spatialization of multiple voices or auditory cues from external devices such as cell phones and warning systems.

Another application of the system is for online gaming, whereby a virtual acoustic environment is shared by a number of different game players in different locations.

Another use of the PHMLS for ear-to-ear communication is for music recording and live performance to allow a musician to monitor a customizable stereo sound mix at a comfortable listening level, with a mix tailored to their personal hearing sensitivity. The absence of a visible microphone and a wireless in-ear transceiver (i.e. combined transmitter and a receiver) would allow for a degree of personal and aesthetic freedom unattainable by existing handheld, lavalier, or headset microphones, and would have the technical benefits of greatly reducing acoustic feedback due to the isolation of the inner microphone. US patent application no. 2006/0182287 describes an audio monitoring system with a suggested application for allowing on-stage musicians to hear an audience using an ambient sound microphone mounted in an ear plug, and to simultaneously monitor the sound mix created by the musician ensemble using an internal loudspeaker receiver in the occluded ear canal.

This sound mix could be different for each musician, who may desire a certain mix from the soundboard as well as a mix from ambient microphones to enable interaction with the crowd. DJ's would also benefit from a hands-free in-ear monitoring system, as reproduction levels in clubs can exceed 100 dBA (Lower and Davis, 2000) and DJ's have to monitor the mix at very high levels to "hear over" the club sound. Active monitoring of SPL Ear Canal using the Ear Canal Microphone ensures PHMLS that the PHMLS user is not exposed to unsafe levels, and also that a signal-to-noise ratio conducive to high intelligibility is maintained.

The present invention improves upon patent application no. 2006/0182287 with a subsystem that detects the presence of a second PHMLS-user (i.e. a musician on stage) by a first PHMLS-user, and the level of the second PHMLS user's instrument is increased in the sound mix received by the first PHMLS user (and vice versa; the level of the first PHMLS user's instrument is increased in level for the mix reproduced by the Ear Canal Microphone of the second PHMLS user). Furthermore, using personalized or semi-personalized HRTF filtering, the musicians are informed of the direction of the other musicians by filtering the signal from the other musician with an HRTF filter corresponding to the direction of the other musicians.

Furthermore, the PHMLS utilizes a system to monitor ambient sound level using a microphone at the entrance to the occluded ear canal, to ensure that the signal reproduced by the receiver in the same occluded ear canal has a suitably high dynamic range conducive for good intelligibility: if the ambient noise level increases, the level of the reproduced audio content correspondingly increases. Furthermore, the SPL in the occluded ear canal is monitored to ensure that reproduction levels will not cause hearing damage. Patent no. 2005/0,254,665 also describes a system for monitoring SPL in the ear canal (or the entrance to it). However, Patent no. 2005/0,254,665 differs from the present invention, as there is no continuous monitoring of SPL Dose exposure to the listener, whereas SPLD Dose is monitored and logged to ensure against exposure to SPL in the PHMLS.

Hearing-aid wireless loop systems (“loops”) are increasingly common, and in some large cities such as London are required by law for communication in public environments like taxis and ticket offices. Using a device to convert the transmitted radio waves used by the loop systems, hearing aid loops can be transmitted to the PHMLS using Bluetooth, or alternatively the hearing aid loop receiver or transceiver (i.e. to allow communication with other people wearing hearing aids with loop system receivers) could be located within the PHMLS Canalphone assembly. The loop-system voice message can be convolved with the user personalized or semi-personalized HRTF to give spatialize the message, such as giving the impression it is coming from a particular direction, and therefore increase intelligibility of the message by spatially separating it from other reproduced audio content. Furthermore, audio content being reproduced at the same time could be paused or attenuated to increase intelligibility of the loop-system voice message.

Ear-to-device (or device-to-ear) communication is likely to be an increasingly common phenomenon, as speech-to-text and text-to-speech processing systems improve. The present invention describes a means to allow the PHMLS user to communicate with a remote computer-based system to allow for two-way control and reading (responding via speech to text) of email, SMS text-messaging, news and weather reports etc. Also, the PHMLS user can be informed of operating conditions of the device itself, such as issuing a warning sound to indicate low-battery or faulty aspects such as microphones clogged by ear detritus. These messages may be processed with the users personalized or semi-personalized HRTF to increase intelligibility, and/or audio content being reproduced at the same time could be paused or attenuated.

The speech-to-text subsystem would be particularly useful for dictating, such as for doctors’ diagnoses and prescriptions or for audio diaries such as Web Logs (Blogs). This could also be an aid for automatic or manual language translation, whereby a translation algorithm may be activated using a user-customizable trigger sound received by the Ambient Sound or Ear Canal Microphones. A particular aspect of the PHMLS which complements speech dictation is an “always on” mode, whereby the local ambient sound is always recording to a circular data buffer, so a user-activated recording will include a preamble of the last 15 seconds or so.

An increasingly common use of ear-to-device technology is that of Auditory Displays for auscultation (i.e. listening to sounds made by the body). The most frequent example of this is electronic stethoscopes for monitoring a patient's heartbeat. The electronic stethoscope described in US patent no. 7,006,638 suggests using a wireless headphone receiver to reproduce signals from a stethoscope on a patient, and uses customizable signal filtering to process the signal so that the physician, anesthesiologist etc. hears the heartbeat in a familiar way, as was learnt. Patent no. 7,006,638 suggests that the signal is also processed with HRTF filters to spatialize different aspects of the detected heartbeat, such as heart murmurs, to increase intelligibility.

A particular embodiment of the present invention improves on extant auscultation devices by actively monitoring the ambient sound field using the Ambient Sound Microphone, located the entrance to the ear canal, for two purposes: Firstly to ensure that an adequate signal-to-noise ratio is maintained (the Ear Canal SPL is also monitored using the Ear Canal Microphone to ensure against Noise Induced Hearing Loss). And secondly, actively monitoring the ambient sound allows local "signature sounds" to be transmitted through the Canalphone to the PHMLS user(s) - such as audio warning sounds from other auditory display devices in the users local environment (e.g. fire-alarms). In the second case, these "signature sounds" are recognized by a sound recognition system which are taught the signature sounds either by the user or from an online database.

This is particularly advantageous in high noise environments, such as in Mobile Army Service Hospitals or in helicopters. The PHMLS can be used by doctors in remote locations, in conjunction with a PHMLS user in a local operating room to allow for remote operations or consultations to be undertaken with a high degree of acoustic realism.

Furthermore, the PHMLS can be used to simultaneously monitor multiple auditory display devices, such as pulse oximeters, and can additionally normalize their pitch and loudness response to a timbre the user is familiar with, and process them with the PHMLS user's unique HRTF and hearing response to further increase intelligibility of their auditory scene.

A GPS system can be used in conjunction with the PHMLS to help direct a user to a specific location, which can be enhanced using spatial signal processing to direct the user in a particular direction or provide a warning for a hidden or dangerous object (e.g. an out-of-bounds area or one containing unexploded ordinance). Furthermore, this GPS information and a unique PHMLS user identify can be transmitted with a wireless data transmission such as Bluetooth. Therefore, other PHMLS users with access permissions can locate the first PHMLS user, and/or informed if this first person is close to the second person (i.e. a node-to-node alert). This could be beneficial for parents monitoring their children's whereabouts, or for friends to meet at social events such as concerts.

Background noise can greatly reduce the intelligibility of a two-way speech communication device. A common way of dealing with background noise is with a user-controlled or automatic "push-to-talk" button.

Other prior art (EP 1503 368 A1) uses facial expressions, bone or throat vibration for detecting if the communication device user is speaking, and activates this “push to talk” button. Besides being cumbersome to use, such speech detection methods use an additional transducer (i.e. a microphone) for the local speech sensor. In a particular embodiment, the present invention can be used to detect PHMLS user speech with a microphone within the occluded ear canal, and this microphone can also be used to receive the user speech for transmission to a second person. This approach differs from VOX triggering circuit of US patent no. 6,175,633, which detects user speech independently in each of a pair of earphone microphones, by instead comparing the output of both ear canal microphones using a system to ensure against “false positives” whereby nearby speech may incorrectly trigger the VOX circuit (i.e. which should only be triggered by user speech).

The acoustic isolation of the inner microphone allows for an improvement in signal-to-noise ratio for the signal that is transmitted from the PHMLS. Furthermore, by combining the outputs of the Ear Canal and Ambient Sound Microphones in both the left and right Canalphone, directional acoustic beamforming is possible. This differs from extant art such as disclosed in patent 2006/0140416, which uses beam-forming to identify a nearby dominant sound source by comparing the SPL monitored by two closely-spaced microphones in just a single earphone assembly. The combined acoustic sensitivity to sound can be tailored for a particular direction, and the effect of a local noise source can be reduced by reducing sensitivity in the direction of the noise source and thereby increasing the received signal-to-noise ratio and increasing intelligibility of the audio signal transmitted from the Canalphone using a wireless data communication system (e.g. Bluetooth).

Imperfections in the acoustic seal in a canal phone can cause ambient sound to leak into the ear canal, and thereby increase the SPL in the ear canal, and increase SPL Dose. US patent application no. 2006/0204014 discloses a method to test if a probe microphone forms a proper seal in the ear canal. This is accomplished by measuring the ear canal SPL at low-frequencies; as if the seal is compromised, the low frequency SPL (usually <500 Hz) is decreased (the SPL is compared with a previously measured low frequency SPL where the seal was intact). The present invention improves upon this seal-leak detection system by continuously monitoring ear canal SPL using the Ear Canal Microphone, as well as monitoring for sound leakage from the Ear Canal Receiver (ECR) loudspeaker using the Ambient Sound Microphone to detect sound components correlated with that signal radiated by the ECR.

Otoacoustic emissions (OAEs) are low-level acoustic signals that emanate from the cochlea and can be detected by a microphone in the ear canal (Kemp, 1978). Distortion product and transient evoked otoacoustic emissions are the most studied. These OAE signals are generated as a byproduct of the active mechanisms associated with the processes of hearing. OAEs can be elicited in the vast majority of ears with normal hearing sensitivity, and are absent in ears with greater than a mild degree of cochlear hearing loss (Lonsbury-Martin, et al., 1987). Present OAEs are thus thought to reflect normal cochlear function.

Screening for normal hearing function in difficult-to-test populations has utilized OAEs; universal newborn hearing screening programs very often make use of OAEs to determine if a newborn's hearing is likely to be normal, or if they need a more thorough hearing evaluation. In recent years, studies have shown that OAEs change in response to insults to the cochlear mechanism from noise and from ototoxic medications, prior to changes in the pure-tone audiogram (Hall and Lutman, 1999; Katbamna, et al., 1999). In a study by LePage and Murray (1998), 80 dB transient clicks were presented to 1724 people and their OAEs were recorded. Subjects were grouped according to their reported noise exposure histories (including groups separated according to recreational noise exposure, occupational noise exposure, and concomitant recreational and occupational exposures). An additive effect of reduction of OAEs was observed in groups exposed to recreational and/or occupational noise.

OAEs have not found their way into occupational hearing conservation programs, as the OAE has yet to successfully predict absolute hearing threshold, and due to inter-subject variability. However, it has been shown that OAEs are highly repeatable within the same subject over time (Dreisbach, et al., 2006). In PHMLS users with normal cochlear function, a baseline OAE measure could be obtained at registration of the device, and subsequent OAE measures could be made at intervals following sound overexposures or at scheduled intervals to monitor cochlear status. If OAEs in the individual PHMLS user differed from baseline, the user could be prompted to repeat the threshold equalization measure and/or seek medical attention. The threshold test will be invoked with the user's consent, using the Ear Canal Receiver and Ear Canal Microphone to generate and monitor an ear canal SPL at a given frequency, and using the Ambient Sound Microphone to ensure that background noise levels are low enough to conduct the audiogram.

As previously discussed, the system disclosed in US patent 7,037,274 also monitors ear canal SPL whilst simultaneously conducting a hearing threshold audiogram administered using a loudspeaker on an ear-canal probe. However, the later system does not measure the audiogram in-situ in a self-contained portable system, whereas the present invention allows the user to relocate to a quieter environment to conduct the audiogram.

3. Brief summary of the invention

The present incorporates an Ambient Sound Monitoring system, whereby sound in the local environment of the user is monitored by microphones at the entrance to an occluded ear canal. The same occluded ear canal is exposed to audio content reproduced from an ear-canal receiver, which may reproduce audio content from a personal media player whilst the Personalized Hearing Measurement and Listening System (PHMLS) simultaneously monitors the ambient sound field for warning sounds. If a warning sound is detected in the local environment, the reproduced audio content is attenuated automatically, on a frequency-selective basis. Alternatively, a warning message may be reproduced to alert the user to the presence of a local auditory warning device.

The Personalized Hearing Measurement and Listening System (PHMLS) introduce a new degree of personalized audio delivery. Anatomical measurements of the PHMLS user's ear allow for a measurement and determination of the personalized Head Related Transfer function. This can be measured by reproducing a test audio signal from a loudspeaker (which may be the Ambient Sound Receiver in the PHMLS assembly) in the proximity of the listening, and measuring the Sound Pressure Level (SPL) at each of the users ear canal using the Ambient Sound Microphone (ASM). Alternatively, the PHMLS provides a means to determine the semi-personalized HRTF by using a remote server to analyze photographs of the users ears for outer ear structural properties that affect the HRTF. The Ear Canal Receiver and Ear Canal microphone can be used together to measure the persons unique occluded inner Ear Canal Transfer Function ECTF, which is often taken in combination with the HRTF to provide a high spatial fidelity auralization experience by filtering audio content signals with the HRTF's.

Furthermore, personalization applies to tailoring sound reproduction to the users' hearing sensitivity. This measurement of unique features of a users auditory system are undertaken with a self-contained system comprising the following fundamental subsystems: A subsystem to measure sound pressure level within the occluded ear canal to (a) determine hearing sensitivity and the Ear Canal SPL which is preferred for a given frequency and a particular Ambient SPL (Ambient SPL is measured using the Ambient Sound Microphones, located at the entrance to the external ear canal).

(b) Using a feedback system, such as a graphical user interface, or with buttons or a dial on the PHMLS Canalphone, the user can provide a continuous feedback to a control system to ensure that occluded Ear Canal SPL is low enough to ensure against Temporary Hearing Threshold Shift; and (c) spectrally equalize the reproduced sound to compensate for the listeners hearing threshold curve and the Canalphones electroacoustic spectral SPL response; (d) An active subsystem to process Audio Content according to properties unique to the personalized features of the users auditory system, such as Preferred Listening Levels and Ear-Canal-Transfer-Function; and (e) a passive acoustic seal in the ear canal, within which the two microphones and at up to two loudspeakers are mounted.

In a preferred embodiment of the system, a Sound Pressure Level Dose (SPLD) is continuously and simultaneously monitored in the Ear Canal using the Ear Canal Microphone, and accumulated (it is not reset at the end of every day). "SPL Dose" is a measurement indicating an individual's cumulative exposure to sound pressure levels over time. This term is derived from the term Noise Dose, but unlike Noise Dose, SPL Dose accounts for exposure to Audio Playback as well as exposure to Environmental Noise. An SPL Dose is expressed as a percentage of some maximum time-weighted average for sound pressure level exposure (i.e. a SPL Dose of 100% indicates the user has reached the time-weighted average SPL exposure limit). Although the time period for Noise Dose measurements usually corresponds to the length of a workday (usually 8 hours), SPL Dose is cumulative and therefore has no time period.

The SPL Dose increases during Audio Playback and for moderate to high Environmental Noise levels, above Effective Quiet. “Effective Quiet” is defined as sound levels considered to not contribute to noise-induced permanent threshold shift (NIPTS); 76-78 dBA (Melnik, 1991) for broadband noise. In the current invention, a value of 78 dBA may represent the level that will not contribute to increasing or decreasing SPL Dose, but rather hold the SPL Dose steady. Alternatively, the value defined as Effective Quiet may be frequency-dependent; Effective Quiet may be defined as 74 dB SPL for the octave band centered at 4000 Hz, 78 dB SPL for the octave band centered at 2000 Hz, and 82 dB SPL for the octave bands centered at 500 Hz and 1000 Hz. The SPL Dose decreases over time when Environmental Noise levels are negligible and no Audio Playback is present, below Effective Quiet. The thresholds and rates of change associated with the SPL Dose are derived from acoustic overexposure criteria and recovery time data (Ear Recovery Function) found in audiological research. If the SPL Dose nears a value, which can cause hearing damage (e.g. can cause a hearing threshold shift), the PHMLS user is actively informed. If the PHMLS determines that the recommended noise dose is exceeded, an Otoacoustic Emission test or hearing threshold test may be automatically invoked as a screening for potential hearing damage.

The SPLD is continually logged as an “SPL Exposure History”; which is a record of a user’s exposure to sound pressure levels over time. This record is very detailed, including Real-Ear Level data, listening duration data, time between listening sessions, absolute time, SPL Dose data, number of acoustic transients and crest-factor, and other data. SPL Exposure History includes both Listening Habits History and Environmental Noise Exposure History.

The “SPL Exposure History” log can be uploaded from the PHMLS system every time the PHMLS detects a wireless signal capable of transferring the SPLD log to a remote server along with any other data and download analyses of the log (which may be undertaken either automatically or with the assistance of an audiologist). The PHMLS considers the Listening History profile of the user to determine expected listening times. The user is informed about remaining time before their SPLD is exceeded (i.e. at which point hearing damage may occur) using a particular system of the present invention called an In Situ Hearing Health Monitoring System.

The In Situ Hearing Health Monitoring System is a process for estimating the risk of hearing damage from sound exposure based on empirical data of the dose-effect relationship of the impact of noise on the inner ear, and auditory system’s capacity to recover from excessive sound pressure level exposures.

When the ear is regularly overexposed to sound, auditory injuries, such as noise-induced TTS, permanent threshold shift, tinnitus, abnormal pitch perception, and sound hypersensitivity, may occur. These injuries have application in Audio Playback using Acoustic Transducers.

Unlike current strategies to reduce risk for auditory injury from excessive sound presented by Acoustic Transducers by placing sound level limiters in the pathway between Playback Hardware and the Acoustical Transducers, this platform described here within, monitors chosen level and duration of Audio Playback, make necessary gain adjustments to reproduced Audio Content to ensure safe listening levels, and provides the system with ongoing information related to the accumulated SPL dose which would lead to hearing damage risk. Unlike current sound exposure monitoring strategies, which simply monitor ambient noise sound pressure levels over a given time period (typically a work-day of eight hours) and reset to zero at the beginning of the next time period, or are averaged across several time periods (as in an average weekly noise exposure), the In Situ Hearing Health Monitoring System describes a process for estimating the time required for recovery from the sound exposure based on exposure history and scientific evidence of the physiologic basis for this process.

In another embodiment of the system, the ambient noise level is monitored to ensure a signal-to-noise ratio conducive for high intelligibility of the reproduced Audio Content. If an increase in ambient SPL is detected, then the reproduction level of the audio content is accordingly increased. This module uses data from empirical psychoacoustic studies to ensure that the timbral balance is maintained by any spectral modification; for instance, by taking into account level dependant equal loudness curves. This ambient noise level monitoring system maintains intelligibility in fluctuating ambient noise environments and is actively monitored by the SPLD system to ensure safe sound exposure limits are not exceeded, and tailors the reproduction level to that hearing zone personalized for a particular individual; taking into account their preferred listening level.

The PHMLS detects when the Canalphone is inserted in the ear of a user by monitoring SPL differences between the two microphones in the same Canalphone (i.e. the difference in SPL between the Ambient Sound Microphone and the Ear Canal Microphone): the differences will be smaller if the Canalphones are removed. Also, the PHMLS can determine if the Canalphones are inserted in the ear by determining the distance between the two Canalphones using either an acoustic SONAR mechanism or by analyzing the signal strength of transmitted radio waves between each Canalphone. Therefore, the operating state of the system can be logged; such as if the user has the Canalphones inserted in their ear or not.

Another embodiment of the system provides a means to confirm correct mechanical insertion or faulty transducer operation of the Canalphone system by providing feedback about spectral level response as measured within the ear canal with reference to a predicted spectral response, and further more, by monitoring the SPL at the outside of the occluded ear canal using the Ambient Sound Microphone to detect for leakage of sound reproduced with the Ear Canal Receiver.

This System Integrity Monitoring Platform is activated whenever the user inserts the PHMLS Canalphones and may also be invoked manually by the user.

This aspect of the invention can mitigate problems such as caused by a leakage in the acoustic seal between the Canalphone and the ear canal as well as ear debris contaminating and compromising the internal transducers.

