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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
60/885,917	01/22/2007		100		10		

CONFIRMATION NO. 8316

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

FILING RECEIPT

Date Mailed: 02/07/2007

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Applicant(s)

Steven Wayne Goldstein, Delray Beach, FL;

Power of Attorney: The patent practitioners associated with Customer Number 32146

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The country code and number of your priority application, to be used for filing abroad under the Paris Convention, is **US60/885,917**

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

Non-Publication Request: No

Early Publication Request: No

** SMALL ENTITY **

Title

Acute Sound Detector and Constant Signal-to-Noise Headset System

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Application Data Sheet 37 CFR 1.76		Attorney Docket Number	
		Application Number	
Title of Invention	Acute Sound Detector and Constant Signal-to-Noise Headset System		
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			
Mailing Address of Applicant:					
Address 1	16390 Maddalena Place				
Address 2					
City	Delray Beach	State/Province	FI		
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Customer Number	32146		
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Email Address	sgoldstein@thebestmedicine.com	Add Email	Remove Email
Email Address	mhbhehb@aol.com		Remove Email

Application Data Sheet 37 CFR 1.76		Attorney Docket Number	
		Application Number	
Title of Invention	Acute Sound Detector and Constant Signal-to-Noise Headset System		

Application Information:

Title of the Invention	Acute Sound Detector and Constant Signal-to-Noise Headset System		
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)	10	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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Please Select One:	☒ Customer Number	☐ US Patent Practitioner	☐ US Representative (37 CFR 11.9)
Customer Number	32146		

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Prior Application Status		Remove	
Application Number	Continuity Type	Prior Application Number	Filing Date (YYYY-MM-DD)
Additional Domestic Priority Data may be generated within this form by selecting the Add button.			Add

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Application Data Sheet 37 CFR 1.76		Attorney Docket Number	
		Application Number	
Title of Invention	Acute Sound Detector and Constant Signal-to-Noise Headset System		

			Remove
Application Number	Country ⁱ	Parent Filing Date (YYYY-MM-DD)	Priority Claimed
			☒ Yes ☐ No
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Signature	/stevegoldstein/		Date (YYYY-MM-DD)	2007-01-22
First Name	Steven	Last Name	Goldstein	Registration Number

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Acute Sound Detector and Constant Signal-to-Noise headset system.

Abstract

An automatic and adaptive, self-contained Acute-level Sound Pass-through System (ALSPTS) which allows high level (acute) ambient sound detected by either or both the left and right ambient sound microphones (ASMs) located near the entrance to fully or partially occluded ear canal to be reproduced with either or both Ear Canal Receivers (ECR) mounted in the CanalPhone assembly which forms a full or partial seal of the User's ear canal. The system activates "sound pass-through" (i.e. reproduction of the ASM signal with the ECR) when the SPL measured at the ASM(s) is greater than the ambient Background Noise Level (BNL) by a user-defined or system-defined value (the SPL analysis may be frequency dependant). The ALSPTS works in combination with, or separate to an automatic and adaptive, self-contained Constant Signal-to-Noise Ratio System (CSNRS) to maintain maximal intelligibility of reproduced Audio Content (e.g. speech audio from a cell-phone or radio, or music audio from a PMP, or audio warning signals) in response to a changing ambient noise environment. Background Noise Level (BNL) is measured using either or both an Ear Canal Microphone (ECM) mounted in a CanalPhone assembly in a fully or partly occluded ear canal, or/and using the Ambient Sound Microphone(s) signals.

Background

1. Field of the invention

This invention pertains to sound reproduction, sound recording, audio communications and hearing protection using an acoustic device containing multiple electroacoustic transducers mounted in a User's ear canal, 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.

US Pat. No. 2006/0182287 describes a CanalPhone monitoring system, with microphones mounted at the entrance to an occluded ear canal (i.e. ASMs), and Ear-Canal Receivers (ECRs) mounted in the same earphone. In US Pat. No. 2006/0182287, the ambient microphone signals are reproduced using with ear-canal receivers with a user-defined gain. The present invention differs from US Pat. No. 2006/0182287 by automatically affecting the gain of the ASM signal which is reproduced with the ECRs, on a frequency-selective basis, to allow the user to hear high-level (or acute) sounds in the ambient sound field. In other words, this particular aspect of the present invention (called an Acute-level Sound Pass-through System-ALSPTS) is an audio squelch system whereby acute sound detected by at least one ASM is reproduced with an Ear Canal Receiver (ECR) in a “pass-through” mode, with the level of the ASM signal used as a gate (or squelch switch) for the audio pass-through.

An audio squelch system is described by US Pat. No. 2,770,721 for a radio communication device, with the squelch control activated by the received signal level (i.e. not using a microphone signal- as with the present invention). Similar, US Pat. No. 5,708,970 describes a wireless sound monitoring system which is activated using a

squelch control affected by analysis of ambient sound levels- but differs from the present invention by not involving an earphone based level monitoring system.

The transmission of detected loud sounds by the ALSPTS to the user via the ECR is designed in accordance with ISO-7731 to ensure that the level of the warning signal is above the effective masked threshold value for a given background noise level. Generally, this involves ensuring that the warning signal is at least 10 dB above the level of the ambient noise within the frequency range of interest.

The ALSPTS works in combination with, or separate to, an automatic and adaptive, self-contained Constant Signal-to-Noise Ratio System (CSNRS) to maintain maximal intelligibility of reproduced Audio Content (e.g. speech audio from a cell-phone or radio, or music audio from a PMP, or audio warning signals) in response to a changing ambient noise environment. The CSNRS maintains a substantially constant (or minimum) level ratio between audio content Signal (e.g. voice or music audio reproduced with the Ear Canal Receiver) and the background noise level measured in the ear canal.

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.

US Pat. No. 3,098,121 describes a circum-aural headphone system that monitors the ambient sound level using one sampling microphone mounted on the head-band. Reproduction level of audio content is modulated by the ambient sound level, so as to maintain a high signal-to-noise ratio. The present system derives an estimate for background noise level (BNL) by analysis of SPL either or both at the entrance to an occluded ear canal (i.e. using the ASM) or/and within the occluded ear canal using the Ear Canal Microphone (ECM). This allows for a accurate representation of the BNL as exposed to the listener’s auditory system (i.e. the SPL at the ear drum). Furthermore, the present invention includes a system to detect if overall sound level exposure to the

listener is too high- i.e. high enough to potentially cause temporary or permanent hearing damage to the User.

Acute Sound Detector and Constant Signal-to-noise headset system.

Drawings

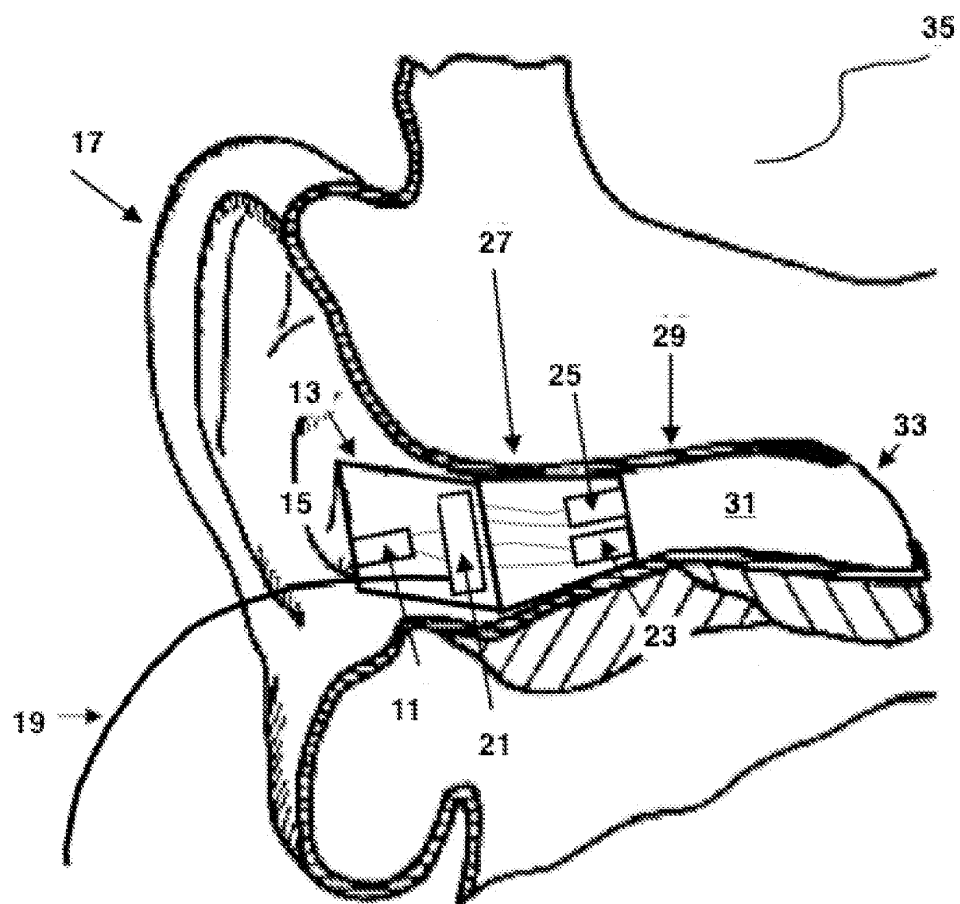


Figure 1.

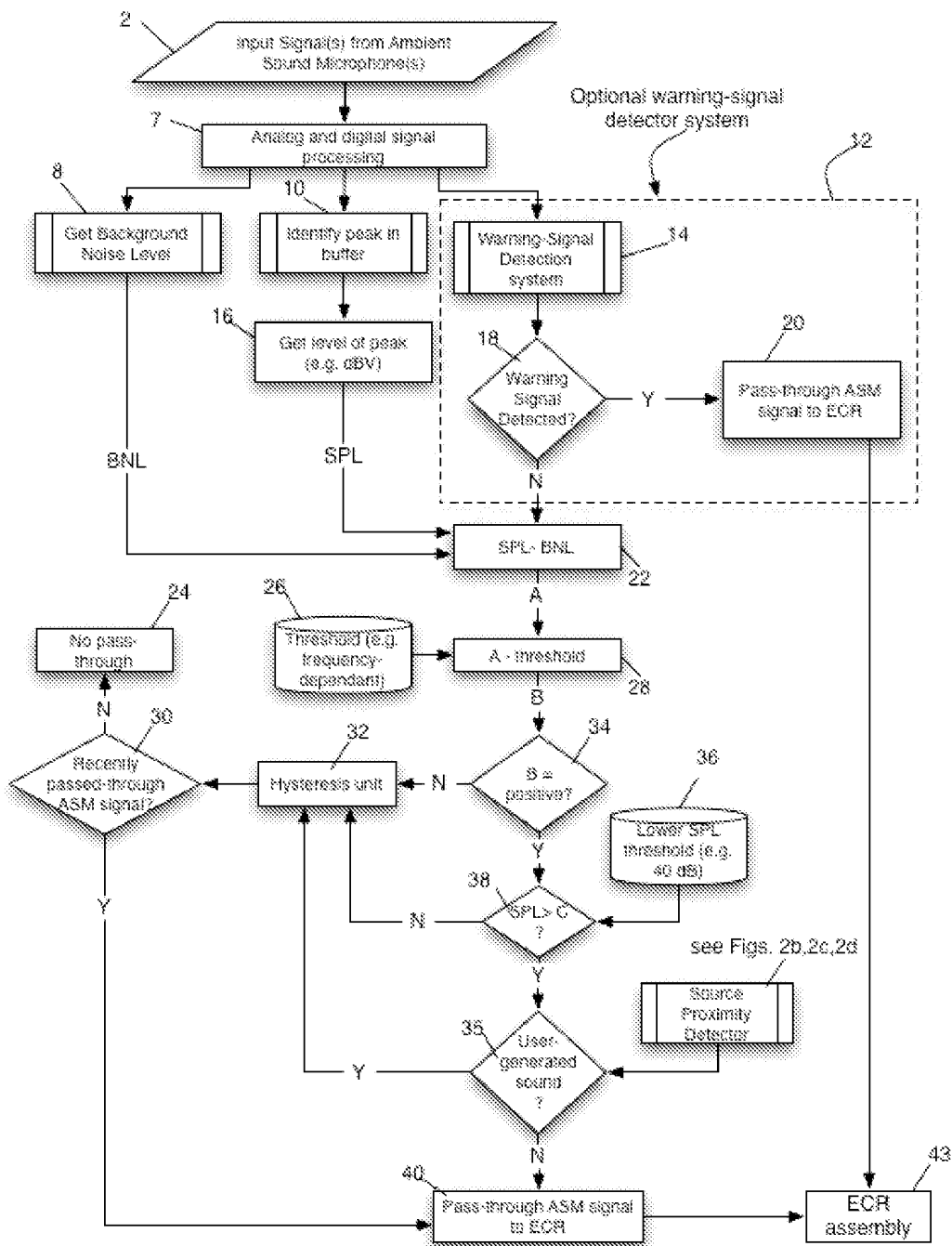


Figure 2a:
Acute sound Pass-through system

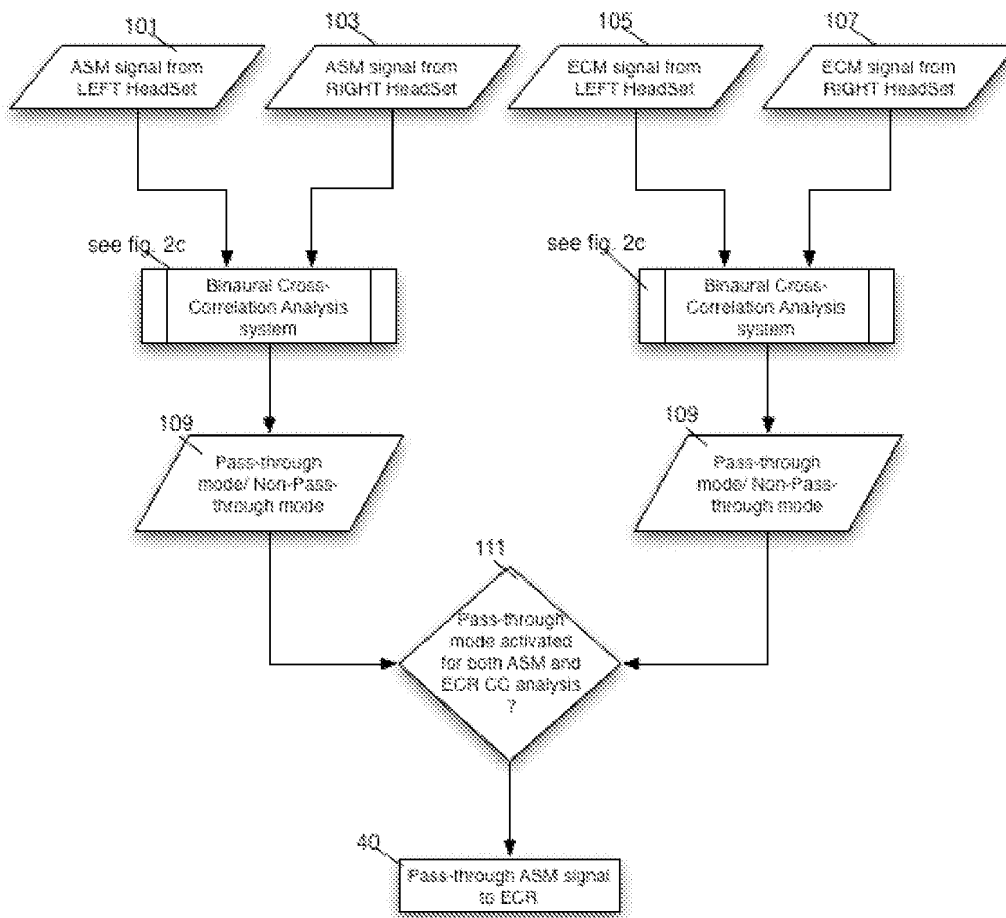


Figure 2b:
Source Proximity Detector: Binaural
Analysis System for Acute sound
Detector, if ASM and ECM are used.