The PHMLS can detect if the Canalphone is correctly inserted in the ear. Leakage of ambient sound into the ear canal occurs when the Canalphone is incorrectly fitted. A test signal to detect leakage may be in the form of an Earcon, which is activated when the PHMLS system is initialized (i.e. switched on), or with a manual user interaction. Alternatively, the leakage may be measured by a comparison of SPL within in the occluded Ear Canal (i.e. using the ECM) and at the entrance to the same ear canal (i.e. using the Ambient Sound Microphone). Spectral compensation can be automatically applied, or the user informed of the physical problem using an auditory or visual message system. Alternatively, the PHMLS may detect sudden or intermittent performances changes, such as caused by faulty electrical systems, low battery or variable damping in the acoustic barrier of the Canalphone, and the PHMLS will inform the listener accordingly (e.g. using a sound warning or sending an email with an error report).

In a further aspect, a means to record the Environmental Audio Sound Fields (EASF) is provided. This ability has applications for local sound replay, such as for oral dictating, and to provide feedback to the user and a database system regarding user listening habits. Using an “always recording” operating mode, the ambient sound is recorded to a circular data buffer allowing the user to record a 15 second (or so) preamble after the recording has been activated. This further provides the ability or for peer-to-peer hands-free wireless communication of the users’ voice and EASF, for instance using the Bluetooth Headphone protocol link. Peer-to-peer communication is also provided for data such as GPS-based information on the user’s location and personal identifying data, which may or may not be encrypted with a private key.

In a further aspect, a PHMLS users’ unique Head Related Transfer Function (HRTF) can be measured using either a removal miniature loudspeaker or an external speaker. The outer microphone within the Canalphone assembly can therefore measure the acoustic transfer function, and the Ear Canal Microphone (ECM) and Ear Canal Receiver (ECR) can measure the Ear Canal Transfer Function (ECTF). The ECTF may be measured by reproducing a test signal with the ECR, such as an MLS signal, or a music audio signal, which may be the Earcon (a sound created by reproducing an audio signal stored in data memory with the ER, that may be pre-processed with the users HRTF to give a personalized listening experience).

Alternatively, the a semi-personalized HRTF is approximated by a visual analysis of the user's ear shape with reference to a database of empirically measured HRTF's and their corresponding ear photograph. This database is accessed using a Web-based interface program, which allows the user to upload a photograph of their ear (e.g. via email or sent from a camera-equipped cell phone).

Once the personalized or semi-HRTF is attained, this can be used to process audio content to enhance spatial sound quality, as well as with applications for auditory displays such as to inform the listener of potentially damaging sound levels.

In a further aspect of the invention, the ability to process speech through an automatic speech to text recognition system followed by a text to speech synthesizer for purposes of signal to noise ratio enhancement is given. By training the PHMLS with a set of example sounds in “learn mode”, the PHMLS also allows a known sound occurring in the local sound scene to be recognized (such as an emergency siren), and to attenuate unwanted sounds as desired.

Related to this learn mode is a User Voice Timbral Equalization System (UVTES). This applies a filter to the signal reproduced by the Ear Canal Microphone whenever the user’s voice is detected, so that the user hears himself/ herself with a similar timbral quality as if the Canalphones were not inserted. This overcomes the occlusion effect; whereby an individual with a hearing protection device may experience a change in the sound quality of their spoken voice due to a change in bone-conducted sound, particularly below 2 kHz.

The UVTES is activated by the user, and informs the user with an auditory message to repeat a sequence of words, such as numbers. The UVTES may process the sound reproduced by the ECM in a number of different ways, and allow the user to select the filter that yields a preferred sound quality. In conjunction with the user selected subjective customization of the UVTES, an automatic system predicts an approximation of an equalizing filter by comparing the Ambient Sound Microphone signal and Ear Canal Microphone signal in response to user-generated speech.

In a further aspect of the invention, a process is provided as a failsafe for a Wireless Connection Break (WCB) between the Canalphones and the Media Communicator (e.g. telephone, personal audio device etc.). A WCB may occur if the Media Communicator and Canalphones are too far away, or if either the Canalphones or Media Communicator has a power interruption (e.g. low battery). A WCB can be detected if the Media Communicator does not respond to a “Received transmission” request command sent by the Canalphone (i.e. an Rx). In such a case of WCB, either or both the Canalphone and the Media Communicator will inform the user using an auditory, visual or tactile message.

In a similar aspect to this, a Device Locating function is provided, whereby if the user loses either the Canalphone(s) or Media Communicator, the user can activate a calling routine on the component they have not lost and an acoustic alarm will sound on the lost item to help the user located it.

In a further aspect of the invention, the ability to measure an individual’s Otoacoustic Emissions (OAEs) is described, using the Ear Canal Receiver to generate a test stimulus (such as an impulse or a pair of tones) and the Ear Canal Microphone to measure the cochlear response in the ear canal.

This measured OAE is then compared with normative data or within subject, and the person is informed about any signs of abnormal deviation. The test stimulus for the OAE may be a number of different signals: a number of Dirac clicks (in which case, the Transient Evoked OAE is measured); a burst of noise a pseudo-random sequence (i.e. an MLS, which has a random noise-like sequence but repeats regularly, e.g. every 256 samples); or it may be a music sequence, such as embedded in the Earcon which may be reproduced when the PHMLS is activated, and the OAE in this case is derived using an adaptive filter to model the Earcon signal with that signal measured with the ECM.

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A Personalized Hearing Measurement and Listening System (PHMLS.2):

Embodiments

- 1. A system for monitoring and recording Ambient Sound in the local environment of the CanalPhone user, using an always-on Ambient Sound Microphones (ASM) mounted at the entrance to an occlude ear canal, and a circular recording buffer.**
 - A. An assembly to monitor the SPL at the entrance to an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ambient Sound Microphones; ASM).
 - B. A signal processing circuit to amplify the signal from the ASM to equalize for the frequency sensitivity of the microphone.
 - C. A system to record or transmit the signal received at the ASM on a continuous basis with an ability to rewind and monitor recorded audio content at will. This process would use (for example) a circular (or ring) sound data buffer of several seconds so as to allow a recording to start several seconds before a start command is issued. The system is always in the record mode, and upon activating the manual record feature; the contents of the circular buffer are transmitted to a second device.
 - D. The ability to active the always-on recording device using a user-defined Signature Sound, such as a user-defined word or hand-clap.
- 2. An Ambient Sound Warning-Detection System, to detect Signature Sounds in the Ambient Sound Field or a CanalPhone user, detected by an Ambient Sound Microphone at the entrance to an occluded ear canal, so as to alert the user about local warning sounds such as car horns or fire alarms, the system comprising:**

- A. An assembly to monitor the SPL in an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the ECM.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- E. An assembly to monitor the SPL at the entrance to an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ambient Sound Microphone; ASM).
- F. A signal processing circuit to amplify the signal from the ASM to equalize for the frequency sensitivity of the ASM.
- G. A signal processing system to monitor the ASM signal to detect Signature sounds, using pattern recognition software which are based on a exemplar sounds (i.e. sounds which have been recorded and stored in a database, which can be online, or stored on a Personal Media Player, or stored in memory in the CanalPhones), or using pattern recognition software which are based on an algorithm (i.e. a software system which analyzes the ASM signal to detect Signature sound patterns based on spectral patterns, such as harmonic analysis, and temporal patterns, such as time envelope).
- H. A system to record Signature Sounds using the ASM.
- I. A system to upload Signature Sounds to an online database, and to use this online database to detect Signature Sounds in the Ambient Sound Field.
- J. A system to reduce the effective Noise Reduction Rating of the CanalPhone at selective frequencies, to allow the listener to hear Signature sounds in the Ambient Sound Field.
- K. A system to attenuate the reproduced Audio Content (i.e. reproduced with the ECM) on a frequency-selective basis whenever Signature Sounds are detected by the ASM, to allow the listener to hear Signature sounds in the Ambient Sound Field.

- 3. An Active Ambience Attenuation system, providing the ability to measure and manually or automatically adjust the Noise Reduction Rating (NRR) or Real-Ear Attenuation provided by the CanalPhone assembly when inserted and sealed in the ear canal. NRR can be varied automatically or by a user controlled amount from approximately +30 dB to -30 dB.**

If the NRR is such that the listener is exposed to dangerously loud SPL's in the occluded ear canal using the Ear Canal Microphone, the user is informed about this using an auditory warning system, and an audiogram recommended to test for hearing damage.

- A. An assembly to monitor the SPL in an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the ECM.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- E. An assembly to monitor the SPL at the entrance to an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ambient Sound Microphone; ASM).
- F. A signal processing circuit to amplify the signal from the ASM to equalize for the frequency sensitivity of the ASM.
- G. An assembly to generate a desired SPL at the entrance to an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the External Receiver; ER).
- H. A signal processing circuit to amplify the signal to the ER to equalize for the frequency sensitivity of the transducer.
- I. A system to calculate NRR by measuring the difference in SPL in an occluded ear canal (i.e. using the ECM) and at the entrance to the ear canal (i.e. using the ASM). If the Ambient Noise Level is deemed too low, then an external noise stimulus is generated by the ER.
- J. A user interface to allow a user to manually adjust NRR on a spectral level. The customization can be undertaken using sliders on a multiband graphic equalizer, or selected from a library of presets.

- K. The ability to store a history of reports to the user of NRR measurements and to upload the history to a central database on a remote server, using a wireless data link.
- L. A system that may analyze a history of NRR of the CanalPhone, and to notify the user if abnormal or degrading performance is detected.

4. An apparatus to monitor SPL in an occluded ear canal to ensure ear canal SPL is not sufficient to cause temporary or permanent hearing damage, comprising:

- A. An assembly to monitor the SPL in an SPL in an occluded ear canal ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the ECM.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- E. An assembly to monitor the SPL at the entrance to an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ambient Sound Microphone; ASM).
- F. A signal processing circuit to amplify the signal from the ASM to equalize for the frequency sensitivity of the ASM.
- G. An assembly to generate a desired SPL at the entrance to an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the External Receiver; ER).
- H. A signal processing circuit to amplify the signal to the ER to equalize for the frequency sensitivity of the transducer.
- I. The ability to automatically measure the user's Audiogram, by generating and monitoring Sound Pressure Level (SPL) in a user's ear canal for purposes of measuring Hearing Threshold values vs. frequency. The test stimulus is generated using the ECR, using pure tones, band-limited noise or band-limited music. The inner microphone measures Real-Ear Level (REL) (or eardrum SPL or tympanic membrane SPL - TMSPL), and the user provides feedback to the system about whether a stimulus is detected using either; a Bluetooth-equipped device such as a mobile phone or computer, using a speech call-sign, or using a buttons located on either CanalPhone.
- J. A system to monitor ear canal SPL and determine if the user is likely to suffer hearing damage due to ear canal SPL (i.e. Noise Induced Hearing Loss).
- K. An assembly to attenuate the level of Audio Content reproduced with the ECM if ear canal SPL is deemed excessive.

5. Another embodiment of the invention measures, predicts and controls sound pressure levels in the occluded ear canal for purposes of minimizing risk of threshold shift be it permanent or temporary. The systems comprises:

- A. An assembly to monitor SPL in an occluded ear canal in response to both environmental (local ambient) sound stimuli and sound created by either the Ear Canal Receiver (ECR) and/or the External Receiver (ER).
- B. The ability to monitor and record in-ear SPL (Real Ear Level, REL) exposure over time, and to inform the user of Noise Dose.
- C. If SPL Dose exceeds levels that may cause temporary or permanent hearing threshold shift, a recovery function is invoked to spectrally attenuate reproduction levels. The Recovery Function considers user history, such as average daily usage, to predict remaining time until the user's safe daily noise dose is exceeded.
- D. The ability to monitor and record in-ear SPL (Real Ear Level, REL) using the Ear Canal Microphone (ECM) exposure over time and to upload the exposure history profile to a data storage system for analysis by the user and/or an audiologist and/or geographic noise profiling system.
- E. The ability to invoke an Audiogram if SPL Dose is deemed excessive, and to alert the user of a need to consult an audiologist.

6. An Always-On Dosimeter assembly to determine and inform the user if SPL exposure (measured in an occluded ear canal) may cause hearing damage, and to attenuate the Audio Content Reproduction level to minimize the risk of hearing damage.

- A. A Dynamic Hearing Damage Risk Monitoring System that determines SPL Dose according to sound levels presented to the ear. These SPL Dose data are stored to a memory storage system and provides information to the user via an interface or to a feedback loop within a system that influences the sound output level or duration of sound output.
- B. Audio input microphones positioned near the occluded ear canals to sample the sound levels that arrive at the ear canals.
- C. Acoustic transfer functions to correct near-field levels measured at the ear canal to free-field equivalent.
- D. A digital timer that starts timing when the combination of the level of Environmental Sound and Audio Playback:
 - i. Increases from below Effective Quiet, or decreases from a value above Effective Quiet + 0.9 dB, to a value between Effective Quiet and Effective Quiet + 0.9 dB.
 - ii. Increases from a value between Effective Quiet and Effective Quiet + 0.9 dB to a value of Effective Quiet + 1.0 dB or higher

- iii. Decreases from a value between Effective Quiet and Effective Quiet + 0.9 dB to a value of Effective Quiet – 0.1 dB or lower
- F. A process for calculating SPL, based on the free-field equivalent Environmental Sound and Audio Playback level and the duration recorded by the digital timer.
- G. A process by which the user is prompted to the Safe Listening Duration time based on the combination of the free-field equivalent Audio Playback and Environmental Sound.
- H. A process by which the user is prompted that SPL Dose has exceeded 100% and reducing Environmental Sound and Audio Playback exposure below Effective Quiet is indicated.
- I. A process by which the Audio Playback Hardware automatically attenuates output level when an SPL Dose criterion is exceeded. This process can be overridden by user control.
- J. A process by which the Audio Playback is automatically suspended (so that Audio Playback contribution to the combination of Environmental Sound and Audio Playback is zero) for a duration dictated by the SPL Dose equation, until SPL Dose decreases to a criterion level. This process can be overridden by user control.
- K. A process by which a parent or guardian is notified via email or other electronic messaging that their dependent's SPL Dose has exceeded 100%.
- L. A process for updating the user's SPL Exposure History with the proportion of time SPL Dose exceeds a criterion level (such as 100%).
- M. A process for prompting the user (or parent or guardian if the user is a dependent) to seek the professional services of a licensed audiologist upon first initiation of using the system, and again if the user's SPL Exposure History consistently exceeds a criterion level over a period of months, as determined by existing and future audiological research.

7. A self-contained, portable system for measuring and inserting necessary compensation of an individual's personal hearing response; comprising of a measure for Ear Canal Transfer Function (ECTF) for an occluded ear; hearing threshold for a measured ambient SPL; equal loudness curves for different stimulus ear-canal SPL; a measure for speech intelligibility, and transmit the data to a remote location (e.g. computer server) for analysis (e.g. by an automatic algorithm or an audiologist or an ENT physician, for diagnosis and prognosis).

- A. An assembly to monitor the SPL in an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the microphone.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECM to equalize for the frequency sensitivity of the ECM.
- E. A system to determine if the Ear Canal Sound Pressure Level (ECSPL) is low enough to conduct an audiogram (such as to measure the user Hearing Threshold), using the ECM. If the noise level is too high, the user is informed using either an auditory or visual sound cue.
- F. A system to automatically measure the user's Audiogram, by generating and monitoring ECSPL for purposes of measuring Hearing Threshold values versus frequency. The test stimulus is generated with the ECR, using pure tones, band-limited noise or band-limited music. The inner microphone measures Real-Ear Level (REL) (or eardrum SPL or tympanic membrane SPL - TMSPL), and the user provides feedback to the system about whether a stimulus is detected by the user using a user interface which may be either; a Bluetooth-equipped device such as a mobile phone or computer, using a speech call-sign; using a buttons located on either CanalPhone; a voice-activated system whereby the user indicates if the auditory stimulus is heard or not.
- G. An assembly to automatically measure the user's Level Dependant Equal Loudness Curves, by generating and monitoring Sound Pressure Level (SPL) in a user's ear canal. This is done using pure tones, band-limited noise, or band-limited Audio content (speech or music). A reference stimulus is played, and the user is asked to match the loudness of a test stimulus using the previously described user interface feedback methods
- H. An assembly to automatically measure the user's frequency-dependant Most Comfortable Listening Level, by generating and monitoring Sound Pressure Level (SPL) in a user's ear canal. The user is asked to provide feedback about the perceived loudness of a test stimulus using the previously described system feedback methods. This feedback may be on a 5-point scale with end-point perceptual anchors of "too loud" and "too soft".
- I. An assembly to generate and monitor a desired SPL in a user's ear canal for purposes of measuring and testing Speech Intelligibility in Noise, e.g. using %ALCONS vs. Signal-to-noise ratio, or using a RASTI measurement using the as a noise source. In the %ALCONS, a number of test stimuli (e.g. speech phonemes) are presented with the ECR in the presence of noise also reproduced from the ECR (the contribution from ambient noise is also taken into account- as measured using the Ambient Sound Microphone (ASM)).

The user provides feedback about the perceived phoneme using either a Bluetooth enabled device (e.g. by typing the perceived phoneme), or by speaking the perceived phoneme, which is detected by the PHMLS using a speech-to-text translator program. The percentage of correctly understood phonemes as a function of signal-to-noise ratio is then measured. The intelligibility index is then reported to the user, and may be used as a basis to suggest a consultation with an audiologist if it is abnormally low, or if the user's speech score in the speech intelligibility test is significantly worse than previous tests.

- J. The ability to store and monitor an individual's audiograms and hearing sensitivity over time so as to provide comparative and trend measures, and to inform the user of need to consult an audiologist if a degradation in hearing ability is detected. The audiogram is stored in the CanalPhone assembly and is downloaded to a central server with a Bluetooth communication link whenever an appropriate signal is detected.
- K. A system to confirm the proper operation of the CanalPhone system when worn by a user; to automatically compensate for improper fitting (e.g. caused by ear detritus) and/or to inform the user about corrective measures.
- L. An assembly to measure and predict the user's Ear Canal Transfer Function. The Ear Canal Transfer function can be measured in-situ using a noise-like test signal (e.g. with Maximum Length Sequence Analysis) or impulses reproduced by the Ear Canal Receiver loudspeaker (ECR). Alternatively, the ECTF test signal may be created using Dirac clicks reproduced by the ECR, and recorded with the ECM. The recorded clicks may be average to improve the signal-to-noise ratio of the ECTF measurement. Alternatively the ECTF may be measured using a test signal present with the ECR composed of an EarCon; a customizable sound reproduced from the ECR when the CanalPhone system is inserted in an occluded ear canal and activated. The ECTF is then measured using an adaptive filter to predict the impulse transfer function of the signal measured by the ECM and the test signal.
- M. The ECTF may be stored to Data Memory (and/or transmitted to a central database), and can be recalled later. Once the ECTF has been measured and recalled to data memory, Audio Content reproduced with the ECR is modified with an inverse response of the measured ECTF to provide a user-compensated frequency response.
- N. A wireless and/or wired data transmission system to transmit the personalized hearing threshold and/or ECTF and/or equal loudness curve and/or speech intelligibility rating from the data Memory of the CanalPhone to a second data processing system, e.g. on a remote server, for diagnosis of potential hearing damage by either a self-contained computer algorithm and/or a physician, such as an ENT, and/or an audiologist.

8. A self-contained system for measuring a user's preferred listening levels for different styles of audio content (e.g. different musical genres, different speech reproduction such as news reports, telephone communication).

- A. An Ear Canal Microphone (ECM) and an Ear Canal Receiver (ECR) to monitor and generate a desired sound field in a user's occluded ear canal.
- B. An External Receiver loudspeaker and (ECR) Ambient Sound Microphone to generate and monitor a desired sound field at the entrance to a user's occluded ear canal.
- C. A system to determine if the noise level in an occluded ear canal and ambient environment is low enough to enable measurement of user's preferred listening level.
- D. A manual user interface to report a degree of satisfaction (or comfort) of the current reproduction level. This may be using a speech detection system, wherein the user reports the degree of satisfaction with speech. Alternatively, this may be using buttons on a separate media device such as a cell phone, or with buttons mounted on the CanalPhone.
- E. A system to store user's preferred listening levels for particular classes of audio content (music, speech etc), and to upload the data to a database on a server, media storage device etc.

9. A self-contained Personalized Hearing Zone system to automatically maintain ear canal SPL at levels above threshold level for absolute detection, or at a certain level above the background noise level in the ear canal, yet below the ear canal SPL at which loudness discomfort is reported. The system comprising:

- A. An assembly to monitor the SPL in an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the ECM.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- E. An assembly to monitor the SPL at the entrance to an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ambient Sound Microphone; ASM).
- F. A signal processing circuit to amplify the signal from the ASM to equalize for the frequency sensitivity of the ASM.