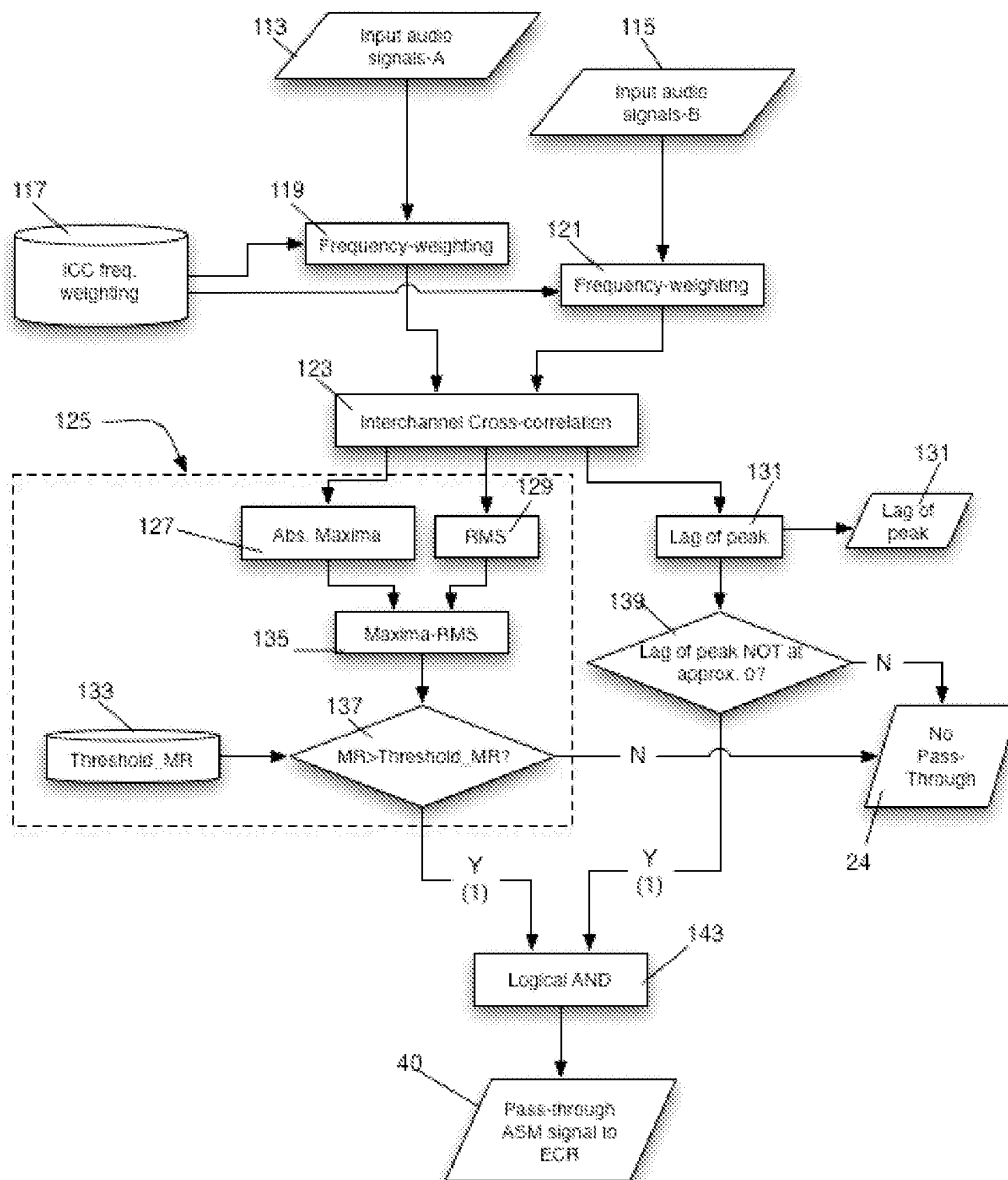


Figure 2c:
Binaural Analysis System for Acute
sound Detector: CC system

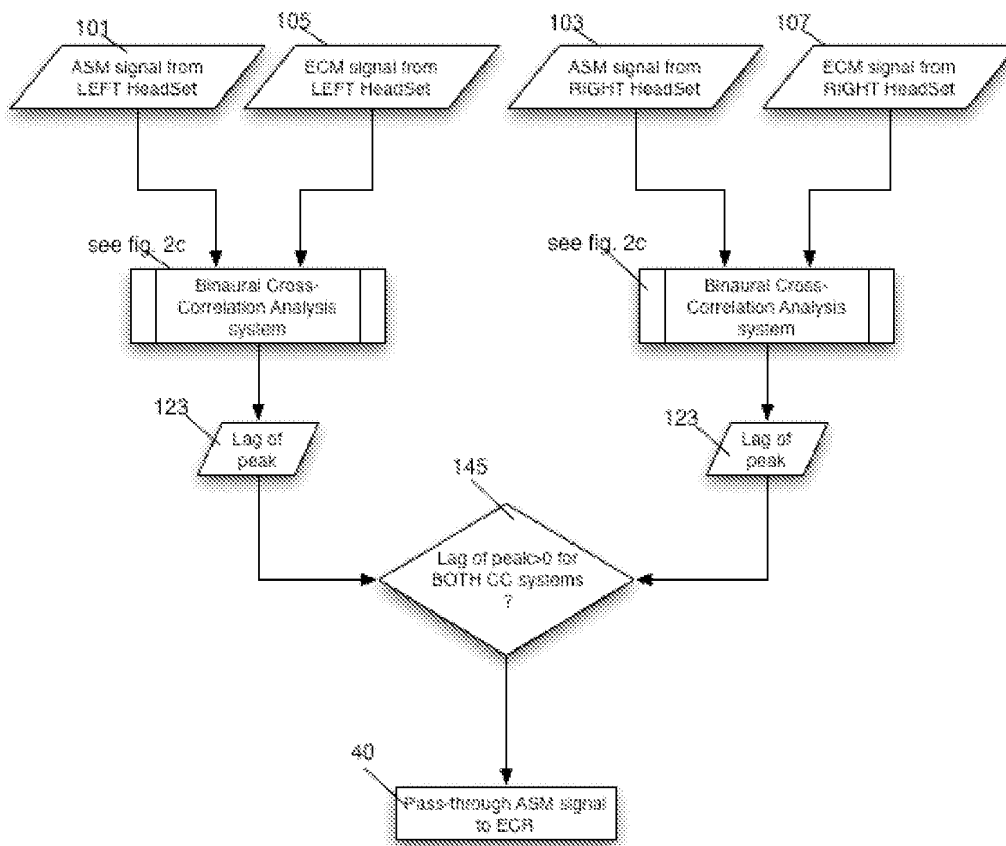


Figure 2d:
Source Proximity Detector: Binaural Analysis System for Acute sound Detector, if ASM and ECM used. In-head detector

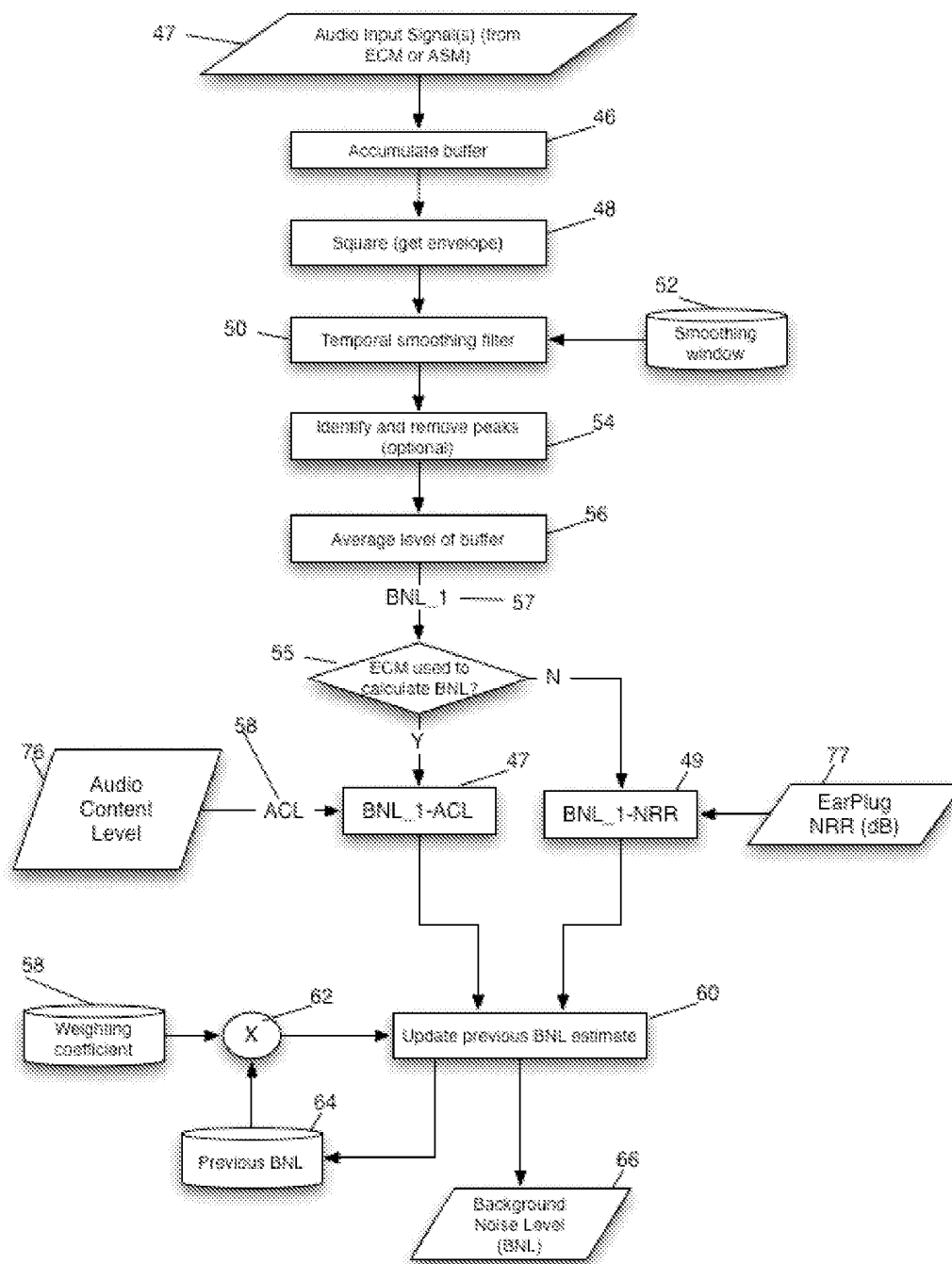


Figure 3: BNL estimation

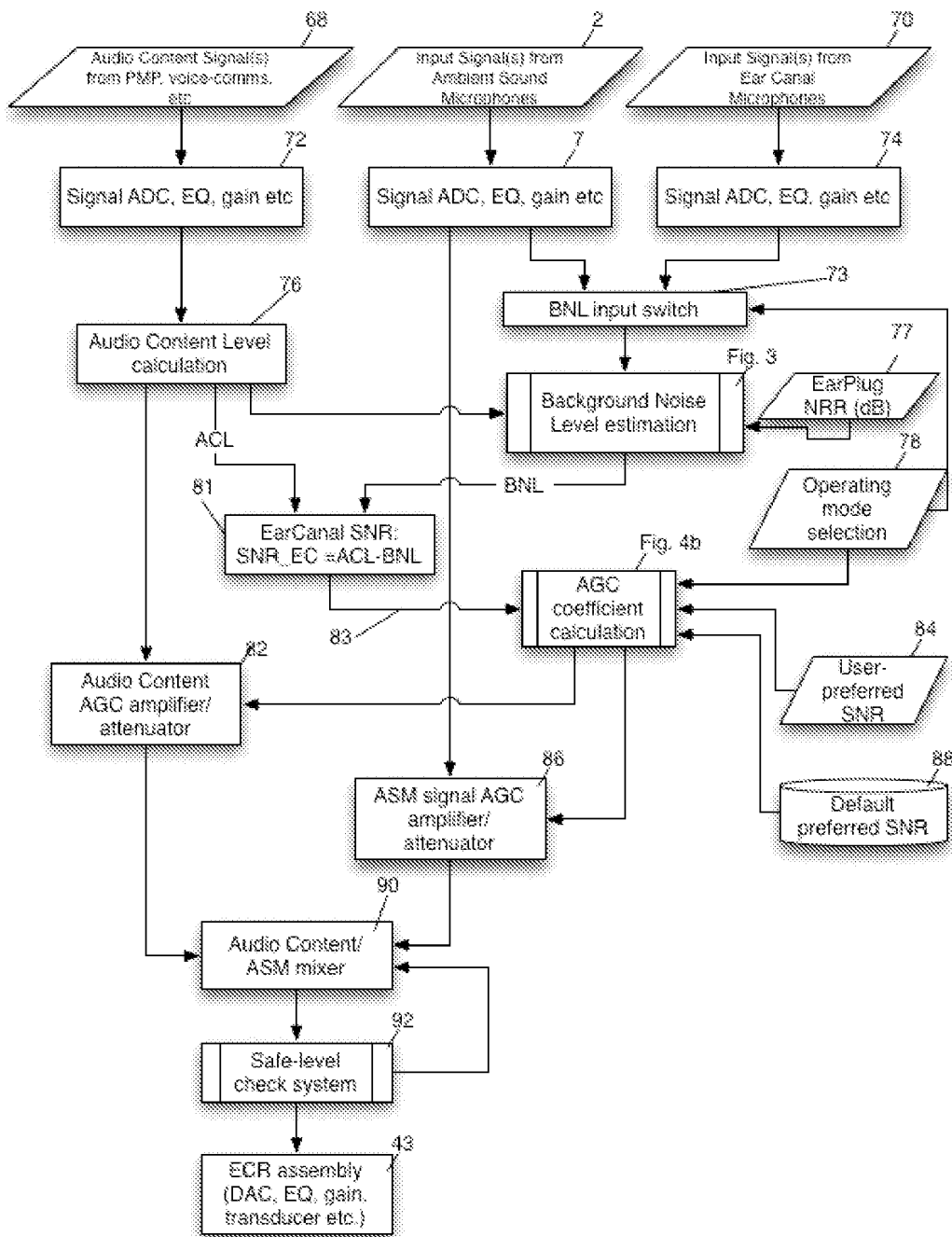


Figure 4a:
Constant SNR system

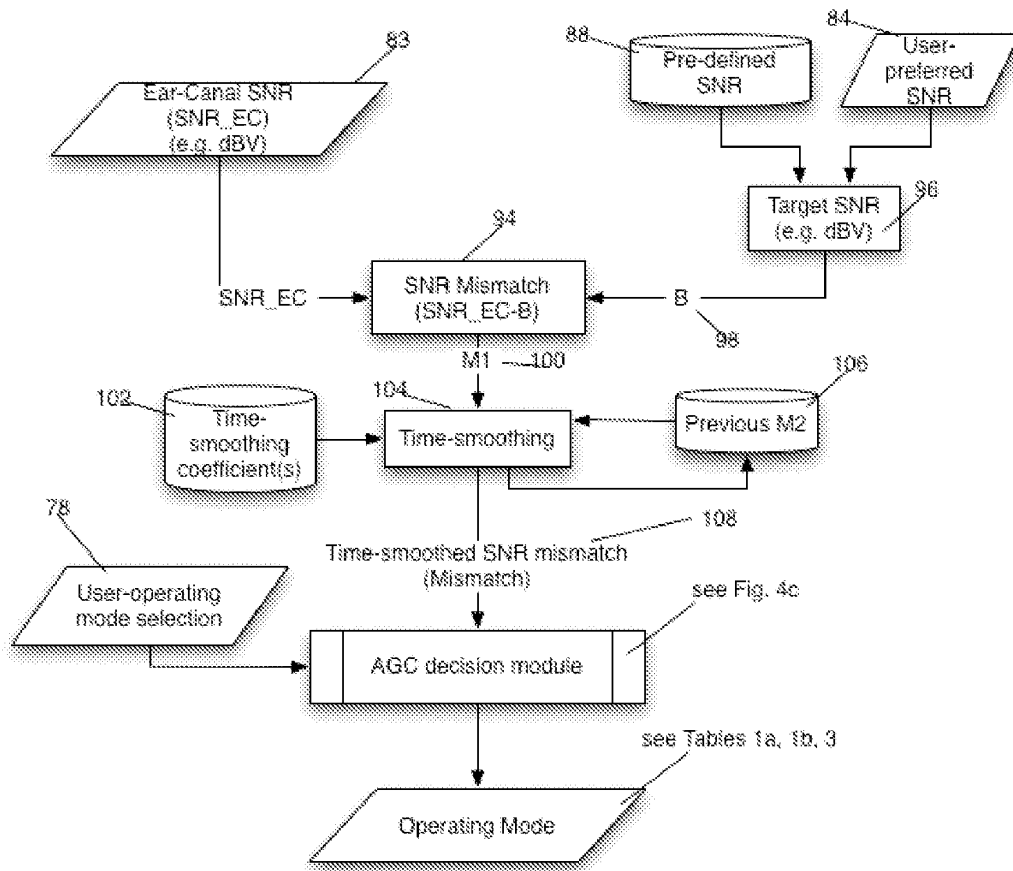


Figure 4b:
AGC coefficient calculation

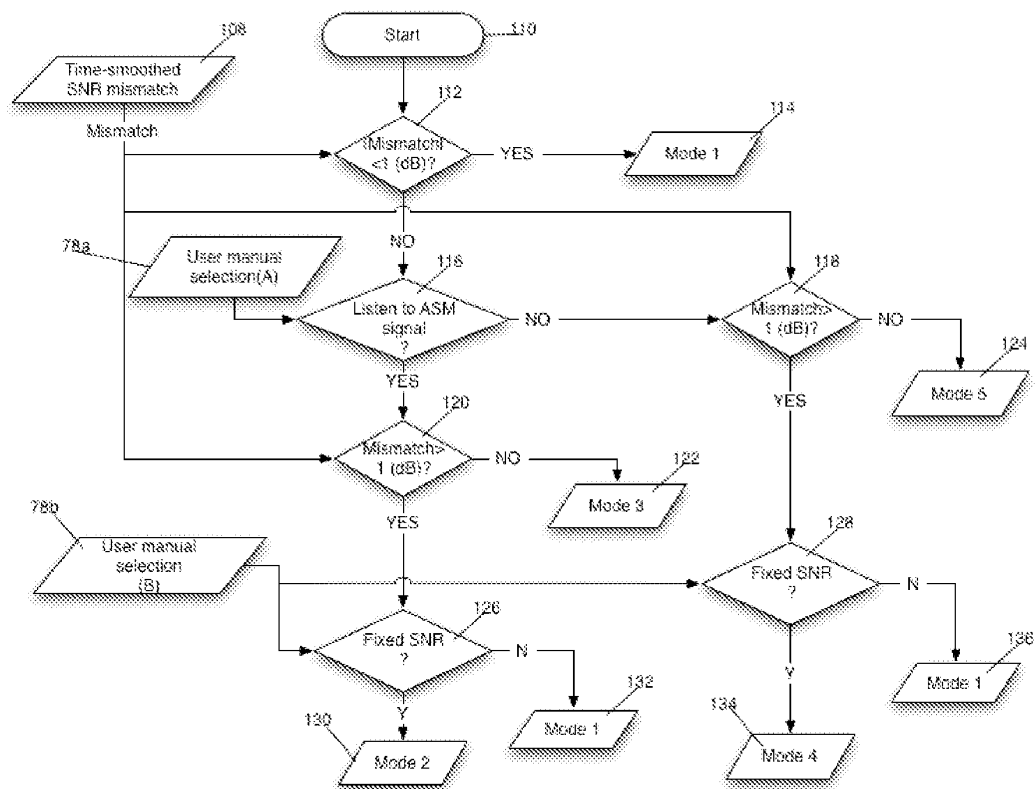


Figure 4c:
AGC mode decision module

Table 1a:
ASM Pass-through mode

	Fixed SNR?	
Ear Canal SNR	Yes	No
High	Mode 2	Mode 1
Low	Mode 3	Mode 3
OK	Mode 1	Mode 1

Table 1b:
ASM Non- Pass-through mode

	Fixed SNR?	
Ear Canal SNR	Yes	No
High	Mode 4	Mode 1
Low	Mode 5	Mode 5
OK	Mode 1	Mode 1

Table 3:
For SNR mismatch of M (dB)

Mode:	Audio Content Gain	ASM Gain
1	NU	NU
2	"-M/2"	"+M/2"
3	"+M/2"	"-M/2"
4	"-M"	"-∞"
5	"+M"	"-∞"

NU

No Update

Acute Sound Detector and Constant Signal-to-noise headset system.

Embodiments

1. An automatic and adaptive, self-contained Acute-level Sound Pass-through System (ALSPTS) which allows high level (acute) ambient sound detected by either or both the left and right ambient sound microphones (ASMs) located near the entrance to an occluded ear canal to be reproduced with either or both Ear Canal Receivers (ECR) mounted in a CanalPhone assembly which forms a full or partial seal of the User's ear canal. The system activates "sound pass-through" (i.e. reproduction of the ASM signal with the ECR) when the SPL measured at the ASM(s) is greater than the ambient Background Noise Level (BNL) by a user-defined or pre-defined value (the SPL analysis may be frequency dependant). The BNL is constantly updated (which may be on a frequency-selective basis) and is estimated using the ASM. The ALSPTS 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 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 estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ASM(s), which may be first band-limited (e.g. to attenuate low and high frequencies). The SPL

history buffer is smoothed using a temporal window (e.g. using a Hanning-shaped 10 ms averaging window) to remove transient peaks, and the resulting buffer is squared and averaged.

- F. Alternative to, or in some embodiments in addition to, the system in (E), there is a system to estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ECM(s), which may be first band-limited (e.g. to attenuate low and high frequencies).
- G. A system to identify transient peaks in the ASM and/or ECM audio data buffer and to calculate the instantaneous Sound Pressure Level (SPL) or voltage value of these peaks. Transient peaks are identified by analyzing the change in voltage (e.g. by detecting a change in voltage gradient above a given value, such as a slew-rate of 1 V per ms) and these transient parts are removed before the SPL buffer average level is calculated (i.e. the parameter created by the system in E and F).
- H. An optional warning detection system to detect the presence of an acoustic warning signal in the User's environment, by analyzing the ASM signal. When a warning signal is detected, the ASM signal is "passed-through" to the Ear Canal Receiver- i.e. the ASM signal is reproduced with the ECR, which may include additional processing such as noise reduction circuitry to increase perceptual intelligibility of the warning signal.
- I. A comparator system to compare the peak SPL (or voltage) measured in (G) with the Background Noise Level, and to detect if the difference is greater than a predefined threshold (the threshold may be frequency-dependant, and the threshold may be user defined and manually adjusted by the user).