- G. An assembly to generate a desired SPL at the entrance to an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the External Receiver; ER).
- H. A signal processing circuit to amplify the signal to the ER to equalize for the frequency sensitivity of the ER.
- I. A system to determine if the noise level in an occluded ear canal and ambient environment is low enough to enable measurement of user's preferred listening level.
- J. A manual user interface to report a degree of satisfaction (or comfort) of the current reproduction level. This may be using a speech detection system, wherein the user reports the degree of satisfaction with speech. Alternatively, this may be using buttons on a separate media device such as a cell phone, or with buttons mounted on the CanalPhone.
- K. A system to store user's preferred listening levels for particular classes of audio content (music, speech etc), and to upload the data to a database on a server, media storage device etc.
- L. A compander gain system to maintain ear-canal SPL within the PHZ.
- M. A system to monitor the SPL in an occluded Ear Canal using an Ear Canal microphone, and to ensure that playback level does not exceed levels that may cause hearing damage. In Ear Canal SPL is deemed unsafe, the user is informed and/or the audio content playback level attenuated and/or an audiogram is invoked to investigate if hearing damage has occurred.
- N. A system to measure ambient SPL using an Ambient Sound Microphone at the entrance to an occluded ear canal, to ensure that Audio Content reproduced by the Ear Canal Receiver in the same CanalPhone has a suitable signal-to-noise ratio to ensure high intelligibility and timbral fidelity of the reproduced content. If ambient SPL is deemed too high, low-level components of the Audio Content may be amplified (i.e. using an expander circuit), whilst maintaining supervision on overall Ear Canal SPL to ensure against level that could cause hearing damage.

10. A system for Personalization of the frequency response of the signal reproduced by an Ear Canal Receiver, using a signal processing assembly such as a digital convolution engine, to maximize timbral fidelity and/or to increase intelligibility of the reproduced audio content by monitoring sound pressure level an occluded ear canal using an Ear Canal Microphone.

- A. An Ear Canal Microphone (ECM) and an Ear Canal Receiver (ECR) to monitor and generate a desired sound field in a user's occluded ear canal.
- B. An External Receiver loudspeaker and (ECR) Ambient Sound Microphone to generate and monitor a desired sound field at the entrance to a user's occluded ear canal.

- C. A system to determine if the noise level in an occluded ear canal and ambient environment is low enough to enable measurement of aspects of a user's auditory system, such as listening thresholds, user HRTF or Ear Canal Transfer Function.
- D. The apparatus for the personalization of the ECR response may be undertaken by pre-processing the record audio content. Alternatively the filtering may take place in situ in the signal processing circuits in the CanalPhone using a convolution engine, or the filtering may be undertaken in real-time on an external processing device, such as a mobile phone or external signal-processing device.
- E. The audio signal reproduced by the ECR may be adjusted depending on the Ambient Noise Level (ANL). The ANL is detected by the Ambient Sound Microphone, and the reproduced audio content signal gain may be increased by decreased on a frequency-specific basis to maintain a signal-to-noise ratio conducive to good intelligibility or to maintain a user-defined signal-to-noise ratio. Alternatively, just the low-level sound may be boosted in response to an increased ANL (i.e. an expander). The sound pressure level in an occluded ear canal is constantly monitored with the ECM to ensure levels do not exceed either the safe daily SPL dose or are not loud enough to cause temporary or permanent threshold shift.

11. Another embodiment of the invention facilitates the measurement or determination of a user's unique HRTF values with the following capabilities:

- A. A detachable External Receiver loudspeaker (ER) and an Ambient Sound Microphone (ASM) to generate and monitor a desired sound field at the entrance to a user's occluded ear canal.
- B. A system to measure the user's personalized HRTF simultaneously for both the left and right ear, by reproducing a test signal
- C. The ability to upload the measured RIR to a database which the user can recall at a later date, or to allow other user's access to pre-stored RIRs, or for use by acousticians for analysis of acoustical properties of rooms (such as reverberation time).

12. A system to obtain a semi-personalized HRTF from a user, by analysis of a photograph of the user's ear, which may contain a reference object (such as a coin or credit card) to aid scaling the picture to real-world units. This system comprises:

- A. A database system containing various personalized HRTF's or generic HRTF's.
- B. A database system containing photographs of the ear corresponding to the empirically determined HRTF, and associated anthropomorphic information such as age and gender.

- C. An interface to allow user's to submit a photograph of each of their ears. The photograph may be taken with, for example, a cell-phone, or a Web-Camera, and uploaded to the Semi-Personalized HRTF determining system with a messenger or email service.
- D. A system to determine if the submitted photograph is of a high enough quality to be processed to determine semi-personalized HRTF of the user. This may take into account light intensity, or if hair is covering the ear.
- E. An interpolation algorithm for determining the best fitting HRTF set in the database from the anthropometrical measurements of the subject;
- F. A system to extract features from the ear photographs in the database and compare with features in the uploaded photograph, to obtain a closest-fit. This feature-matching system determines measures such as the cavum concha height, cymba concha height, cavum concha width, fossa height, pinna height, pinna width, and intertragal incisure pinna width and cavum concha depth.
- G. A system that returns an auditory cue to the subject, Convolved with the best fitting HRTF data, placing the cue virtually in five or more spatial locations, to determine the efficacy of the interpolation process.

13. A head-tracking system that modifies the auditioning of Audio Content, Personalized Audio Content or Psychoacoustically Personalized Content as to change the positioning of the Auditory Spatial Image in response to user's head movement, by monitoring changes in SPL measured at the entrance to each occluded ear of the user, using the Ambient Sound Microphone, in response to an external sound source generated for, for instance, a laptop or cell-phone loudspeaker, comprising:

- A. A system for monitoring the orientation of the user's head in real-time, using a pair of Ambient Sound Microphones to monitor sound pressure levels at the entrance to both of the user's occluded ear canals in response to a dominant sound stimulus located at a fixed (non-moving) location, such as a musical sound source reproduced by a loudspeaker on a cell-phone, or on a lap-top. The head-tracking system calculates head orientation by an analysis of spectral level and phase differences of the two ASM signals.
- B. A method for modifying Personalization Processing dynamically on any combination of Client, Server, or Embedded Devices based on the input of the head-tracking system.
- C. A system for processing Audio Content using the user's personalized or semi-personalized HRTF and ECTF measurements.

14. Another embodiment of the invention provides an ability to confirm and adjust the correct acoustic energy transfer of the CanalPhone system when inserted and sealed in the ear of the user.

- A. The degree of performance of the acoustic seal between the CanalPhone system and the user ear canal can be confirmed as it relates to the CanalPhone systems ability to reproduce adequate bass response. Bass response is measured monitored by continually measuring low frequency SPL in an occluded ear canal using the Ear Canal Microphone (ECM), and comparing with a predicted SPL based on the user's ECTF (which is stored in data memory). Alternatively, low frequency response may be measured using a test signal generated by the Ear Canal Receiver. A deviation in the actual low frequency SPL will caused the system to either or both inform the listener or/and actively compensate for the bass mismatch by applying an adaptive digital filter. SPL in an occluded ear canal is monitored using the ECM to ensure that equalization of the audio content does not exceed safe listening levels.
- B. The degree of performance of the CanalPhone system seal and the user's ear canal can be evaluated as it relates to the sound isolation capabilities of the system. Seal integrity can be measured by creating a test signal using the External Receiver loudspeaker and measuring the difference in SPL between the ear canal SPL and external ambient sound level (the latter level measured using the Ambient Microphone). Alternatively, a comparison between the inner ear SPL and ambient SPL is undertaken without the ER active, as an indication of Noise Reduction Rating (NRR).
- C. The degree of performance of the individual (internal) receiver and microphone components can be confirmed as to their ability to provide proper sound reproduction and monitoring functions. An example of a situation in which system performance could be compromised would be due to a blockage of a microphone or receiver sound port due to cerumen. If a reduced performance is detected, the listener may be informed (e.g. using a visual or auditory display, or by email) and cleaning recommendations given.
- D. The ability to store a history of reports to the user about improper fitting of the CanalPhone, and to upload the history to a central database on a remote server, using a wireless data link.

15. A system to detect whether the CanalPhones (which comprise a left and right pair) are inserted in the ears of a user, the system comprising:

- A. An Ear Canal Microphone (ECM) and an Ear Canal Receiver (ECR) to monitor and generate a desired sound field in a user's occluded ear canal.
- B. An External Receiver loudspeaker and (ER) Ambient Sound Microphone (ASM) to generate and monitor a desired sound field at the entrance to a user's occluded ear canal.
- C. A system to compare the SPL measured by the ECM and ASM in one or two pairs of CanalPhones. If the SPL's of one or both CanalPhones are measured to be less than threshold, and/or correlated with a certain degree, then the CanalPhone is determined to be not inserted in the user's ear.
- D. A SONAR system to determine if the Canal phone is inserted in the ear, which generates a "ping", which may be an EarCon audio signal, or an ultrasonic ping, is radiated from the ER, and if it is detected by the ECR at a certain SPL, which is considerably less than the Noise Reduction Rating of the acoustic seal provided by the CanalPhone when inserted in the ear, the system determines that the CanalPhone is not inserted in the user's ear canal.
- E. A system for determining the distance between the two CanalPhones of a left-right pair for a particular user. This is obtained either by emitting an acoustic signal from the ER and/or ECM of a first CanalPhone, and measuring the SPL with the ECM and/or ASM. If the later SPL is significantly different than would be expected if the CanalPhone were inserted in the ear, or if the ECM and/or ASM of the second CanalPhone predict that the time-of-arrival for the received sound is different, then the system deems that the CanalPhone is not inserted in the user's ear.
- F. A system for determining the distance between a left and right CanalPhone (belonging to the same pair) by analyzing the strength of a radio signal (e.g. Bluetooth signal) emitted by a first CanalPhone and received by a second CanalPhone.

16. Another embodiment of the invention provides a means by which the operating state of the system is conveyed to the user by means of personalized or semi-personalized, spatially positioned sound cues (Sonification):

- A. The ability to warn a system user via a readily identifiable acoustic signal, that a listening level exists that is potentially damaging unless corrective action is taken.
- B. The ability to inform the user via a readily identifiable acoustic signal, as to a low battery condition.

- C. The ability to inform the user that the local sound field has exceeded a set threshold sound pressure level using a broadband weighting function.
- D. The ability to inform the user that the local sound field has exceeded a set threshold sound pressure level within one of several narrow frequency bands.
- E. The ability to inform a system user that the EASF monitoring system is in a record or playback mode.
- F. The ability to for device to ear communication. For example, using a text-to-speech program to inform the user of email; location (using GPS); time and date.
- G. A system to monitor ambient SPL using the ASM to ensure affect the reproduction level of the sound cue to maintain intelligibility.
- H. A system to process the sound cue in accordance with the listeners personalized or semi-personalized HRTF and ECTF so as to give the perception of sound cues at different locations around the listener.
- I. A system to process the sound cue in accordance with the listeners personalized hearing comfort level.

17. The use of the ECM and/or the ASM as a switch to activate audio signal transmission and/or recording.

- A. An Ear Canal Microphone (ECM) and an Ear Canal Receiver (ECR) to monitor and generate a desired sound field in a user's occluded ear canal.
- B. An External Receiver loudspeaker and (ER) Ambient Sound Microphone (ASM) to generate and monitor a desired sound field at the entrance to a user's occluded ear canal.
- C. A system that determines if the SPL measured by either or both the ECM and ASM are above a certain threshold-curve for a given range of frequencies. If so, the "Talk" switch is activate, and sound received by either or both the ASM and ECM are transmitted from the CanalPhone to a second recording or forwarding audio data system.

18. A means for measurement of a user's unique Otoacoustic Emission (OAE) values, for diagnosis and prognosis of potential hearing damage, using the Ear Canal Microphone and Ear Canal Receiver.

- A. An Ear Canal Microphone (ECM) and an Ear Canal Receiver (ECR) to monitor and generate a desired sound field in a user's occluded ear canal.
- B. A system to generate a test sound field by reproducing a signal with the ECR. The signal may be: Either a number of clicks (i.e. the Transient Evoked OAE would be obtained); or a sustained-burst signal such as an MLS signal. Alternatively, the OAE test signal may be present in an Earcon, which is activated, for example, every time the PHMLS is activated, and the OAE in this case is derived using an adaptive filter to model the Earcon signal with that signal measured with the ECM.

- C. The test may alternatively presented to the user in a benign manor, such that the user would not be aware of the test being conducted. Once such approach would be to insert or mix the necessary test stimuli with music files, such that the stimuli would be recognizable to the software based on its signature and repetition yet undetectable or innocuous to the user. The stimuli may need to be inserted in a sufficient durations as to obtain a meaningful S/N relationship. The user would use the headphones, as they normally would while listening to their digital music files, whilst the system would be engaged in producing a background OAE test.
- D. The OAE stuffy may be invoked based on SPL does or duration or by a predetermined set of time using the system.
- E. A system to detect if ear-canal SPL is low enough to conduct an OAE test, and to inform the user of the SPL is adequate.
- F. The ability to compare the measured OAE with an OAE database of people with no hearing damage to predict hearing damage. The reference OAE could be one from an individual with a similar personal profile, e.g. gender, and age.
- G. The ability to warn a system user via a readily identifiable acoustic signal, that their OAE predicts hearing damage (e.g. with a comparison with a healthy OAE, or an OAE of the user from a past measurement). If such an OAE were detected, the user would be advised to visit an audiologist.
- H. The ability to send the user's unique OAE to a central database of OAE measurements. This could be used by audiologists to look for trends in hearing health for the particular user, or could be used for analysis of geographic or work-related trends in OAE.
- I. A system to detect if the OAE suggests hearing damage (e.g. with a comparison with a healthy OAE, or an OAE of the user from a past measurement), and to evoke an audiogram to measure hearing threshold using the self-contained in-situ audiogram function.

19. A system for recording sound created by the user, using an always-on sound recorder, using the inner-microphones and a circular recording buffer.

- A. An assembly to monitor the SPL in an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the ECM.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.

- E. A system to record or transmit the signal received at the microphones on a continuous basis with an ability to rewind and monitor recorded audio content at will. This process would use (for example) a circular (or ring) sound data buffer of several seconds – several minutes so as to allow a recording to start several seconds before a start command is issued. The system is always in the record mode, and upon activating the manual record feature; the contents of the circular buffer are transmitted to a second device.
- J. The ability to active the always-on recording device using a user-defined Signature Sound, such as a user-defined word or hand-clap.

20. A system to detect the presence of one or more other PHMLS user's and to alert the first PHMLS user and/or inform them of the direction of a second user.

- A. The ability of a CanalPhone to encode and transmit a unique identifier with at intervals with Bluetooth.
- B. The ability of a first CanalPhone to detect the transmitted unique identifier of a second CanalPhone.
- C. The ability of a first CanalPhone to detect the location of a second CanalPhone by detecting the difference in signal strength of the transmitted signal at each Bluetooth receiver on each of the first CanalPhones.
- D. The ability of a CanalPhone to transmit the user's global coordinates, measured using a GPS system, to an online database, or to other CanalPhones in the local area.
- E. The ability to inform the user of a first CanalPhone of the bearing (e.g. azimuth) and range (i.e. distance) to a second CanalPhone user, by either a voice message with bearing instructions (e.g. 20 meters ahead), or using a voice or non-voice message processed with the personalized or semi-personalized HRTF of the user corresponding to the bearing of the second CanalPhone.

21. A User Voice Timbral Equalization System (UVTES) to reduce the subjective effects of the Occlusion Effect. This system applies a filter to the signal reproduced by the Ear Canal Microphone whenever the user's voice is detected, so that the user hears himself/ herself with a similar timbral quality as if the CanalPhones were not inserted. The UVTES is activated by the user, and informs the user with an auditory message to repeat a sequence of words, such as numbers. The UVTES may process the sound reproduced by the ECM in a number of different ways, and allow the user to select the filter that yields a preferred sound quality. The system comprises:

- A. An Ear Canal Microphone (ECM) and an Ear Canal Receiver (ECR) to monitor and generate a desired sound field in a user's occluded ear canal.
- B. An Ambient Sound Microphone (ASM) to monitor a sound field at the entrance to a user's occluded ear canal.
- C. A system to determine if the noise level in an occluded ear canal and ambient environment is low enough for the UVTES.
- D. A user-operated, self-contained system that prompts the user to repeat certain audio cues, such as number.
- E. A system that records the user's vocal response to the prompt, using the ECM and ASM.
- F. A signal processing system that filters the recorded user vocal response with a number of equalization filters, and presents these filtered recordings to the user.
- G. A user interface, wherein the user selects which of the processed recordings is preferable.
- H. A data storage system, wherein the user's preferred timbral equalization filter is stored to allow it to be recalled in the future.

22. A self-test system to automatically test the electroacoustic operating efficacy of a canal phone system, and to self-calibrate to account for misalignment of transducers or signal processing assemblies, comprising:

- A. An assembly to monitor the SPL in an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Microphone; ECM).
- B. A signal processing circuit to amplify the signal from the ECM to equalize for the frequency sensitivity of the ECM.
- C. An assembly to generate a desired SPL in an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ear Canal Receiver; ECR).
- D. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- E. An assembly to monitor the SPL at the entrance to an occluded ear canal consisting of a microphone mounted in an earplug that forms an acoustic seal of the ear canal. (This is the Ambient Sound Microphone; ASM).
- F. A signal processing circuit to amplify the signal from the ASM to equalize for the frequency sensitivity of the microphone.
- G. An assembly to generate a desired SPL at the entrance to an occluded ear canal consisting of a loudspeaker receiver mounted in an earplug that forms an acoustic seal of the ear canal. (This is the External Receiver; ER).
- H. A signal processing circuit to amplify the signal to the ER to equalize for the frequency sensitivity of the transducer.

- I. A system to detect if the CanalPhone is inserted in a test cradle, such as a CanalPhone carrying and/or storage assembly, by generating a test signal with the ECR and/or ER, and monitoring the signal detected by the ASM and/or ECM, and comparing with a signal in data memory corresponding to a signal recording made with the CanalPhone in a test cradle in response to a signal generated by the ECR and/or ER.
 - J. A system to detect improper functioning of the ECM and/or ECR and/or ER and/or ASM by comparing the signal recording of the ECR and/or ER in response to a test-signal generated by the ECR and/or ER when the CanalPhone is inserted in a test cradle, or when the CanalPhone is inserted in the user's ear canal(s).
 - K. A system to detect improper functioning of the ECM and/or ECR and/or ER and/or ASM with a frequency-dependant electronic impedance measurement analysis, and comparison with a database of correct-functioning impedance measurements.
 - L. A system to inform the user and/or an online analysis system about improper functioning of the ECM and/or ECR and/or ER and/or ASM using, for example, an Auditory Cue reproduced with the ECR, or with an automatically generated email alert, and to inform the user about corrective procedures, e.g. customer support contact information.
-

23. A self contained Signature Sound Identification System (SSIDS) which monitors a User's local ambient sound field using one or more Ambient Sound Microphones (ASMs) resident in a CanalPhone (that forms a seal in the ear canal) in the locality to the entrance of one or both fully or partly occluded ear canal(s). Additional Ear Canal Microphone (ECM) sound sensors may be housed in the CanalPhone assembly to monitor sound within the User's fully or partly occluded ear canal(s). The SSIDS detects the presence of one or more particular sounds, called Signature Sounds, by analysis of the signal generated by the ASM. Signature sounds include speech critical words such as "help", "stop", "police" (in multiple languages) and non-speech such as klaxons, warning-signals.

The SSIDS detects Signature Sounds in two ways; using an Algorithmic Pattern Recognition system that detects signature sounds with a specific or general spectral and temporal profile analysis, and a Prototype Recognition System that detects specific signature sounds using a database of stored signature sounds. The two systems may be active simultaneously. The SSIDS comprises:

- A. An Ambient Sound Microphone (ASM) to monitor the sound field at the entrance to a user's occluded ear canal.
- B. An ASM amplification and signal processing (signal conditioning) system to compensate for the frequency response of the ASM.
- C. An optional Ear Canal Microphone (ECM) to monitor the sound field within a user's occluded ear canal.
- D. An ECM amplification and signal processing (signal conditioning) system to compensate for the frequency response of the ECM.
- E. A signature sound communication system to transfer stored signature sound audio data and meta-data (meta data consists of e.g. SPL data, spectral and temporal profiles) from a database. The database may be resident within computer memory within the CanalPhone assembly or on a physically separate second data storage device such as a PMP.
- F. An Algorithmic Signature Sound Recognition System (ASSRS) that detects signature sounds by analysis of the ASM (and in some embodiments, the ECM) audio signal. The DSP assembly for the ASSRS may be on an ASIC DSP device mounted within the CanalPhone assembly or on a second DSP device such as a PMP, which may be communicated with using a wireless data transmission system such as Bluetooth. The ASSRS detects signature sound using spatial cues for the sound source by analysis of the conditioned left and/or right ASM signal to determine if the sound source is close to the listener or if it is approaching the listener. The distance and bearing of a sound source to a User may be calculated by analyses of the frequency-dependant magnitude and phase between the left and right ASM (and in some embodiments, the ECM) signal(s). The speed and bearing of sound source may be calculated using a pitch analysis to detect changes predicted by the Doppler effect, or alternatively by an analysis in changes in relative phase and magnitude between the two ASM signals. The ASSRS may also detect general warning signals such as car horns or emergency sirens (and other signals referenced by ISO 7731) using a spectral and temporal profile analysis.

- G. A Prototype Signature Sound Recognition System (PSSRS) that detects specific signature sounds by analysis of the ASM audio signal(s) (and in some embodiments, the ECM) signal(s)). The DSP assembly for the ASSRS may be on an ASIC DSP device mounted within the CanalPhone assembly or on a second DSP device such as a PMP, which may in a shared-mode embodiment be communicated with using a wireless data transmission system such as Bluetooth. The PSSRS may detect speech (critical words such as “help” etc) and non-speech (e.g. klaxon sound) signature sounds by detecting commonalities between the database of specific signature sounds and the ASM signal using, for example, a computer-based neural network to match the sound (such as a specific alarm sound, rather than a general alarm sound as with the ASSRS).

24. A Signature Sound Communication Assembly to transfer audio data or audio meta-data to a SSIDS, comprising:

- A. A database of signature sound audio data and meta data. The database is stored in computer memory housed either within the CanalPhone assembly, or on a second data storage device (e.g. a PMP), or on both the CanalPhone and a second data storage device. Audio data may be stored in a data-compressed format (e.g. using a perceptual encoding method such as mp3). Meta-data may be included in addition or instead of the audio data, which may include such information as spectral-level profile, real-world loudness (e.g. SPL at 1 meter), sound-source size, typical sound-source location (e.g. airborne objects).
- B. A Learn Mode system whereby signature sounds are acquired recorded and analyzed to give a user-defined signature sound database comprising audio data and meta-data.
- C. A data communication system for transmitting audio and meta-audio data from the database of signature sounds in (A) to the SSIDS system.

25. A Signature Sound Identification System (SSIDS) operating mode module to determine an appropriate action when a signature Sound is detected by a SSIDS, comprising:

- A. A User Mode selection interface, such as a software system on a computer, PDA, PMP etc to select which of a plurality operating modes should be assigned to particular signature sounds.
- B. An automatic Mode Selection system including an automatic self-contained prioritizing system to invoke a particular signal processing system when a particular signature sound is detected.

- C. A signature Sound Pass through Mode (SSPTM) wherein a signature sound detected by one or more ASMs (and in some embodiments, the ECM signal(s)) is reproduced with an Ear Canal Receiver, mounted in a fully or partially sealing CanalPhone. The gain and timbral quality of the SSPTM is such that the perceived loudness and timbral character of the Signature Sound (by the User) is the same as if the CanalPhones were not inserted in the ear canal(s) of the User.
- D. A Signature Sound Boost Mode (SSBM) wherein a signature sound detected by one or more ASMs is reproduced with an Ear Canal Receiver, mounted in a fully or partially sealing CanalPhone. The gain of the SSBM is such that the perceived loudness of the Signature Sound (by the User) is increased compared to the case when the CanalPhone(s) are not inserted in the ear canal(s) of the User.
- E. A Signature Sound Rejection Mode (SSRM) wherein a signature sound detected by one or more ASMs is reproduced with an Ear Canal Receiver, mounted in a fully or partially occluding CanalPhone assembly. The gain of the SSRM is such that the perceived loudness of the Signature Sound (by the User) is decreased (attenuated) compared to the case when the CanalPhones are not inserted in the ear canal(s) of the User.
- F. A Signature Sound Replacement Mode (SSRM) wherein a signature sound detected by one or more ASMs triggers reproduction of a second computer-generated warning sound with an Ear Canal Receiver, mounted in a fully or partially sealing CanalPhone. The second sound may be reproduced from an audio data buffer mounted within or external to the CanalPhone assembly, or the second sound may be generated computationally using a DSP-based audio generation algorithm.
- G. A User Generated Signature Sound Detection System (UGSSDS) wherein a signature sound detected by either or both the ASM and ECM activates a specific operation, such as recording the ASM and/or ECM and/or ECR signal to a data storage device.
- H. An automatic time-stamping system that generates a record of the time of occurrence when a signature sound is detected by the SSIDS. The time record may be stored on a database within computer memory that resides either within the CanalPhone assembly or on a second data storage device.

26. A system embed a Transient Evoked OtoAcoustic Emission (TEOAE) audio test signal embedded within a host music or non-music audio signals to create a new audio signal with minimal perceptible change to the perceived sound quality of the new audio signal, for the purposes of changes in hearing sensitivity or detecting potential hearing damage in a user, particularly following loud sound exposure. The TEOAE signal embedding comprises:

- A. An audio signal analysis system to identify regions of a host audio test signal with low voltage levels preceding loud transient voltages, where the loud transient voltages contents a substantially flat frequency spectrum.
- B. An audio signal generator for creating transient test signals for insertion into the host audio test signal. The signal generator may be separate from the analysis system in (A) and the test signals may be stored in computer memory.
- C. A system for inserting the audio test signal in (B) into the host audio signal at the identified time location.
- D. A post-processing signal processing module to dynamically adjust the gain and spectral response of the new signal generated on (C) so as to reduce the perceptual detection of the test signal generated in (B).