- J. A user interface assembly to allow the BNL threshold to be adjusted manually by a user- so as to affect the sensitivity of the comparator in (I).
- K. A system to activate the Sound Pass-Through Mode, whereby the ASM signal is reproduced by the ASM when the output of the comparator in (I) determines that the instantaneous SPL is greater than the BNL by a predefined value.
- L. A Hysteresis system to ensure that the Sound Pass-Through Mode system is not activated or deactivated within a small time period (e.g. 10 ms) to ensure against switching artifacts which may be audible.
- M. An optional comparator system to compare the current SPL calculated in (I) with a fixed or adaptive low-level threshold (which may be frequency dependant, and is approximately 40 dB), and ensures that the current SPL is above a given level before the "pass-through" mode is activated.

2. An automatic and adaptive, self-contained Acute-level Sound Pass-through System (ALSPTS) which allows high level (acute) ambient sound detected by either or both the left and right ambient sound microphones (ASMs) located near the entrance to an occluded ear canal to be reproduced with either or both Ear Canal Receivers

mounted in a CanalPhone assembly which forms a full or partial seal of the User's ear canal. A system detects if the acute sound was generated by the User's voice, using a combination of the ASM signals, and signals generated by Ear Canal Microphones (ECMs), which are embedded in an ear plug in a fully or partly sealed ear canal. The system activates "sound pass-through" (i.e. reproduction of the ASM signal with the ECR) when the SPL measured at the ASM(s) is greater than the ambient Background Noise Level (BNL) by a user-defined or pre-defined value (the SPL analysis may be frequency dependant), and when the detected acute sound is not generated by the User's voice. The BNL is constantly updated (which may be on a frequency-selective basis) and is estimated using the ASM. The ALSPTS 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 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 Ear Canal Microphone (ECM) to monitor the sound field at the entrance to a user's occluded ear canal.
- F. An ECM amplification and signal processing (signal conditioning) system to compensate for the frequency response of the ASM.
- G. A system to estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ASM(s), which may be first band-limited (e.g. to attenuate low and high frequencies). The SPL history buffer is smoothed using a temporal window (e.g. using a Hanning-shaped 10 ms averaging window) to remove transient peaks, and the resulting buffer is squared and averaged.
- H. Alternative to, or in some embodiments in addition to, the system in (E), there is a system to estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ECM(s), which may be first band-limited (e.g. to attenuate low and high frequencies).
- I. A system to identify transient peaks in the ASM and/or ECM audio data buffer and to calculate the instantaneous Sound Pressure Level (SPL) or voltage value of these peaks. Transient peaks are identified by analyzing the

change in voltage (e.g. by detecting a change in voltage gradient above a given value, such as a slew-rate of 1 V per ms) and these transient parts are removed before the SPL buffer average level is calculated (i.e. the parameter created by the system in E and F).