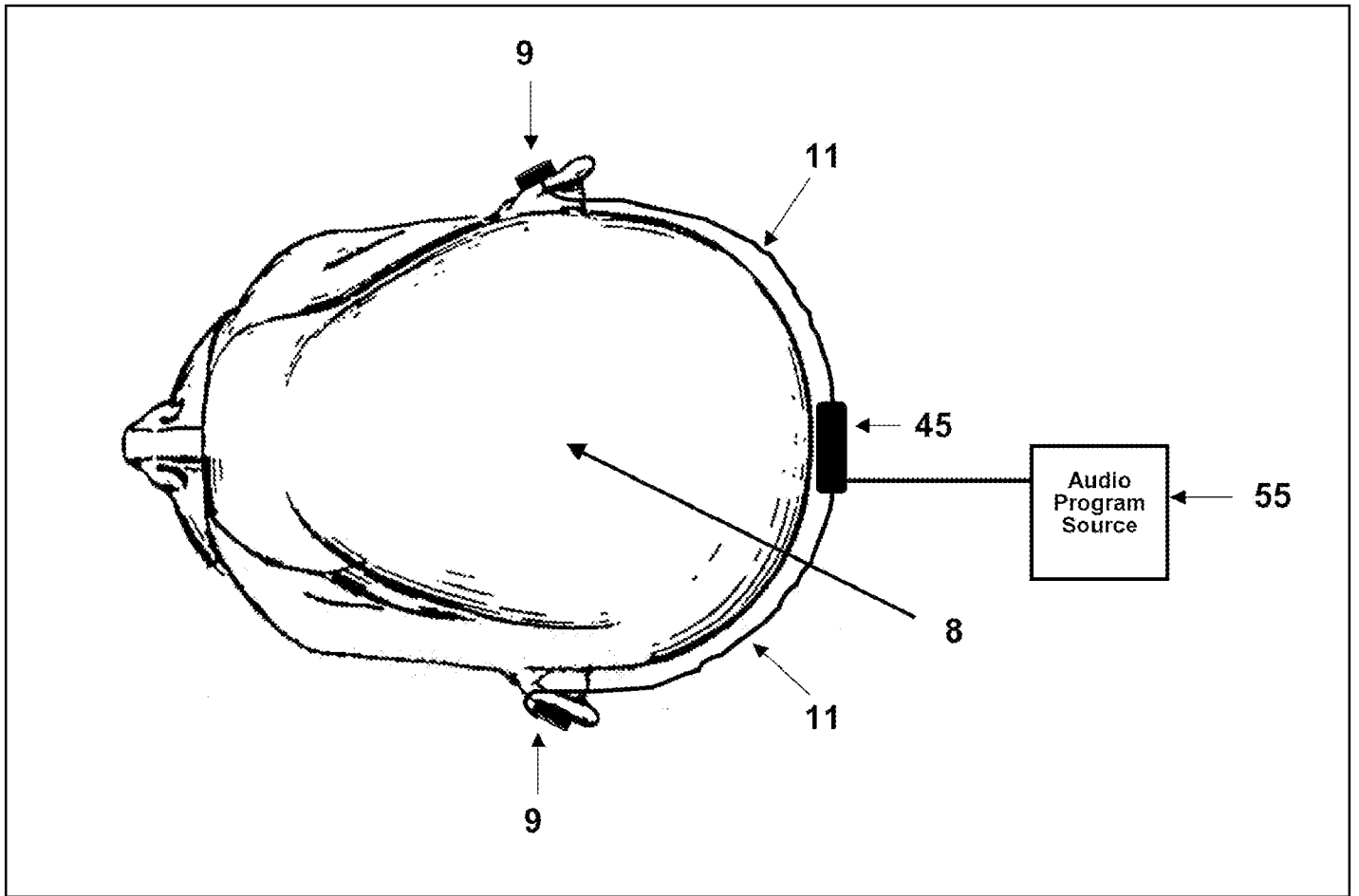


Figure 1b

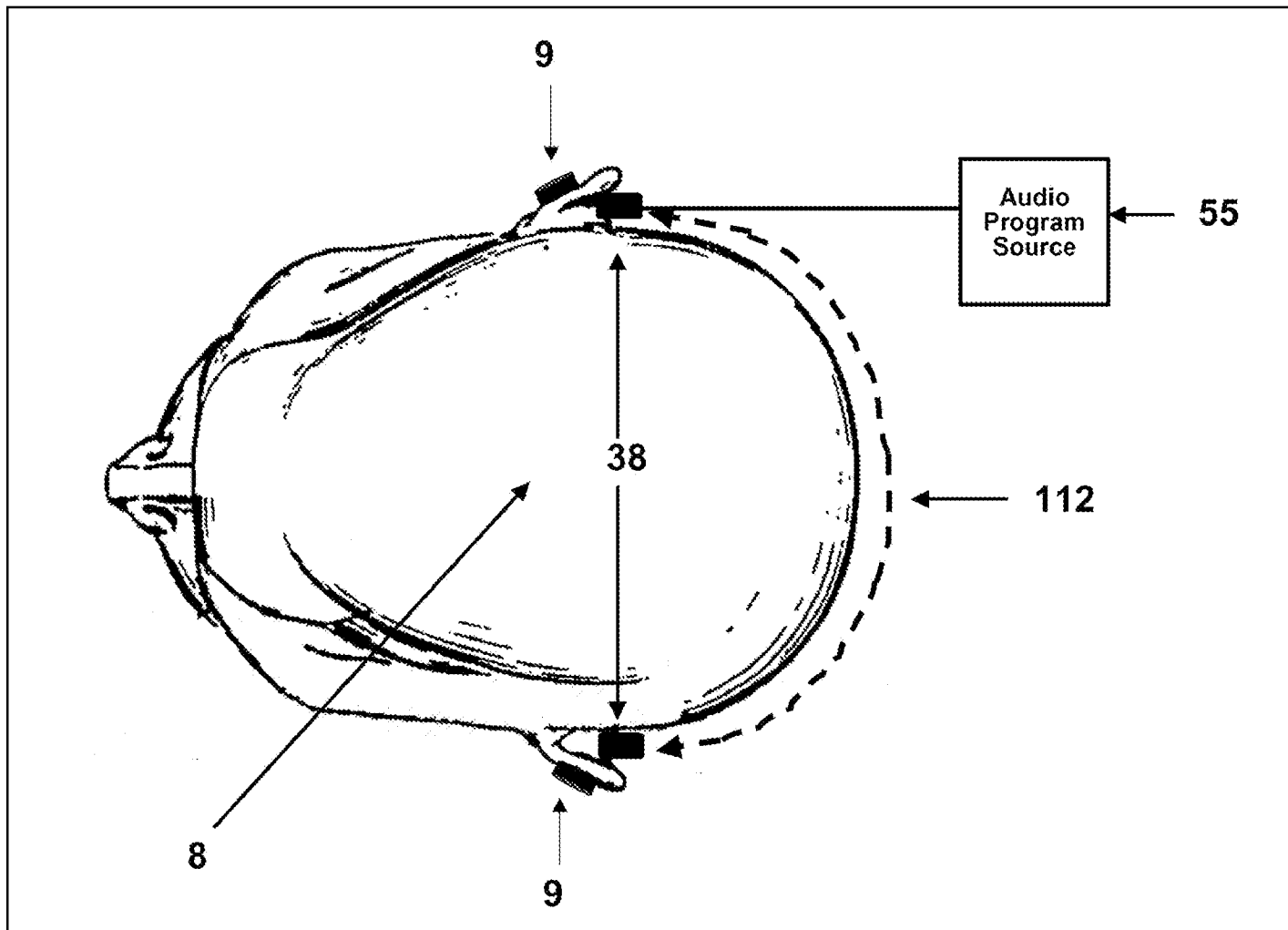


Figure 1c

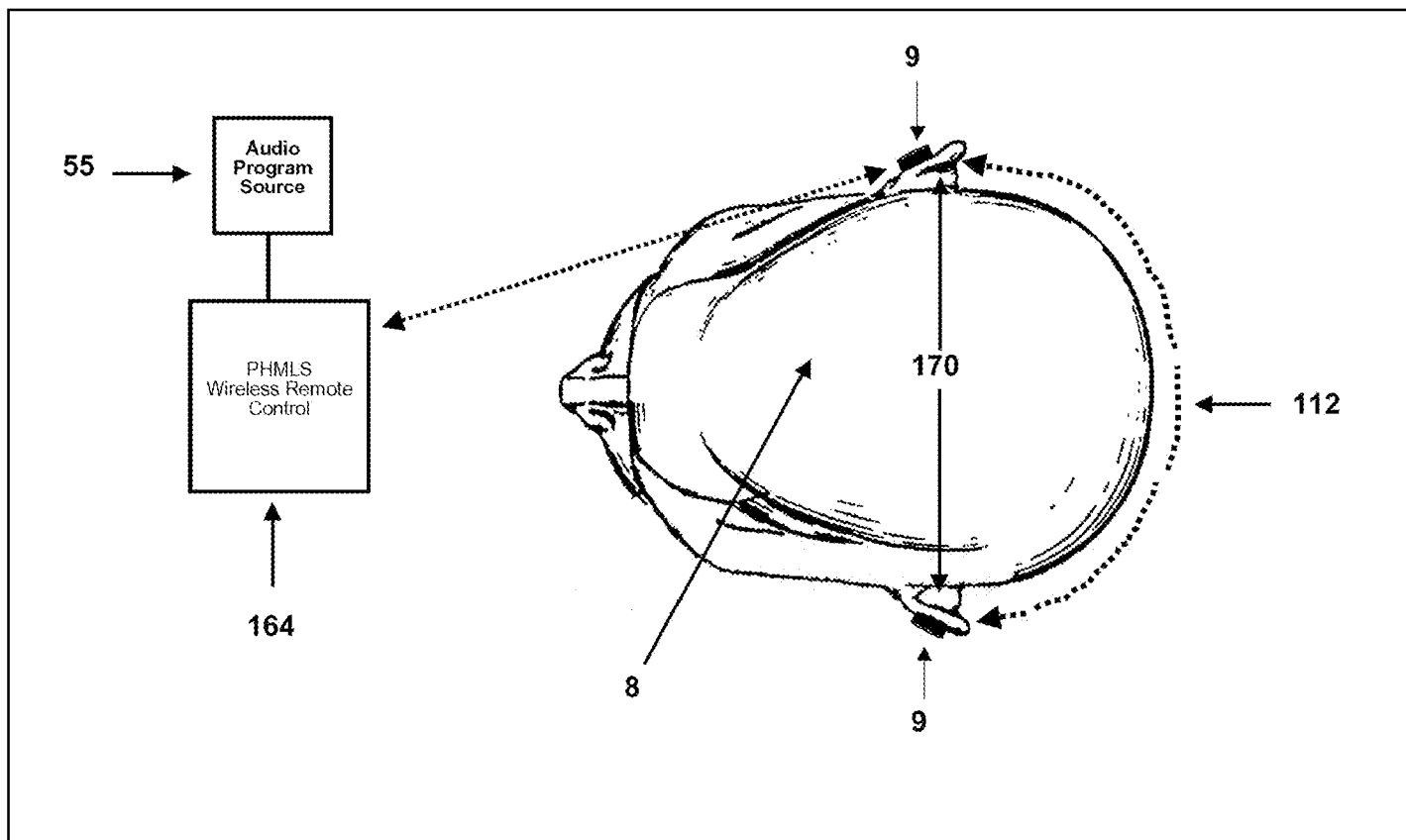


Figure 1d

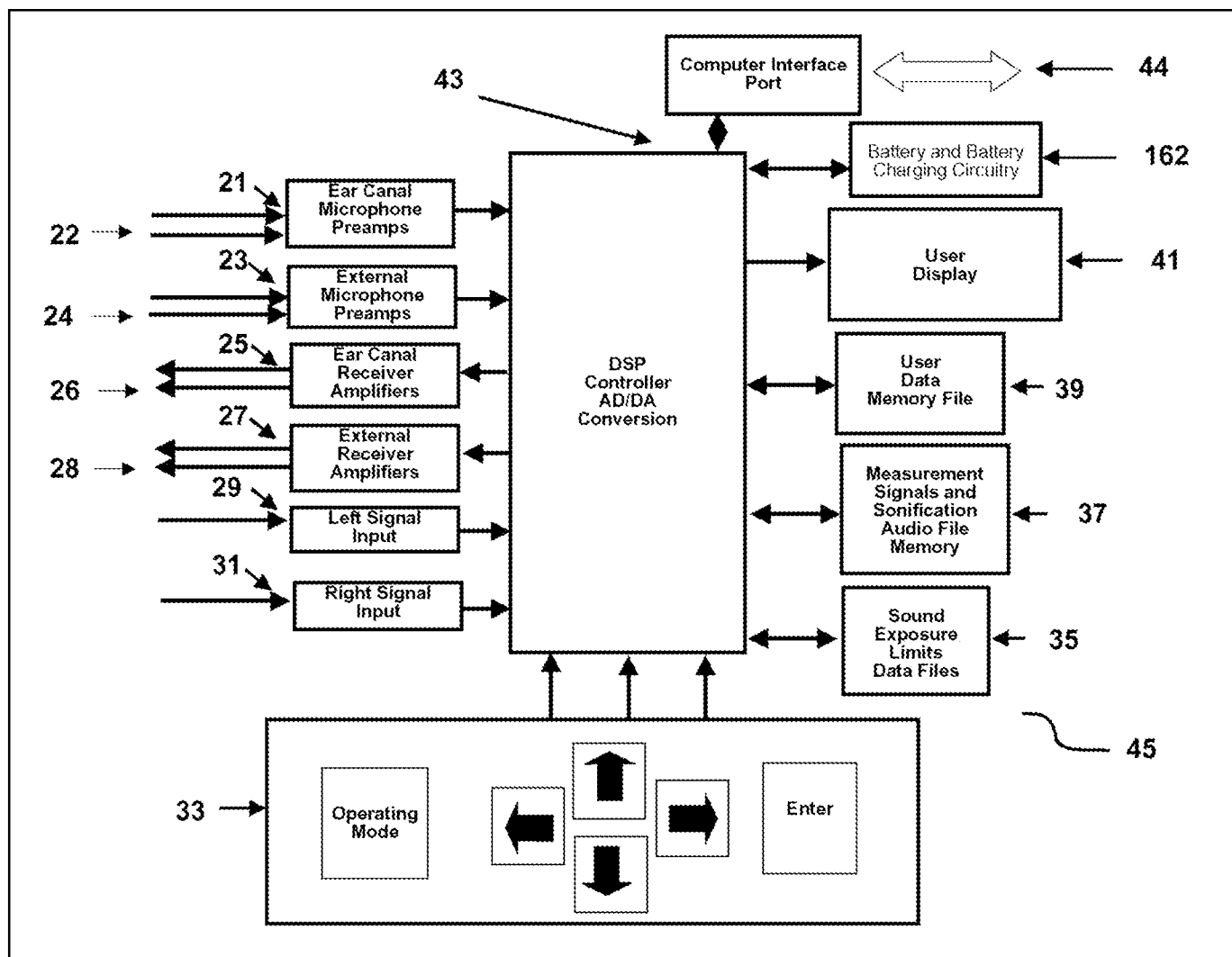


Figure 2a

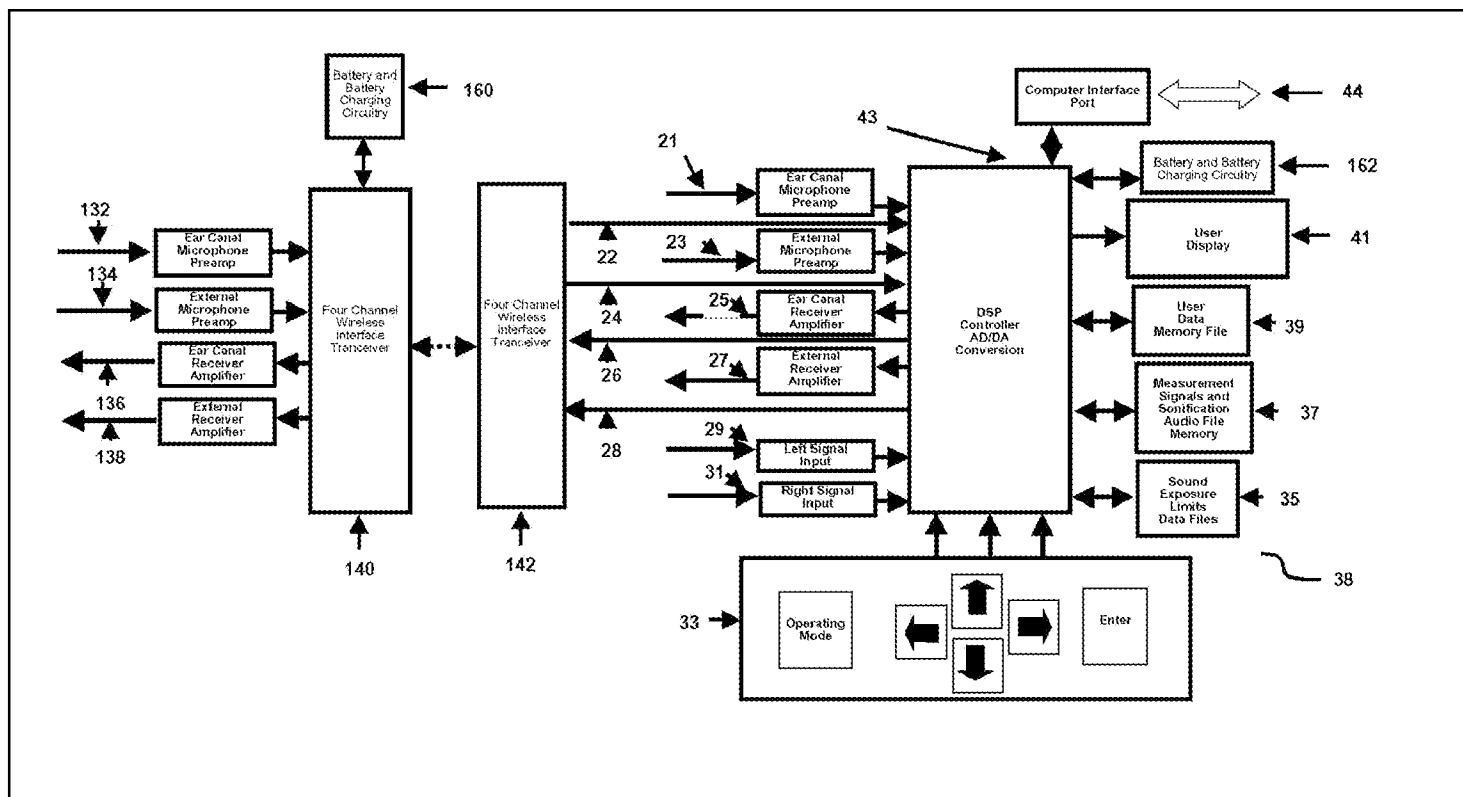


Figure 2b

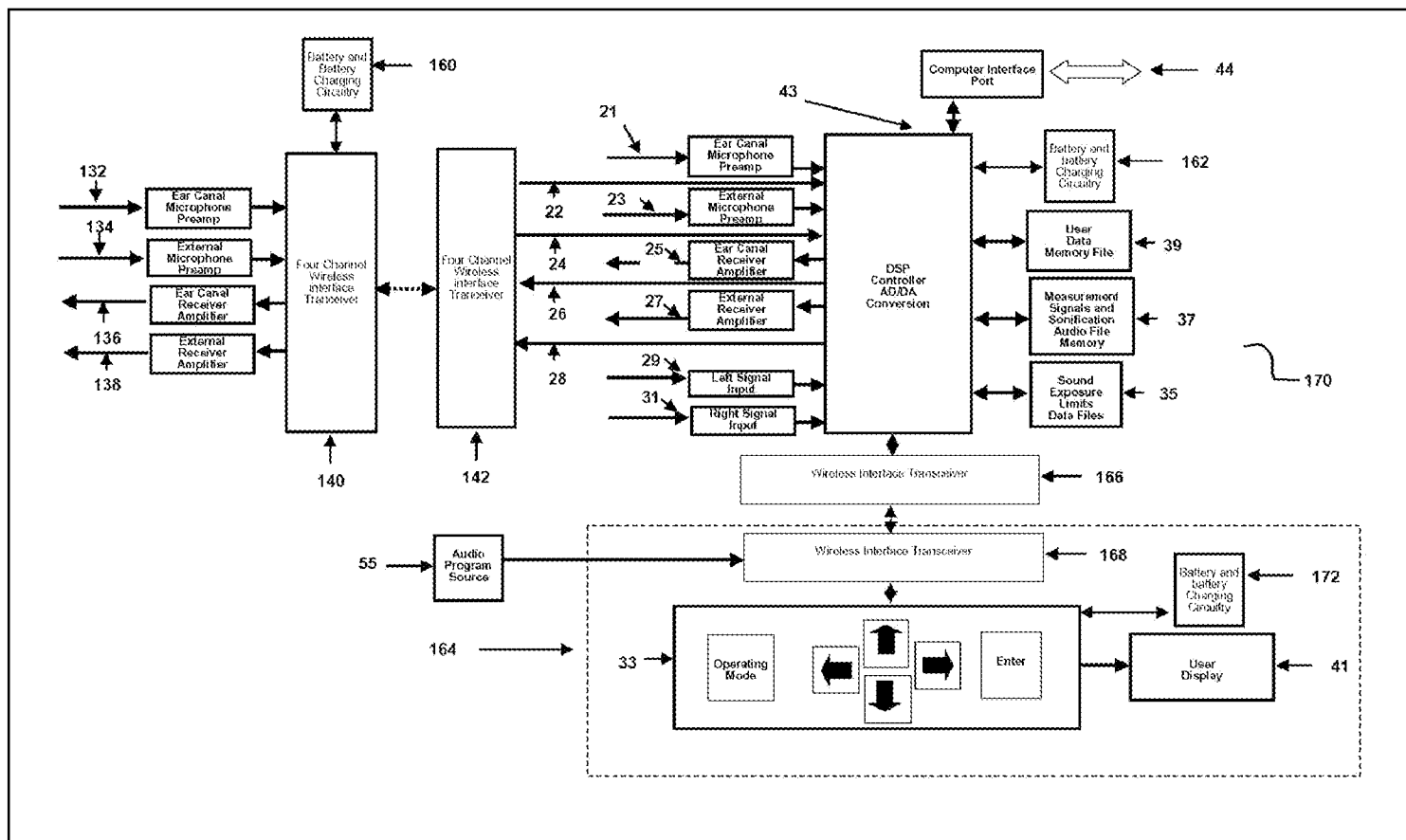


Figure 2c

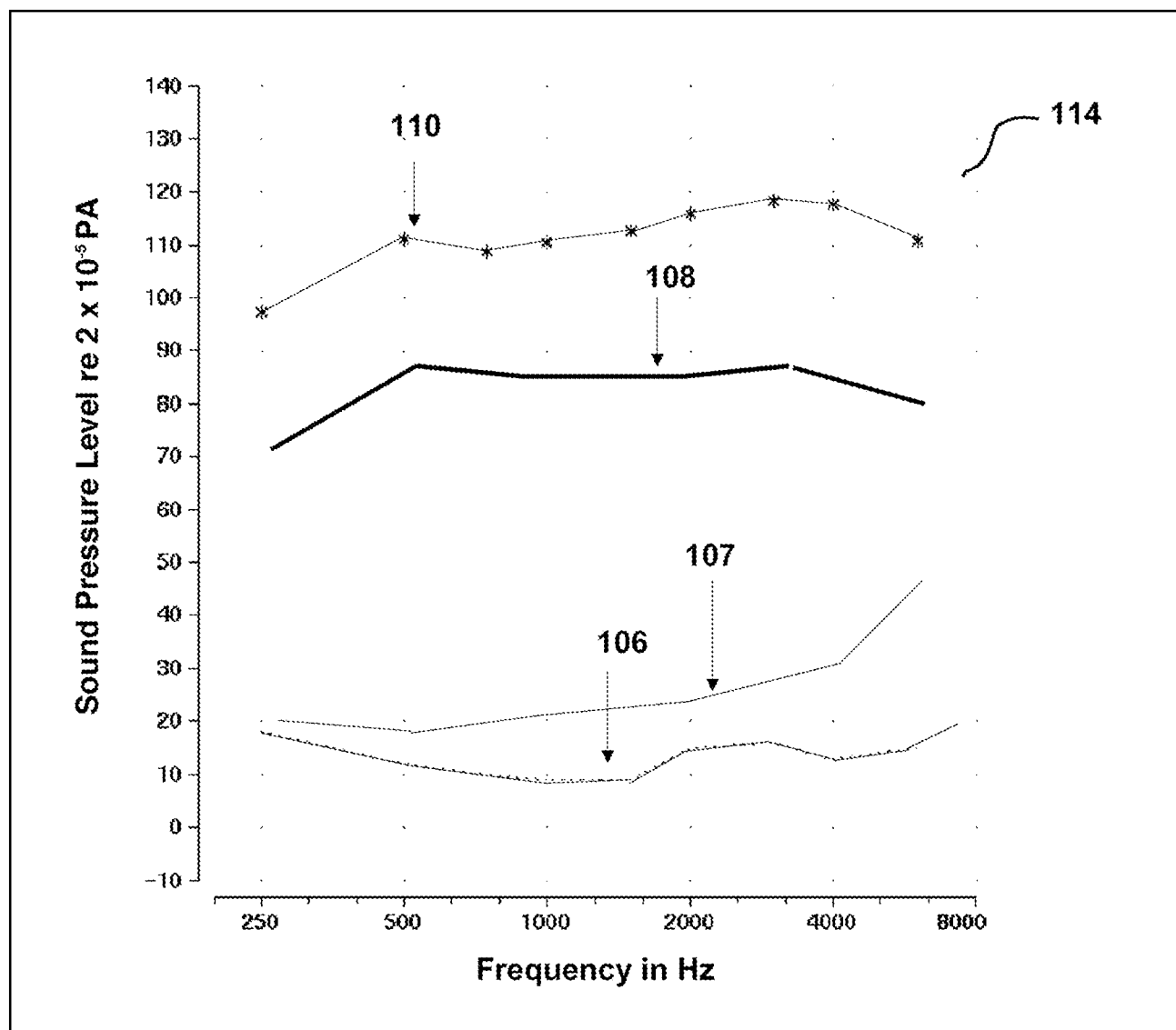


Figure 3

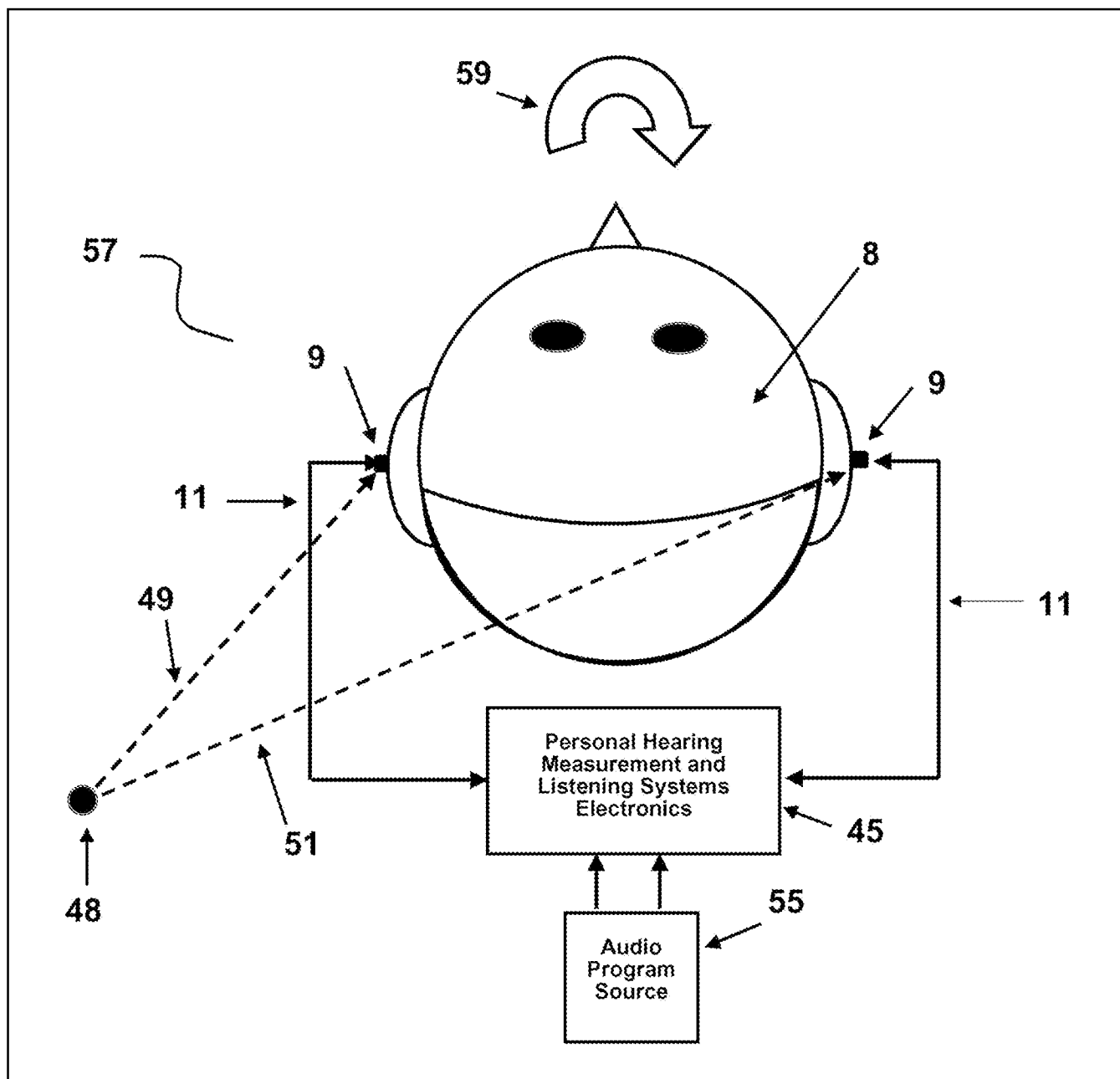


Figure 4

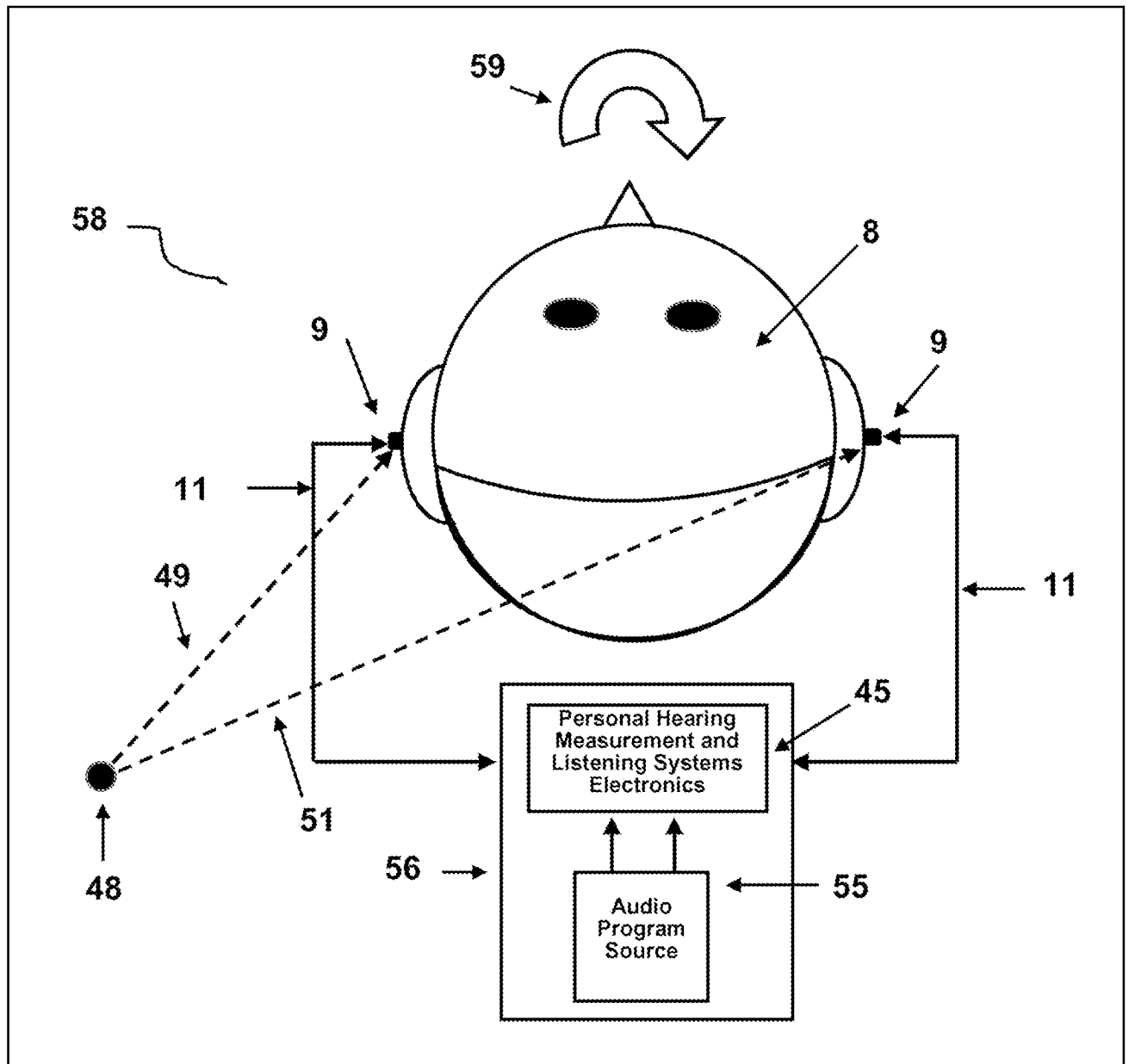


Figure 5

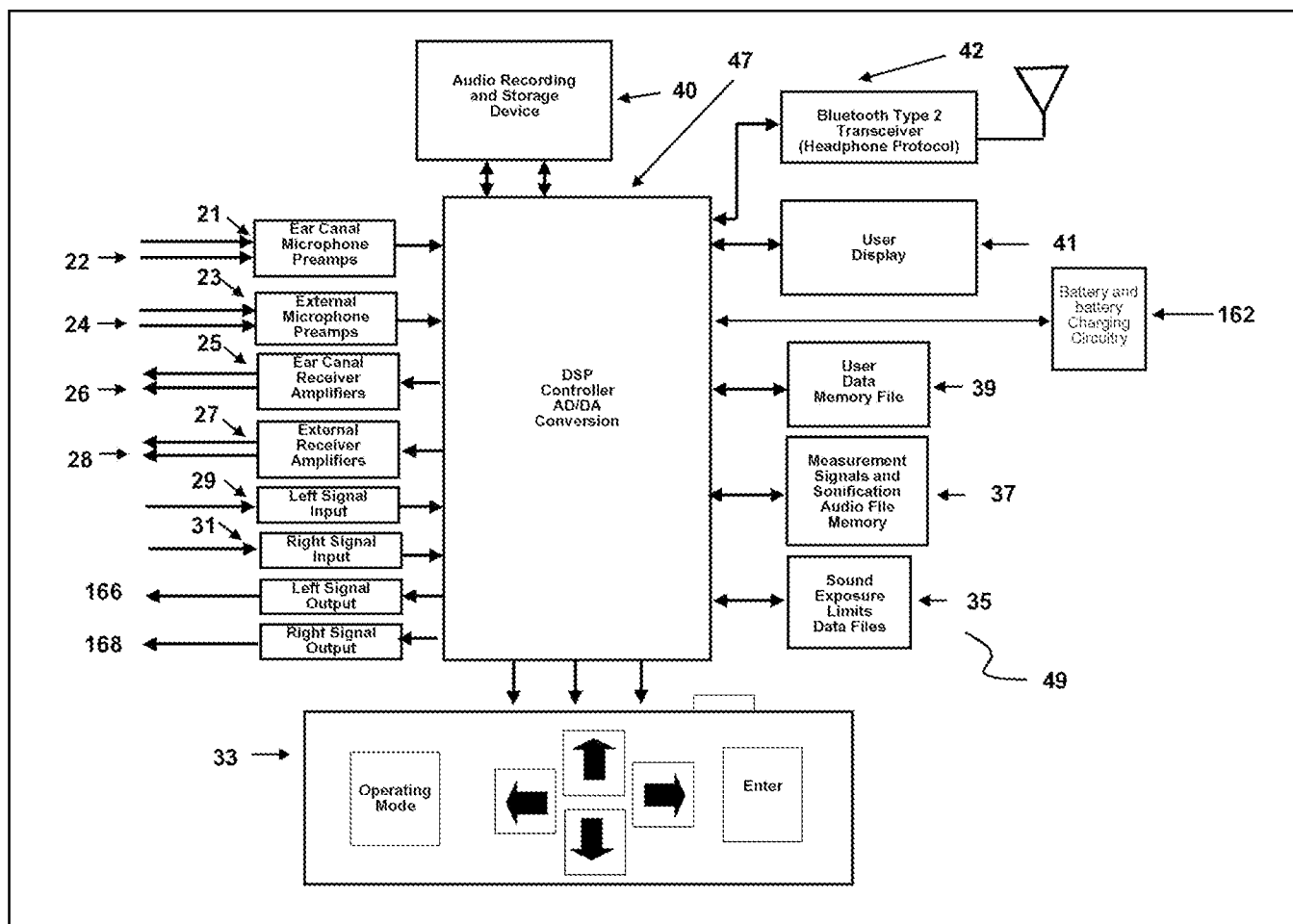


Figure 6

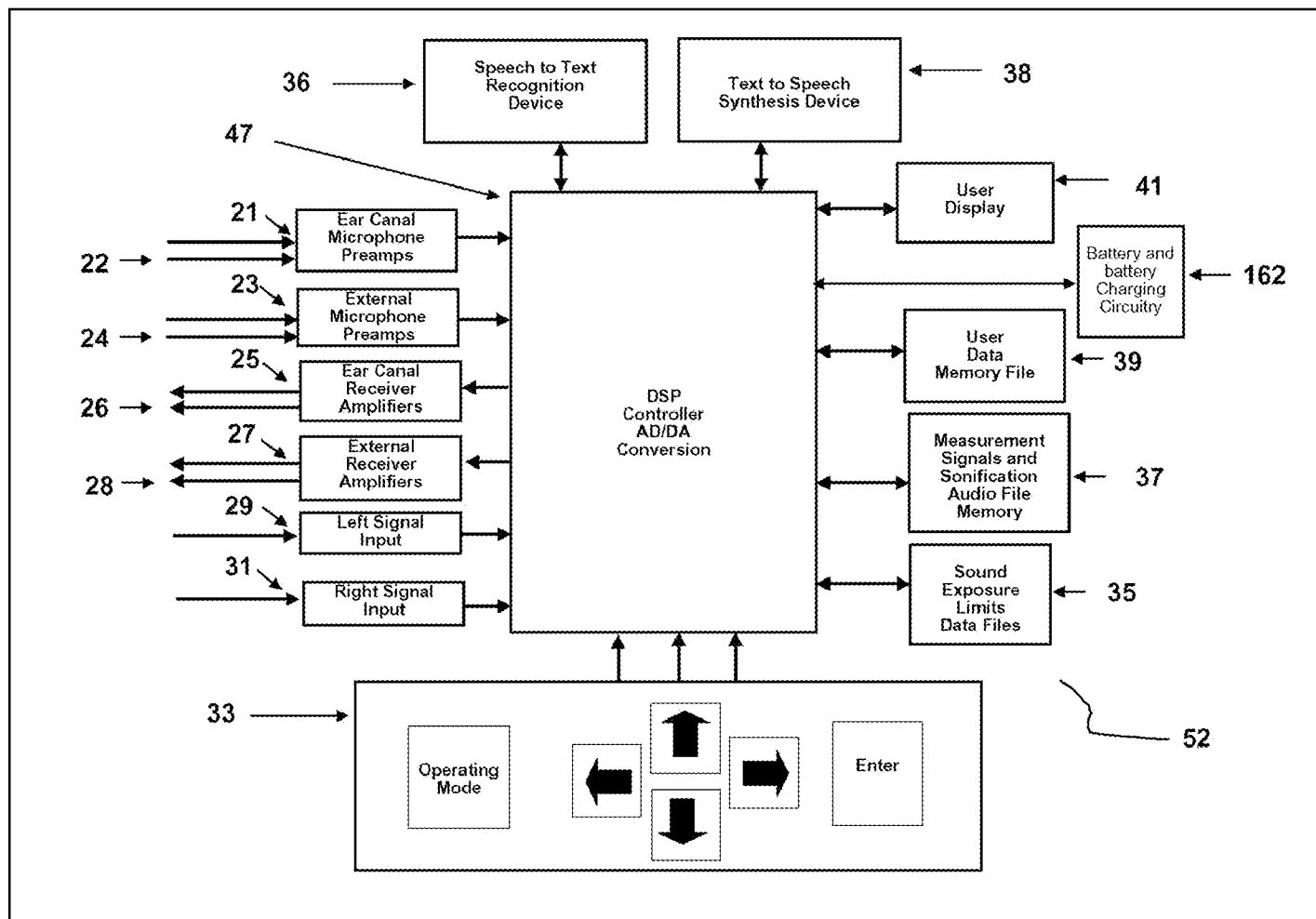


Figure 7

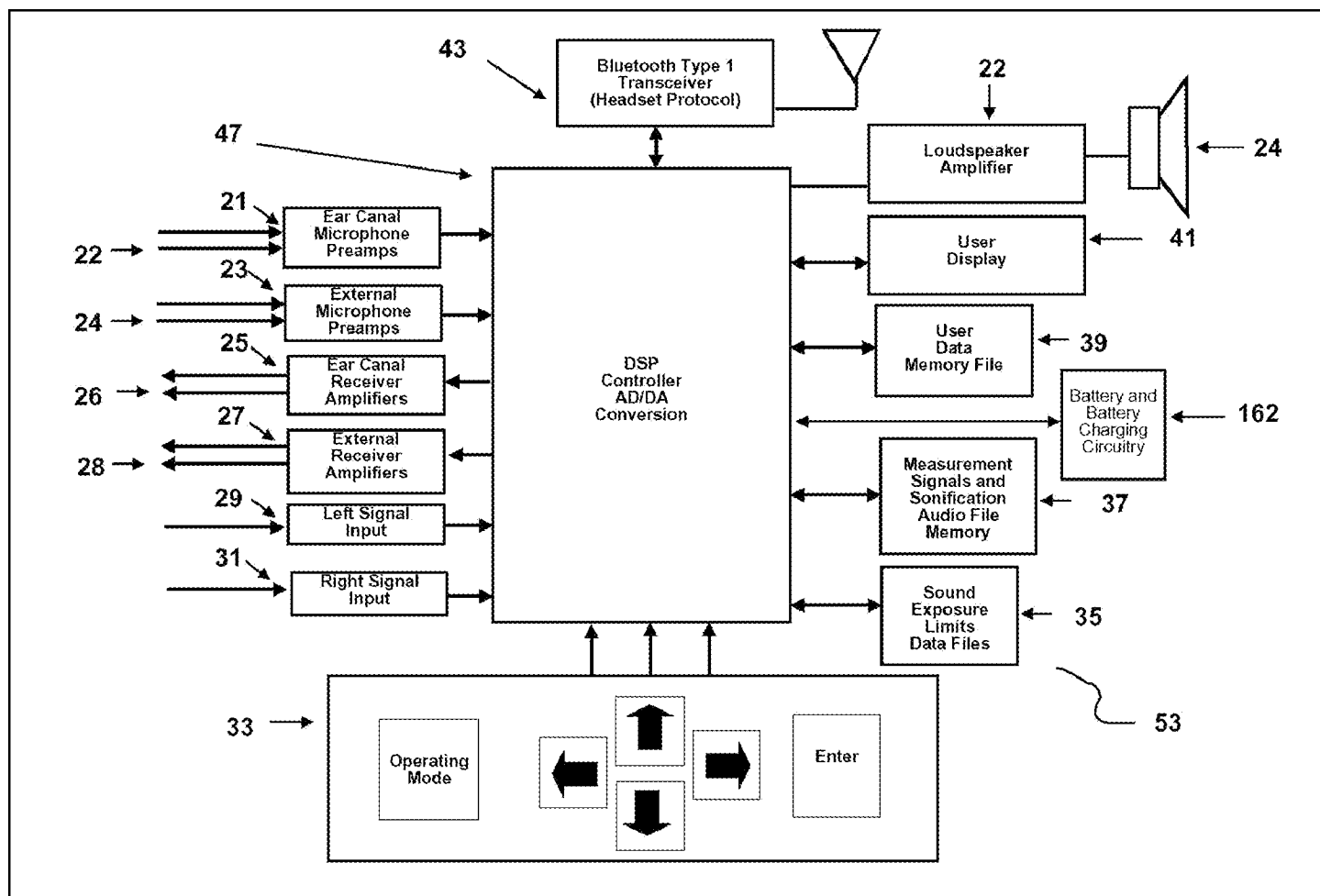


Figure 8a



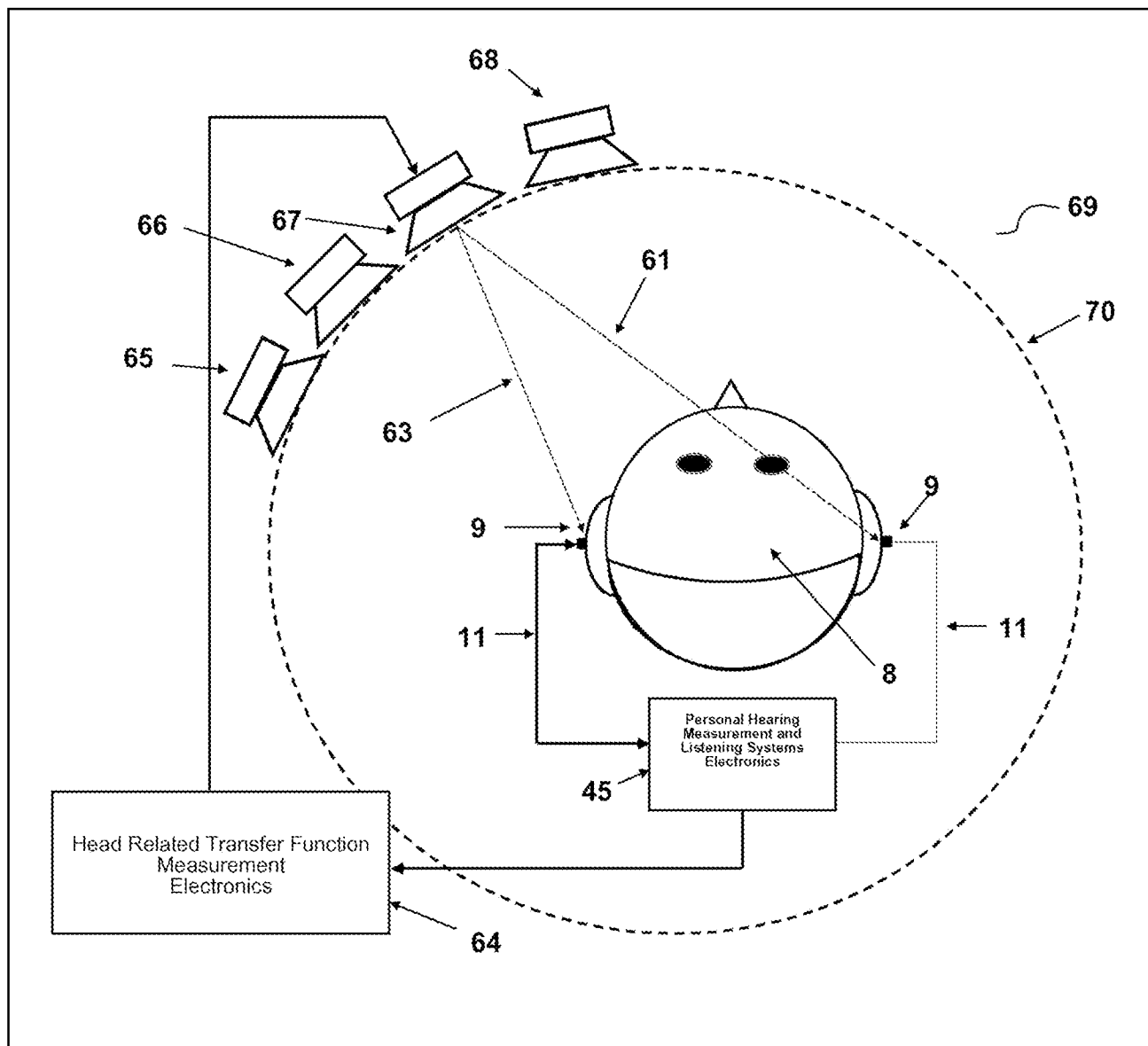


Figure 9a

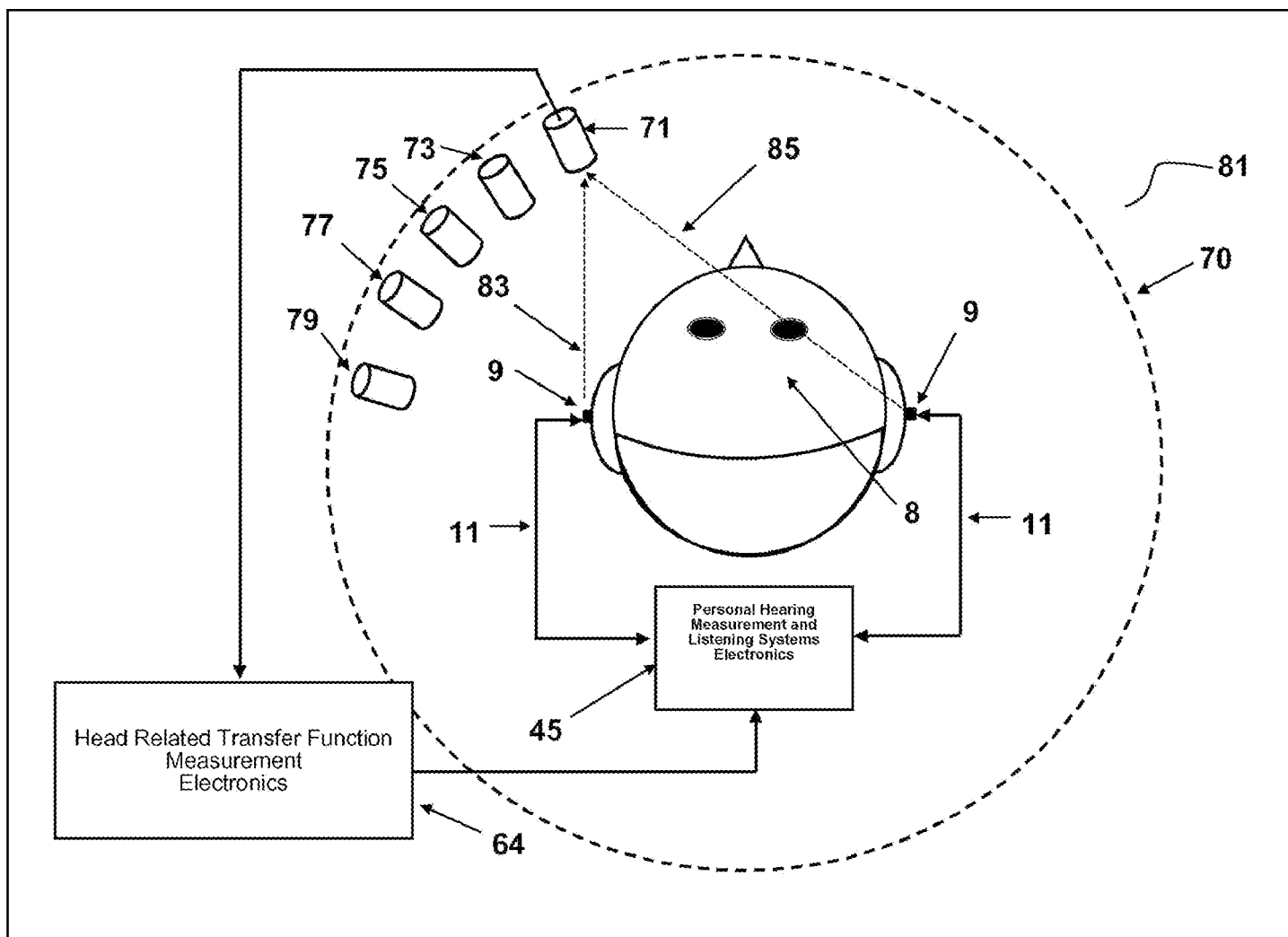


Figure 9b

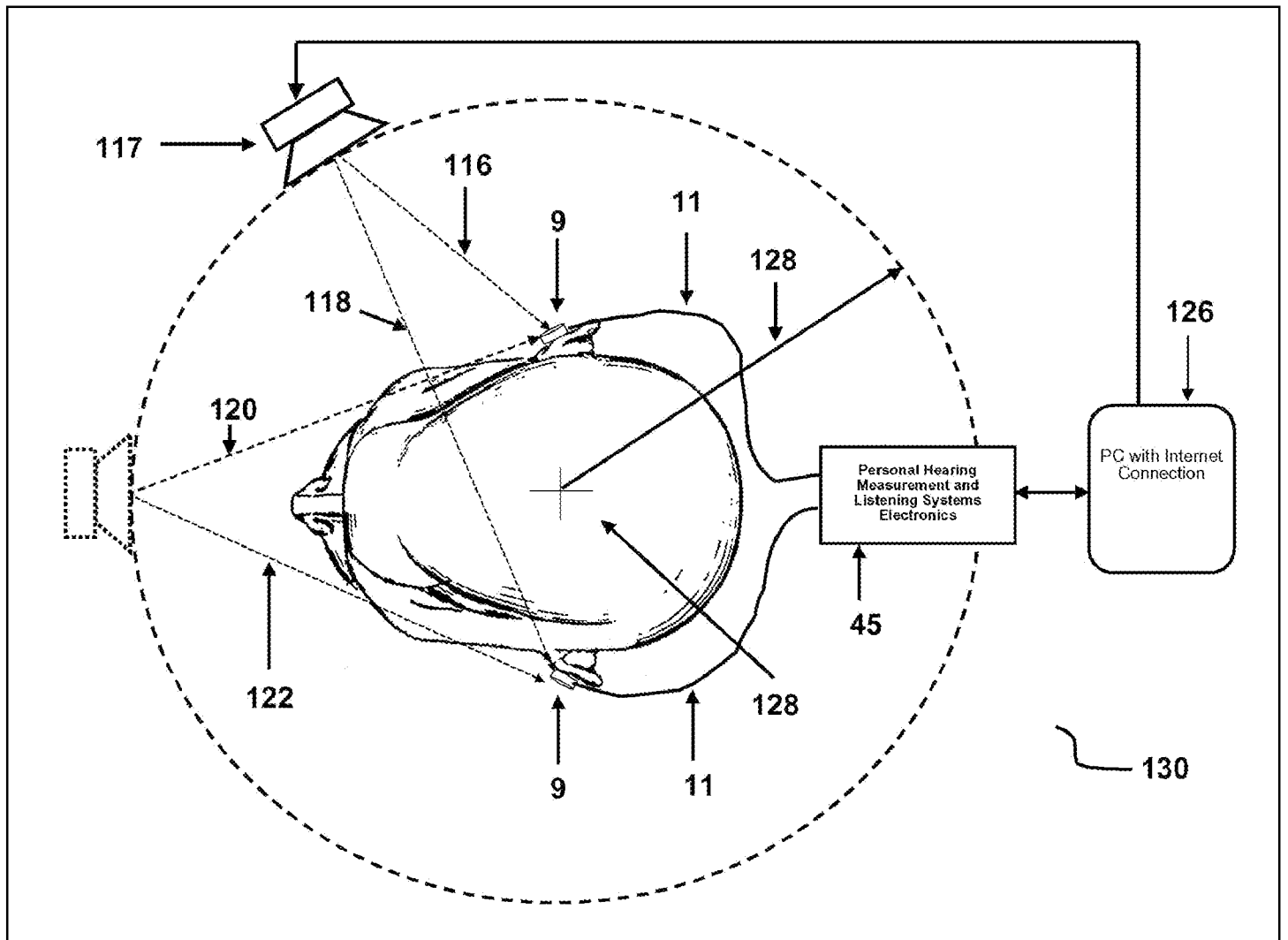


Figure 10

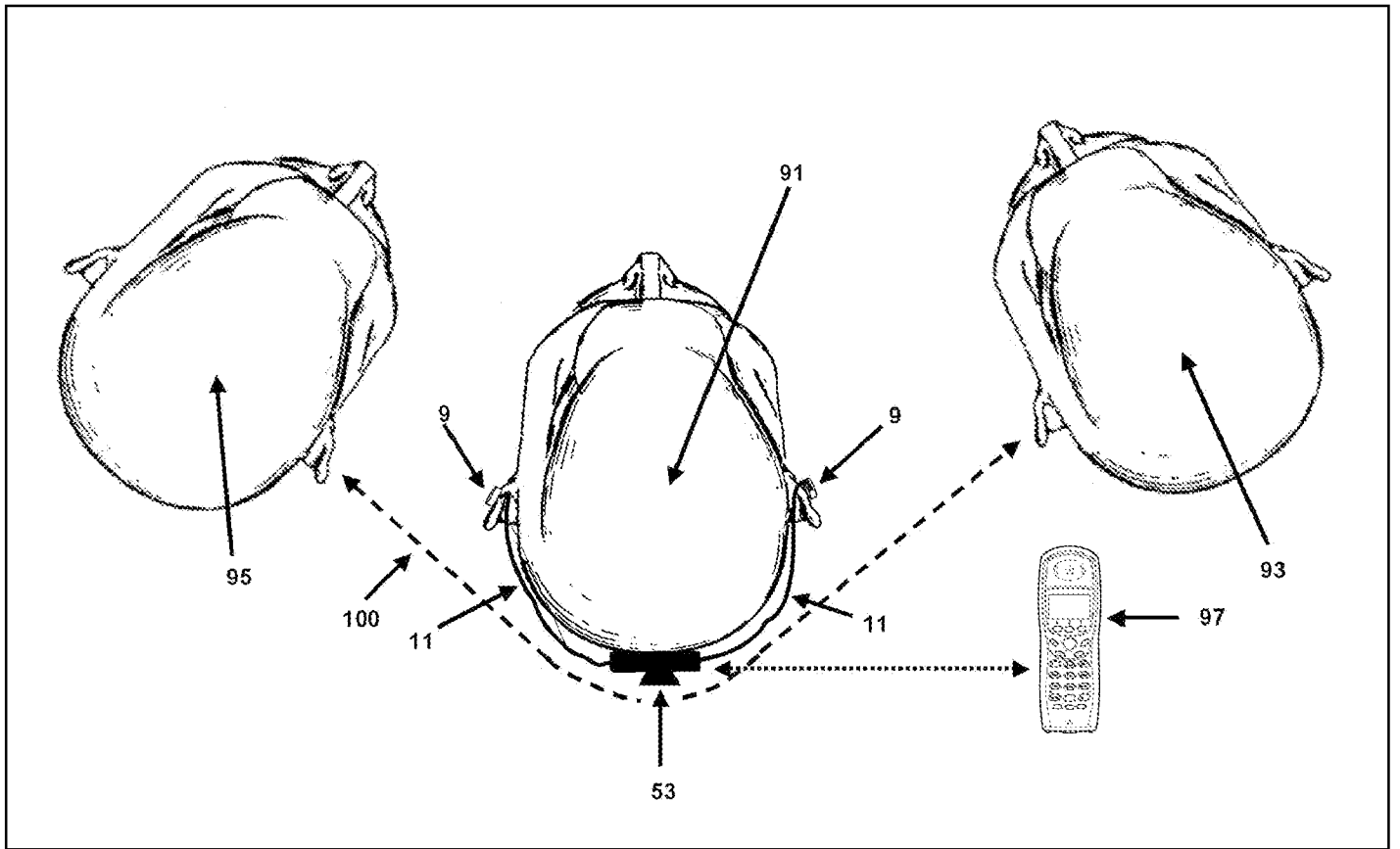


Figure 11a

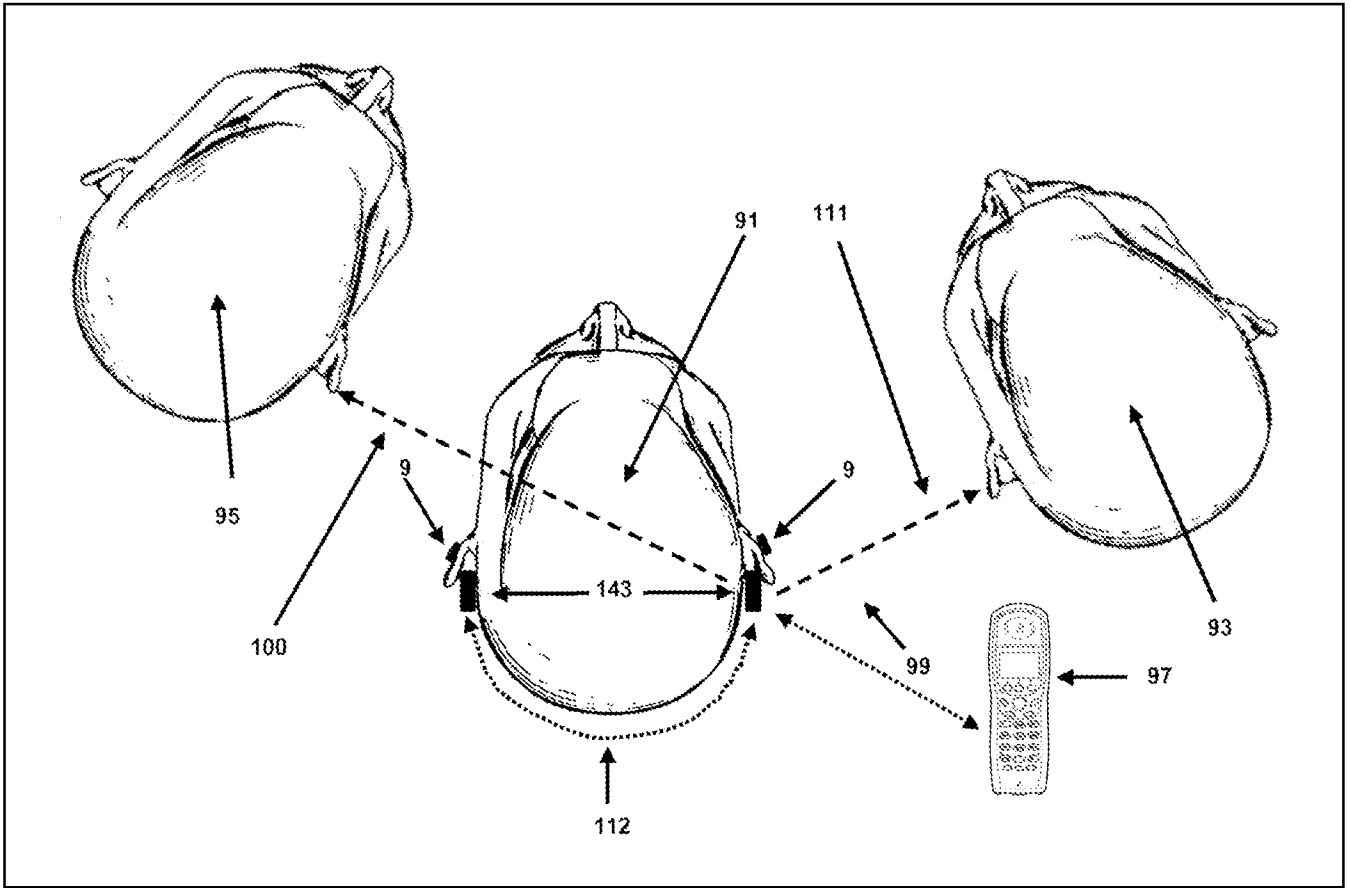
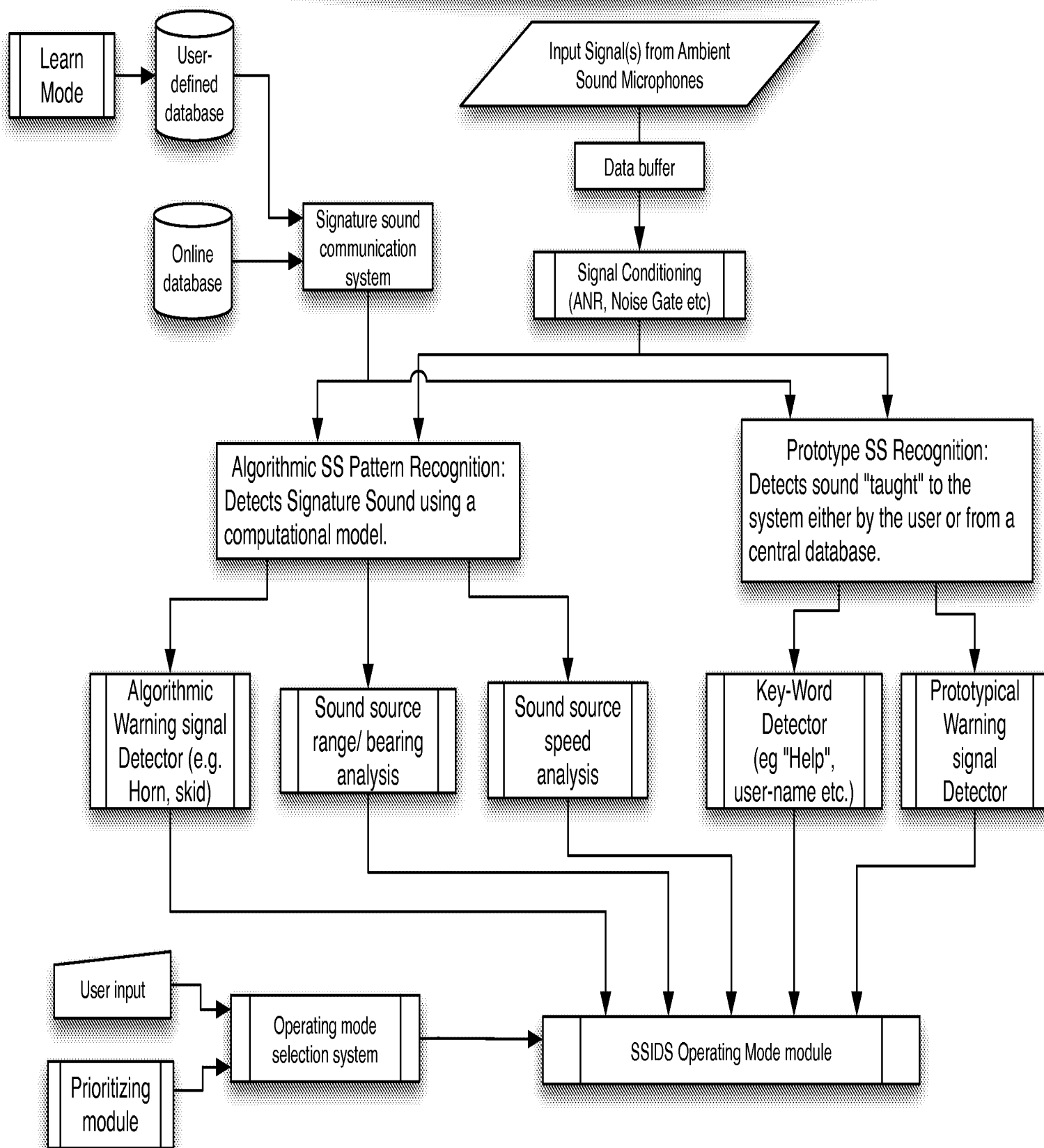
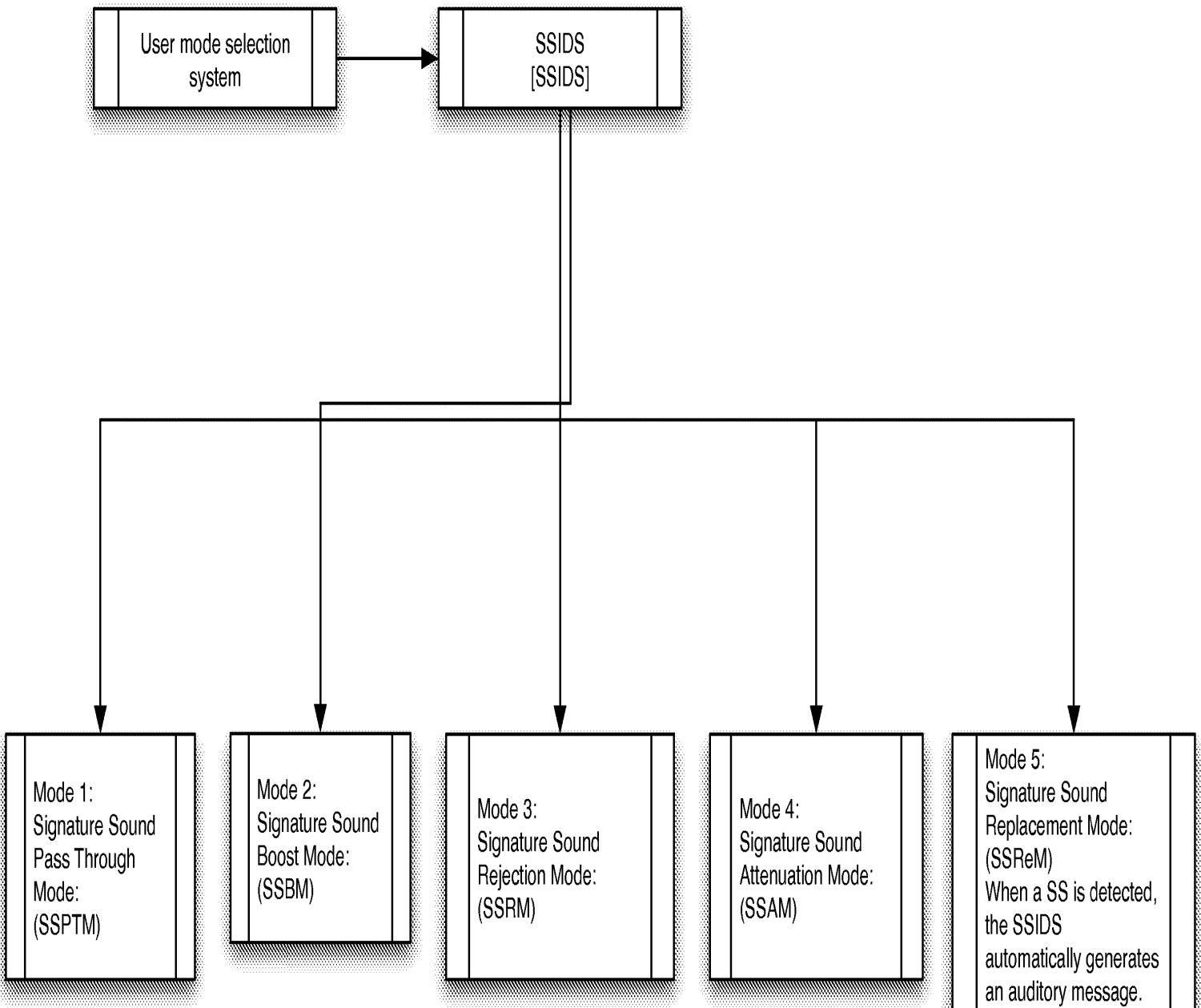


Figure 11b

Automatic Signature Sound Identification System (SSIDS):
 This module monitors the ambient sound field (using the Ambient Sound Microphones) for Signature Sounds. When Signature Sounds are detected, a number of things can happen depending on the SSIDS Operating Mode.



SSIDS Operating Mode module.



A Personal Hearing Measurement and Listening System (PHMLS.2)

Detailed Description of Drawings

Detailed Description of Drawings:

Figure 1a depicts the electroacoustical assembly **9** of a PHMLS, as it would typically be placed in the ear canal **12** of a user **5**. The assembly is designed to be inserted into the users ear canal **12**, and to form an acoustic seal with the walls of the ear canal at a location **18**, between the entrance to the ear canal **19**, and the tympanic membrane or ear drum **15**. Such a seal is typically achieved by means of a soft and compliant housing of assembly **9**. Such a seal is critical to the performance of the system in that it creates a closed cavity **13** of approximately .5cc between the in-ear assembly **9** and the tympanic membrane **15**. As a result of this seal, the Ear Canal Receiver (ECR), or Speaker **1** is able to generate a full range bass response when reproducing sounds for the system user. This seal also serves to significantly reduce the sound pressure level at the users eardrum resulting from the sound field at the entrance to the ear canal **19**. This seal is also the basis for the sound isolating performance of the PHMLS. Located adjacent to the ECR, is an Ear Canal Microphone (ECM) **3**, which is also acoustically coupled to closed cavity **13**. One of its function is that of measuring the sound pressure level in the cavity 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.

Since a reference transfer function can be established between ECR**1** and ECM **3**, for each user, changes in this relationship due to seal integrity, wear or contamination by ear wax or cerumen can be identified when desired or automatically such as when the PHMLS is turned on. Once such a change is detected, the user can be made aware of such a condition by an audible warning reproduced via ECR **1** and or display message **41** as shown in **Figure 2a**. Furthermore once a change in seal integrity, the PHMLS will attempt to auto correct the problem through equalization, so long as the output capabilities of the system are not significantly compromised.

In another iteration, ECR **1** and ECM **3** can be used as a part of an Otoacoustic Emissions (OAE) measurement system. Such a measurement would involve pure tones or transient signals to be reproduced by ECR **1** that are down loaded from the “Measurement Signals and Sonification Audio File Memory” **37** as shown in **Figure 2a**. These signals would be subsequently introduced into the ear canal **13** of the user, and the resulting acoustical emissions from hair cells of the inner ear detected, analyzed and stored in the “User Data Memory File” **39** also shown in **Figure 2a**. Published research indicates that such measurements can be used to identify previous exposure to very loud acoustic sound pressure levels that are capable of producing short term hearing threshold shifts, in addition to leading to a permanent loss in hearing over time.

An additional receiver, the External Receiver (ER) **2**, and Ambient Sound Microphone (ASM) **4** are located at the opposite end of electroacoustical assembly **9**, so as to create and monitor sound pressure levels at the entrance to the ear canal **19**. Circuit module **7** serves as a means of connecting signals to and from transducers **1, 2, 3, 4** to multi-conductor cable **11**. Cable **11** may then be connected to a variety of signal processing circuitry modules as further described in **Figures 2a, 2b, 6, 7, 8a and 8b**, to perform a variety of listening, hearing protection, communications, and Environmental Audio Sound Field (EASF) monitoring tasks.

Furthermore by comparing the sound pressure levels inside the ear canal **13** as measured by ECM **3** and outside the canal **19**, as measured by ASM **4**, a PHMLS can calculate the degree of sound isolation or protection provided by each inserted electroacoustical assembly **9**. These characteristics can be represented as attenuation values vs. frequency, and stored in the User Data Memory File **39**. If the attenuation values are considered to be inadequate for a desired level of hearing protection, the PHMLS user will be notified by a combination of audible warnings presented by ECR **1** and display graphics on User Display **41**.