- J. An optional warning detection system to detect the presence of an acoustic warning signal in the User's environment, by analyzing the ASM signal. When a warning signal is detected, the ASM signal is "passed-through" to the Ear Canal Receiver- i.e. the ASM signal is reproduced with the ECR, which may include additional processing such as noise reduction circuitry to increase perceptual intelligibility of the warning signal.
- K. A comparator system to compare the peak SPL (or voltage) measured in (G) with the Background Noise Level, and to detect if the difference is greater than a predefined threshold (the threshold may be frequency-dependant, and the threshold may be user defined and manually adjusted by the user).
- L. A user interface assembly to allow the BNL threshold to be adjusted manually by a user- so as to affect the sensitivity of the comparator in (I).
- M. A Source Proximity Detector, that uses the outputs of the ASM's and ECM's to determine the location of the sound source, by an analysis of the interchannel cross-correlation between ASM signals from the left and right electroacoustic assemblies, and in some embodiments by an analysis of the cross-correlation between the ASM and ECM signals of the same electroacoustic assembly.
- N. A system to activate the Sound Pass-Through Mode, whereby the ASM signal is reproduced by the ASM when the output of the comparator in (I) determines that the instantaneous SPL is greater than the BNL by a predefined value, and when the Source Proximity Detector in (M) determines that the acute sound source was not the user's voice.
- O. A Hysteresis system to ensure that the Sound Pass-Through Mode system is not activated or deactivated within a small time period (e.g. 10 ms) to ensure against switching artifacts which may be audible.
- P. An optional comparator system to compare the current SPL calculated in (I) with a fixed or adaptive low-level threshold (which may be frequency dependant, and is approximately 40 dB), and ensures that the current SPL is above a given level before the "pass-through" mode is activated.

3. An automatic and adaptive, self-contained Constant (or Minimum) Signal-to-Noise Ratio System (CSNRS) to maintain maximal intelligibility of reproduced Audio Content (specifically- speech audio, e.g. from a cell-phone or radio) in response to a changing ambient noise environment. Background Noise Level (BNL) is measured using either or both Ear Canal Microphone(s) (ECM)

mounted in a CanalPhone assembly in a fully or partly occluded ear canal, or/and using an Ambient Sound Microphone (ASM) located near the entrance to a fully or partly occluded ear canal. Audio Content is reproduced using an Ear Canal Receiver (ECR) mounted in the same CanalPhone assembly. The reproduced level of the Audio Content signal is modulated by the level of the estimated BNL. In addition to audio content, the signal detected by the ASM may be simultaneously reproduced by the ECR. The CSNRS 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 Ear Canal Microphone (ECM) to monitor the sound field at the entrance to 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. 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.)
- F. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- G. An audio data transmission system to receive audio data (Audio Content) from, for example, from a voice communication device (e.g. radio).
- H. A signal amplification and processing (signal conditioning) system to process the audio content in (G), e.g. to equalize the audio content to emphasize speech-frequencies and attenuate non-speech frequencies.
- I. An Audio Content Level Calculation system to estimate the level (e.g. in dBV) of the Audio content (e.g. using a 10 ms smoothing-average window).
- J. A system to estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ASM(s) and/or ECR(s), which may be first band-limited (e.g. to attenuate low and high frequencies).
- K. An Automatic Gain Control (AGC) coefficient generation system to determine the appropriate gain (which may be on a frequency-selective basis) to apply to either or both the Audio Content signal and ASM signal in order to maintain a user-defined or automatically defined Signal-to-Noise Ratio (SNR).
- L. An optional User Input System to allow a User to adjust the SNR in (K), e.g. a software system on a PMP, or a manual system with buttons mounted on the CanalPhone assembly.

- M. An AGC system to adjust the gain (which may be on a frequency-selective basis) to the Audio Content Signal according to the gain coefficients in (K).
- N. An AGC system to adjust the gain (which may be on a frequency-selective basis) to the ASM signal according to the gain coefficients in (K).
- O. An audio data signal mixing system, which combines the Audio Content Signal and ASM signal to produce a new single (mono) or two-channel (stereo) signal.
- P. A “Safe-Level Check System” (SLCS) which takes the new signal in (O) as its input and determines if the level of the new signal (e.g. measured in dBV) is high enough to potentially cause hearing damage to the User if it was reproduced with the ECR. The SLCS attenuates either or both the Audio Content signal or/and ASM signal if the new signal level is deemed too high.

4. An automatic and adaptive, self-contained Constant (or Minimum) Signal-to-Noise Ratio System (CSNRS) to maintain maximal clarity of reproduced Audio Content (specifically- music audio, e.g. from a PMP) in response to a changing ambient noise environment. Background Noise Level (BNL) is measured using either or both an Ear Canal Microphone (ECM) mounted in a CanalPhone assembly in a fully or partly occluded ear canal, or/and using an Ambient Sound Microphone (ASM) located near the entrance to a fully or partly occluded ear canal. Audio Content is reproduced using an Ear Canal Receiver (ECR) mounted in the same CanalPhone assembly. The reproduced level of the Audio Content signal is modulated by the level of the estimated BNL. In addition to audio content, the signal detected by the ASM may be simultaneously reproduced by the ECR. The CSNRS 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 Ear Canal Microphone (ECM) to monitor the sound field at the entrance to 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. 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.)
- F. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- G. An audio data transmission system to receive audio data (Audio Content) from, for example, from music audio from a PMP.
- H. A signal amplification and processing (signal conditioning) system to process the audio content in (G), e.g. to equalize the audio content to emphasize speech-frequencies and attenuate non-speech frequencies.
- I. An Audio Content Level Calculation system to estimate the level (e.g. in dBV) of the Audio content (e.g. using a 10 ms smoothing-average window).
- J. A system to estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ASM(s) and/or ECR(s), which may be first band-limited (e.g. to attenuate low and high frequencies).
- K. An Automatic Gain Control (AGC) coefficient generation system to determine the appropriate gain (which may be on a frequency-selective basis) to apply to either or both the Audio Content signal and ASM signal in order to maintain a user-defined or automatically defined Signal-to-Noise Ratio (SNR).
- L. An optional User Input System to allow a User to adjust the SNR in (K), e.g. a software system on a PMP, or a manual system with buttons mounted on the CanalPhone assembly.
- M. An AGC system to adjust the gain (which may be on a frequency-selective basis) to the Audio Content Signal according to the gain coefficients in (K).
- N. An AGC system to adjust the gain (which may be on a frequency-selective basis) to the ASM signal according to the gain coefficients in (K).
- O. An audio data signal mixing system, which combines the Audio Content Signal and ASM signal to produce a new single (mono) or two-channel (stereo) signal.
- P. A "Safe-Level Check System" (SLCS) which takes the new signal in (O) as its input and determines if the level of the new signal (e.g. measured in dBV) is high enough to potentially cause hearing damage to the User if it was reproduced with the ECR. The SLCS attenuates either or both the Audio Content signal or/and ASM signal if the new signal level is deemed too high.