Figure 1b shows how left and right in-ear assemblies as shown in **Figure 1a** may be combined with the signal processing electronics as shown in **Figure 2a** to form a wearable PHMLS. In this example the left and right in-ear electroacoustical assemblies **9** are directly wired by means of cables **11** to connections **21, 22, 23, 24, 25, 26, 27, 28 and 29** of electronic circuitry **45** detailed in **Figure 2a**.

Figure 1c depicts an alternate method of configuring a PHMLS by means of wireless transmission techniques described in blocks **180** and **182** as shown in **Figure 2b**. In this case left and right in-ear electroacoustical assemblies **9** are individually connected at one ear to inputs **132 and 134**, and outputs **136**, and **138 associated** with wireless transceiver **140**, which broadcasts and receives signals from wireless transceiver **142** located on the other ear. Connection from electroacoustical assembly **9** for this ear are made through inputs **21** and **23**, and outputs **25**, and **27**.

Figure 1d depicts still another method of configuring a PHMLS by adding additional wireless connection links as further described in blocks **180, 170** and **164** in **Figure 2c**. In this configuration the in-ear electroacoustical components **9** as shown in **Figure 1a** are combined with components **180** in one ear. In the opposite ear a second set of in-ear electroacoustical components **9** are combined with components **170**. Furthermore these sides of the system communicate with each other by means of wireless transceivers **140** and **142**. In addition the user control panel and display are wirelessly linked to the main PHMLS signal processing circuitry by means of transceivers **166** and **168**. The wireless remote control **164** further provides a means for wirelessly linking an external audio data and communication device **55** to the PHMLS. Examples of such devices include; an audio player, an audio recorder, a data file storage device, and a computer.

When configured as a PHMLS a number of functions can be performed with respect to both listening as well as measuring the hearing acuity of a system user.

Referring to system **45** shown in **Figure 2a**, user control switches **33**, can be used to start, stop and perform tests as well as save, recall and transfer data within the system or to an external device through files **35**, **37** and **39** and computer interface port **44**, which may also be wireless.

Two particular tests of value are the determination of a users hearing threshold sensitivity vs. frequency or audiogram, and the users ability to understand speech in speech like noise. The PHMLS allows each of these measurements to be performed by means of a self-administered test.