5. An automatic and adaptive, self-contained Constant (or Minimum) Signal-to-Noise Ratio System (CSNRS) to maintain maximal detectability of reproduced Audio Content (specifically- speech audio, e.g. from a cell-phone or radio) in response to a changing ambient noise environment. Background Noise Level (BNL) is measured using either or both an Ear Canal Microphone (ECM)

mounted in a CanalPhone assembly in a fully or partly occluded ear canal, or/and using an Ambient Sound Microphone (ASM) located near the entrance to a fully or partly occluded ear canal. Audio Content is reproduced using an Ear Canal Receiver (ECR) mounted in the same CanalPhone assembly. The reproduced level of the Audio Content signal is modulated by the level of the estimated BNL. In addition to audio content, the signal detected by the ASM may be simultaneously reproduced by the ECR. The CSNRS 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 Ear Canal Microphone (ECM) to monitor the sound field at the entrance to 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. 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.)
- F. A signal processing circuit to amplify the signal to the ECR to equalize for the frequency sensitivity of the ECR.
- G. An audio data transmission system to receive audio data (Audio Content) from a device which creates a warning audio signal.
- H. A signal amplification and processing (signal conditioning) system to process the audio content in (G), e.g. to equalize the audio content to emphasize speech-frequencies and attenuate non-speech frequencies.
- I. An Audio Content Level Calculation system to estimate the level (e.g. in dBV) of the Audio content (e.g. using a 10 ms smoothing-average window).
- J. A system to estimate the ambient sound field Background Noise Level (BNL) by integrating a recent SPL history as measured with the ASM(s) and/or ECR(s), which may be first band-limited (e.g. to attenuate low and high frequencies).
- K. An Automatic Gain Control (AGC) coefficient generation system to determine the appropriate gain (which may be on a frequency-selective basis) to apply to either or both the Audio Content signal and ASM signal in order to maintain a user-defined or automatically defined Signal-to-Noise Ratio (SNR).
- L. An optional User Input System to allow a User to adjust the SNR in (K), e.g. a software system on a PMP, or a manual system with buttons mounted on the CanalPhone assembly.

- M. An AGC system to adjust the gain (which may be on a frequency-selective basis) to the Audio Content Signal according to the gain coefficients in (K).
- N. An AGC system to adjust the gain (which may be on a frequency-selective basis) to the ASM signal according to the gain coefficients in (K).
- O. An audio data signal mixing system, which combines the Audio Content Signal and ASM signal to produce a new single (mono) or two-channel (stereo) signal.
- P. A “Safe-Level Check System” (SLCS) which takes the new signal in (O) as its input and determines if the level of the new signal (e.g. measured in dBV) is high enough to potentially cause hearing damage to the User if it was reproduced with the ECR. The SLCS attenuates either or both the Audio Content signal or/and ASM signal if the new signal level is deemed too high.

6. A single electroacoustic transducer assembly containing a collection of Micro-Electro-Mechanical Systems (MEMS) and an Application-Specific Integrated Circuit (ASIC) in a single assembly. The MEMS comprising; least one Ambient Sound Microphone (ASM), at least one Ear Canal Receiver (ECR) and at least one Ear Canal Microphone, and one or more optional Ambient Sound Receivers (ECR). The ASIC undertakes audio signal processing such as analog to digital (or vice versa) conversion, analog or digital filtering, automatic gain control. The combined unit comprising:

- A. A transducer 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 transducer 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).
- C. A transducer 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).
- D. An optional transducer 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 Ambient Sound Receiver; ASR).
- E. A battery-powered ASIC.

Acute Sound Detector and Constant Signal-to-Noise Ratio headset system.

Detailed Description of Drawings

Figure 1 depicts the electroacoustical assembly **13** of an in-the-ear acoustic assembly, as it would typically be placed in the left ear canal **31** of a user **35**. The assembly is designed to be inserted into the users ear canal **31**, and to form an acoustic seal with the walls of the ear canal at a location **27**, between the entrance to the ear canal **15**, and the tympanic membrane or ear drum **33**. Such a seal is typically achieved by means of a soft and compliant housing of assembly **13**. Such a seal is critical to the performance of the system in that it creates a closed cavity **31** of approximately .5cc between the in-ear assembly **13** and the tympanic membrane **33**. As a result of this seal, the Ear Canal Receiver (ECR), or Speaker **25** 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 **15**. This seal is also the basis for the sound isolating performance of the electroacoustic assembly. Located adjacent to the ECR, is an Ear Canal Microphone (ECM) **23**, which is also acoustically coupled to closed cavity **31**. One of its functions 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. The Ambient Sound Microphone (ASM) **11** is housed in the ear seal **13** and monitors sound pressure at the entrance to the occluded ear canal. All transducers receive or transmit audio signals to the ASIC **21** that undertakes audio signal processing and provides a transceiver for audio via the wired or wireless communication path **19**.

Figure 2a depicts an Acute-Sound Pass-Through System (ACPTS). The input signal **2** is from the ASM **11**, which may be from one or both left and right earphone assemblies **13**. The input signal is processed with system **7**; comprising analog filtering **4** to compensate for the frequency response of the transducer, and digital filtering **6** after ADC **5**. An optional warning signal detector system **12** detects warning signals in the ASM signal with system **14** (e.g. car horns, klaxons). When a warning signal is detected, decision unit **18** invokes a pass-through mode **20** whereby the ASM signal is reproduced with the ECR **25**. The Background Noise Level (BNL) at the entrance to the occluded ear canal is calculated with system **8** (described in **Figure 3**). If a transient, high-level sound (or acute sound) is detecting in the ASM input signal buffer with system **10**, then it is converted to a level with system **16**, and it's magnitude compared with the BNL is calculated using the differencing system **22**. The magnitude of this resulting difference (signal A in **Figure 2**) is compared with a threshold **26** stored in computer data memory, using the comparator **28**. If this value is deemed positive by comparator **34**, and the level of the transient is

greater than a predefined threshold **36**, then comparator **38** invokes an optional Source Proximity Detector (see **Figures 2b, 2c and 2d**), which determines if the acute sound was created by the User's voice, using decision unit **35**. If a User-generated sound is NOT detected, then Pass-through system **40** is invoked, whereby the ASM signal is reproduced with the ECR **25**. If the difference signal from system **28** is not positive, or the level of the identified transient is too low, then a hysteresis unit **32** is invoked. This unit decides if the pass-through mode of units **40** or **20** were recently used (e.g. in the last 10 ms). If pass-through mode was recently activated, then decision unit **30** invokes pass-through system **40**; otherwise there is no pass-through of the ASM signal to the ECR (i.e. system **24** is invoked).

Figure 2b describes a Source Proximity Detector (SPD), to determine if the Acute (i.e. loud) sound detected by the system described in **Figure 2a** was created by the User's voice. The SPD uses as its inputs the ASM signals from the left **101** and right **103** electroacoustic headset assemblies, and in some embodiments the Ear Canal Microphone (ECM) signals from the left **105** and right **107** headsets. The electronic cross-correlation between the ASM signals is calculated with the system described in **Figure 2c**, which determines a Pass-through or Non Pass-through operating mode. In the specific embodiment whereby the cross-correlation of both the ASM and ECM signals is involved, pass-through mode is only invoked when both cross-correlation analysis systems return a "Pass-through" operating mode **40**, as determined by the logical AND unit **111**.

Figure 2c describes a component of the SPD wherein the cross-correlation of two input audio signals **113** and **115** (e.g. the left and right ASM signal) is calculated. The input signals may first be weighted using a frequency-dependant filter (e.g. an FIR-type filter) using filter coefficients **117** and filtering network **119, 121**. Alternatively, the interchannel cross-correlation calculated with function **123** may return a frequency-dependant correlation such as a coherence function. The absolute maximum peak of the calculated cross-correlation **127** may be subtracted from the mean (or RMS) **129** correlation, with subtractor **135**, and compared **137** with a predefined threshold **133**, to determine if the peak is significantly greater than the average correlation (i.e. a test for peakedness). Alternatively, the maxima of the peak may simply be compared with the threshold **133** without the subtraction process **135**. If the lag-time of the peak **131** is at approximately lag-sample 0, then the sound source is determined as being on the interaural axis- indicative of User-generated speech, and a no-pass through mode is returned **24** (a further function described in **Figure 2d** may be used to confirm that the sound source originates in the User-head, rather than external to the user- and further confirming that the acute sound is a User-generated voice sound). The logical AND unit **143** activates the pass-through mode **40** if both criteria in the decision units **137** and **139** confirm that the absolute maxima of the peak is above a predefined threshold **133**, AND the lag of the peak is NOT at approximately lag sample zero.

Figure 2c describes a further component of the SPD, which is optional to confirm that the acute sound source is from a location indicative of user-generated speech; i.e. inside the head. The cross-correlation system in **Figure 2c** is used to provide the time-lag of the

maximum absolute peak **123** for a pair of input signals; the ASM **101** and ECM **105** signals for the same headset (e.g. the ASM and ECM for the left headset). If the lag of the peak is greater than zero- this indicates that the sound arrived at the ECM signal before the ASM signal. Decision unit **145** determines if the lag is greater than zero for both the left and right headsets- and activates the pass-through mode **40** if so.

Figure 3 describes the Background Noise Level estimation system, which takes as its input **47** either or both the ASM **11** signal or signal from the Ear Canal Microphone (ECM) **23**. An audio buffer **46** of the input audio signal is accumulated (e.g. 10 ms of data), which is then processed by unit **48** to obtain the temporal envelope. This envelope is smoothed using the smoothing window stored in data memory **52** (e.g. a Hanning or Hamming shaped window) and the smoothing system **50** (e.g. an FIR-type low-pass digital filter). Transient peaks in the input buffer may be identified removed using system **54** to determine the “steady-state” BNL by averaging the buffer with system **56** (obtaining a value similar to, or the same as, the RMS)- giving signal BNL_1 **57**, which may be frequency dependant or may be a single value averaged over all frequencies. If the ECM was used to determine the BNL, then decision unit **55** adjusts the ambient BNL estimation to give an equivalent ear-canal BNL SPL, by deducting the Ear Plug Noise Reduction Rating **77** from the BNL estimate **49**. Alternatively, if the ECM **23** is used, then the SPL level of any reproduced Audio Content **58** is deducted from the ECM level with unit **47**. The updated BNL estimate is then integrated **60** with the previous BNL estimate **64**, by averaging a weighted previous BNL (weighted with coefficient **58**), to give a new ear-canal BNL **66**.

Figure 4a describes a Constant Signal-to-Noise Ratio System (CSNRS). It takes as its input an Audio Content signal **68**, e.g. a music audio signal from a portable Media Player or mobile-phone, which is processed with analog and digital signal processing system **72** and it's level calculated with unit **76**. The level is converted to an equivalent SPL value (ACL), which may use either a look-up-table or algorithm to calculate the ear-canal SPL of the signal if it was reproduced with the ECR **43**. The BNL is estimated according to **Figure 3**, which takes as it's inputs either or both the ASM signal **2**, processed with signal conditioning unit **7**, and/or the ECM signal **70**, processed with conditioning unit **74**. These signals are selected using the input switch **73**, which may be controlled automatically or with a specific user-generated manual operation **78**. The Ear-Canal SNR is calculated by differencing the ACL and BNL **81** and the resulting SNR **83** is passed to the AGC coefficient calculation system described in **figure 4b**. The AGC coefficient calculation system calculates the gains for the Audio Content signal and ASM signal from the Automatic Gain Control units **82** and **86** (for the Audio Content and ASM signals, respectively). After the ASM signal and Audio content signal have been processed by the AGC's **82** and **86**, the two signals are mixed with the audio signal mixer **90**. A safe-level check system **92** determines if the resulting mixed signal is too high, if it were reproduced with the ECR assembly **43**. The safe-level check system **92** may use information regarding the Users listening history to determine if their sound exposure is such that it may cause temporary or permanent hearing threshold shift. If such high levels are measured, then the safe-level check system **92** reduces the signal level of the mixed

signals via the feedback path to the mixer **90**. The resulting audio signal from system **92** is then reproduced with the ECR **25** via the signal processing assembly **43**.

Figure 4b describes the AGC coefficient calculation system, that takes as its inputs the Ear Canal SNR **83** and a target SNR **96** to give a SNR mismatch **94**. The target SNR is chosen from a pre-defined SNR, stored in computer memory **88** or a user, manually defined SNR **84**. The difference between the actual ear-canal SNR and the target SNR gives the mismatch **100**. The mismatch level is smoothed over time using the time averaging system **104**, which uses a previous mismatch **106** that is weighted using single or multiple weighting coefficients **102**, to give a new time-smoothed SNR mismatch **108**. Depending on the magnitude of this mismatch, various operating modes may be invoked, as described by the AGC mode decision module in **Figure 4c**.

The operating modes affect the gains of the Audio Content and ASM signals in AGC units **82** and **86** (respectively), before these audio signals are mixed and reproduced with the ECR using assembly **43**. The AGC mode decision module in **Figure 4c** is invoked when a new mismatch **108** is calculated. A decision unit **112** first determines if the absolute magnitude of the mismatch is small- in the order of ± 1 dB. If mismatch **108** is small, then mode 1 is invoked; as described in **table 3** that there is no change made to the AGC coefficients. Following a specific automatic or user-generated manual selection **78a**, decision unit **116** selects whether the ASM signal is monitored with the ECR, or alternatively only the Audio Content may be reproduced with the ECR. If the ASM signal is reproduced with the ECR comparator **120** determines if the ear-canal SNR mismatch is less than approximately -1 dB; if so, then mode 3 is invoked, whereby the Audio Content signal is boosted, and the ASM signal reduced, in order to increase the ear-canal SNR to a level closer to the desired ear-canal SNR. Likewise, if the user does not audition the ASM signal with the ECR, and the mismatch is less than approximately 1 dB, then mode 5 is invoked; whereby the Audio Content signal level is boosted to increase the ear-canal signal-to-noise ratio (as described in **table 3**). The user manual selection **78b** determines if the user desires to maintain a substantially constant ear-canal SNR, or whether the user desires to maintain a substantially *minimum* ear-canal SNR. If the ECR reproduces the ASM signal, and the mismatch is greater than approximately 1 dB, and the user selects a *constant* SNR, then mode 2 is invoked. Mode 2 operation reduces the gain of the Audio Content AGC and increases the ASM AGC gain- so as to reduce the SNR to the desired level. Alternately, if the ECR reproduces the ASM signal, and the mismatch is greater than approximately 1 dB, and the user selects a *minimum* SNR; then mode 1 is invoked. (As discussed, this implies that the Audio Content or ASM signal gains are not updated). And if the ECR does *not* reproduce the ASM signal, and the mismatch is greater than approximately 1 dB, and the user selects a *constant* SNR, then mode 4 is invoked. Mode 4 operations reduce the gain of the Audio Content - so as to reduce the ear-canal SNR to the desired level. Alternately, if the ECR does *not* reproduce the ASM signal, and the mismatch is greater than approximately 1 dB, and the user selects a *minimum* SNR; then mode 1 is invoked.

Table 3:
For SNR mismatch of M (dB)

Mode:	Audio Content Gain	ASM Gain
1	NU	NU
2	"-M/2"	"+M/2"
3	"+M/2"	"-M/2"
4	"-M"	"-∞"
5	"+M"	"-∞"

NU

No Update

Electronic Patent Application Fee Transmittal				
Application Number:				
Filing Date:				
Title of Invention:		Acute Sound Detector and Constant Signal-to-Noise Headset System		
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:	1453636
Application Number:	60885917
International Application Number:	
Confirmation Number:	8316
Title of Invention:	Acute Sound Detector and Constant Signal-to-Noise Headset System
First Named Inventor/Applicant Name:	Steven Wayne Goldstein
Customer Number:	32146
Filer:	Steven Wayne Goldstein
Filer Authorized By:	
Attorney Docket Number:	
Receipt Date:	22-JAN-2007
Filing Date:	
Time Stamp:	01:27:27
Application Type:	Provisional

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Submitted with Payment	yes
Payment was successfully received in RAM	\$ 100
RAM confirmation Number	1428
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	901398	no	4
Warnings:					
Information:					
2	Abstract	Acute_Sound_Detector_and _Constant_SNR_systems_ Abstract_Introduction.pdf	91464	no	4
Warnings:					
Information:					
3	Drawings	Acute_Sound_Detector_and _Constant_SNR_systems_ Drawings.pdf	477268	no	10
Warnings:					
Information:					
4	Specification	Acute_Sound_Detector_and _Constant_SNR_systems_ Embodiments.pdf	156664	no	9
Warnings:					
Information:					
5	Specification	Acute_Sound_Detector_and _Constant_SNR_systems_D etailed_Description_of_Draw ings.pdf	132870	no	5
Warnings:					
Information:					
6	Fee Worksheet (PTO-06)	fee-info.pdf	8101	no	2
Warnings:					
Information:					
Total Files Size (in bytes):			1767765		
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