In order to first activate and configure a PHMLS so as to make this and other measurements a customer will be required to electronically contact a customer support web site for purposes of registering and activating the PHMLS product. This process will involve matching an electronic serial number supplied with the product with a companion serial number electronically stored in the User Data Memory File **39**. This data may also be distributed to a server location for further analysis. The user will also be given an opportunity to share their audiogram and other measured data with the customer support site in exchange for commercial products or services.

The primary objective of determining a users audiogram is that of accessing the lowest level of pure tones hearable by a user as a function of frequency (from 250 Hz to 20 kHz). Prior to running this test, the ECM **3** facilitates the measurement of the background noise in the ear canal so as to determine if an adequate signal to noise ratio exists so as to obtain reliable data. In addition to the use of the ECM **3**, the ASM **4** can be used to monitor the ambience sound level at an ear canal entrance, which may well be contributing to the overall sound level in the test subject's ear canal **13**. If the measured sound level in the ear is sufficiently low to allow accurate threshold sound pressure level measurements, the system will indicate to the user by sonic or visual means via an ECR **4** or display **41** that the system is ready for testing. If desired, the full battery of testing may be performed in one ear at a time or by cycling between ears thus reducing listening-testing fatigue.

Once initiated by means of control switches **33**, the user is presented with audible instructions and test signals from the "Measurement Signals and Sonification Audio File Memory **37**". The user will be asked to respond each time a tone is heard during the test. As the test progresses the users performance will be stored in user data file **39** along with the users identification and date of measurement, for future reference and analysis. This data can be used by a PHMLS to benefit a user by compensating for a threshold hearing loss, through the use of low-level expansion functions. Such functions boost inaudible (as a result of hearing loss) low-level sounds to levels of audibility. For higher-level sounds that are normally audible, no signal boosting will be implemented.

With respect to the measurement of speech intelligibility in speech like noise, the users task is that of properly understanding what is being said during each test in which a nonsense saying such as “her purse was full of useless trash” is voiced over a conflicting level of multi-talker babble. Depending upon the ratio of desired talker level to noise level, the intelligibility task is easy or difficult. Once the test sequence is initiated by means of control switches **33**, by the user, audible instructions and test signals from the “Measurement Signals and Sonification Audio File Memory **37** are presented. As the user proceeds through the test a point will be reached where the desired speech is just understandable to the user, in that most all of the speech is understood. It is at that point where the system will pause and determine the signal to noise ratio of speech to background noise level (typically -5 to +10 dB). The results of this test will be stored in the User Data File **39**, along with the users identification and date of measurement, for future reference and analysis. In addition this data may be downloaded to a server.

In addition to a users ability to perceive very low sound pressure levels, and to understand human speech in a noisy environment, all users have desirable and undesirable preferences for the levels of sound. The terms Most Comfortable Level (MCL), and Uncomfortable Level (UCL) are commonly used to describe such listening attributes. Such levels can be measured and used by a PHMLS. In this process the user activates a test sequence from controls **33**, in which a test signal is reproduced from the Measurement Signals and Sonification Audio File Memory **37**. These files are reproduced at a variety of sound pressure levels and frequencies, and the user makes a selection of both an MCL and UCL condition for each ear. Once selected the PHMLS is able to measure the actual SPL in the users ear canal for these situations. These data are then stored on the Sound Exposure and Limits Data Files, along with the identity of the user and the date. This data can be used to benefit a PHMLS in various ways such as the following:

1. Automatically setting a limiter threshold to avoid the reproduction of sounds that exceed an UCL selection for each ear.
2. Automatically setting the level of reproduced sounds so as to match a MCL selection suitable to both ears.

Individuals using a PHMLS will have a wide variety of hearing capabilities, which are subject to change over time. By having an initial and ongoing record of hearing acuity, a user will be able to detect potential hearing health issues and seek professional advice as well as gain more confidence as to how their choice of listening levels are affecting their hearing abilities. In another embodiment, the provider of the PHMLS may make recommendations to the user about seeking professional audiological consultation based on the results of measurement data.

In addition to the measurement of hearing acuity, a PHMLS provides a means of measuring the actual sound pressure levels that are occurring in a users ear canals resulting from the users choice of listening level of reproduced sounds and non-reproduced sounds from the users environment.

Such sound pressure levels are determined by measurement of the output voltage of the ECMs located in each sealed ear canal. By previous knowledge of the acoustic sensitivity of the left and right ECMs, the signal processing portion of a PHMLS **43** as shown in **Figure 2a** calculates a time weighted average (TWA) measure of sound pressure level with an ability to display its value on a user display **41**, or by means of an audible signal that is reproduced in the users ear by an ECR **1**. In addition these measures are compared with accepted international noise exposure limits as stored in a Sound Exposure Limits Data File **35**. Based on these comparative measures, a PHMLS can inform a user and automatically adjust as to how they are progressing with respect to what is considered a safe daily sound dose. Such a sound dose is typically based on a maximum exposure of an A weighted 85 dB SPL for 8 hours with an exchange rate of 3 dB. By using this dose reference, the PHMLS is able to compare repeated TWA measurements, noting for example, that a SPL of 85dB +3 dB reduces the exposure time by a factor of two (4 hrs), and conversely an SPL of 85dB-3dB increases the exposure time by a factor of two (8hrs). This information may consequently be used by the listener in a variety of ways, by means of user selectable operating modes via controls **33** such as the following:

1. As a training indicator to allow continuous listening at safe levels.
2. As a training indicator to allow limited listening times at higher listening levels.
3. As a means of informing a listener that a daily dose based on a specific reference standard has been received.
4. To initiate audio compression actions on reproduced audio to increase safe listening times.
5. To initiate a parental controlled shut down of the system for the balance of a day once a complete dose has been received.
6. To initiate a parental compression control that initiates increasing levels of compression as a full daily dose is approached.

Due to the fact that a PHMLS has the ability to measure actual sound pressure levels in users ear canals, high level sounds capable of producing “ringing in the ears” and or a Temporary Threshold Shift (TTS) can be detected. Once detected a PHMLS has an ability to inform the system user that such a hearing condition may exists as well as make suggestions as to user settings to avoid such problems in the future.

Figure 3 is a graphical presentation **114**, of several of the functions that a PHMLS can measure and accommodate. The graph shows sound pressure levels of five typical perception characteristics vs. frequency for humans:

1. The lower limit of pure tone audibility for a person with normal hearing **106**, representing the lowest levels of sound perception.
2. A threshold measurement from an individual experiencing a loss of low-level sound perception, which increases for higher frequency sounds **107**, representing an individual that will not be able to hear certain low level sounds.
3. MCL values for an individual **108**, which varies among individuals
4. UCL values for an individual **110**, which varies among individuals

This graphical presentation represents a Personal Hearing Zone or PHZ, in that it defines the dynamic range of sound reproduction over which an individual can both perceive sound and find it to be acceptably loud.

An additional PHMLS capability is that of offering the user a measurement of the sound level exposure on a daily basis. This function, referred to as an Envirogram, is derived in part by monitoring the sound pressure levels in each of the users ear canals. By averaging these sound pressure levels over time, an accumulated dosage calculation can be made and compared to a preset daily dose limit. Based on such a limit, a system user can note at any time in the course of a day where they stand with respect to accumulation a daily dose. A commonly accepted daily dose is equivalent to an 85dBA sound pressure level during a period of 8 hours. In calculating this dose, the sound pressure level must first exceed a lower limit of approximately 80 dBA, which would allow all day safe exposure. On the other hand, for sound pressure levels of 100 dBA, exposure time would be limited to approximately 15 minutes.

In addition to average sound pressure level measurements, left and right ear peak detectors function to monitor peak sound pressure level occurrences. Such short-term peaks have the capability of reducing daily exposure time limits, as well as inducing temporary threshold shifts and in extreme situations producing permanent hearing loss. These peak exposure limits are also compared to accepted exposure limits, and integrated into the daily dose calculation process. If sufficient sound pressure levels are determined that would most likely produce a threshold shift, the user receives an audible warning as well as a display indication.

In a typical user situation where a PHMLS will not be worn all day, the system offers several user defined options. In one option, the dose calculation function is suspended during periods of non use. This would be due to either the user turning the product off, or removing it from its in use position. In one case the system receives a user command to suspend activity with a button push, while in the other, the ECM will see essentially the same signal as the ASM, while at the same time a check of the transfer function between the ECR and the ECM will be that of a open ear canal situation. In either case the Envirogram process is suspended.

Another user definable option is that of a continuous Envirogram function, even if the product is turned off and or removed from the users head. Because of the low power requirements to perform the Envirogram functions, turning the product off, will not inhibit such measurements and calculations. When however the PHMLS is removed from the users head an approximate dose accumulation measurement will take place, based on the assumption that the user is in the same acoustical environment as the PHMLS. In such case the outputs of the ASM and ECM will be quite similar and will thus serve as a measure of the users sound environment.

In addition to listening to reproduced audio a PHMLS system allows a user to monitor, record and transmit ambient sounds at varying levels with a high degree of realism. **Figures 4 and 5** provide insight as to this capability. **Figures 4 and 5** differ in that **Figure 5** shows an integration of a PHMLS signal processor **45** with an audio program source **55** as component **56**.

Figure 4 shows user **8** in a situation where sounds from an arbitrary sound source **48** are picked up by the left **49** and right **51** in-ear electroacoustical assemblies **9** and transmitted to PHMLS electronics **45** as shown in **Figure 2a** by cables **11**. After processing the audio outputs of microphones **4** as shown in **Figure 1a** are returned to receivers **1** in each ear canal for audition by the system user.

As the users head is rotated, sounds **49** and **50** can be seen to vary, as would be the case when the PHMLS is completely removed. The transfer functions between any arbitrary sound and the entrance to each ear canal consequently approximate those of the individual wearing the PHMLS to a high degree. In such case the anatomical head and body related transfer functions (ATFs) of the individual are controlling the sounds picked up by the PHMLS and reproduced in the user ear canals.

Due to the physical placement of ASMs **4** and amplification equalization implemented by DSP Controller AD/DA Conversion module **43**, based on the known frequency response of the ASMs **4**, the ECRs **1**, and the transfer function between a blocked ear canal response and an ear drum response for an average human, ambient sounds picked up and reproduced by a PHMLS have a neutral and life like tonal characteristic. Furthermore based on the location of the ASMs, being near the occluded entrance to the users ear canal, ambient sounds reproduced by a PHMLS will highly approximate real world sounds in terms of localization and directionality as would be heard with the PHMLS removed. A primary difference however is that the level of reproduced audio will be adjustable by the user, so as to accommodate a variety of listening situations. From the users perspective the PHMLS will in essence offer a volume control on ones hearing, with an ability to turn volume levels down limited by the degree of acoustical isolation provided by the in-ear electroacoustical assemblies **9**. Other features enabled by this capability allow a PHMLS user to mix ambient and recorded sounds at desired relative levels as well as switch between either source at the exclusion of the other. A more advanced feature is that of automatically pausing a recorded source when a full ambience listening mode is selected. The ambience feature can also be combined with the PHMLS ability to monitor and control actual sound pressure levels as in the case of attending a live concert event.

In such case the actual listen level can be controlled and reduced if desired with full knowledge of the exposure risk and sound level being reproduced in the user ears.

Additional PHMLS features are detailed in **Figure 6** where an Audio recording and storage component **40**, a secure, wide bandwidth, and high signal to noise ratio audio transmission system **42**, and left **166** and right **168** channel audio outputs have been added to the PHMLS signal processing system. An import characteristic of the recording feature is that it includes a memory buffer that is continually recording the latest few minutes of ambient sound, so that when a user implements a record command, the past few minutes of audio are captured in memory. With the addition of these components, a system user has access to a variety of performance features:

1. An ability to record a live ambient event while doing something else.
2. An ability to record audible notes on command such as in the case of creating an audible record of a personal or business conversation or transaction.
3. An ability to transmit live or recorded conversations, speech and singing to another location for sharing, entertainment, emergency communications, data storage, or transcription.
4. An ability to record a live audio broadcast event in a muted mode while listening to an important ambient event.
5. An ability to connect a PHMLS to an external storage or wireless transmission system.

A PHMLS offers an additional group of user listening options associated with a selected listening level. One common problem is that the ear canal noise level is masking the lower levels portions of a reproduced audio signal, and thus inhibiting a user from enjoying the lower level nuances of an audio program source.

Such a condition is detectable based on a measure of the transfer function between the ECM **3** and the electrical signal feeding the ECR **1**. A linear relationship will exist between these two signals in a quiet ear canal background noise environment. If however the noise in the ear canal is too high due to acoustical leakage around the ear seal or resulting from reproduced ambience level, the ECM will continue to measure a significant signal for conditions of low ECR electrical drive levels.

Once detected several user selectable options are available:

1. The drive level to the ECR may be increased thus raising the reproduced level of low level musical passages sufficiently to over the ear canal background noise level. This however must be done with caution so as to avoid excessive sound pressure levels on musical peaks.
2. If excessive sound levels peaks are a result of such an increase in gain, a user selectable limiter may be engaged, so that the higher level gain setting, will not produce excessive sound levels.
3. If excessive sound levels are not acceptable and limiting is also not desired, an expansion function can be engaged to boost only low-level program passages with out affecting higher-level passages.

4. If a reproduced ambience signal is significantly contributing to the ear canal background noise level, the level of this signal can be reduced either by a fixed amount, or by a variable amount dependent on the level of the reproduced audio program.

A further PHMLS enhancement is shown in **Figure 7**, where a Speech to Text Recognition device **36** and a Text to speech Synthesis Device **38** have been added to the PHMLS signal processing system. It is through the addition of these functions that ambient speech and other machine recognizable sounds can be identified and used in the following ways:

1. By converting recognized speech to text, the text can be converted to synthesized speech with a very high signal to noise ratio for purposes of enhancing the signal to noise ratio for individuals with poor speech in noise intelligibility.
2. By converting recognized speech to text, the text can be translated into another language and then converted to synthesized speech for purposes of translation.
3. By learning to recognize the sound signatures of various non speech sounds that are considered annoying or potentially damaging to a users hearing, the ambient monitoring system can be momentarily muted or attenuated in level until the offended sound is no longer detected. Examples of such sound include a high-speed dental drill, commuter train cabin noise, and the sound of a pneumatic tool.
4. By learning to recognize a variety of sonic alert signals, a PHMLS user can be made aware of an impending danger while listening to an audio program source with the absence of an ambience signal. Such a danger warning could be in the form of turning on the ambient monitoring mode or by means of a warning signal mixed into the audio program being auditioned.

In addition to a PHMLS recognizing and adapting to non-speech annoying sounds, is the special case of a wind disturbance such as when the system is used out of doors or perhaps in front of a fan of air-conditioning system. Under such circumstances the left and right side ASMs will produce outputs with excessive low frequency spectral content, that do not represent the actual sound levels of exposure. If reproduced by the ECRs, a user would be subjected to potentially loud and annoying low frequency sounds. Under such circumstances it would be desirable to avoid such loud in ear sound levels. It can be readily demonstrated that the low frequency content of such wind induced microphone outputs are highly uncorrelated, unlike low frequency sounds under non-windy conditions. This differentiation can consequently be used as a wind presence detector. By comparing the outputs of a correlation function detector and a low frequency spectral content detector, a variable corner frequency high pass filter function can be realized. This function will consequently allow the normal reproduction of natural low frequency sounds while attenuating the low frequency content of annoying wind induced noise.

These detection and filter functions would be implemented by the DSP Controller and AD/DA Converter signal processing 43 as shown in **Figure 2a**.

Figures 8a and 8b detail a further addition to the PHMLS signal processing electronics previously described so as to enable the system to link into a cellular or other form of wireless telephone conversation. In this system configuration, a wireless secure transceiver 43 such as one meeting the Bluetooth Type 1 Headset Protocol, would allow the PHMLS user to engage in a hands free conversation enabled by the ambient sound microphone ASM 4 and the ECR 1. With the further addition of a loudspeaker 22 and amplifier 24, the PHMLS system can provide speaker phone capabilities further enabling a small group of individuals to engage in a conversation with the PHMLS user. The signal processing system shown in **Figure 8b** differs from that of **Figure 8a** in that **Figure 8b** shows the addition of two wireless transceivers to replace direct-wired connections between the in-ear electroacoustical system components. Further details of the use of these system features are shown in **Figures 11a and 11b**. In the case of **Figure 11a** the in-ear electroacoustical assemblies 9 are connected to the PHMLS signal processing electronics 53 by means of cables 11. The PHMLS signal processing electronics 53 also houses the speaker 24 for the speaker phone function and the wireless transceiver 43 for the wireless telephone 97. **Figure 11b** represents the same system functionality, however the in-ear electroacoustical system components are connected by wireless system 143 as shown in **Figure 8b**, located at each ear of the system user. In this case the wireless telephone 97 is linked to one of the ear assemblies, which also houses the speaker for the speaker phone function.

Figures 9a, 9b and 10 relate to a data collection feature of a PHMLS that support its capability of providing a personalized binaural auralization function for the playback recorded music. This function relies in part on its ability to implement a set of Head Related Transfer Functions that are unique to the system user and are stored in the User Data Memory File 39 shown in **Figure 2a**. These functions are the basis for creating sonic images at specific points in space external to and around the system users head. Three possible measurement techniques are shown in these figures. Using the system described in **Figure 9a** the PHMLS user is required to make use of a measurement system involving an array of loudspeakers (65, 66, 67, 68) on a spherical surface that surround the head of a system user. Data collection is facilitated by connecting the PHMLS system computer interface port 44 to the test system 64. The collection process obtains left 63 and right 61 ear head related transfer functions (HRTF), for each desired loudspeaker orientation, by means of the ambient sound microphones (ASM) 4 mounted near the left and right occluded ear canals. Once measured this data is downloaded to the User Data Memory File 39 for use in the binaural auralization reproduction process. The process shown in figure 9b is essentially the same except that sound is radiated from the external receivers located near the occluded ear canal of the PHMLS user and collected by a spherical array of microphones. As in the case for Figure 9a, once data is derived it may be downloaded to the User Data Memory File 39 for use in the binaural auralization reproduction process.

Figure 10 represents an alternate approach to the HRTF collection process in which the PHMLS system user makes a small number of HRTF measurements, which are compared by means of an Internet connection to a large database of HRTF functions. As a result of this comparison process, a selection is made as to a set of HRTF values that closely match those of the PHMLS user. As shown in **Figure 10**, user **128**, places a small test loudspeaker **117** at several points around an imaginary circle of radius **128**. The test signal used to drive the speaker is down loaded from an Internet web site, and delivered to the speaker by means of the computer audio output connection. For each orientation, HRTF data is collected based on left and right ear sound paths **118** and **116** respectively to the ASMs for each ear located near the occluded entrance to the users ear canals. As in the situations shown **Figure 9a and 9b**, once data is derived and data extracted from the HRTF database, it may be downloaded to the User Data Memory File **39** for use in the binaural auralization reproduction process.

Electronic Patent Application Fee Transmittal				
Application Number:				
Filing Date:				
Title of Invention:		Personalized Hearing Measurment & Listening System Ver 2.0		
First Named Inventor/Applicant Name:		Steven Wayne Goldstein		
Filer:		Steven Wayne Goldstein		
Attorney Docket Number:				
Filed as Small Entity				
Provisional Filing Fees				
Description	Fee Code	Quantity	Amount	Sub-Total in USD(\$)
Basic Filing:				
Provisional Application filing fee	2005	1	100	100
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD(\$)
Miscellaneous:				
Total in USD (\$)				100

Electronic Acknowledgement Receipt

EFS ID:	1409451
Application Number:	60883013
International Application Number:	
Confirmation Number:	6535
Title of Invention:	Personalized Hearing Measurment & Listening System Ver 2.0
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	32146
Filer:	Steven Wayne Goldstein
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Application Type:	Provisional

Payment information:

Submitted with Payment	yes
Payment was successfully received in RAM	\$ 100
RAM confirmation Number	1390
Deposit Account	

File Listing:

Document Number	Document Description	File Name	File Size(Bytes)	Multi Part /.zip	Pages (if appl.)
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1	Application Data Sheet	US_ADS_Form_SB_14.pdf	901486	no	4
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Information:					
2	Abstract	PHMLL_ABSTRACT_VER2.pdf	320296	no	21
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4	Drawings	PHMLS_DRAWINGS.PDF	8374413	no	19
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5	Drawings	SSIDS_DRAWING.pdf	124523	no	1
Warnings:					
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Information:					
6	Drawings	SSIDS_operating_mode.pdf	67418	no	1
Warnings:					
The page size in the PDF is too large. The pages should be 8.5 x 11 or A4. If this PDF is submitted, the pages will be resized upon entry into the Image File Wrapper and may affect subsequent processing					
Information:					
7	Specification	PHMLS_DRAWING_DETAIL_DESC_Ver2.pdf	172858	no	11
Warnings:					
Information:					
8	Fee Worksheet (PTO-06)	fee-info.pdf	8095	no	2
Warnings:					
Information:					
Total Files Size (in bytes):			10259446		

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New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.